



Avaya Aura[®] Communication Manager Screen Reference

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Contents

Chapter 1: Introduction	15
Purpose	15
Changes to platform support	15
Discontinued support for IP Server Interface (TN2312, commonly known as “IPSI”)	16
Communication Manager	16
Change history	17
Chapter 2: AAR Digit Analysis Table and ARS Digit Analysis Table	18
Chapter 3: AAR Digit Conversion Table and ARS Digit Conversion Table	23
Chapter 4: Abbreviated Dialing	26
Enhanced List	26
Group List	28
Personal List	29
System List	31
7103A Button Assignment	33
Chapter 5: Access Endpoint	34
Chapter 6: Administered Connection	37
AUTHORIZED TIME OF DAY	38
MISCELLANEOUS PARAMETERS	39
Chapter 7: Agent Login ID	41
Agent Login ID: page 1	41
MWI Served User Type	45
Unicode Name	47
Agent Login ID: page 2	47
Chapter 8: Alias Station	50
Chapter 9: Alphanumeric Dialing Table	51
Chapter 10: Announcements/Audio Sources	52
Live Stream Source	54
Chapter 11: ARS Toll Table	58
Chapter 12: Attendant Console	59
Attendant console: page 1	59
Attendant console: page 2 Softconsole IP Attendant	63
Attendant console: page 2 VIS feature options	65
Attendant console: page 3	65
Attendant console: page 3 feature button assignments	68
Attendant console: page 4 display module button assignments	70
Chapter 13: Audio Group	71
Chapter 14: AUDIX-MSA Node Names	73

Chapter 15: Authorization Code — COR Mapping	74
Chapter 16: Best Service Routing	76
Chapter 17: Bulletin board	79
Chapter 18: Button Type Customization Restriction	80
Chapter 19: Call Type Digit Analysis Table	82
Chapter 20: Call Vector	84
Chapter 21: Calling Party Number Conversion for Tandem Calls	90
Len	93
Outgoing Number Format	94
Outgoing Trunk Group(s)	94
Chapter 22: CAMA Numbering Format	95
Chapter 23: CDR system parameters	97
CDR system parameters: page 1	97
CDR retention (days)	98
CDR System Parameters: page 2	105
Chapter 24: Change Station Extension	108
Chapter 25: Class of Restriction	111
Class of restriction: page 1	111
Class of Restriction: page 2	119
Auto-answer media link calls	119
Class of Restriction: page 3	122
Allow Trunk Seizure On Busy Indicator	123
Homing Policy for Users	124
Internal Users Only	124
Listen Only Talk Path on Auto Answered Intercom Calls	125
Preferred CM List	126
Trunk Alerting Timer (minutes)	127
Class of restriction: page 4 to page 23	127
Chapter 26: Class of service	130
Class of service: page 1	131
Class of service: page 2	134
Bridging Exclusion Override	135
DCC Activation/Deactivation	135
Chapter 27: Cluster Preferred CM List	137
Pref-CM-List Number	138
Pref-CM-List Name	138
1-5	138
Chapter 28: Cluster Session Manager	139
Chapter 29: Code Calling IDs	141
Chapter 30: Configuration Set	142
Call Log Notify	144

EC500Delay Deactivation State.....	146
Feature Invocation by In-Call DTMF Code.....	147
Location for routing incoming overlap calls.....	147
Chapter 31: Console parameters	149
Console parameters: page 1.....	149
SIP.....	153
Console parameters: page 2.....	154
Console parameters: page 4.....	160
Console parameters: page 5.....	160
Chapter 32: Coverage Answer Group	161
Chapter 33: Coverage Path	163
Coverage path description.....	164
Criteria.....	165
Coverage points.....	167
Chapter 34: Crisis Alert System Parameters	170
ALERT PAGER.....	171
SNMP NOTIFICATION.....	173
SNMP Inform to Notify Adjunct When DCP and H.323 Stations Go In-Service.....	173
SNMP Inform to Notify Adjunct When SIP Station Dials Emergency Call.....	174
Chapter 35: CTI Link	175
CTI link: page 1.....	175
AES host.....	176
CTI link: page 2.....	177
IC Adjunct Routing.....	178
CTI link: page 3.....	179
Bridged Appearance Origination Restriction.....	179
SAC/CF Override.....	179
Chapter 36: Date and Time	180
Chapter 37: Daylight Saving Rules	183
Chapter 38: DCS to QSIG TSC Gateway	187
Chapter 39: Dial Plan Analysis Table	189
Chapter 40: Dial Plan Parameters	193
Call Type Analysis Behavior When Extension Not Assigned.....	194
Location-Based Call Type Analysis.....	197
Chapter 41: Digit Absorption	199
Chapter 42: Display Parameters	200
Chapter 43: Duplicate Station	203
Chapter 44: Duplicate Vector	204
Chapter 45: Enable Session	206
Chapter 46: Extended Pick-Up Group	207
Chapter 47: Extensions administered to have an MCT-Control button	208

Chapter 48: Extensions to Call Which Activate Features by Name.....	209
Chapter 49: Failover Group Domain Mapping.....	210
Chapter 50: Feature Access Code (FAC).....	212
Feature Access Code: page 1.....	212
Feature Access Code: page 2.....	216
Feature Access Code: page 3.....	221
Feature Access Code: page 4.....	224
Feature Access Code: page 5.....	228
Feature Access Code: page 6.....	231
Feature Access Code: page 7.....	232
Feature Access Code: page 8.....	233
Feature Access Code: page 9.....	235
WNDP PRECEDENCE ACCESS CODES.....	236
Destination Code Control.....	236
Chapter 51: Feature-related system parameters.....	238
Feature-related system parameters: page 1.....	238
Feature-related system parameters: page 2.....	244
Feature-related system parameters: page 3.....	246
Feature-related system parameters: page 4.....	249
Feature-related system parameters: page 5.....	253
Feature-related system parameters: page 6.....	258
Feature-related system parameters: page 7.....	265
Feature-related system parameters: page 8.....	269
ISDN PARAMETERS.....	269
PARAMETERS FOR CREATING QSIG SELECTION NUMBERS.....	272
Feature-related system parameters: page 9.....	274
Feature-related system parameters: page 10.....	277
Block Service Observe Across Tenants.....	277
ITALIAN DCS PROTOCOL.....	280
Feature-related system parameters: page 11.....	281
CALL CENTER SYSTEM PARAMETERS EAS.....	281
VECTERING.....	282
Service Observing field descriptions.....	285
Feature-related system parameters: page 12.....	287
AGENT AND CALL SELECTION.....	287
ASAI	289
CALL MANAGEMENT SYSTEMS.....	290
OTHER CALL MANAGEMENT SYSTEM FIELDS.....	291
Feature-related system parameters: page 13.....	292
Feature-related system parameters: page 14.....	296
SIP Transfer VDN Counted-Calls Options field descriptions.....	298
Feature-related system parameters: page 15.....	299
FORCED AGENT LOGOUT PARAMETERS.....	299

SIP STATION REACHABILITY CHECKING OPTIONS.....	300
Feature-related system parameters: page 16.....	301
Feature-related system parameters: page 17.....	304
Feature-related system parameters: page 18.....	306
Feature-related system parameters: page 19.....	311
Commercial Cost Avoidance (CCA) Related System Parameters.....	315
Chapter 52: Firmware Station Download	317
Chapter 53: Group Paging Using Speakerphone	320
Chapter 54: Holiday Table	324
Chapter 55: Hospitality	327
Hospitality: page 1.....	327
Hospitality: page 2.....	330
Hospitality: page 3.....	335
Chapter 56: Hunt Group	337
Hunt group: page 1.....	338
SIP URI.....	345
Hunt group: page 2.....	346
Hunt group: page 3.....	355
Interruptible Aux Deactivation Threshold.....	356
Hunt Group: page 4.....	358
Hunt Group: page 5 through X.....	364
Chapter 57: Incoming Call Handling Treatment	366
Chapter 58: Integrated Announcement Boards	370
Chapter 59: Integrated Announcement Translations	372
Chapter 60: Intercom Group	374
Chapter 61: Inter-Exchange Carrier (IXC) Codes	376
Inter-Exchange Carrier (IXC) Codes: page 1.....	376
Inter-Exchange Carrier (IXC) Codes: page 2.....	376
Chapter 62: Intra-Switch CDR	377
Chapter 63: IP Address Mapping	378
Chapter 64: IP Codec Set	380
IP Codec Set: page 1.....	380
Encrypted SRTCP.....	381
IP Codec Set: page 2.....	384
FB Timer.....	386
XMT.....	386
Media Connection IP Address Type Preferences.....	387
Packet Size.....	387
SIP 64K Data.....	388
IP Codec Set: page 3.....	389
Chapter 65: IP Interfaces	393

IP Interface: page 1.....	393
IP Interface: page 2.....	396
Chapter 66: IP network region.....	398
IP network region: page 1.....	398
NR Group.....	398
IP network region: page 2.....	403
ALTERNATIVE NETWORK ADDRESS TYPES.....	403
RTCP Reporting to Monitor Server Enabled.....	404
IP network region: page 3.....	405
IP network region: page 4.....	408
Chapter 67: IP Node Names.....	413
Chapter 68: IP options system parameters.....	414
SYSTEM PARAMETERS IP-OPTIONS: page 1.....	414
SYSTEM PARAMETERS IP-OPTIONS: page 2.....	418
SYSTEM PARAMETERS IP-OPTIONS: page 3.....	420
ALTERNATIVE NETWORK ADDRESS TYPES.....	420
SYSTEM PARAMETERS IP-OPTIONS: page 4.....	422
SYSTEM PARAMETERS IP-OPTIONS: page 5.....	423
BW Management Option.....	423
Signaling-group of Primary SM BW Mgr.....	424
Signaling-group of Secondary SM BW Mgr.....	424
Chapter 69: IP Routing.....	425
Chapter 70: IP Services.....	428
IP Services: page 1.....	428
TLS Encryption.....	430
IP Services: page 3.....	430
IP Services: page 4.....	431
Chapter 71: ISDN Trunk Group.....	433
ISDN Trunk Group: page 1.....	433
ISDN Trunk Group: page 2.....	445
ISDN Trunk Group: page 3.....	450
QSIG Trunk Group Options.....	461
ISDN Trunk Group: Administrable Timers.....	465
Shared UI Feature Priorities.....	466
CBC Service Trunk Group Allocation Plan Assignment Schedule.....	468
CBC Trunk Group Usage Allocation.....	469
Wideband Support Options.....	470
ISDN Trunk Group: Group Member Assignments.....	471
Chapter 72: Language translations.....	474
Chapter 73: Listed Directory Numbers.....	481
Chapter 74: Locations.....	482
Chapter 75: Location Parameters.....	486

Location Parameters: page 1.....	486
Convert ISDN Public Numbers to International for Display.....	487
Line Load Control.....	488
Time Format on Terminals.....	489
Replace International Access Code with '+'.....	491
Location Parameters: page 2.....	492
Location Parameters: page 3.....	493
Chapter 76: Login Administration.....	494
Chapter 77: Logging levels.....	495
Logging Levels: page 1.....	495
Logging Levels: page 2.....	495
Chapter 78: Loudspeaker Paging.....	497
Chapter 79: Maintenance-related system parameters.....	501
Chapter 80: Media-Gateway.....	502
Chapter 81: Media Server.....	503
Signaling Group.....	503
Voip Channel License Limit.....	503
Dedicated Voip Channel Licenses.....	504
Capacity Profile Max Sessions.....	504
Capacity Profile MPU Estimate.....	504
Cluster Type.....	504
Chapter 82: Message Waiting Indication Subscriber Number Prefixes.....	506
Chapter 83: Mode Code Related System Parameters.....	508
Chapter 84: MOH Group.....	511
Chapter 85: Multifrequency-Signaling-Related Parameters.....	513
Multifrequency-Signaling-Related Parameters: page 1.....	513
Multifrequency-Signaling-Related Parameters: page 2.....	520
Multifrequency-Signaling-Related Parameters: page 3.....	523
Multifrequency - Signaling- Related Parameters: page 4.....	528
Chapter 86: Multiple Level Precedence & Preemption (MLPP) Parameters.....	532
ANNOUNCEMENTS.....	532
ASAC Enabled.....	532
Attendant Queue Announcement.....	533
PRECEDENCE CALLING-DIALED DIGIT ASSIGNMENT.....	533
Chapter 87: Music Sources.....	537
Chapter 88: Network Facilities.....	540
Chapter 89: Network Region Group.....	541
Name.....	541
Chapter 90: Node Number Routing.....	542
Chapter 91: Numbering-Private Format.....	543
Chapter 92: Numbering — Public/Unknown Format.....	546

Chapter 93: Off-PBX Telephone Mobile Feature Extensions.....	549
Chapter 94: Partition Routing Table.....	550
Chapter 95: Personal CO Line Group.....	551
Personal CO Line Group: page 1.....	551
Personal CO Line Group: page 2.....	560
Chapter 96: Pickup Group.....	561
Chapter 97: Policy Routing Table.....	562
Chapter 98: Precedence Routing Digit Analysis Table.....	565
Destination Code Control.....	565
Percentage IPBO Blocked.....	566
Chapter 99: Precedence Routing Digit Conversion Table.....	568
Chapter 100: PRI Endpoint.....	570
Chapter 101: Processor Channel Assignment.....	574
Chapter 102: QSIG to DCS TSC Gateway.....	578
Chapter 103: Reason Code Names.....	580
Chapter 104: Remote Access.....	582
Chapter 105: Remote Call Coverage Table.....	586
Chapter 106: Remote Office.....	588
Chapter 107: RHNPA Table.....	590
Chapter 108: Route Pattern.....	591
Used for SIP Stations.....	600
Chapter 109: Security related system parameters.....	601
Security related system parameters: page 1.....	601
Security related system parameters: page 2.....	603
Chapter 110: Service Hours Table.....	606
Chapter 111: Signaling group.....	608
Signaling group: page 1.....	608
Clustered.....	610
Signaling group: page 2.....	627
Enable Failover Event Package Subscription.....	629
Failover/failback Signaling-group Pair.....	629
OPTIONS Request Parameters.....	630
Signaling group: NCA-TSC Assignment page.....	631
Chapter 112: SIT Treatment for Call Classification.....	633
Chapter 113: Site Data.....	637
Chapter 114: Station.....	638
Station: Page 1.....	644
Actual.....	644
Attendant.....	644
Button Module Pages.....	645

Buttons per Page.....	645
Set.....	656
System ID.....	659
Terminal Number.....	665
Unicode Name.....	665
Station: Page 2.....	666
EC500 State.....	672
Station: Page 3.....	677
Bridging Tone for This Extension.....	678
Do not Share Port with Unnamed Registration.....	678
Enhanced Callr-Info display for 1-line display phones.....	678
Offline Call Logging.....	680
Require Mutual Authentication if TLS.....	680
Station: Page 4.....	681
Station :Page 5.....	684
Station: Page 6.....	685
Enable Reachability for Station Domain Control.....	685
Primary Session Manager, Secondary Session Manager, Third Session Manager, and Fourth Session Manager.....	685
Route Selection.....	685
SIP URI.....	686
Chapter 115: Stations With Off-PBX Telephone Integration.....	690
Stations With Off-PBX Telephone Integration: page 1.....	690
Dual Mode.....	692
Stations With Off-PBX Telephone Integration: page 2.....	693
Stations With Off-PBX Telephone Integration: page 3.....	696
Chapter 116: Survivable ARS Analysis Table.....	697
Chapter 117: Survivable Processor.....	699
Survivable processor: page 1.....	699
Serial Number.....	700
Survivable License Grace Period (in days).....	701
Topology.....	702
Survivable Processor: page 3 (IP Services page).....	702
Priority with respect to Media Servers.....	703
Survivable Processor: page 4 (IP Services — Session Layer Timers page).....	704
Chapter 118: System Capacity.....	706
Chapter 119: System Configuration.....	707
Chapter 120: System Parameters Call Coverage/Call Forwarding.....	708
Chapter 121: System Parameters Country Options.....	718
Chapter 122: System Parameters Customer Options.....	724
System parameters customer options: page 1.....	724
Maximum Off-PBX Telephones - EMX.....	726

System parameters customer options: page 2.....	726
System parameters customer options: page 3.....	727
Maximum Number of Avaya Media Server VoIP Channels.....	727
System parameters customer options: page 4.....	728
System parameters customer options: page 5.....	732
System parameters customer options: page 6.....	736
System parameters customer options: page 7.....	739
System parameters customer options: page 8.....	742
System parameters customer options: page 9 (QSIG OPTIONAL FEATURES).....	743
System parameters customer options: page 10 (ASAI FEATURES).....	745
System parameters customer options: page 11 (MAXIMUM IP REGISTRATIONS BY PRODUCT ID).....	745
Chapter 123: System parameters maintenance.....	747
Chapter 124: System Parameters Media Gateway Automatic Recovery Rule.....	748
Chapter 125: Media Server Recovery Rules.....	750
Auto Return.....	750
Day.....	751
Report Expiration (sec).....	751
Report Interval (sec).....	751
Time.....	751
Time Delay.....	752
Chapter 126: System parameters OCM call classification.....	753
System parameters OCM call classification: page 1.....	753
System parameters OCM call classification: page 2.....	754
Chapter 127: System Parameters Special Applications.....	756
Chapter 128: Telecommuting Access.....	757
Chapter 129: Tenant.....	758
Tenant: page 1.....	758
SIP.....	760
Tenant: page 2.....	761
Chapter 130: Terminal Parameters.....	763
Chapter 131: Terminating Extension Group.....	767
Chapter 132: TFTP Server.....	771
Chapter 133: Time of Day Coverage Table.....	773
Chapter 134: Time of Day Routing Plan.....	775
Chapter 135: Time of Day Station Lock Table.....	777
Chapter 136: Toll Analysis.....	779
Chapter 137: Tone Generation.....	783
Tone Generation: page 1.....	783
Tone generation: page 2 (TONE GENERATION CUSTOMIZED TONES page).....	784
Chapter 138: Trunk Group.....	787

Trunk Group.....	787
Trunk Group: page 1.....	787
Trunk Group: page 2.....	806
Trunk Group: page 3.....	823
Administrable Timers.....	840
Trunk Group: Group Member Assignments.....	855
Trunk Group: Protocol Variations.....	857
Resend Display UPDATE Once on Receipt of 481 Response.....	858
Build Refer-To URI of REFER From Contact For NCR.....	859
Request URI Contents.....	862
Chapter 139: Uniform Dial Plan Table.....	864
Chapter 140: User Profile.....	867
Chapter 141: Variables for Vectors.....	868
Chapter 142: Vector Directory Number.....	871
Vector Directory Numbers: page 1.....	871
SIP URI.....	875
Unicode Name.....	875
Vector Directory Number: page 2 (Meet-me Conference disabled).....	876
Used for BSR Polling.....	881
Vector Directory Number: page 2 (Meet-me Conference enabled).....	883
Vector Directory Number: page 3.....	885
Chapter 143: Video Bridge.....	888
Chapter 144: Resources.....	892
Communication Manager documentation.....	892
Finding documents on the Avaya Support website.....	894
Accessing the port matrix document.....	894
Avaya Documentation Center navigation.....	895
Training.....	896
Viewing Avaya Mentor videos.....	897
Support.....	897
Using the Avaya InSite Knowledge Base.....	897
Appendix A: PCN and PSN notifications.....	899
PCN and PSN notifications.....	899
Viewing PCNs and PSNs.....	899
Signing up for PCNs and PSNs.....	900

Chapter 1: Introduction

Purpose

This document provides detailed description of the fields of Avaya Aura® Communication Manager Communication Manager screens.

The primary audience for this document is:

- Avaya field technicians
- Avaya partners
- Technical support personnel
- Solution architects
- Implementation engineers
- Support personnel
- Technical support representatives

Changes to platform support

As of January 1, 2026, Avaya has refined its infrastructure support strategy for Avaya Aura® and Surround Applications. This update specifically impacts Software-Only and Infrastructure as a Service (IaaS) deployment models. To ensure your environment remains compliant and supported, review the following changes to supported platforms:

Discontinued Platforms:

- Hypervisor: Microsoft Hyper-V
- Cloud Platforms: Microsoft Azure, Google Cloud Platform (GCP), IBM Cloud

Supported platforms for Software Only and Infrastructure as a Service (IaaS) deployment models:

- Cloud Platform: AWS
- On-premises platforms: KVM, Nutanix, VMware

Discontinued support for IP Server Interface (TN2312, commonly known as “IPSI”)

With Release 10.2, Communication Manager does not support the IP Server Interface (IPSI). As a result, access and functionality are removed. This means, the IPSI connected cabinets and gateways do not work with Communication Manager Release 10.2. Examples of IPSI connected cabinets and systems include G3cfs, G3csi, G3i, G3r, G3s, G3si, G3vs, G3x, G600, G650, MCC, SCC, CMC, IPSI, IP Server Interface, and IP port network.

Discontinued support also includes the TN8412, which previously paired with the TN8400 blade server. TN8412 was last supported with Communication Manager Release 5.x.

For more information, see the [End of sale G650 document](#) published on the Avaya Support website.

Communication Manager

Communication Manager is an open, scalable, reliable, and secure telephony application. Communication Manager provides call-processing solutions, user and system-management functionality, intelligent call routing, application integration and extensibility, and enterprise communications networking for large and small customer environments. The standard edition of Communication Manager uses H.248 for gateway control.

Note:

From Avaya Aura® Communication Manager Release 10.1 SMI page will have the following options but are for future use:

- Administration > Licensing > Feature Administration > Current Settings > Display > Optional Features > Clustering
- Administration > Server (Maintenance) > Server Configuration > Configure Memory (for LSP) > This Server's Memory Setting > X- Large/Array
- From Avaya Aura® Release 10.1, HP ProLiant DL360p G8 (CSR2), HP ProLiant DL360 G9 (CSR3), Dell™ PowerEdge™ R620 (CSR2), Dell™ PowerEdge™ R630 (CSR3), and Avaya Solutions Platform 120 servers are not supported.

However, in Release 10.1, Avaya Solutions Platform 120 can be upgraded to Avaya Solutions Platform 130 Release 5.x, and S8300E can be upgraded to Avaya Solutions Platform S8300 R5.1.x.

- From Avaya Aura® Release 10.1, Appliance Virtualization Platform is not available for deploying or upgrading the Avaya Aura® applications. To upgrade the Avaya Aura® applications, migrate the Appliance Virtualization Platform to Avaya Solutions Platform 130 (Avaya-Supplied ESXi 7.0) Release 5.x.

Change history

Issue	Date	Summary of changes
6	March 2026	Added the section: Changes to platform support on page 15
5	April 2025	Added the following section for Release 10.2.1.1: • Survivable License Grace Period (in days) on page 701
4	December 2024	Added the following sections for Release 10.2.1: • AES host on page 176. • Preferred CM List on page 126. • Homing Policy for Users on page 124. • Cluster Preferred CM List on page 137. • Pref-CM-List Number on page 138. • Pref-CM-List Name on page 138. • 1-5 on page 138. Updated the following sections for Release 10.2.1: • CTI link: page 1 on page 175. • Class of Restriction: page 3 on page 122.
3	August 2024	Updated the “SIP URI” sub-section under the following sections: • Vector Directory Numbers: page 1 • Hunt group: page 1
2	April 2024	Updated the section Service Observing by Recording Device on page 121.
1	December 2023	Release 10.2

Chapter 2: AAR Digit Analysis Table and ARS Digit Analysis Table

Communication Manager compares dialed numbers with the value of the **Dialed Strings** field and determines the route pattern for the dialed number.

To change the value of the fields on the AAR Digit Analysis Table screen, use the following command:

change aar analysis *n*, where *n* is the digit string.

To change the value of the fields on the ARS Digit Analysis Table screen, use the following command:

change ars analysis *n*, where *n* is the digit string.

ANI Req

Use this field to specify whether the R2-MFC and the Russian MF ANI calls require Automatic Number Identification (ANI).

The system displays the **ANI Req** field only if you disable the **Request Incoming ANI (non-AAR/ARS)** field on the Multifrequency-Signaling-Related Parameters screen.

Valid entry	Usage
n	The R2-MFC or Russian MF ANI calls do not require ANI. The default value is n.
y	The R2-MFC or Russian MF ANI calls require ANI.
r	The call on the Russian Shuttle trunk or the Russian Rotary trunk drops if the ANI request fails. The system displays this option only if you enable the Allow ANI Restriction on AAR/ARS field on the Feature-related system parameters screen.

Related links

[Allow ANI Restriction on AAR/ARS](#) on page 259

[Request Incoming ANI \(non-AAR/ARS\)](#) on page 519

Call Type (AAR only)

Use this field to enter the call type associated with each dialed string.

Valid entry	Usage
aar	Use this option for regular Automatic Alternate Routing (AAR) calls.
intl	Use this option if the Route index contains public network ISDN trunks that require international type of number encoding.
pubu	Use this option if the Route index contains public network ISDN trunks that require unknown type of number encoding.
lev0–lev2	Use this option for ISDN Private Numbering Plan (PNP) number formats.
unku	Use this option to set up an Implicit numbering plan. Users can use the plan to dial an extension without using an Automatic Route Selection (ARS) or AAR access code.

ISDN Protocol:

Call type numbering	Numbering plan identifier	Type of numbering
aar	E.164 (1)	national(2)
intl	E.164 (1)	international(1)
pubu	E.164 (1)	unknown(0)
lev0	PNP(9)	local(4)
lev1	PNP(9)	Regional Level 1(2)
lev2	PNP(9)	Regional Level 2(1)

Call Type (ARS only)

Use this field to enter the call type associated with each dialed string.

Valid entry	Usage	China Number 1, Call Type
intl	Use this option for public-network international calls.	toll-auto
alrt	Use this option to alert attendant consoles or other digital telephones when a user places an emergency call.	normal
emer	Use this option for emergency calls.	normal
fnpa	Use this option for ten-digit North American Numbering Plan (NANP) calls.	attendant
hpna	Use this option for seven-digit NANP calls.	normal
lop	Use this option for international operator calls.	attendant

Table continues...

Valid entry	Usage	China Number 1, Call Type
locl	Use this option for public-network local calls.	normal
lpvt	Use this option local private calls.	normal
natl	Use this option for non-NANP calls.	normal
npvt	Use this option for national private calls.	normal
nsvc	Use this option for national service calls.	normal
op	Use this option for operator calls.	attendant
pubu	Use this option for public-network number (E.164)-unknown calls.	normal
svcl	Use this option for national(2) calls.	toll-auto
svct	Use this option for national(2) calls.	normal
svfl	Use this option for service call first party control calls.	toll
svft	Use this option for service call first party control calls.	local

Dialed String

Communication Manager matches the dialed numbers with the entry in the **Dialed String** field that most closely matches the dialed number. You can enter up to 18 digits that the call-processing server analyzes. You can also enter the wildcard characters, `x` and `X`.

Location

This field displays the location of the telephone that is used to dial the number. For more information about assigning a location and for a list of features that use the location, see *Avaya Aura® Communication Manager Feature Description and Implementation*.

Valid entry	Usage
1 to 250 or 1 to 2000	If there are matching entries in the location of the telephone, the system displays the matched entry.
all	If there are no matching entries in the location of the telephone, the system displays all .

Related links

[Multiple Locations](#) on page 736

Max

Use this field to specify the maximum number of user-dialed digits that the system must collect to match the dialed string.

Min

Use this field to specify the minimum number of user-dialed digits that the system must collect to match the dialed string.

Node Number

Use this field to specify a destination node number when the system uses node number routing or Distributed Communication System (DCS).

Valid entry	Usage
blank	Use this option if you do not want to enter the destination node number.
1 to 999	Use this option to enter the destination node number.

Percent Full

This field displays the percentage of system memory resources used for the AAR and the ARS digit analysis. For more information about the system memory resources, see *Avaya Aura® Communication Manager System Capacities Table*.

Route Pattern

Use this field to specify the route number that Communication Manager must use for the dialed string.

Valid entry	Usage
p1 to p2000	To enter a route index number.
digits	To enter a route pattern. For the maximum allowed entry, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .

Table continues...

Valid entry	Usage
r1 to r32	To enter a Remote Home Numbering Plan Area (RHNP) number if the dialed string requires RHNP translations.
node	To enable the system to use node number routing.
deny	To enable the system to block a call.

Chapter 3: AAR Digit Conversion Table and ARS Digit Conversion Table

Use the AAR Digit Conversion Table and the ARS Digit Conversion Table screens to modify a dialed number for efficient routing. You can insert or delete digits from the dialed number. For example, you can delete 1 and the area code from the dialed number and avoid long distance charges by routing the call over the private network.

To change the value of the fields on the AAR Digit Conversion Table screen, use the following command:

`change aar digit-conversion n`, where *n* is the digit string.

To change the value of the fields on the ARS Digit Conversion Table screen, use the following command:

`change ars digit-conversion n`, where *n* is the digit string.

ANI Reqd

Use this field to specify whether the R2-MFC and the Russian MF ANI calls require Automatic Number Identification (ANI).

The system displays the **ANI Reqd** field only if you disable the **Request Incoming ANI (non-AAR/ARS)** field on the Multifrequency-Signaling-Related Parameters screen.

Valid entry	Usage
n	The R2-MFC or Russian MF ANI calls do not require ANI. The default value is n.
y	The R2-MFC or Russian MF ANI calls require ANI.
r	The call on the Russian Shuttle trunk or the Russian Rotary trunk drops if the ANI request fails. The system displays this option only if you enable the Allow ANI Restriction on AAR/ARS field on the Feature-related system parameters screen.

Related links

[Allow ANI Restriction on AAR/ARS](#) on page 259

[Request Incoming ANI \(non-AAR/ARS\)](#) on page 519

Conv

Use this field to enable additional digit conversion.

Del

Use this field to enter the number of digits that the system must delete from the starting of the dialed string.

Location

This field displays the location of the telephone that is used to dial the number. For more information about assigning a location and for a list of features that use the location, see *Avaya Aura® Communication Manager Feature Description and Implementation*.

Valid entry	Usage
1 to 250 or 1 to 2000	If there are matching entries in the location of the telephone, the system displays the matched entry.
all	If there are no matching entries in the location of the telephone, the system displays all .

Related links

[Multiple Locations](#) on page 736

Matching Pattern

Use this field to specify the pattern that the system must use to match the dialed number. The pattern can be up to 18 characters long and consist of digits and the wildcard characters, `x` and `X`.

Max

Use this field to specify the maximum number of user-dialed digits that the system must collect to match the dialed string.

Min

Use this field to specify the minimum number of user-dialed digits that the system must collect to match the dialed string.

Net

Use this field to specify the method that the call processing server must use to analyze the converted number.

Valid entry	Usage
ext	The call processing server analyzes the converted number as an extension number.
aar	The call processing server analyzes the converted number as an AAR number.
ars	The call processing server analyzes the converted number as an ARS number.

Percent Full

This field displays the percentage of system memory resources used for the AAR and the ARS digit analysis. For more information about the system memory resources, see *Avaya Aura® Communication Manager System Capacities Table*.

Replacement String

Use this field to specify the digit string that must replace the deleted portion of the dialed number.

Valid entry	Usage
blank	To delete the digits without replacement.
0 to 9, *	To type the digit string. The digit string can have up to 18 characters.
#	To indicate the end of dialing.

Chapter 4: Abbreviated Dialing

Use the Abbreviated Dialing screen to store and retrieve dialing lists that simplify or eliminate dialing.

The types of Abbreviated Dialing lists are:

- Enhanced list
- Group list
- Personal list
- System list

Users can use these lists to:

- Make local, long distance, and international calls.
- Activate or deactivate features.
- Access the remote system.

Related links

[Enhanced List](#) on page 26

[Group List](#) on page 28

[Personal List](#) on page 29

[System List](#) on page 31

[7103A Button Assignment](#) on page 33

Enhanced List

Use this screen to create systemwide or personal lists for speed dialing.

Note:

You must activate dialing in the license file before the system programs Abbreviated Dialing Enhanced List.

To modify the value of the fields on the Enhanced List screen, use the following command:

`change abbreviated-dialing enhanced n`, where *n* is an enhanced list number.

Related links

[Abbreviated Dialing](#) on page 26

[DIAL CODE](#) on page 27

[Privileged](#) on page 27

[Size \(multiple of 5\)](#) on page 27

[Abbreviated Dialing Enhanced List](#) on page 728

DIAL CODE

Use this field to specify the number that the system must dial when a user enters the dial code. The number can have up to 24 characters. You can also assign a feature access code.

Valid entry	Usage
0 to 9	To specify the number.
* or #	To specify the feature access code.
~p	To enable the system to pause for 1.5 seconds.
~w	To enable the system to wait for dial tone.
~m	To enable the system to output DTMF digits at the end-to-end rate.
~s	To enable the system to suppress the display of the digits being outputted.
~W	To enable the system to wait indefinitely for the dial tone if the network response time is more than 30 seconds. This entry is unavailable for Avaya Solutions Platform S8300 servers.

Related links

[Enhanced List](#) on page 26

[Group List](#) on page 28

[Personal List](#) on page 29

[System List](#) on page 31

Privileged

Use this field to allow users to dial any number in the list, regardless of the class of restriction number. The default value is n.

Related links

[Enhanced List](#) on page 26

[Group List](#) on page 28

[System List](#) on page 31

Size (multiple of 5)

Use this field to specify the number of dial code entries required in the list. You can enter a value from 5 to 100 in multiples of 5. The default value is 5.

Related links

[Enhanced List](#) on page 26

[Group List](#) on page 28

[Personal List](#) on page 29

[System List](#) on page 31

Group List

Use this screen to create a group list. You can assign up to 100 numbers to every group list.

To change the value of the fields on the Group List screen, use the following command:

change abbreviated-dialing group *n*, where *n* is the group list number.

Related links

[Abbreviated Dialing](#) on page 26

[DIAL CODE](#) on page 27

[Group List](#) on page 29

[Privileged](#) on page 27

[Program Ext](#) on page 29

[Size \(multiple of 5\)](#) on page 27

DIAL CODE

Use this field to specify the number that the system must dial when a user enters the dial code. The number can have up to 24 characters. You can also assign a feature access code.

Valid entry	Usage
0 to 9	To specify the number.
* or #	To specify the feature access code.
~p	To enable the system to pause for 1.5 seconds.
~w	To enable the system to wait for dial tone.
~m	To enable the system to output DTMF digits at the end-to-end rate.
~s	To enable the system to suppress the display of the digits being outputted.
~W	To enable the system to wait indefinitely for the dial tone if the network response time is more than 30 seconds. This entry is unavailable for Avaya Solutions Platform S8300 servers.

Related links

[Enhanced List](#) on page 26

[Group List](#) on page 28

[Personal List](#) on page 29

[System List](#) on page 31

Group List

This field displays the number that you assign to the group list.

Related links

[Group List](#) on page 28

Privileged

Use this field to allow users to dial any number in the list, regardless of the class of restriction number. The default value is n.

Related links

[Enhanced List](#) on page 26

[Group List](#) on page 28

[System List](#) on page 31

Program Ext

Use this field to enter the extension that has permissions to program the Group list.

Related links

[Group List](#) on page 28

Size (multiple of 5)

Use this field to specify the number of dial code entries required in the list. You can enter a value from 5 to 100 in multiples of 5. The default value is 5.

Related links

[Enhanced List](#) on page 26

[Group List](#) on page 28

[Personal List](#) on page 29

[System List](#) on page 31

Personal List

Use this screen to create a personal dialing list for telephone or data module users.

To change the value of the fields on the Personal List screen, use the following command:

change abbreviated-dialing personal *n*, where *n* is a personal list number.

Related links

[Abbreviated Dialing](#) on page 26

[DIAL CODE](#) on page 27

[List Number](#) on page 30

[Personal List](#) on page 30

[Size \(multiple of 5\)](#) on page 27

DIAL CODE

Use this field to specify the number that the system must dial when a user enters the dial code. The number can have up to 24 characters. You can also assign a feature access code.

Valid entry	Usage
0 to 9	To specify the number.
* or #	To specify the feature access code.
~p	To enable the system to pause for 1.5 seconds.
~w	To enable the system to wait for dial tone.
~m	To enable the system to outpulse DTMF digits at the end-to-end rate.
~s	To enable the system to suppress the display of the digits being outpulsed.
~W	To enable the system to wait indefinitely for the dial tone if the network response time is more than 30 seconds. This entry is unavailable for Avaya Solutions Platform S8300 servers.

Related links

[Enhanced List](#) on page 26

[Group List](#) on page 28

[Personal List](#) on page 29

[System List](#) on page 31

List Number

This field displays which of the three personal lists is defined for the telephone.

Related links

[Personal List](#) on page 29

Personal List

Use this field to enter the extension of the telephone using this list.

Related links

[Personal List](#) on page 29

Size (multiple of 5)

Use this field to specify the number of dial code entries required in the list. You can enter a value from 5 to 100 in multiples of 5. The default value is 5.

Related links

[Enhanced List](#) on page 26

[Group List](#) on page 28

[Personal List](#) on page 29

[System List](#) on page 31

System List

Use this screen to create the system list. You can create only one system list.

To change the value of the fields on the System List screen, use the following command:

change abbreviated-dialing system *n*, where *n* is the system list number.

Related links

[Abbreviated Dialing](#) on page 26

[DIAL CODE](#) on page 27

[Label Language](#) on page 32

[LABELS FOR 2420/4620 STATIONS](#) on page 32

[Privileged](#) on page 27

[Size \(multiple of 5\)](#) on page 27

DIAL CODE

Use this field to specify the number that the system must dial when a user enters the dial code. The number can have up to 24 characters. You can also assign a feature access code.

Valid entry	Usage
0 to 9	To specify the number.
* or #	To specify the feature access code.
~p	To enable the system to pause for 1.5 seconds.
~w	To enable the system to wait for dial tone.
~m	To enable the system to output DTMF digits at the end-to-end rate.
~s	To enable the system to suppress the display of the digits being outputted.
~W	To enable the system to wait indefinitely for the dial tone if the network response time is more than 30 seconds. This entry is unavailable for Avaya Solutions Platform S8300 servers.

Related links

[Enhanced List](#) on page 26

[Group List](#) on page 28

[Personal List](#) on page 29

[System List](#) on page 31

Label Language

Use this field to administer custom labels on the 2420 and 4620 telephones. The system can display the labels in one of the following languages:

- english
- french
- italian
- spanish

Note:

Unicode is not a valid language.

If you change the language, the system saves all administered labels in the new language and displays the labels in the new language.

To translate the labels to a different language, type `user-defined` in the **Label Language** field. You can use Avaya Message Editing Tool (AMET) to assign translations to the labels. For more information about AMET, see *Job Aid: Telephone Message Translations*.

Related links

[System List](#) on page 31

[Display Character Set](#) on page 722

LABELS FOR 2420/4620 STATIONS

Use this field to customize the labels for the Abbreviated Dialing buttons on the 2420 and 4620 telephones. The labels can have up to 15 alphanumeric characters.

Related links

[System List](#) on page 31

Privileged

Use this field to allow users to dial any number in the list, regardless of the class of restriction number. The default value is n.

Related links

[Enhanced List](#) on page 26

[Group List](#) on page 28

[System List](#) on page 31

Size (multiple of 5)

Use this field to specify the number of dial code entries required in the list. You can enter a value from 5 to 100 in multiples of 5. The default value is 5.

Related links

[Enhanced List](#) on page 26

[Group List](#) on page 28

[Personal List](#) on page 29

[System List](#) on page 31

7103A Button Assignment

Use this screen to assign abbreviated dialing numbers to the 7103A telephone buttons. You can use this list only for 7103A fixed feature telephones. You can implement only one 7103A abbreviated dialing list and this list applies to all 7103A fixed feature telephones in the system.

To modify the value of the fields on the 7103A Button Assignment screen, use the following command:

`change abbreviated-dialing 7103A-buttons n` where *n* is a 7103A button assignment abbreviated dialing list.

Related links

[Abbreviated Dialing](#) on page 26

[DIAL CODE](#) on page 33

DIAL CODE

Use this field to assign a number to each dial code button. Any additions or changes apply to all 7103A fixed feature telephones. A call progress tone receiver is tied up while the system is waiting. The outgoing calling capability is impaired as there are a limited number of receivers in the system.

You can also assign a VDN.

Valid entry	Usage
0 to 9	To specify the number.
* or #	To specify the feature access code.
~p	To enable the system to pause for 1.5 seconds.
~w	To enable the system to wait for dial tone.
~m	To enable the system to output DTMF digits at the end-to-end rate.
~s	To enable the system to suppress the display of the digits being outputted.
~W	To enable the system to wait indefinitely for the dial tone if the network response time is more than 30 seconds. This entry is unavailable for Avaya Solutions Platform S8300 servers.

Related links

[7103A Button Assignment](#) on page 33

Chapter 5: Access Endpoint

Use the Access Endpoint screen to administer an access endpoint or a Wideband Access Endpoint (WAE).

WAE is an endpoint application connected to line-side non-ISDN T1 or E1 facilities. WAEs do not have signaling interface with the system. You can administer WAE only if **Wideband Switching** is enabled for the system.

An access endpoint is a nonsignaling trunk that neither responds to signaling nor generates signaling. You can use access endpoints to assign an extension number to a single trunk and eliminate the need to dedicate an entire trunk group for a single trunk. You can specify an access endpoint as the originator or the destination endpoint of an administered connection.

 Note:

Access endpoints and WAEs use the same resources that the trunks use. Therefore, the sum of access endpoints and trunks cannot exceed the maximum number of trunks available in the system.

To change the value of the fields on the Access Endpoint screen, use the following command:

add access-endpoint *next*, where *next* is the next available extension.

Related links

[Wideband Switching](#) on page 738

Communication Type

Use this field to specify the communication type for the access endpoint.

Valid entry	Usage
56k-data	For a DS1 access endpoint. You can use this option for the robbed-bit trunks. The default value is 56k-data.
64K-data	For a DS1 access endpoint. Do not use this option for the robbed-bit trunks.
voice-grade-data	For an analog tie trunk access endpoint.
wideband	For a WAE.

COR

Use this field to assign a class of restriction (COR) number to the access endpoint. You must administer this field so that only an Administered Connection (AC) endpoint can be connected to another AC endpoint.

COS

Use this field to assign a class of service (COS) number to the access endpoint. You must administer this field so that Communication Manager restricts the use of the **Call Forwarding All Calls** feature on the access endpoint.

Extension

Use this field to assign an extension number to the nonsignaling trunk.

ITC

Use this field to assign a transmission facility to the access endpoint. The system displays this field only if the **Communication Type** field is set to 56k-data, 64k-data, or Wideband.

Valid entry	Usage
restricted	Restricted transmission facilities are used to complete the call. Before using this option, ensure that the associated port of the access endpoint is a channel of a DS1 media module and the Line Coding field of the DS1 media module is set to B8ZS. The default value is restricted.
unrestricted	Unrestricted transmission facilities are used to complete the call.

Name

Use this field to provide a name for the access endpoint.

 Note:

Avaya BRI deskphones support only ASCII characters because non-ASCII characters, such as Eurofont and Kanafont, show up incorrectly.

(Starting) Port

Use this field to assign a port number to the access endpoint.

You can assign the port number by using one the following formats:

- *gateway number : module number : port number*

Valid entry	Usage
001 to 250	The first three characters represent the gateway number.
V1 to V9	The fourth and fifth characters represent the module number.
01 to 31	The sixth and seventh characters represent the port number.

 Note:

For WAEs, use the DS1 media module, not analog tie trunks.

TN

Use this field to assign a tenant number to the access endpoint. You can enter a value from 1 to 250.

Width

The system displays this field only if the **Communication Type** field is set to wideband. Use this field to provide the number of adjacent DS0 ports beginning with the specified starting port that make the WAE. You can enter a value between 2 to 31. The default entry is 6.

Related links

[Communication Type](#) on page 34

Chapter 6: Administered Connection

Use the Administered Connection screen to assign an end-to-end Administered Connection (AC) between two access endpoints or data endpoints. The system establishes the AC after a restart or when the AC is active.

To change the value of the fields on the Administered Connection screen, use the following command:

change administered-connection *n*, where *n* is the AC number.

Connection Number

This field displays the AC number.

Destination

Use this field to specify the address of the destination access endpoint or data endpoint. The endpoint can be remotely connected to the server on which the AC is assigned.

You can enter one of the following:

- A local extension that is assigned to an access endpoint or a data endpoint
- An Abbreviated Dialing entry

The value must be consistent with the dial plan.

Enable

Use this field to enable Communication Manager to establish the AC when the AC is active.

Valid entry	Usage
y	Communication Manager attempts to establish the AC when the AC is active. The default value is y.
n	Communication Manager does not attempt to establish the AC when the AC is active.

Name

Use this field to enter the name of the AC. You can enter up to 27 alphanumeric characters.

 Note:

Avaya BRI deskphones support only ASCII characters because non-ASCII characters, such as Eurofont and Kanafont, show up incorrectly.

Originator

Use this field to specify an assigned access endpoint extension or data module extension. The endpoint must be registered with the same instance of Communication Manager on which you administered the nonsignaling DS1 trunk or the analog tie trunk.

AUTHORIZED TIME OF DAY

Continuous

Use this field to schedule the AC to be active for the entire time. The AC is reestablished if the connection fails.

Valid entry	Usage
y	The AC is active for the entire time. The default value is y.
n	The AC is active only for the scheduled days and times.

Duration

Use this field to specify the duration for which the scheduled AC must remain active. The maximum duration is 167:59. The system displays this field only if you set the **Continuous** field to n.

Valid entry	Usage
000 to 167	To enter the duration in hours.
00 to 59	To enter the duration in minutes.

Start Days (Sun through Sat)

Use the following fields to specify the days when an attempt is made to establish the AC:

- Sun
- Mon
- Tue
- Wed
- Thu
- Fri
- Sat

The system displays these fields only if you set the **Continuous** field to n. The start days can be the days when the AC is inactive. A scheduled AC might be active over a number of days. In this situation, use the fields only to specify the days when the AC starts.

Related links

[Continuous](#) on page 38

Start Time

Use this field to specify the time of day when Communication Manager must establish the AC. You can enter a value from 00:00 to 23:59. The system displays this field only if you set the **Continuous** field to n.

Related links

[Continuous](#) on page 38

MISCELLANEOUS PARAMETERS

Alarm Threshold

Use this field to specify the number of attempts to establish or reestablish an AC that must fail consecutively before the system generates an AC alarm. For example, If you set this field to 1, the system generates an alarm after the first failure. You can enter a value from 1 to 10. The default value is 5.

The system displays this field only if you set the **Alarm Type** field to a value other than none.

Related links[Alarm Type](#) on page 40

Alarm Type

Use this field to specify the type of alarm generated when:

- The AC cannot be established.
- The AC cannot be reestablished after a failure.
- The number of consecutive failures equals the alarm threshold.

Valid entry	Usage
warning	For failures that do not cause significant degradation of service. These alarms are not reported to the attendant console or Initialization and Administration System (INADS).
major	For failures that cause critical degradation of service and require immediate attention.
minor	For failures that: • Cause degradation of service, but continue to support a crucial part of the system. • Require action, but the consequences of the failure are gradual. • Interfere with a feature or impair the services to trunks or stations.
none	For disabling the alarm notification.

Auto Restoration

Use this field to enable the system to reestablish the AC that failed. The **Auto restoration** field is available only for the AC that is established over an ISDN Software Defined Data Network (SDDN) trunk group.

Priority

Use this field to specify the order in which the ACs are established. You must enter a value from 1 to 8. The highest priority is 1, and the lowest priority is 8. The default priority is 5.

Retry Interval

Use this field to specify the duration between the attempts to establish or reestablish the AC. You must enter the duration in minutes. The default value is 2 minutes.

Chapter 7: Agent Login ID

In an Expert Agent Selection (EAS) environment, use the Agent Login ID screen to:

- Add an agent login ID.
- Change the options of an agent login ID.
- Assign skills to an agent login ID.

To change the value of fields on the Agent Login ID screen, use one of the following commands:

- **change agent-loginID** *n*, where *n* is the agent login ID.

For any changes to take effect, the agent must log out and log in again.

- **change agent-loginID** *n* auto, where *n* is the agent login ID.

For the changes related to the skill set to take effect, the system logs out and logs in the agent. However, for the other changes to take effect, the agent must log out and log in again.

Agent Login ID: page 1

AAS

Use this field to enable the system to use the agent login ID as a port for Auto Available Split/Skill (AAS).

Valid entry	Usage
n	To use the agent login ID as a port for the switch adjunct equipment.
y	To use the agent login ID as a port for AAS. If you use this option, you must run the remove Agent-LoginID command.

ACW Agent Considered Idle

Use this field to include the After Call Work (ACW) agents in the Most-Idle Agent (MIA) queue.

Valid entry	Usage
system	To apply the systemwide values.

Table continues...

Valid entry	Usage
n	To exclude the ACW agents from the MIA queue.
y	To include the ACW agents in the MIA queue.

Related links

[ACW Agents Considered Idle](#) on page 287

AUDIX

Use this field to enable the system to use the agent login ID as a port for voice messaging.

If an agent login ID is used as a port for AAS, the agent login ID cannot be used as a port for voice messaging.

Audix Name for Messaging

Use this field to assign a name to the messaging system that is used for Leave Word Calling (LWC) reception. The messaging system also provides call coverage to the agent login ID.

Auto Answer

Use this field to enable the **auto answer** setting of the agent login ID.

Valid entry	Usage
station	The auto answer setting on the station overrides the auto answer setting of the agent.
acd	The auto answer setting applies only to the ACD and direct agent calls. The agent extension gets a ringing tone for non-ACD calls.
none	The auto answer setting is disabled. The agent extension gets a ringing tone for all calls.
all	The auto answer setting applies to all ACD and non-ACD calls to the agent. For non-ACD calls, the agent extension receives a single ring. To prevent the ring, use the ringer-off button.

Related links

[Allow Ringer-off with Auto-Answer](#) on page 293

[Auto Answer](#) on page 668

Aux Work Reason Code Type

Use this field to specify whether the agent must enter a reason code when changing to the AUX work mode.

Valid entry	Usage
system	The systemwide settings apply.
none	The agent must not enter a reason code when changing to the AUX work mode.

Table continues...

Valid entry	Usage
requested	The agent can enter a reason code when changing to the AUX work mode.
forced	The agent must enter a reason code when changing to the AUX work mode.

Related links

[Aux Work Reason Code Type](#) on page 296

[Reason Codes](#) on page 740

COR

Use this field to assign a class of restriction (COR) number to the agent login ID. You can enter a value from 0 to 995. The default value is 1.

Coverage Path

Use this field to specify the coverage path used by the calls to the agent.

The coverage path is used when:

- The agent is logged out.
- The agent does not answer.
- The agent is attending a personal call while logged in.

Valid entry	Usage
blank	Use this option if you do not want to enter a coverage path.
1 to 199	Use this option to enter a coverage path number. The upper limit might change depending on the system configuration. For more information, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .
t1 to t199	Use this option to enter a time-of-day table number.

Direct Agents Calls First

Use this field to enable the system to select the direct agent calls before other ACD calls.

The system displays this field only if you set the **Call Handling Preference** field to `percent-allocation`.

Related links

[Call Handling Preference](#) on page 47

Forced Agent Logout Time

Use this field to specify the time at which the system must log out the agent.

Valid entry	Usage
blank	Use this option if you do not want to specify the time at which the system must log out the agent.
00, 15, 30, or 45	Use this option to enter the time in minutes.
01 to 23	Use this option to enter the time in hours.

Related links

[Timezone Offset](#) on page 485

Login ID

This field displays the login id for the agent.

LoginID for ISDN Display

Valid Entry	Usage
y	The Agent LoginID CPN (Calling Party Number) and Name are included in ISDN messaging over network facilities.
n	Default. The calling party name and number are determined by the conditions administered for the trunk group.

Related links

[Name](#) on page 46

Logout Reason Code Type

Use this field to specify whether the agent must enter a reason code when logging out.

Valid entry	Usage
system	The systemwide settings apply.
none	The agent must not enter a reason code when logging out.
requested	The agent can enter a reason code when logging out.
forced	The agent must enter a reason code when logging out.

Related links

[Logout Reason Code Type](#) on page 296

[Reason Codes](#) on page 740

LWC Reception

Use this field to specify the location where the system must store the LWC messages.

Valid entry	Usage
spe	Use this option to store the LWC messages on Switch Processor Element (SPE).
none	Use this option if you do not want to store the LWC messages.
audix	Use this option to store the LWC messages on the voice messaging system.

Related links

[AUDIX Name](#) on page 358

Maximum time agent in ACW before logout (sec)

Use this option to enter the maximum time an agent can be in ACW on a per agent basis.

Valid entry	Usage
system	With this option, system-wide settings apply. This is the default option.
none	With this option, ACW timeout is inapplicable to the agent.
30 to 9999	With this option, you can provide the number of seconds in a specific timeout period. The setting takes precedence over the system setting for maximum time in ACW.

Related links

[Maximum Time Agent in ACW before Logout \(sec.\)](#) on page 299

Messaging Server Name for Messaging

Use this field to assign a name for the messaging system that is used for LWC reception and that provides coverage for an agent login id. By default, the field is blank.

MIA Across Skills

Use this field to specify if the system must remove an agent from the MIA queue.

Valid entry	Usage
system	With this option, system-wide settings apply. This is the default option.
y	With this option, the system removes an agent from the MIA queues for all the splits or skills for which an agent is available when the agent answers a call from any assigned splits or skills.
n	With this option, the system removes ACW agents from the queue.

Related links

[MIA Across Splits or Skills](#) on page 289

MWI Served User Type

Controls the auditing or interrogation of a served user's message waiting indicator (MWI).

Valid Entries	Usage
fp-mwi	The station is a served user of an fp-mwi message center.
qsig-mwi	The station is a served user of a qsig-mwi message center.
sip-adjunct	Used to audit message waiting lamps.
blank	The served user's MWI is not audited or if the user is not a served user of either an fp-mwi or qsig-mwi message center.

Name

Use this field to enter the name of the agent. You can enter up to 27 alphanumeric characters.

 Note:

Supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, 1600-Series, 9600-Series IP Deskphones, 9600-Series H.323 Deskphones, and Avaya J100 Series IP Phones. Unicode is also an option for the 2420J telephone when the **Display Character Set** is katakana. For more information about 2420J, see *2420 Digital Telephone User's Guide*.

Related links

[Display Character Set](#) on page 722

Password

Use this field to assign a password to the agent login ID.

The **Minimum Agent-LoginID Password Length** field specifies a minimum length for the password.

Related links

[Minimum Agent-LoginID Password Length](#) on page 282

Password (enter again)

Use this field to enter the same password that you entered for the **Password** field.

Port Extension

Use this field to assign an extension to the port used for AAS or voice messaging.

You cannot use a vector directory number (VDN) or an agent login ID as a port extension.

Security Code

Use this field to enter the four-digit station security code for the **Demand Print messages** feature.

TN

Use this field to assign a tenant number (TN) to the agent login ID. The default value is 1. The upper limit might change depending on the system configuration. For more information, see *Avaya Aura® Communication Manager System Capacities Table*.

Unicode Name

This read-only field shows the assignment of Unicode Name to the agent.

Valid entry	Usage
n	Unicode Name is not assigned to the agent.
y	Unicode Name is assigned to the agent.

For more information on Unicode Language Display, see *Administering Avaya Aura® Communication Manager*.

Agent Login ID: page 2

add agent-loginID 3600004 Page 2 of 3

AGENT LOGINID

Direct Agent Skill: _____

Call Handling Preference: skill-level Local Call Preference? n

SN	SL	SN	SL	SN	SL	SN	SL
1: _____	_____	16: _____	_____	31: _____	_____	46: _____	_____
2: _____	_____	17: _____	_____	32: _____	_____	47: _____	_____
3: _____	_____	18: _____	_____	33: _____	_____	48: _____	_____
4: _____	_____	19: _____	_____	34: _____	_____	49: _____	_____
5: _____	_____	20: _____	_____	35: _____	_____	50: _____	_____
6: _____	_____	21: _____	_____	36: _____	_____	51: _____	_____
7: _____	_____	22: _____	_____	37: _____	_____	52: _____	_____
8: _____	_____	23: _____	_____	38: _____	_____	53: _____	_____
9: _____	_____	24: _____	_____	39: _____	_____	54: _____	_____
10: _____	_____	25: _____	_____	40: _____	_____	55: _____	_____
11: _____	_____	26: _____	_____	41: _____	_____	56: _____	_____
12: _____	_____	27: _____	_____	42: _____	_____	57: _____	_____
13: _____	_____	28: _____	_____	43: _____	_____	58: _____	_____
14: _____	_____	29: _____	_____	44: _____	_____	59: _____	_____
15: _____	_____	30: _____	_____	45: _____	_____	60: _____	_____

Call Handling Preference

Use this field to specify the call that the agent must receive when calls are in queue.

Valid entry	Usage
skill-level	For the oldest high priority call waiting for an agent with the highest skill level.
greatest-need	For the oldest high priority call waiting for an agent with any skill level.
percent-allocation	For the call waiting for an agent with a skill that deviates the most from the administered allocation. This option is available only with Avaya Business Advocate.

Direct Agent Skill

Use this field to specify the skill number used to handle direct agent calls. You can enter a value from 1 to 99. The upper limit might change depending on the system configuration. For more information, see *Avaya Aura® Communication Manager System Capacities Table*.

Local Call Preference

Use this field to enable the Local Call Preference feature. The system displays this field only if you set the **Multiple Locations** field to y. For calls queued in more than one skill for a multiskilled EAS agent, the system gives preference to the calls whose trunk location number matches the location number of the previously busy agent. This feature is available only with Call Center Release 3.0 and later.

Related links

[Call Center Release](#) on page 739

[Multiple Locations](#) on page 736

PA

Use this field to allocate a percentage of the agent time for each skill that you assign to the agent. The total allocation for all agent skills must be equal to 100%.

RL

Use this field to administer the agent as a reserve agent for a skill or a group of skills. You can assign a reserve level or an interruptible level.

The system displays this field only if you enable the **Business Advocate** feature.

Valid entry	Usage
1 or 2	Use this option to assign a reserve level to the agent.
a	Use this option to assign the auto-in-interrupt interruptible level to the agent.
m	Use this option to assign the manual-in-interrupt interruptible level to the agent.
n	Use this option to assign the notify-interrupt interruptible level to the agent.
blank	Use this option if you do not want to assign a reserve level or an interruptible level to the agent.

Related links

[Interruptible Aux Threshold](#) on page 356

Service Objective

Use this field to enable the Service Objective feature. The agent receives calls according to the ratio of Predicted Wait Time (PWT) or Current Wait Time (CWT) and administration of the Service Objective feature.

The Service Objective feature is available only with Avaya Business Advocate. The system displays this field only when you set the **Call Handling Preference** field to `greatest-need` or `skill-level`.

Related links

[Call Handling Preference](#) on page 47

SL

Use this field to specify a skill level for every skill that you assign to the agent.

If you disable the **Expert Agent Selection-Preference Handling Distribution (EAS-PHD)** feature, you can enter up to two priority levels. If you enable the **Expert Agent Selection-Preference Handling Distribution (EAS-PHD)** feature, you can enter up to 16 priority levels.

SN

Use this field to assign a skill number to the agent.

If you disable the **Expert Agent Selection-Preference Handling Distribution (EAS-PHD)** feature, you can enter up to four skills. If you enable the **Expert Agent Selection-Preference Handling Distribution (EAS-PHD)** feature, you can enter up to 20 or 60 skills depending on the system configuration.

Chapter 8: Alias Station

Use the Alias Station screen to:

- Administer new telephone types that are not supported by the system.
- Map new telephone types to a telephone type that supports Alias Station.
- Assign names to telephone types.

To change the value of the fields on the Alias Station screen, use the following command:

change alias station

Alias Set Type

Use this field to assign a name to the nonsupported telephone type. You can enter up to five alphanumeric characters.

Supported Set Type

Use this field to enter a supported telephone type. For example, DCP, H.323, SIP, or analog telephone types.

Chapter 9: Alphanumeric Dialing Table

Use the Alphanumeric Dialing Table screen to associate Alpha-name to digit strings. Users can make a data call by entering the Alpha-name instead of the digit string.

To change the value of the fields on the Alphanumeric Dialing Table screen, use the following command:

```
change alphanumeric-dial-table
```

Alpha-name

Use this field to assign a unique name to the digit string. The name must begin with an alphabet and can have up to eight alphanumeric characters.

Mapped String

Use this field to enter a string that you must use to generate the dialing string. You can enter up to 24 characters. You can also use the facility access codes in the string.

Valid entry	Usage
0 to 9	To enter a digit.
A to Z, a to z	To enter an alphabet. The system maps the uppercase alphabets to the lowercase alphabets.
(,), /, -, ^, space	To enter a special character.
+	To enable the system to wait for a dial tone.
%	To indicate that rest of the digits are for end-to-end signaling.
,	To enable the system to pause for 1.5 seconds.
#	To enter a DTMF digit pound.
*	To enter a DTMF digit asterisk.

Chapter 10: Announcements/Audio Sources

Use this screen to assign announcements to port locations.

To change the value of the fields on the Announcements/Audio Sources screen, use the following command:

add announcement *n*, where *n* is the extension number.

In Communication Manager 7.1.3, you can configure up to 9000 announcements for a single Avaya Aura Media Server instance.

```
add announcement 3600004                                     Page 1 of 1
```

ANNOUNCEMENTS/AUDIO SOURCES	
Extension: 360-0004	COR: <u>1</u>
Annc Name: _____	TN: <u>1</u>
Annc Type: _____	Queue? <u>y</u>

Annc Name

Use this field to assign a name to the announcement. You can enter up to 27 characters. Announcement names on a single vVAL must be unique. You can enter alphabets, digits, and the following special characters: /, :, *, ?, <, >, \, @, and space.

The system saves the value of this field as the filename of the announcement.

 Note:

Avaya BRI deskphones support only ASCII characters because non-ASCII characters, such as Eurofont and Kanafont, show up incorrectly.

Annc Type

Use this field to specify the type of announcement.

Valid entry	Usage
analog	To play the announcement from an external device for a specific period and then disconnect. The caller hears a click when the device is disconnected.
analog-m	To play the announcement continuously from an external announcement device.
analog-fd	To play the announcement from an external device for a specific period and then disconnect. The caller hears a click when the device is disconnected.
aux-trunk	To play the announcement from an external announcement device with a 4-wire aux interface.
aux-trk-m	To play the announcement from an external announcement device that does not indicate that the playback is active.
ds1-fd	To play the announcement from a DS1 port on the media module. The caller does not hear a click when the device is disconnected. The system sends a disconnect message to the port to stop the playback when the announcement is complete.
ds1-ops	To play the announcement from a DS1 port on the media module. The caller does not hear when the device is disconnected.
ds1-sa	To play the announcement from a DS1 port on the media module. The caller does not hear a click when the device is disconnected. The system sends a disconnect message to the port to stop the playback when the announcement is complete.
integrated	To play the announcement stored internally on a special integrated announcement board. Use this option for general and VDN announcements.
integ-mus	To play the announcement from an integrated music source, or to play music stream.
integ-rep	To play the announcement with integrated repeating.

COR

Use this field to assign a class of restriction (COR) number to the announcement. You can enter a value from 0 to 995.

Extension

This field displays the extension number of the announcement.

 Note:

If you enter a shortened extension from the multilocation dial plan while adding the announcement, then the system does not perform certain administration and validation tasks. Therefore, the system might not display the resultant warnings or submittal denials.

Group/Port/Media Server

Use this field to assign a board location or an audio group number to the announcement.

For an integrated announcement, integ-mus and integ-rep announcements type, the system displays the field name as **Source**.

You must enter the value by using one of the following formats:

- *Gnn*, where *nn* is an audio group number.
- *ggg*v9, where *ggg* is a gateway number.
- *Mxx*, where *xx* is a media server number.

 Note:

To administer DID Intercept announcements in a multilocation system where each location requires a different announcement, you must enter an audio group number instead of a VAL port number.

Live Stream Source

With Live Stream source, you can source music from external sources. **Live Stream Source** is visible only when **Annc-type** is set to *integ-rep* or *integ-mus*.

Live Stream Source allows CM to use an Avaya Aura® Media Server (MS) stream source as an *annc* for music or as a repeating annc. Additional configuration is needed on AMS to define the stream source.

Valid Entry	Usage
y	Music on hold is streamed live from an external source. Live Stream Source can be set to <code>y</code> , only if Queue is set to <code>b</code> and the Source is set to <code>media-server</code> or <code>audio_group</code> having all the members as media-server only. If Live Stream Source is set to <code>y</code> then the Protected field becomes <code>Y</code> and non-editable.
n	Identifies that music on hold is not streamed live from external source. The default value is <code>n</code> .

 Note:

Avaya Aura® Media Server (MS) streams music from an external source, using the concept of a StreamSource. A StreamSource is designed to simulate a music source such as a CD for a natural music-on-hold experience. Live StreamSource provides a radio broadcast effect and enables multiple endpoints to listen same audio stream, that plays continuously in the background.

Protected

Use this field to set the protection mode for an integrated announcement.

Valid entry	Usage
y	A user cannot change or delete the integrated announcement using a telephone session or an FTP session. When you record or transfer an announcement file by using FTP, the system sets the announcement file to read only.
n	A user can change or delete the integrated announcement using a telephone session or an FTP session.

Queue

Valid Entry	Usage
y	Queues calls for the announcement if the announcement type is integrated, integ-rep or aux-trunk. The caller is always connected to the beginning of the announcement. ACD, vectoring delay announcements and call centers should always use this option. This is the default.
n	No queue nor barge-in. The caller is always connected to the beginning of the announcement. The announcement does not play if a port is unavailable.

Table continues...

Valid Entry	Usage
b	Sets up barge-in if the announcement type is integrated, integ-rep or aux-trunk. When the announcement type is integ-mus, this field defaults to b. Callers are connected to the announcement at any time while it is playing.  Note: The same non-barge-in announcement can be played through more than one port (or all ports) of an integrated board. The initial request to play an announcement selects an available port on the board on which the announcement resides. If there are additional requests to play the announcement while it is playing on another port(s), another port is selected. If all ports are busy, new requests to play announcements go to the integrated announcement system queue. Otherwise, the request to play is denied, and processing continues without the caller hearing the announcement. When a port becomes available, all queued calls (up to the platform “calls connected” limit) are connected at the same time to hear the announcement play from the beginning. A barge-in announcement starts playing when first requested and continues playing through a port, repeating until there are no more requests. Call processing simultaneously connects calls to the playing barge-in announcement. Each call remains connected until the requesting feature operation removes the call (for example, wait step times out). Barge-in type announcements never select another port to play the same announcement once it is playing on a specific port.

Queue Length

Use this field to specify the number of calls that can queue for the announcement. The maximum allowed entry depends on the system configuration.

Rate

The recording rate speed in 1000 bits/seconds for TN750 or ISSPA integrated announcements. A different recording speed can be used for each integrated announcement. With VAL type sources, the default is 64 and cannot be changed.

Valid Entry	Usage
16	16 kbps • 8 minutes and 32 seconds of announcement time • 240 minutes of storage time for the ISSPA This rate does not provide a high-quality recording. Do not use this rate for customer announcements, but it is adequate for VDN of Origin announcements.
32	32 kbps • 120 minutes of storage time for the ISSPA
64	64 kbps • 2 minutes and 8 seconds of announcement time • 60 minutes of storage time for the ISSPA This is the default for vVAL

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Chapter 11: ARS Toll Table

Use this screen to assign ARS toll tables used by Subnet Trunking. This screen specifies whether calls to the local central office codes listed on the table are toll calls.

To change the value of the fields on the ARS Toll Table screen, use the following command:

change ars toll

Chapter 12: Attendant Console

Use this screen to assign an attendant console to the system.

To add an attendant, use the following command:

add attendant *n*, where *n* is the attendant console number.

Attendant console: page 1

Attendant Console x

Number assigned to the attendant console. The attendant console is the main answering position for an organization.

Auto Answer

Use this field to administer the setting of the **auto answer** button on the attendant console.

Valid entry	Usage
all	Enables automatic answering for all incoming calls.
acd	Enables automatic answering only for ACD and direct agent calls.
none	Disables automatic answering.

Console Type

Use this field to specify the type of attendant console.

You can administer only one night-only or day/night type console in the system unless the Tenant Partitioning feature is enabled.

Night Service can be activated from the principal attendant console or the station on which the **nite-serv** button is administered.

Valid entry	Usage
principal	To enable the attendant console to activate Night Service.
day-only	To enable the attendant console to handle only day service calls.

Table continues...

Valid entry	Usage
night-only	To enable the attendant console to handle only night service calls.
day/night	To enable the attendant console to handle day or night service calls.

COR

Use this field to assign a class of restriction (COR) number to the attendant console. You must enter a value from 0 to 995.

COS

Use this field to assign a class of service (COS) number to the attendant console. You must enter a value from 0 to 15.

Data Module

Use this field to enable the connection between the attendant console and the data terminal.

Disp Client Redir

Use this field to specify whether the attendant console must display the redirection information for a call originating from the station with the Client Room Class of Service feature.

Note:

You must set the **Disp Client Redir** field to *y* for:

- The stations with the audix type.
- The stations with the audix voice power ports or the ports that require the display information.

The system displays this field only if the Hospitality feature is enabled.

Related links

[Hospitality \(Basic\)](#) on page 733

[Hospitality \(G3V3 Enhancements\)](#) on page 733

Display Language

Use this field to specify the language in which the system must display the console messages. You must enter one of the following values:

- english
- french
- italian
- spanish

- user-defined
- unicode

 Note:

The unicode option is available only for unicode-supported telephones, such as 4610SW, 4620SW, 4621SW, 4622SW, 2420J, Sage, Spice, Spark, 9600-series telephones, and Avaya J100 Series IP Phones.

Related links

[Display Character Set](#) on page 722

Extension

Use this field to assign an extension to the attendant console.

Users can call the attendant console by dialing this extension. If an attendant console does not have an extension, the attendant console can only be addressed as a member of the attendant group.

You must not leave this field blank for the attendant console with a data module.

Group

Use this field to assign an attendant group number to the attendant console.

H.320 Conversion

Use this field to enable the conversion of H.320-compliant calls to voice-only calls for the attendant console.

 Note:

The system can handle only a limited number of conversion calls. Therefore, the number of attendant consoles with H.320 conversion must be limited.

Name

Use this field to assign a name to the attendant console. You can enter up to 27 alphanumeric characters.

 Note:

Avaya BRI deskphones support only ASCII characters because non-ASCII characters, such as Eurofont and Kanafont, show up incorrectly.

Port

Use this field to assign a port number to the attendant console. For reliability, the attendant consoles must not be assigned to ports on the same digital line media module.

You must enter one of the following values:

- A port number.
- **ip** for a soft attendant console.
- **X** if no hardware is associated with the port assignment.

You can assign the port number using one of the following formats:

- *gateway number : module number : circuit number*

Valid entry	Usage
001 to 250	The first three characters represent the gateway number.
V1 to V9	The fourth and fifth characters represent the module number.
01 to 31	The sixth and seventh characters represent the circuit number.

Related links

[IP Attendant Consoles](#) on page 733

[Extension](#) on page 61

Security Code

The station security code required by the SoftConsole IP attendant.

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Type

Valid Entry	Usage
console	The extension is used with any type of console.
302	302B/C/D or SoftConsole IP attendant

Direct Trunk Group Select Button Assignments (Trunk Access Codes)

Use this field to assign the local and remote trunk access codes (TACs) to the attendant console. A local TAC refers to a trunk group or a Loudspeaker Paging zone on the local server. A remote TAC refers to a trunk group on the remote server.

The local TAC can have up to four digits and the remote TAC can have up to three digits. You can use the asterisk (*) and the pound sign (#) as the first digit.

Note:

The local TAC between the local and remote servers must be a DCS trunk. If not, Communication Manager cannot monitor the remote trunk.

Hundreds Select Button Assignments

Use this field to assign a hundreds group to a **Hundreds Select** button on an attendant console. You can enter up to 11 digits.

Use **1** to **8** fields for 24A consoles, and use **1** to **8** fields for 26A consoles.

Attendant console: page 2 Softconsole IP Attendant

Always Use

Use this field to enable a soft attendant console to register, regardless of the emergency call handling settings on the console. If the user dials **911**, the system uses the administered value in the **Emergency Location Ext** field.

The **Always Use** field is inapplicable to SCCAN wireless consoles and the consoles administered as the H.323 type.

Related links

[Emergency Location Ext](#) on page 672

Direct IP-IP Audio Connections

Use this field to enable direct audio connections between IP endpoints. Direct audio connections save bandwidth resources and improve the sound quality of voice over IP transmissions.

Note:

This field has no impact on a SIP station (OPS) on shuffling. For SIP stations, this field is controlled by the "Direct IP-IP Audio Connections" field in the Signaling group form.

Emergency Location Ext

Use this field to specify the emergency location extension for the soft attendant console. You can enter up to 16 digits. This extension is used as a starting point for identifying the location when an emergency call is made.

The system uses:

- The CAMA Numbering format to modify the extension before sending the extension over the CAMA trunk.
- The Numbering-Public/Unknown format to modify the extension before sending the extension over the ISDN trunk.

Remote Softphone Emergency Calls

Use this field to specify how Communication Manager must handle emergency calls from an IP telephone. The system displays this field only when **IP Softphone** is enabled for the system.

Valid entry	Usage
as-on-local	If the administrator populates the IP Address Mapping screen with emergency numbers, the value as-on-local functions as follows: • If the Emergency Location Extension field in the Station screen is the same as the Emergency Location Extension field in the IP Address Mapping screen, the feature sends the console's own extension number to the Public Safety Answering Point (PSAP). • If the Emergency Location Extension field is different from the Emergency Location Extension field in the IP Address Mapping screen, the value as-on-local sends the extension in the IP Address Mapping screen to the Public Safety Answering Point (PSAP).
block	Prevents the completion of emergency calls. Used for users who are mobile but always have a circuit-switched telephone nearby, and for those who are farther away from the media server or switch than an adjacent area code served by the same 911 Tandem office. When users attempt to dial an emergency call from an IP Telephone and the call is blocked, they can dial 911 from a nearby circuit-switched telephone instead.
cesid	Communication Manager can send the CESID information supplied by the IP Softphone to the PSAP. The end user enters the emergency information into the IP Softphone. Used for IP Softphones with road warrior service that are near enough to the Avaya S8XXX server that an emergency call routed over the trunk reaches the PSAP that covers the server or switch. If the Avaya S8XXX server uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the IP Softphone. If the Avaya S8XXX server uses CAMA trunks for emergency calls, the end user enters a specific digit string for each IP Softphone location, based on advice from the local emergency response personnel.
option	Users can select the option (extension, block, or cesid) that the user selected during registration and the IP Softphone reported. Used for extensions that can be swapped back and forth between IP Softphones and a telephone with a fixed location. Users choose between block and cesid on the softphone. A DCP or IP telephone in the office automatically selects extension.

Related links

[IP Softphone](#) on page 649

Service Link Mode

Use this field to specify the duration of a service link connection. The service link is the combined hardware and software multimedia connection between an H.320 Desktop Video Conferencing (DVC) system and Communication Manager.

The service link is established when a user receives or makes a call during a multimedia, IP softphone, or IP telephone session.

Valid entry	Usage
as-needed	For multimedia, IP softphone, and IP telephone users. The service link remains connected for 10 seconds after the user disconnects a call so that the user can immediately make or receive another call. After 10 seconds, the link is disconnected, and a new link must be established to make or receive a call.
permanent	For call center agents who are constantly making or receiving calls during the multimedia, IP softphone, or IP telephone session. The service link remains connected for the entire duration of the session.

Attendant console: page 2 VIS feature options

VIS FEATURE OPTIONS

These fields administer Visually Impaired Services options.

Auto Start

Use this field to allow or deny permission to an attendant to start a call by pressing any key on the keypad without the need to first press the **Start** button.

Echo Digits Dialed

Enables and disables voiced confirmation of dialed digits.

Attendant console: page 3

Displays if the attendant console is to be connected to a data terminal using a 7400B or 8400 data module.

Related links

[Data Module](#) on page 60

ATTENDANT DATA MODULE

Bcc

Determines compatibility when non-ISDN facilities are connected to ISDN facilities (ISDN Interworking feature).

COR

Valid Entry	Usage
0 to 995	The Class Of Restriction (COR) number for the data module.

COS

Valid Entry	Usage
0 to 15	The (COS) number used to designate allowed features.

Data Extension

Use this field to assign an extension to the data module. You can enter up to five digits. This value must conform to the dial plan.

Name

The name assigned to the data module extension number.

Related links

[Ext](#) on page 67

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

ABBREVIATED DIALING

List1

Valid Entry	Usage
s	System
g	Group — A group number is also required
p	Personal — A personal list number also is required
e	Enhanced

SPECIAL DIALING OPTION

Use this field to specify the destination of a call when the data module originates the call. Use this field to specify the destination of a call that starts from the data module.

Valid entry	Usage
hot-line	Single-line telephone users can automatically make a call to an extension, a telephone number, or a feature access code.
default	The system dials an Abbreviated Dialing number when the user disconnects.

HOT LINE DESTINATION — Abbreviated Dialing Dial Code

Abbreviated Dialing Dial Code is the Abbreviated Dialing number dialed when the user goes off-hook on a Data Hotline call.

Using Hotline Service, IP telephone users can place a call to a preassigned destination by going offhook. The preassigned destination can be an extension, a telephone number, or a feature access code. The Hotline Service feature does not work for SIP endpoints.

The Hotline Service destination number is stored in Abbreviated Dialing List.

A Hotline Service destination can be a Direct Department Calling (DDC), a Uniform Call Distribution (UCD), a Terminating Extension Group (TEG) extension, or any individual extension within a group. Also, any extension within a DDC group, UDC group, or TEG can have Hot Line Service assigned.

Use Hotline Service when you require very fast service and when you use a telephone only for gaining access to a certain facility. You can use Loudspeaker Paging Access with Hotline Service to provide automatic access to paging equipment.

This field is available only for a default or hotline special dialing option.

Related links

[SPECIAL DIALING OPTION](#) on page 66

DEFAULT DIALING Abbreviated Dialing Dial Code

The AD number dialed when the user goes off-hook and enters a carriage return following the “DIAL” prompt. The data call originator also can perform data terminal dialing by specifying a dial string that might or might not contain alphanumeric names. Available only for a default special dialing option.

Related links

[SPECIAL DIALING OPTION](#) on page 66

ASSIGNED MEMBER

Ext

The extension number of a previously-administered user who has an associated **Data Extension** button and who shares the use of the module.

Name

The name assigned to the data module extension number.

Related links

[Ext](#) on page 67

Attendant console: page 3 feature button assignments

FEATURE BUTTON ASSIGNMENTS

Administers the feature buttons assigned to the attendant console. The split and forced release buttons are administered in a fixed location. The **hold**, **night-serv**, and **pos-busy** buttons have default locations. The following table provides descriptions of feature buttons that are unique to the attendant console.

Audible Tones On/Off

Valid Entry	Usage
cw-ringoff	Call waiting ringer off; turns on/off the audible tone for call waiting on attendant console (1 per console).
in-ringoff	Incoming call ringer off; turns on/off the audible tone for incoming call ringer (1 per console).
re-ringoff	Timed reminder ringer off; turns on/off the audible tone for timer reminder ringer (1 per console).

Attendant Control of Trunk Group Access

Valid Entry	Usage
act-tr-grp	Activate trunk group access. The attendant can control a trunk group. All calls going to the trunks are routed to the attendant (one per console).
deact-tr-g	Deactivate trunk group access. The attendant can release control of a trunk group (one per console).
class-rstr	Display Class of Restriction. Used to display the COR associated with a call (one per console).
em-acc-att	Emergency Access to the Attendant. The associated status lamp is flashed when there are one or more calls on the emergency attendant queue (one per console).
hold	Hold. When the Hold button is pressed while the attendant is active on a loop, the party on the loop is put on hold and the call type button associated with the loop is lit (one per console).
pos-busy	Position Busy. When this button is pushed, the attendant is put into position busy mode, the "Pos Avail" light is turned off, and the light associated with the pos-busy button is lit. Pushing the pos-busy button a second time takes the console out of "position busy" mode, turns on the "Pos Avail" light and turns off the light associated with the pos-busy button. If the pos-busy button is administered on a 2-LED button, the top LED flashes when the last attendant goes into "Position Busy" mode. Otherwise, if the button has only one LED, the single LED associated with the pos-busy button flashes (one per console).
serial-cal	Serial Call. The attendant-extended calls can return to the same attendant if the trunk remains off-hook (one per console).

Table continues...

Valid Entry	Usage
override	Attendant Override. The attendant can override diversion features such as, Call Forwarding, Call Coverage, and so on (one per console).
intrusion	Call Offer. The attendant can extend a call when the called party is active on another call (one per console).
dont-split	Don't Split. The attendant cannot split away a call when dialing (one per console).
vis	Visually Impaired Attendant Service (vis). Activates visually impaired service for the attendant. When this service is activated, the attendant can listen to console status or messages by pressing buttons that have been translated as follows: • “con-stat” repeats the console status. • “display” calls out display contents. • “dtgs-stat” calls out the DTGS status. • “last-mess” repeats the last message. • “last-op” calls out the last operation.

Trunk Group Select

Up to 12 **DTGS** buttons can be administered. The status lamp associated with the feature button is used to monitor the busy/idle status of the trunk. Trunk groups administered on these buttons cannot be controlled using **Attendant Control of Trunk Group Select** buttons.

Valid Entry	Usage
local-tgs	Local trunk group select. The attendant can access trunk groups on the local server running Communication Manager.
remote-tgs	Remote trunk group select. The attendant can access trunk groups on a remote server running Communication Manager.

Other

Valid Entry	Usage
alt-frl	Alternate facility restriction level. The attendant can activate or deactivate the AFRL feature. When activated, the originating device (lines or trunks) can use an alternate set of the facility restriction levels to originate a call (one per console).
hundrd-sel	Hundreds group select. Additional administered hundreds group select feature buttons. When a feature button is administered as “hundrd-sel”, the system displays a subfield that must be administered with a 1 to 3 digit hundreds group plus prefix, if needed. Administered hundrd-sel feature buttons operate in the same manner as fixed HGS buttons. The total number of hundreds group select buttons (fixed and administered) allowed on a console is 20. Therefore, if all 20 fixed HGS buttons have been administered, no additional hundrd-sel feature buttons can be administered. If 12 HGS buttons are assigned, reassign the night, pos-busy, and hold buttons to locations 20, 21, and 3, respectively. The HGS buttons should then be assigned to the right-most three columns, as required.

Table continues...

Valid Entry	Usage
group-disp	Group Display. The attendant can see a display of extensions currently being tracked on the DXS module.
group-sel	Group Select. The attendant can select a specific group of hundreds by dialing the first 2 or 3 digits of the hundreds group.
occ-rooms	Occupied rooms. The attendant can see which rooms are occupied.
maid-stat	Maid status. The attendant can see which rooms are in one of six specified states.
vu-display	VuStats (vu-display). Users with display telephones and attendants can turn on the VuStats display. The limit to the number of VuStats feature buttons depends on how many feature buttons are available on the attendant console. You can set up a separate VuStats display format for each feature button. Therefore, agents can change the type of measurements on their display by selecting a different VuStats feature button.

Attendant console: page 4 display module button assignments

Display Module Button Assignments

Display-type buttons obtain display functions on the associated alphanumeric display. Also, several feature buttons can be administered so that their associated status lamps can be used to provide visual indications of the associated feature or function. In some cases, the button itself is inoperative. These buttons are noted as [status lamp]. If a **Call Cover Msg Rt** (cov-msg-rt) button is assigned, a Leave Word Calling Delete Msg (delete-msg) button and a **Next** (next) button must also be assigned.

Chapter 13: Audio Group

Use this screen to assign an audio group to the system. The audio group is a collection of recorded audio sources that have been placed in the group to facilitate the selection.

To add an audio group, use the following command:

add audio-group *n*, where *n* is the audio group number.

add audio-group 1

Page 1 of 5

AUDIO GROUP 1

Group Name: _____

AUDIO SOURCE LOCATION

1: _____	16: _____	31: _____	46: _____	61: _____	76: _____
2: _____	17: _____	32: _____	47: _____	62: _____	77: _____
3: _____	18: _____	33: _____	48: _____	63: _____	78: _____
4: _____	19: _____	34: _____	49: _____	64: _____	79: _____
5: _____	20: _____	35: _____	50: _____	65: _____	80: _____
6: _____	21: _____	36: _____	51: _____	66: _____	81: _____
7: _____	22: _____	37: _____	52: _____	67: _____	82: _____
8: _____	23: _____	38: _____	53: _____	68: _____	83: _____
9: _____	24: _____	39: _____	54: _____	69: _____	84: _____
10: _____	25: _____	40: _____	55: _____	70: _____	85: _____
11: _____	26: _____	41: _____	56: _____	71: _____	86: _____
12: _____	27: _____	42: _____	57: _____	72: _____	87: _____
13: _____	28: _____	43: _____	58: _____	73: _____	88: _____
14: _____	29: _____	44: _____	59: _____	74: _____	89: _____
15: _____	30: _____	45: _____	60: _____	75: _____	90: _____

Audio Source Location

Use this field to assign a board location to the audio group. You must enter the value using one of the following formats:

- *ggg*∇*x*, where *ggg* is a gateway number and ∇*x* is the module number.
- *M*_{*x*}, where *x* is the media server number.

Group Name

Use this field to assign an alphanumeric name to the audio group.



Note:

Avaya BRI deskphones support only ASCII characters because non-ASCII characters, such as Eurofont and Kanafont, show up incorrectly.

Chapter 14: AUDIX-MSA Node Names

Example command: `change node-names audix`

AUDIX Names

The name of the voice messaging node consisting of a one- to seven-character string. Used as a label for the associated IP address. The node names must be unique on each server running Communication Manager.

IP Address

The IP address associated with the node name.

MSA Names

Identifies the name of the MSA node. Accepts a one- to seven-character string. The MSA names must be unique on each server running Communication Manager.

Chapter 15: Authorization Code — COR Mapping

Use this screen to administer an authorization code and the class of restriction (COR) number associated with the authorization code.

Security alert:

To maximize security, you must:

- Administer authorization codes to the acceptable maximum length.
- Create random authorization codes.
- Change authorization codes at least quarterly.
- Deactivate authorization codes immediately if a user leaves the company or changes assignments.
- Assign authorization codes with the minimum level of calling permission.

To change the value of the fields on the Authorization Code — COR Mapping screen, use the following command:

`change authorization-code n`, where *n* is the authorization code.

AC

Use this field to administer an authorization code (AC). The AC can be any combination of numbers and can consist of 4 to 13 digits. To enhance the system security, you must administer the AC of 13 random digits.

Related links

[Authorization Code Length](#) on page 252

COR

Use this field to assign a class of restriction (COR) number to the authorization code. You can enter a value from 0 to 995.

Number of Codes Administered

The number of authorization codes already administered. There is a maximum number of authorization codes depending on system configuration.

Valid entry	Usage
0 to 9, *, #	To enter the digit.
~p	To enable the system to pause.
~w or ~W	To enable the system to wait.
~m	To enable the system to mark.
~s	To enable the system to suppress.

Location Name

Use this field to assign a name to the location where the Best Services Routing feature is enabled. You can enter up to 15 alphanumeric characters.

Lock

Use this field to provide additional security by preventing the system from sending any Best Service Routing (BSR) information to Call Management System.

TG

Valid entry	Usage
y	Does not send the BSR information to Call Management System.
n	Sends the BSR information to Call Management System.

Maximum Suppression Time

Use this field to prevent callers from connecting to a VDN during the specified period after receiving a busy signal.

Valid entry	Usage
0 to 60	The maximum poll suppression time in seconds.

Net Redir

Use this field to enable Network Call Redirection.

Num

Use this field to specify a location number. You can enter a value from 1 to 255.

Number

This field displays the current BSR plan number. The system displays a value from 1 to 255.

Status Poll VDN

Use this field to specify the dial access code that the user must use to access the Status Poll vector directory number (VDN) at the remote location. You can enter up to 16 alphanumeric characters.

Valid entry	Usage
0 to 9, *, #	To enter the digit.
~p	To enable the system to pause.
~w or ~W	To enable the system to wait.
~m	To enable the system to mark.
~s	To enable the system to suppress.

Switch Node

Valid Entry	Usage
1 to 32767 blank	The Network Node ID for each switch if the Universal Call ID is being used. This is optional.

Chapter 17: Bulletin board

Use this screen to provide the messages that the system must display when a user logs in. The first 10 lines are reserved for high-priority messages, which only an init or an inads user can enter. The system also displays the date on which the message was posted.

To change the value of the fields on the Bulletin board screen, run the following command:

change bulletin-board

Date

The date the bulletin board information was entered or last changed.

Text lines

Used for high priority messages on the bulletin board. Anyone with an init or inads login can enter high-priority information to trigger the high-priority message at login time. Additional lines can be used by anyone with access.

Chapter 18: Button Type Customization Restriction

Use this screen to:

- Prevent users from accessing the customized button labels.
- Manage system allocation of the customized button labels.

To change the value of the fields on the Button Type Customization Restriction screen, run the following command:

```
change button restriction
```

Restrict Customization Of Button Types

Valid Entry	Usage
y	Restricts the use of customized feature button labels. This is the default.
n	Users can customize labels for all buttons on their telephones without any restrictions.

Restrict Customization Of Labels For the Following Button Types

The button type restricted from being customized.

 Note:

The **abr-spchar** and the **abrv-dial** button types require a special associated character.

- abrv-dial

An abbreviated dialing list type: e = enhanced, g = group, p = personal, and s = system

- abr-spchar

A ~-character in an abbreviated dialing digit string. ~, ~m, ~p, ~s, ~w, and ~W. For more information on digit strings, see Dial Code.

Available only when **Restrict Customization of Button Types** is enabled.

Related links

[Restrict Customization Of Button Types](#) on page 80

[DIAL CODE](#) on page 33

Chapter 19: Call Type Digit Analysis Table

Use this screen to specify how the system must modify telephone numbers when a user makes a call to an internal contact using the call log or a corporate directory.

To change the value of the fields on the Call Type Digit Analysis Table screen, run the following command:

```
change calltype analysis
```

Delete

Use this field to specify the number of digits that Communication Manager must delete from the starting of the dialed string. You can enter a value from 0 to 28.

 Note:

The value in the **Delete** field must not exceed the value in the **Min** field.

Dialed String length (Min, Max)

Communication Manager compares digit strings of this length to the original digit string, looking for a match to complete analysis and routing.

Dialed String Match

Use this field to specify the string that Communication Manager must use to match the dialed number for analysis and routing. The string can be up to 18 characters long and can consist of:

- Digits from 0 to 9
- The plus sign (+)
- The wildcard characters, x and X

 Note:

The plus sign (+), if used, must be the first digit in the string.

Insert

Communication Manager inserts these digits into the left-hand side of the original digit string to complete analysis and routing.

Location

Valid Entry	Usage
<i>Numeric value</i> all	Telephones dialing from this location use the entries in the Call Type Digit Analysis Table. If there are matching entries in the telephone's location, those entries are used. If there are no matching entries in the telephone's location, the Communication Manager tries the entries in location all.

Type

Use this field to specify the call type associated with the dialed string.

Valid entry	Usage
aar	For Automatic Alternate Routing (AAR) calls.
ars	For Automatic Route Selection (ARS) calls.
ext	For calls with extensions in the dial plan analysis table.
udp	For calls with extensions in the uniform dial plan table.

Chapter 20: Call Vector

Use this screen to run a series of commands to specify how the system must handle calls directed to a vector directory number (VDN).

For example, to change the value of the fields on the screen, run the following command:

change vector *n*,

where *n* is the VDN.

change vector 1

Page 1 of 6

CALL VECTOR

Number: 1

Name:

Multimedia? n

Attendant Vectoring? n

Meet-me Conf? n

Lock? n

Basic? y

EAS? y

G3V4 Enhanced? y

ANI/II-Digits? y

ASAI Routing? y

Prompting? y

LAI? y

G3V4 Adv Route? y

CINFO? y

BSR? y

Holidays? y

Variables? y

3.0 Enhanced? y

01

02

03

04

05

06

07

08

09

10

11

12

Press 'Esc f 6' for Vector Editing

01 through XX

Use this field to enter the vector commands. The maximum number of commands that you can enter depends on the system configuration.

Valid entry	Usage
adjunct routing	To send a message to an adjunct for routing instructions.
announcement	To provide an announcement to the caller.
busy	To provide a busy signal to the caller and to end the vector processing.
check	To check the status of a split or skill.
collect	To collect the Caller Information Forwarding (CINFO) digits. The user can enter up to 16 digits.
consider	To define the resource that the system selects as part of a Best Service Routing (BSR) consider series and to get the data that BSR uses to compare resources.
converse-on	To direct a call to a split or skill and activate the voice response script in a Voice Response Unit (VRU).
disconnect	To disconnect a call. The system plays an announcement before the call is disconnected.
goto	To move the vector processing to an earlier or later step in the vector.
messaging	To enable the user to leave a message for the specified extension or the active VDN extension.
queue-to	To queue a call to a split or skill and assign a priority level to the call.
reply-best	To return the data to the primary vector on the server. Use this option only in status poll vectors and BSR applications.
return	To return the vector processing to the step following the goto command after a subroutine call is processed.
route-to	To route a call to the destination specified in the administered digit string.  Note: Communication Manager prepends a plus sign (+) to the called number when Network Call Redirection is enabled.
set	To perform arithmetic and string operations and assign value to a vector variable.
stop	To stop the processing of subsequent vector steps.
wait-time	To delay the vector processing of the next step.

ANI/II-Digits

Indicates whether the ANI and II-Digits vector routing commands can be used or not. ANI/II-Digits Routing is a **G3V4 Enhanced** feature.

Related links

[G3V4 Enhanced](#) on page 87

ASAI Routing

Indicates whether or not the **CalVisor Adjunct/Switch Applications Interface (ASAI) Routing** option is active or inactive.

Related links

[Attendant Vectoring](#) on page 730

Attendant Vectoring

Indicates whether or not Attendant Vectoring is selected on this VDN. Attendant Vectoring does not support Call Center features.

Related links

[Attendant Vectoring](#) on page 730

Basic

Indicates whether Vectoring (Basic) is enabled for system.

Related links

[Vectoring \(Basic\)](#) on page 741

BSR

Indicates if the Best Service Routing option is enabled on the system. BSR commands and command elements can be used in vectors only if this option is enabled.

Related links

[Vectoring \(Best Service Routing\)](#) on page 741

CINFO

Indicates whether Caller Information Forwarding (CINFO) Routing is enabled on the system. A call with CINFO routes based on the digits supplied by the network in an ISDN-PRI message.

Related links

[Vectoring \(CINFO\)](#) on page 741

EAS

Indicates if Expert Agent Selection (EAS) is enabled on the system.

 Note:

When EAS is enabled, the terminology changes from “Split” to “Skill” for help messages, error messages, and vector commands. For example, check backup split becomes check backup skill.

Related links

[Expert Agent Selection \(EAS\)](#) on page 740

G3V4 Adv Route

Indicates whether G3V4 Advanced Vector Routing commands are allowed.

Related links

[Vectoring \(G3V4 Advanced Routing\)](#) on page 742

G3V4 Enhanced

Indicates whether G3V4 Enhanced Vector Routing commands and features are allowed.

Related links

[Vectoring \(G3V4 Enhanced\)](#) on page 742

Holidays

Indicates whether Holiday Vectoring features are allowed.

Related links

[Vectoring \(Holidays\)](#) on page 742

LAI

Indicates if Look-Ahead Interflow is enabled.

Related links

[Lookahead Interflow \(LAI\)](#) on page 740

Lock

Controls access to the vector from the Call Management System.

 Security alert:

Always lock vectors that contain secure information (for example, access codes).

Valid Entry	Usage
y	This vector is inaccessible from the CMS. Locked vectors can only appear and be administered through the SAT or a terminal emulator. If Meet-me Conference is enabled, Lock must also be enabled.
n	Gives CMS users the ability to administer this vector.

Meet-me Conf

Indicates if the Meet-me Conference feature is enabled. If enabled, designates the VDN as a Meet-me Conference VDN. Attendant Vectoring and Meet-me Conference cannot be enabled at the same time.

Related links

[Enhanced Conferencing](#) on page 732

Multimedia

Valid Entry	Usage
y	Multimedia calls are set to be answered and billed at the start of vector processing. Multimedia Call Handling must be enabled.
n	The vector stops receiving multimedia calls. Multimedia Call Handling is disabled. This is the default.

Related links

[Multimedia Call Handling \(Basic\)](#) on page 735

[Multimedia Call Handling \(Enhanced\)](#) on page 736

Name

Use this field to assign a name to the vector. You can enter up to 27 alphanumeric characters.

The system displays this field only if you enable the Network Call Redirection feature.

Related links

[ISDN/SIP Network Call Redirection](#) on page 735

[Display Character Set](#) on page 722

Number

The vector number.

Prompting

Indicates whether Vectoring (Prompting) is enabled for the system.

Related links

[Vectoring \(Prompting\)](#) on page 742

Chapter 21: Calling Party Number Conversion for Tandem Calls

Use this screen to administer calling party number formats for tandem calls. Tandem calls are not originated locally on the switch.

For tandem calls to the public network, the system might not provide the correct calling party information. To rectify this problem, Communication Manager changes the calling party number. Communication Manager compares the incoming calling party number with the calling party digit lengths, calling party prefixes, and trunk groups. When a match is found, Communication Manager creates the calling party number by deleting the digits specified in the **Delete** field, and then inserting the digits specified in the **Insert** field or the values of Numbering Plan Identifier (NPI) and Type of Number (TON) for Calling Party Number. Communication Manager uses the numbering format specified in the Number Format field.

 Note:

Communication Manager does not update Calling Party Number in the NCA-TSC SETUP message.

To map numbers for ISDN-SIP interworking, Communication Manager uses the values of the fields on the Calling Party Number Conversion for Tandem Calls screen or sends the call to Session Manager. Session Manager uses the number adaptation module for the mapping.

 Note:

During ISDN to SIP mapping, the number adaptation module does not have access to NPI and TON values in the incoming ISDN number. During SIP to ISDN mapping, Communication Manager uses the leading plus (+) sign to process an incoming SIP number as an international number and insert the plus (+) sign in the resulting number. To process an incoming SIP number, Communication Manager uses the values of the fields on the Calling Party Number Conversion for Tandem Calls screen.

The following rules apply to an incoming calling party number with a leading plus (+) sign:

- For an incoming SIP calling party number with a leading plus (+) sign, Communication Manager includes the plus (+) sign when searching for a match.
- For an incoming SIP calling party number with a leading plus (+) sign, if a match is not found with the plus (+) sign, Communication Manager removes the plus (+) sign.
- For an incoming ISDN calling party number reconstructed as an international number, Communication Manager searches for a match. If a match is not found, Communication Manager adds a leading plus (+) sign.

Use one of the following ways to modify calling party numbers for a SIP endpoint:

- * Note:**

If you choose the System Parameters and Location Parameters forms, you can use the Calling Party Number Conversion for Tandem Calls form for other purposes. To do so, use the **Outgoing Trunk Group (s)** column to ensure that no entry on that form affects the trunk groups used for SIP stations.

change tandem-calling-party-num

Related links

Insert on page 92

March 2026

[Modify Tandem Calling Number](#) on page 454

CPN Prefix

Valid Entry	Usage
0 to 9, +	Any combination of digits 0 to 9, up to 16 digits to indicate the calling party prefix. The '+' sign is also valid.
any	Any incoming calling number matches the entry.

Delete

Valid Entry	Usage
1 to 15	The number of digits deleted when modifying the tandem calling party number.
all	All digits are deleted.
blank	No digits are deleted from the received calling party number. This is the default value.

Insert

Valid Entry	Usage
0 to 9, +	Any combination of digits 0 to 9, up to 16 digits. The '+' sign is also valid.
natl-prefix	Communication Manager adds the Long distance access code number as the prefix to the calling-party number. The Long distance access code field is on page 1 of the Location parameter screen. If the Long distance access code field is blank, Communication Manager uses the prefix number in the National CPN Prefix field on the Feature related system parameter screen.
intl-prefix	Communication Manager adds the International access code number as the prefix to the calling-party number. The International access code field is on page 1 of the Location parameter screen. If the International access code field is blank, Communication Manager uses the prefix number in the International CPN Prefix field on the Feature related system parameter screen.
blank	The received calling party number is not prefixed with any digits. This is the default value.

Incoming Number Format

Use this field to specify the numbering format of the calling party number.

*** Note:**

Unlike ISDN calling party numbers, SIP calling party numbers do not include a numbering format. The only descriptor is the optional leading plus sign (+). Therefore, to match the calling number in an incoming SIP trunk call, you must do one of the following:

- To match a calling number that does not start with a plus sign (+), you must leave the **Incoming Number Format** field blank.
- To match a calling number that starts with a plus sign (+), you can leave the **Incoming Number Format** field blank, or you can set the field to `intl-pub`.

Valid Entry	Usage
intl-pub, lev0-pvt, lev1-pvt, lev2-pvt, locl-pub, natl-pub, pub-unk, unk-unk	The numbering format to use in modifying the tandem calling party number.
blank	All numbering formats match this entry.

Len

Valid entry	Usage
1 to 16	A number between 1 and 16 that indicates the length of calling party number.
any	A calling party number of any length. * Note: If after insertion and deletion the resulting calling number is longer than 21 digits, the system will not perform the conversion.

Outgoing Number Format

Valid Entry	Usage
intl-pub, lev0-pvt, lev1-pvt, lev2-pvt, locl-pub, natl-pub, pub-unk, unk-unk	The numbering format to use in modifying the tandem calling party number.
blank	All numbering formats match this entry.

Outgoing Trunk Group(s)

Valid entry	Usage
Trunk group number	Enter an ISDN or a SIP outgoing trunk group number.
blank	If you leave this field blank, any ISDN and SIP trunk group matches this entry, if it has the Modify Tandem Calling Number field enabled. By default, this field is blank.

Chapter 22: CAMA Numbering Format

Administers the Centralized Automatic Message Accounting (CAMA) trunks and provides Caller's Emergency Service Identification (CESID) information to the local community's Enhanced 911 system through the local tandem office.

Provides the CESID format by extension number or number blocks. Multiple CESID formats can be sent over multiple CAMA trunk groups allowing for mixed station numbering plans and some limited conversion from non-DID to DID numbers typically required by the Private Switch/Automatic Location Interface (PS/ALI) database.

Example command: `change cama-numbering`

CESID

The number used to identify the calling terminal within an emergency service system. This field can represent a prefix to an extension or the entire CESID. Accepts up to 16 digits. The maximum number of allowed digits depends on the equipment.

Ext Code

The leading digits or all of the digits in the extension for the specified CESID. If the extension length is greater than the number of digits in the extension code, the extension code is interpreted as a block of digits.

Example: If the extension length is 4 and the extension code is 11, the CESID serves extensions 1100 through 1199.

Ext Len

The number of digits in the extension. Accepts up to 16 digits.

System CESID Default

The number sent over the CAMA trunk if the Extension Code for the CESID is unspecified.
Accepts up to 16 digits.

Related links

[Ext Code](#) on page 95

Total Length

The total number of sent digits. Accepts up to 16 digits.

Chapter 23: CDR system parameters

Administers the Call Detail Recording (CDR) feature used to record information on incoming, outgoing, and tandem calls for each trunk group administered for CDR, including auxiliary trunks. The system records information on each trunk-group call and each station-to-station call.

Example command: `change system-parameters cdr`

CDR system parameters: page 1

```
change system-parameters cdr                                     Page 1 of 1
CDR SYSTEM PARAMETERS
Node Number (Local PBX ID):                                     CDR Date Format: month/day
Primary Output Format:                     
Secondary Output Format:                     
CDR Retention (days): 20
Use ISDN Layouts? n                                         Enable CDR Storage on Disk? n
Use Enhanced Formats? n                                     Condition Code 'T' For Redirected Calls? n
Use Legacy CDR Formats? y                                     Remove # From Called Number? n
Modified Circuit ID Display? n                               Intra-switch CDR? n
Record Outgoing Calls Only? n                               Outg Trk Call Splitting? y
Suppress CDR for Ineffective Call Attempts? y               Outg Attd Call Record? y
Disconnect Information in Place of FRL? n                   Interworking Feat-flag? n
Force Entry of Acct Code for Calls Marked on Toll Analysis Form? n
Calls to Hunt Group - Record: member-ext
Record Called Vector Directory Number Instead of Group or Member? n
Record Agent ID on Incoming? n                               Record Agent ID on Outgoing? y
Inc Trk Call Splitting? n
Record Non-Call-Assoc TSC? n                               Call Record Handling Option: warning
Record Call-Assoc TSC? n                                     Digits to Record for Outgoing Calls: dialed
Privacy - Digits to Hide: 0                                   CDR Account Code Length: 15
Remove '+' from SIP Numbers? y
```

Calls to Hunt Group — Record

Valid Entry	Usage
member-ext	Records the extension of the telephone or data terminal where the call terminated.
group-ext	Records the extension that was dialed.

CDR Account Code Length

Valid Entry	Usage
1 to 15	The number of digits to record when a user enters an account code. For some record formats, a long account code overwrites spaces on the record that are usually assigned to other fields.

CDR Date Format

Valid Entry	Usage
month/day day/month	The format for the date stamp that begins each new day of call records.

CDR retention (days)

Valid Entry	Usage
1 to 20	Specify the number of days for which you want to retain the CDR logs. The default value is 20. To retain the CDR logs, you must: • Enable the Enable CDR Storage on Disk option. • Enter <code>DISK</code> in the Primary Output Endpoint field.  Important: - If you enter CDR1 or CDR2 in the Primary Output Endpoint field, then the CDR logs are not retained on the Primary Controller (main server). - In case of survivable server, if the survivable server is active, then the CDR logs are retained on the local disk.

Condition Code 'T' for Redirected Calls

Enables or disables identification of CDR records of calls that have been redirected automatically off the server running Communication Manager.

Valid Entry	Usage
y	The Condition Code of both CDR records for the call is 'T.'
n	Special identification is applied.

Digits to Record for Outgoing Calls

Valid Entry	Usage
dialed	Record the digits a user dials.

Table continues...

Valid Entry	Usage
outpulsed	Record the digits that Communication Manager sends out over the trunk, including any additions or deletions that take place during routing.

Disconnect Information in Place of FRL

Valid Entry	Usage
y	Replace the Facility Restriction Level (FRL) with information about why a call disconnects.
n	Record the call FRL.

Enable CDR Storage on Disk

Enables or disables the Survivable CDR feature for the main server, Survivable Remote Server (Local Survivable Processor), and Survivable Core Server (Enterprise Survivable Server). Default is disabled.

Force Entry of Acct Code for Calls Marked on Toll Analysis Form

Specifies whether or not an account code is required when making a toll call. Account codes are not required on all charged calls, and are required on some non-charged calls.

Valid Entry	Usage
y	Restricts all toll calls unless the user dials an account code. Available only if Forced Entry of Account Codes is enabled for the system.
n	Users can place a call without an account code. This does not override other calling restrictions.

Related links

[Forced Entry of Account Codes](#) on page 733

Inc Attd Call Record

Enables or disables separate recording of attendant portions of outgoing calls that are transferred or conferenced.

Available only if Incoming Trunk Call Splitting is enabled.

Related links

[Inc Trk Call Splitting](#) on page 99

Inc Trk Call Splitting

Enables or disables the creation of separate records for each portion of incoming calls that are transferred or conferenced.

Available only if **Record Outgoing Calls Only** is disabled.

Related links

[Record Outgoing Calls Only](#) on page 103

Interworking Feat-flag

Valid Entry	Usage
y	The feature flag indicates interworked outgoing ISDN calls.
n	The feature flag indicates no answer supervision for interworked calls.

Intra-Switch CDR

Enables or disables the recording of calls within Communication Manager. Requires administration of intra-switch CDR extensions.

Related links

[Intra-Switch CDR](#) on page 377

Modified Circuit ID Display

Affects the “printer”, “teleseer”, and “59-character” output formats.

Valid Entry	Usage
y	Displays the circuit ID in its actual format (100's, 10's, units). For example, circuit ID 123 displays as 123.  Note: Requires verification that an output device can accept this format.
n	Displays the circuit ID in its default format (10's, units, 100's). For example, the system displays circuit ID 123 as 231.

Node Number (Local PBX ID)

Displays the DCS switch node number in a network of switches.

Outg Attd Call Record

Enables or disables separate recording of attendant portions of outgoing calls that are transferred or conferenced.

Available only if Outgoing Trunk Call Splitting is enabled.

Related links

[Inc Trk Call Splitting](#) on page 99

[Outg Trk Call Splitting](#) on page 101

Outg Trk Call Splitting

Enables or disables the creation of separate records for each portion of outgoing calls that are transferred or conferenced

Primary Output Endpoint

Primary Output Endpoint determines where the server running Communication Manager sends the CDR records. You require Primary Output Endpoint if a Primary Output Format is specified.

Valid Entry	Usage
eia	The EIA port is used to connect the CDR device.
<i>Extension number</i>	The extension of the data module that links the primary output device to the server running Communication Manager.
CDR1, CDR2	The CDR device is connected over a TCP/IP link, and this link is defined as either CDR1 or CDR2 for IP Services.
DISK	If you configure the CDR Retention (Days) field, then you must enter <code>DISK</code> in the Primary Output Endpoint field so that the CDR logs are retained. If you configure the retention policy, the retention policy is applicable if the Primary Output Endpoint field is set to <code>DISK</code> only.

Related links

[IP Services](#) on page 428

Primary Output Format

Indicates the format of the call records sent to the primary output device.

Valid Entry	Usage
customized	For special call accounting needs that are prevented for standard record formats. Requires call accounting software that is also customized to receive these records. Call accounting vendors should be consulted before using this option.
printer	Call detail records are sent to a printer rather than to a record collection or call accounting system.
59-char expanded lsu lsu-expand int-direct int-isdn int-process teleseer unformatted	For standard record formats. The selection must be compatible with your call accounting software. Verify this through your vendor or the accounting system documentation.

Privacy — Digits to Hide

Indicates how much of the dialed number to hide on the CDR record for stations administered for CDR privacy.

Valid Entry	Usage
0 to 7	The number of digits to hide, counting from the end right to left.

Example

For a value of 4, when the user dials 555-1234, the system displays only “555” in the CDR record.

Related links

[CDR Privacy](#) on page 670

Record Agent ID on Incoming

Determines whether the EAS agent login ID must be included instead of the physical extension in the CDR record.

Available only if Expert Agent Selection (EAS) is enabled for the system. Cannot be enabled if **Record Called Vector Directory Number Instead of Group or Member** is enabled.

Related links

[Record Called Vector Directory Number Instead of Group or Member](#) on page 102

Record Agent ID on Outgoing

Determines whether the EAS agent's LoginID must be included instead of the physical extension in the CDR record.

Available only if Expert Agent Selection (EAS) is enabled for the system.

Related links

[Expert Agent Selection \(EAS\)](#) on page 740

Record Call-Assoc TSC

Enables or disables the creation of records for call-associated temporary signaling connections. A large number of data connections could increase the number of records, so call collection device capacity must be taken into consideration.

Record Called Vector Directory Number Instead of Group or Member

Determines whether or not to include the Vector Directory Number (VDN) in the CDR record. If enabled, the called VDN overrides the group or member information that normally the system displays in the CDR record. If a call is directed through more than one VDN, the first VDN used for

the call is stored. This applies only to calls routed to a hunt group by a vector, not to calls routed directly to an extension by a vector.

Cannot be enabled if **Record Agent ID on Incoming** is enabled.

Related links

[Record Agent ID on Incoming](#) on page 102

Record Non-Call-Assoc TSC

Enables or disables the creation of records for non-call-associated temporary signaling connections. A large number of data connections could increase the number of records, so record collection device capacity must be taken into consideration.

Record Outgoing Calls Only

Valid Entry	Usage
y	Record only outgoing calls.  Note: This can save space if you are only concerned with charges for outbound calls.
n	Record both outgoing and incoming calls.

Remove '+' from SIP Numbers

If you set the **Remove '+' from SIP Numbers** field on the CDR System Parameters screen to y and if a leading plus (+) sign is present in a calling or called number sent over a CDR adjunct, Communication Manager removes the leading plus (+) sign from the number. The default value is y.

Remove # From Called Number

Valid Entry	Usage
y	Removes the “#” or “E” symbol from the Dialed Number field of the call detail record. The output device must be able to accept this format.
n	Retains the trailing “#” or “E” symbol in the Dialed Number field whenever an inter-digit time-out occurs or users dial # to indicate the end of dialing.

Secondary Output Endpoint

Available only if **Secondary Output Format** is administered.

Valid Entry	Usage
eia	The secondary output device is connected to the eia port.

Table continues...

Valid Entry	Usage
<i>Extension number</i>	The extension of the data module that links the secondary output device to the server running Communication Manager.
CDR1, CDR2	The CDR device is connected over a TCP/IP link, and this link is defined as either CDR1 or CDR2 for IP Services.

Related links

[Secondary Output Format](#) on page 104

[Service Type](#) on page 429

Secondary Output Format

⚠ Caution:

Only qualified Avaya service personnel should administer a secondary output device. This option might cause loss of data when the buffer contains large amounts of data.

Valid Entry	Usage
customized int-direct int-process lsu unformatted	The format of the call records sent to the secondary output device. Only these formats can be used for a secondary output device. The format must be compatible with call accounting software. Verify this through the vendor or the accounting system documentation.

Suppress CDR for Ineffective Call Attempts

Ineffective call attempts are the calls that are blocked because the user has insufficient calling privileges or all outgoing trunks are busy. These calls appear on the CDR record with a condition code "E".

Valid Entry	Usage
y	Ignores ineffective call attempts. Used when there is limited storage space for CDR records and when records often overrun the buffer.
n	Reports ineffective call attempts. Used when users are often unable to place outgoing calls, or if a large number of incoming calls are incomplete. Also used to keep records of attempts to contact a client, and when ISDN trunks are used. This option requires more space for records.

Use Enhanced Formats

Enables or disables the use of the Enhanced version of the specified primary output formats. Enhanced formats provide additional information about time in queue and ISDN call charges, where available. This affects the expanded, teleseer, lsu, printer, and unformatted output formats. Enhanced formats and ISDN formats cannot be used at the same time.

Use ISDN Layouts

Enables or disables using the ISDN version of the specified primary output format. ISDN Layouts provide more accurate information about the inter-exchange carrier and ISDN network services used for a call. This affects lsu and printer output formats, as well as any format with ISDN layouts, such as teleseer. ISDN formats and Enhanced formats cannot be used at the same time.

Use Legacy CDR Formats

Enables or disables the use of pre-Communication Manager 4.0 legacy Call Detail Recording (CDR) formats for CDR records. The default is enabled. Listed below are the CDR formats that are impacted by this field. All other CDR formats remain unchanged.

CDR Format	Communication Manager 3.1 and earlier length	Communication Manager 4.0 and later length
ISDN Teleseer	80	82
Enhanced Teleseer	81	83
ISDN Printer	84	86
Enhanced Printer	85	87
ISDN LSU	59	61
Enhanced LSU	59	61
Expanded	135	139
Enhanced Expanded	151	155
Unformatted	105	109
Enhanced Unformatted	119	123
Int-ISDN	136	140

CDR System Parameters: page 2

Used if there is an arrangement with the vendor to customize the call accounting system to receive these records.

Available only if **Primary Output Format** is customized.

Related links

[Primary Output Format](#) on page 101

Data Item

Use this field to itemize the data items that appear on the customized record.

You must include at least one field for a record. The last two data items in the record must be line-feed and return.

Data Item	Length
acct-code	15
attd-console	2
auth-code	7
bandwidth	2
bcc	1
calltype	1
calling-num	15
cca-id	20
clg-pty-cat	2
clg-num/in-tac	10
code-dial	4
code-used	4
cond-code	1
contact-uri	20
country-from	3
country-to	3
date	6
dialed-num	23
duration	4
end-date(4d)	8
feat-flag	1
fri	1
from-uri	20
in-crt-id	3
in-trk-code	4
ins	3
internal-codec	2
isdn-cc	11
isdn-cc/ppm	11
ixc-code	4
line-feed	1
local-time-from	
local-time-to	
location-from	3
location-to	3

Table continues...

Data Item	Length
ma-uui	1
node-num	2
null	3
out-crt-id	3
ppm	5
request-uri	20
res_flag	1
return	1
sec-dur	5
seq-num	10
space	1
start-date	6
start-date(4d)	8
start-time	6
time	4
timezone-from	3
timezone-to	6
to-uri	20
tsc_ct	4
tsc_flag	1
trunk-codec	2
ucid	20
vdn	5

Length

The length of each data item. The maximum record length depends on the call accounting system. Check with the vendor. The date field should be six digits to ensure proper output. Certain fields default to the required length.

Record Length

Displays the accumulated total length of the customized record, updated each time the length of a data item changes.

Chapter 24: Change Station Extension

Changes extensions on the switch from one extension to another all at once. Specifically:

- All administration that was associated with the current extension is associated with the new extension.
- Any administration references on the changing extension, such as references used in a vector, coverage, and so on, are now referenced to the new extension.
- All references to the previous extension are removed from the system.

If an extension is changed that is also administered on an adjunct, the extension on the adjunct must also be changed to ensure proper functionality.

For fields that are changing, current information prior to the change displays under **From Extension**.

A forwarded extension administered as a button remains.

Exceptions

Station extensions cannot be changed if that station is administered as the emergency location extension for another station. For example, if station A is administered as the emergency location extension for station B, then:

- First assign station B to a different emergency location extension.
- First change station B. The system displays station A's extension as the emergency location extension.

Example command: `change extension-station n`, where *n* is the extension assigned to a station.

change extension-station 3600000		Page 1 of 1
CHANGE STATION EXTENSION		
Station Name: ip1	Port: S00000	
FROM EXTENSION	TO EXTENSION	
-----	-----	
Station: 360-0000		
Message Lamp: 360-0000	<u>360-0000</u>	
Emergency Location Ext: 360-0000	<u>360-0000</u>	
WARNING: Submitting this form does not update the extension stored in the station itself. After submitting this command, be sure to reprogram the station with the new extension.		

Emergency Location Extension

The Station screen displays the new extension for the **Emergency Location Ext.** field. Up to sixteen numbers can be used to make up a valid extension number for the dial plan.

IP Parameter Emergency Location

The emergency location extension from the IP Address Mapping screen associated with the current extension.

Related links

[IP network region](#) on page 398

[Emergency Location Extension](#) on page 378

Message Lamp

The new extension for the message lamp extension. Up to sixteen numbers can make up a valid extension number for the dial plan.

Station

The new extension. Up to seven numbers can make up a valid extension number for the dial plan.

Station Name

The name of the existing extension.

Chapter 25: Class of Restriction

Establishes classes of restriction (COR). Classes of restriction restrict users from originating or terminating certain types of calls. A system might use only one COR or as many as necessary to control calling privileges.

Example command: `change cor n`, where *n* is the COR number.

Class of restriction: page 1

change cor 1 Page 1 of 23

CLASS OF RESTRICTION

COR Number: 1

COR Description: _____

FRL: 0

Can Be Service Observed? n

Can Be A Service Observer? n

Time of Day Chart: 1

Priority Queuing? n

Restriction Override: none

Restricted Call List? n

APLT? y

Calling Party Restriction: outward

Called Party Restriction: none

Forced Entry of Account Codes? n

Direct Agent Calling? n

Facility Access Trunk Test? n

Can Change Coverage? n

Access to MCT? y

Group II Category For MFC: 7

Send ANI for MFE? n

MF ANI Prefix: _____

Hear System Music on Hold? y

Fully Restricted Service? n

Hear VDN of Origin Annc.? n

Add/Remove Agent Skills? n

Automatic Charge Display? n

PASTE (Display PBX Data on Phone)? n

Can Be Picked Up By Directed Call Pickup? y

Can Use Directed Call Pickup? y

Group Controlled Restriction: inactive

Access to MCT

Grants or denies access to the Malicious Call Trace feature.

Valid Entry	Usage
y	With this, user can activate a request to trace a malicious call.

Table continues...

Valid Entry	Usage
n	With this, user cannot request a malicious call trace. If this extension is the subject of a malicious call trace, the MCT History report does not display the extension.

Add/Remove Agent Skills

User can add or remove skills.

APLT

Users can gain access to APLT trunk group Enhanced Private Switched Communications System (EPSCS) or Common Control Switched Arrangement (CCSA) off-net facilities.

Automatic Charge Display

Shows the cost of an active outgoing call using Periodic Pulse Metering (PPM) or ISDN Advice of Charge (AOC) on Digital Communications Protocol (DCP) or Avaya BRI stations. Unavailable in the U.S.

Valid Entry	Usage
y	Call charges display during and at the end of the call.
n	Call charges display only when users press the disp-chrg button before the call drops.

Called Party Restriction

Valid Entry	Usage
Inward	Blocks the calling party from receiving incoming exchange network calls, attendant originated calls, and attendant completed calls.
Manual	Blocks the called party from receiving all calls except for those originated or extended by the attendant.
Public	Blocks the called party from receiving public network calls. Attendant calls can go through the called party and attendant-assisted calls if administered for this COR.
Termination	Blocks the called party from receiving any calls at any time.
none	No called party restrictions.

Related links

[Restriction Override](#) on page 117

Calling Party Restriction

Determines the level of calling restriction associated with this COR.

Important:

Limit calling permissions as much as possible.

Valid Entry	Usage
Origination	Blocks the calling party from originating a call from the facility at any time. The party can only receive calls. A telephone with this COR can initiate Remote Access calls, if the COR of the barrier code supports it.
Outward	Blocks the calling party from calling outside the private network. Users can dial other users on the same server running Communication Manager or within a private network. To enhance security, you can use outward restrictions when practical.
All-toll	Blocks the calling party from making ARS and trunk access calls to certain toll areas as defined by the Dialed String on the ARS Toll Analysis table. The call completes if the facility COR is also associated with an Unrestricted Call List and whose Dialed String also matches the dialed number.
Tac-toll	Blocks the calling party from making trunk access calls to certain toll areas as defined by the Dialed String on the ARS Toll Analysis table. The call completes if the facility COR is also associated with an Unrestricted Call List and whose Dialed String also matches the dialed number.
none	No calling party restrictions.

Related links

[Dialed String](#) on page 20

Can Be a Service Observer

A user with this CO permission can be a service observer.

Available only if Basic Service Observing is enabled for the system.

** Caution:**

Service Observing might be subject to federal, state, or local laws, rules, or regulations; or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves with and comply with all applicable laws, rules, and regulations before using these features.

Related links

[Service Observing \(Basic\)](#) on page 741

Can Be Picked Up By Directed Call Pickup

Calls for this station or EAS agent can be picked up using the Directed Call Pickup Up feature.

Available only if Directed Call Pickup is enabled for the system.

Related links

[Directed Call Pickup](#) on page 313

Can Be Service Observed

Service observing can be activated for physical extensions and logical agent IDs and VDNs.

Can Change Coverage

Station users can: Or Station users can have permission to:

- Select one of two previously-administered coverage paths
- Activate, change, or deactivate call forward all calls or call forward busy/don't answer from any on-site or off-site location

Can Use Directed Call Pickup

The station, attendant, or EAS agent permission can pick up calls using the Directed Call Pickup feature.

Available only if Directed Call Pickup is enabled for the system.

Related links

[Directed Call Pickup](#) on page 313

COR Description

The description of the COR. Accepts up to 35 characters.

Tip:

A clear description makes it easier to remember which COR to assign when adding users.

Example

Two examples of COR descriptions include “Customer Service” and “Legal Department”.

COR Number

The COR number.

Direct Agent Calling

Users can use the Direct Agent Calling feature. Using Direct Agent Calling, users can dial an ACD agent extension directly, rather than anyone in the agent pool. If the system is in Night Service, the call routes to the Night Service extension. If the extension with this COR belongs to an agent, the agent can receive calls directly.

Facility Access Trunk Test

Users can perform Facility Access Trunk Tests. When this feature is active, the **trk-ac-alm** feature button status lamp lights when a successful test attempt occurs.

Forced Entry of Account Codes

If enabled, the Forced Entry of Account Codes (FEAC) feature is active on this COR. Any telephone assigned the associated COR must dial an account code before making an outgoing call. If a call is being routed by ARS, the system prevents account code checking on the COR.

Available only when the FEAC feature is enabled for the system.

Important:

If a COR using the FEAC feature is assigned to a VDN, the route-to commands executed by the associated vector will be unsuccessful.

Related links

[Forced Entry of Account Codes](#) on page 733

[CDR FEAC](#) on page 779

FRL

Security alert:

Assign the lowest possible FRL to enhance system security.

Valid Entry	Usage
0 to 7	Assigns a Facilities Restriction Level (FRL) to the COR to enhance system security. AAR or ARS features use this entry to determine call access to an outgoing trunk group. Outgoing call routing is determined by a comparison of the FRLs in the AAR/ARS Routing Pattern and the FRL associated with the COR of the call originator. The call originator is typically a telephone user. An originating FRL of 0 has the least calling privileges.

Fully Restricted Service

Unavailable for Enhanced Private Switched Communications System (EPSCS) or Common Control Switched Arrangement (CCSA) off-net facilities.

Valid Entry	Usage
y	Stations do <i>not</i> have access to the public network for either incoming or outgoing calls.
n	Stations have access to the public network for either incoming or outgoing calls.

Group II Category For MFC

Valid Entry	Usage
1 to 10	Controls categories for Russian signaling trunks. Also controls categories for R2-MFC signaling trunks if administered system-wide. The server running Communication Manager sends this value as the Calling or Called Party Category for telephones or trunks that use this COR. This administered Calling Party Category digit is included as part of the ANI information sent to the local telephone company central office on request using R2-MFC signaling.

Related links

[Use COR for Calling Party Category](#) on page 519

Group Controlled Restriction

Valid Entry	Usage
active	The current COR is under controlled restriction.
inactive	The current COR is beyond controlled restriction.

Hear System Music on Hold

If you enable the **Music on Hold** feature on the telephone, callers that the user puts on hold will hear music.

Hear VDN of Origin Announcement

Agents can receive VDN of Origin Announcement (VOA) messages that provide agents with a short message about the caller's city of origin or requested service based on the VDN used to process the call.

MF ANI Prefix

The prefix applied to an extension number when ANI is sent to the local telephone company central office. Accepts up to seven digits. This COR-specific value overrides any ANI prefix administered system-wide for multifrequency signaling. The prefix is inapplicable when ANI is tandemed through the Communication Manager server on tandem calls. Also applies to the ANI for the server when the originating side is a trunk and there was no ANI.

Related links

[ANI Prefix](#) on page 513

Partitioned Group Number

Available only if AAR/ARS Partitioning is enabled for the system and partition groups are administered.

Valid Entry	Usage
1 to 8	The AAR/ARS partitioned group number associated with this COR.

Related links

[ARS/AAR Partitioning](#) on page 729

PASTE (Display PBX Data on telephone)

Use this field to enable the PASTE feature.

Valid entry	Usage
y	Downloads all lists.
n	Disables the PASTE feature.

Priority Queuing

If enabled, a telephone user's calls are placed ahead of non-priority calls in a hunt group queue.

Related links

[ACD](#) on page 739

Restricted Call List

Use this field to allow or restrict access to the Restricted call list.

Valid entry	Usage
y	Users cannot make calls to numbers on the Restricted call list from a facility assigned to this COR.
n	Users can make calls to numbers on the Restricted call list from a facility assigned to this COR.

Related links

[RCL](#) on page 781

Restriction Override

Users assigned to this COR can override inward restriction during conference, transfer or call forwarding operation to a telephone that is inward restricted.

Valid Entry	Usage
attendant	Attendants might override inward restrictions. A telephone with a COR that is inward restricted cannot receive public network, attendant-originated, or attendant-extended calls.
all	All users can override inward restrictions
none	No users can override inward restrictions

Send ANI for MFE

Applicable only:

- For Spain
- For 2/6 signaling, but not 2/5 signaling

Valid Entry	Usage
y	Enables Automatic Number Identification (ANI) to send the calling party's number to the public or IBERCOM network so that charges are broken down by line.
n	Charges are not itemized by line. The company receives a single bill for the total number of calls made; also called <i>block charging</i> .

Time of Day Chart

Available only if Time of Day Routing is enabled for the system.

Valid Entry	Usage
1 to 8	The AAR/ARS time-of-day-chart number associated with this COR.

Related links

[Time of Day Routing](#) on page 738

Unrestricted Call List

Valid Entry	Usage
1 to 10 blank	Overrides specified toll call restrictions for a COR otherwise restricted from making ARS or trunk access calls.

Related links

[Dialed String](#) on page 20

Class of Restriction: page 2

change cor 1		Page 2 of 43
CLASS OF RESTRICTION		
Auto-Answer Media Link Calls?	<u>n</u>	
MF Incoming Call Trace?	<u>n</u>	
Brazil Collect Call Blocking?	<u>n</u>	
Block Transfer Display?	<u>n</u>	
Block Enhanced Conference/Transfer Displays?	<u>y</u>	
Remote Logout of Agent?	<u>n</u>	
Station Lock COR: <u>1</u>	TODSL Release Interval (hours):	
	ASAI Uses Station Lock? <u>n</u>	
Line Load Control: <u>1</u>		
Maximum Precedence Level: <u>ro</u>	Preemptable? <u>y</u>	
MLPP Service Domain: _____		
Station-Button Display of UUI IE Data?	<u>n</u>	
Service Observing by Recording Device?	<u>n</u>	
Can Force A Work State Change?	<u>n</u>	
Work State Change Can Be Forced?	<u>n</u>	
Restrict Second Call Consult?	<u>n</u>	

ASAI Uses Station Lock

Valid Entry	Usage
y	Enables the ASAI originated calls to follow the restriction or permission assigned to the locked station, via the Station Lock COR feature.
n	Enables the ASAI originated calls to follow the permission or restrictions of the COR assigned to the endpoint. The default value is n.

Auto-answer media link calls

You can configure the **Auto-Answer Media Link Calls** field to enable the Communication Manager to answer the calls automatically or not. **Auto-Answer Media Link Calls** field applies to J169, J179, and Avaya SIP (AvayaSIP) set types.

Valid entry	Usage
Y	Communication Manager enables the endpoints to auto-answer the incoming calls automatically.
N	Communication Manager enables the endpoints to ring but disables the endpoints to auto-answer the incoming calls.

Block Enhanced Conference/Transfer Display

Valid entry	Usage
y	Blocks all the enhanced conference and transfer display messages on digital telephones, except the Transfer Completed message.
n	Does not block the enhanced conference and transfer display messages.

Block Transfer Displays

Valid Entry	Usage
y	Users of DCP, Hybrid, ISDN-BRI, or wireless display telephones cannot receive a confirmation message when they transfer a call.
n	Users can receive a confirmation message when they transfer a call.

Brazil Collect Call Blocking

If enabled, all Brazilian trunk calls that terminate to a station send back a double answer to the local telephone company central office (CO). This double answer tells the CO that this particular station cannot accept collect calls. The CO then tears down the call if it is a collect call.

Line Load Control

Valid Entry	Usage
1 to 4	Specifies the line load control level for this COR, where 1 has no restrictions and 4 is the most restrictive.

Maximum Precedence Level

Assigns a maximum precedence level for extensions with this COR for use with the Multiple Level Precedence and Preemption feature.

Valid Entry	Usage
fo	Flash Override
fl	Flash
im	Immediate
pr	Priority
ro	Routine. This is the default value.

MF Incoming Call Trace

If enabled, a Call Trace can be assigned to a station. Communication Manager then generates an MFC backward signal during call setup instead of the “free” signal. This triggers the local telephone company central office to collect trace information before releasing the calling party.

MLPP Service Domain

Valid Entry	Usage
0 to 16777215	The service domain for users and trunks to which this COR is assigned

This field is visible only if **Multi level precedence preemption** is set to *y* on the system-features customer-options form.

Outgoing Trunk Disconnect Timer (minutes)

Disconnects an outgoing trunk automatically after an administrable amount of time.

Valid Entry	Usage
2 to 999	The number of minutes to wait before automatically disconnecting the call. A warning tone is given to all parties on the trunk call 1 minute before the administered value and a second warning tone is heard 30 seconds later. The call is automatically disconnected 30 seconds after the second warning tone.
blank	Outgoing trunk calls disconnect only when dropped by one or all parties. This is the default.

Preemptable

If enabled, makes extensions with this COR preemptable for Multiple Level Precedence and Preemption calls.

Remote Logout of Agent

If enabled, users can use a feature access code to logout an idle ACD or EAS agent without being at the agent's telephone.

Restrict Second Call Consult

Use this field to enable the Restrict Second Agent Consult feature. With this feature, Communication Manager allows the agents of Avaya Aura® Contact Center to use only one consult operation at a time. The consult operations include transfer and conference.

Note:

If you set the **Restrict Second Call Consult** field to *n*, after you dial the **Restrict Second Consult** feature access code, the feature access code overrides the setting of the **Restrict Second Call Consult** field on the COR group of the station.

Service Observing by Recording Device

If enabled, the service observer associated with this COR can use an audio recording device, such as the Witness product.

For more information on Service Observing by Recording Device, see *Avaya Aura® Communication Manager Feature Description and Implementation*

Station-Button Display of UI IE Data

If enabled, a station user can push a **uui-info station** button and see up to 32 bytes of ASAI-related User-User-Information Information Element (UUI-IE) data. Pressing the **uui-info** button displaces the incoming call/collected digits display. Pressing **callr-info** re-displays the collected digits.

Available only with Call Center release 3.0 or later.

Related links

[Call Center Release](#) on page 739

Station Lock COR

Valid Entry	Usage
0 to 995	If the user dials the station lock Feature Access Code (FAC), the user's extension applies the Station Lock COR for call restrictions instead of the station's usual COR. For information on call restrictions, see the Station Lock section of <i>Avaya Aura® Communication Manager Feature Description and Implementation</i> . This field defaults to the current COR.

Related links

[Station Lock Activation/Deactivation](#) on page 226

Class of Restriction: page 3

The Send All Calls and Call Forwarding (SAC/CF) Override feature overrides active rerouting. This feature overrides these active rerouting settings:

- Send All Calls (SAC)
- Call Forwarding (CF) all
- Enhanced Call Forwarding (ECF) unconditional

```

PuTTY
change cor 21                                     Page 3 of 43

CLASS OF RESTRICTION

ERASE 24XX/14XX/94XX USER DATA UPON
Dissociate or unmerge this phone: none
EMU login or logoff at this phone: none
Mask CPN/NAME for Internal Calls? n
Allow Agent to Activate Call Forward? n
SAC/CF Override by Team Btn? n Priority Call? n Dialing? n
SAC/CF Override Protection for Team Btn? n Priority Call? n Dialing? n

one-X Server Access? y
Team Btn Silent if Active? n Priority Ring? n
Team Btn Display Name? n Pick Up by Going Off Hook? n
Listen Only Talk Path on Auto Answered Intercom Calls? n
Outgoing Trunk Disconnect Timer (minutes):
Outgoing Trunk Alerting Timer (minutes): Internal Users Only? n
Allow Trunk Seizure on Busy Indicator? y
Homing Policy for Users: 2 Preferred CM List: 0

ESC-x=Cancel Esc-e=Submit Esc-p=Prev Pg Esc-n=Next Pg Esc-h=Help Esc-r=Refresh

```

Allow Trunk Seizure On Busy Indicator

This field specifies whether the trunk is seized when the user presses the **Busy indicator** button.

Valid entry	Usage
y	When the user presses the Busy indicator button for TAC: - The trunk is seized. - On pressing the digits, the call goes through.
n	When the user presses the Busy indicator button for TAC: - The trunk is not seized. - The user hears reorder tone with fluttering lamp. - The system logs a denial event.

Auto Answer

Enables or disables automatic answer on team button calls.

Erase 24xx/14xx/94xx User Data Upon: Dissociate or unmerge this phone

Administers what local terminal data items are erased when the 24xx, 14xx, or 94xx is dissociated or unmerged.

Valid Entry	Usage
none	Default value. No local terminal data is erased.
log	Terminal's local call Log data is erased.
customizations	Call Log, Button labels, Speed Dial List, Local Terminal Options are erased.
all	All local terminal data is erased (Call Log, Button Labels, Speed Dial List, Options, Language).

Erase 24xx/14xx/94xx User Data Upon: EMU login or logoff at this phone

Administers what local terminal data items are erased upon Enterprise Mobility User (EMU) login or logoff.

Valid Entry	Usage
none	This is the default. No local terminal data is erased.
log	Terminal's local call Log data is erased.
customizations	Call Log, Button labels, Speed Dial List, Local Terminal Options are erased.
all	All local terminal data is erased (Call Log, Button Labels, Speed Dial List, Options, Language).

Homing Policy for Users

A homing policy specifies the Communication Manager node to which a station connects. A homing policy value of one indicates that there is a dynamic assignment of Communication Manager nodes which does not follow any fixed or static pattern. A homing policy value of two indicates that there is a strict node hierarchy.

These values are important for the successful operation of the Communication Manager for Avaya Aura® Distributed Architecture (AADA) using Kafka. This integration requires the movement from one Communication Manager instance to another in a specific order.

Note:

The Kafka integration feature is reserved for future use.

Valid entry	Usage
1	Dynamic assignment of Communication Manager nodes.
2	Static assignment of Communication Manager nodes in a strict node hierarchy.

Internal Users Only

Use this feature to enable or disable the zip tone for internal users for all incoming and outgoing trunk calls. The zip tone plays for all calls, such as one-to-one mapping and conference calls.

When you enable the **Internal Users Only** field, Communication Manager disables the **Outgoing Trunk Alerting Timer (minutes)** field and enables the **Trunk Alerting Timer (minutes)** field with an interval ranging from 0 to 999.

Valid entry	Usage
y	The zip tone plays only for the internal users for all incoming and outgoing trunk calls.
n	The zip tone plays for internal and external users.

Listen Only Talk Path on Auto Answered Intercom Calls

If enabled, the intercom call receiver can listen to another party but cannot speak to the other party. By default, this field is disabled.

Mask CPN/Name for Internal Calls

Valid entry	Usage
y	Hides the display of calling and called party numbers and administered name on internal calls.
n	Shows the display of calling and called party numbers and administered name on internal calls.

one-X Server Access

Enables or disables the one-X Server Access for one-X control.

Outgoing Trunk Alerting Timer (minutes)

Applies an zip tone to an outgoing trunk call after an administrable amount of time.

Valid Entry	Usage
2 to 999	The number of minutes to wait before an zip tone is applied to the call.
blank	The Zip Tone for Outgoing Trunk Calls feature is disabled and the zip tone is not applied to the call. This is the default.

Related links

[Trunk Alerting Tone Interval \(seconds\)](#) on page 263

Outgoing Trunk Disconnect Timer (minutes)

Disconnects an outgoing trunk automatically after an administrable amount of time.

Valid Entry	Usage
2 to 999	The number of minutes to wait before automatically disconnecting the call. A warning tone is given to all parties on the trunk call 1 minute before the administered value and a second warning tone is heard 30 seconds later. The call is automatically disconnected 30 seconds after the second warning tone.
blank	Outgoing trunk calls disconnect only when dropped by one or all parties. This is the default.

Preferred CM List

A list of Communication Manager nodes in a particular order. This grouping of nodes is a Communication Manager cluster. You can configure this list using `change cluster preferred-CM-list <list#>`. Each node in the list must match the node name that you configured using the System Manager web console.

These values are important for the successful operation of the Communication Manager for Avaya Aura® Distributed Architecture (AADA) using Kafka. This integration requires the movement from one Communication Manager instance to another in a specific order.

 Note:

The Kafka integration feature is reserved for future use.

Priority Ring

Enables or disables priority ringing for speed dialing on team button calls.

SAC/CF Override by Priority Call and Dialing

Use this field to enable or disable the SAC/CF override by Priority Call and Dialing feature.

SAC/CF Override by Team Btn

Use this field to allow the user of a monitoring station to override any active rerouting, such as Send All Calls or Call Forwarding, on the monitored station.

The monitoring station can override the rerouting only if the following conditions are met:

- The **SAC/CF Override by Team Btn** field is set to y.
- The **SAC/CF Override Protection for Team Btn** field is set to n.
- The monitoring station uses the **team** button to make the call to the monitored station.

 Note:

Communication Manager Release 6.2 and earlier supported this feature on H.323 and DCP stations. Communication Manager Release 6.3.6 extends this support to SIP stations.

SAC/CF Override Protection for Priority Call and Dialing

Use this field to enable or disable the SAC/CF Override Protection for Priority Call and Dialing feature.

SAC/CF Override Protection for Team Btn

Use this field to restrict the user of a monitoring station from overriding the active rerouting, such as Send All Calls or Call Forwarding, on the monitored station. This field works only when the monitoring station uses the **team** button to make a call to the monitored station.

Note:

Communication Manager Release 6.2 and earlier supported this feature on H.323 and DCP stations. Communication Manager Release 6.3.6 extends this support to SIP stations.

Team Btn Display Name

Enables or disables the display of station name on team button calls.

Team Btn Silent if Active

Enables or disables audible ringing on team button calls.

Team Pick Up by Going Off Hook

Enables or disables pick-up by going off hook on team button calls.

Trunk Alerting Timer (minutes)

With Release 10.2, when you enable **Internal Users Only**, the Communication Manager disables the **Outgoing Trunk Alerting Timer (minutes)** field and enables the **Trunk Alerting Timer (minutes)** field with an interval ranging from 0 to 999.

Valid entry	Usage
0 to 999	The number of minutes to wait before Communication Manager applies a zip tone to the call.
blank	The zip tone for Outgoing Trunk Calls feature is disabled, and Communication Manager does not apply the zip tone to the call. This is the default value.

Class of restriction: page 4 to page 23

You can use the screens from page 4 to page 13 to assign calling permissions and the screens from page 14 to page 23 to assign service observing permissions to the COR that you administer.

CALLING PERMISSION

change cor 1

Page 4 of 23

CLASS OF RESTRICTION

CALLING PERMISSION (Enter "y" to grant permission to call specified COR)

0? <u>y</u>	15? <u>y</u>	30? <u>y</u>	44? <u>y</u>	58? <u>y</u>	72? <u>y</u>	86? <u>y</u>
1? <u>y</u>	16? <u>y</u>	31? <u>y</u>	45? <u>y</u>	59? <u>y</u>	73? <u>y</u>	87? <u>y</u>
2? <u>y</u>	17? <u>y</u>	32? <u>y</u>	46? <u>y</u>	60? <u>y</u>	74? <u>y</u>	88? <u>y</u>
3? <u>y</u>	18? <u>y</u>	33? <u>y</u>	47? <u>y</u>	61? <u>y</u>	75? <u>y</u>	89? <u>y</u>
4? <u>y</u>	19? <u>y</u>	34? <u>y</u>	48? <u>y</u>	62? <u>y</u>	76? <u>y</u>	90? <u>y</u>
5? <u>y</u>	20? <u>y</u>	35? <u>y</u>	49? <u>y</u>	63? <u>y</u>	77? <u>y</u>	91? <u>y</u>
6? <u>y</u>	21? <u>y</u>	36? <u>y</u>	50? <u>y</u>	64? <u>y</u>	78? <u>y</u>	92? <u>y</u>
7? <u>y</u>	22? <u>y</u>	37? <u>y</u>	51? <u>y</u>	65? <u>y</u>	79? <u>y</u>	93? <u>y</u>
8? <u>y</u>	23? <u>y</u>	38? <u>y</u>	52? <u>y</u>	66? <u>y</u>	80? <u>y</u>	94? <u>y</u>
9? <u>y</u>	24? <u>y</u>	39? <u>y</u>	53? <u>y</u>	67? <u>y</u>	81? <u>y</u>	95? <u>y</u>
10? <u>y</u>	25? <u>y</u>	40? <u>y</u>	54? <u>y</u>	68? <u>y</u>	82? <u>y</u>	96? <u>y</u>
11? <u>y</u>	26? <u>y</u>	41? <u>y</u>	55? <u>y</u>	69? <u>y</u>	83? <u>y</u>	97? <u>y</u>
12? <u>y</u>	27? <u>y</u>	42? <u>y</u>	56? <u>y</u>	70? <u>y</u>	84? <u>y</u>	98? <u>y</u>
13? <u>y</u>	28? <u>y</u>	43? <u>y</u>	57? <u>y</u>	71? <u>y</u>	85? <u>y</u>	99? <u>y</u>
14? <u>y</u>	29? <u>y</u>					

Users with this COR can call users with the CORs from 0 to 995.

Valid entry	Usage
y	A station user with the COR that you administer can call stations with the other COR. The default value is y.
n	A station user with the COR that you administer cannot call stations with the other COR.

SERVICE OBSERVING PERMISSION

change cor 1		Page 14 of 23	
CLASS OF RESTRICTION			
SERVICE OBSERVING PERMISSIONS			
(Enter "y" to grant permission to service observe the specified COR)			
0? <u>y</u>	15? <u>y</u>	30? <u>y</u>	44? <u>y</u>
1? <u>y</u>	16? <u>y</u>	31? <u>y</u>	45? <u>y</u>
2? <u>y</u>	17? <u>y</u>	32? <u>y</u>	46? <u>y</u>
3? <u>y</u>	18? <u>y</u>	33? <u>y</u>	47? <u>y</u>
4? <u>y</u>	19? <u>y</u>	34? <u>y</u>	48? <u>y</u>
5? <u>y</u>	20? <u>y</u>	35? <u>y</u>	49? <u>y</u>
6? <u>y</u>	21? <u>y</u>	36? <u>y</u>	50? <u>y</u>
7? <u>y</u>	22? <u>y</u>	37? <u>y</u>	51? <u>y</u>
8? <u>y</u>	23? <u>y</u>	38? <u>y</u>	52? <u>y</u>
9? <u>y</u>	24? <u>y</u>	39? <u>y</u>	53? <u>y</u>
10? <u>y</u>	25? <u>y</u>	40? <u>y</u>	54? <u>y</u>
11? <u>y</u>	26? <u>y</u>	41? <u>y</u>	55? <u>y</u>
12? <u>y</u>	27? <u>y</u>	42? <u>y</u>	56? <u>y</u>
13? <u>y</u>	28? <u>y</u>	43? <u>y</u>	57? <u>y</u>
14? <u>y</u>	29? <u>y</u>		

Users with this COR can observe users with the CORs from 0 to 995.

Valid entry	Usage
y	A station user with the COR that you administer can observe stations with the other COR. The default value is y.
n	A station user with the COR that you administer cannot observe stations with the other COR.

Chapter 26: Class of service

Administrators access permissions for call processing features that require dial code or feature button access.

 Note:

Class of Service (COS) is inapplicable to trunk groups except for the Remote Access feature.

A COS assignment defines which features and functions a telephone user can access. Up to 16 different COS numbers can be administered (0 to 15). When Tenant Partitioning is enabled for the system, you can administer up to 100 COS groups, each with 16 Classes of Service.

The screen lists the default values for each COS or feature combination. For a particular combination, y is for access to the feature and n is for denying access.

 Caution:

Because many hunt groups are set up with COS 1, be careful when you assign restrictions to COS 1.

Example command: `change cos-group n`, where *n* is the COS group number.

Related links

[Tenant Partitioning](#) on page 738

Class of service: page 1

change cos-group 1															
Page 1 of 2															
CLASS OF SERVICE	COS Group: 1				COS Name: _____										
	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14
Auto Callback	n	y	y	n	y	n	y	n	y	n	y	n	y	n	y
Call Fwd-All Calls	n	y	n	y	y	n	n	y	y	n	n	y	y	n	y
Data Privacy	n	y	n	n	n	y	y	y	y	n	n	n	n	y	y
Priority Calling	n	y	n	n	n	n	n	n	n	y	y	y	y	y	y
Console Permissions	n	y	n	n	n	n	n	n	n	n	n	n	n	n	n
Off-hook Alert	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Client Room	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Restrict Call Fwd-Off Net	y	y	y	y	y	y	y	y	y	y	y	y	y	y	y
Call Forwarding Busy/DA	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Personal Station Access (PSA)	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Extended Forwarding All	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Extended Forwarding B/DA	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Trk-to-Trk Transfer Override	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
QSIG Call Offer Originations	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Contact Closure Activation	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n

Automatic Callback

Enables or disables Automatic Callback. Using Automatic Callback, internal users who place a call to a busy or an unanswered internal telephone can be automatically called back when the called telephone becomes available.

Automatic Exclusion

Available when Automatic Exclusion is enabled for the system and the applicable station has an assigned **Exclusion** button.

Valid Entry	Usage
y	Activates Automatic Exclusion when a user goes off hook on a station. Automatically prevents other multi-appearance users from bridging onto a call.
n	Denies Automatic Exclusion. Manual exclusion is available when the user presses the Exclusion button before dialing or during a call.

Related links

[Automatic Exclusion by COS](#) on page 304

Call Forwarding All Calls

Used to activate the call forwarding feature for all calls to any extension.

Call Forwarding Busy/DA

If enabled, users can forward calls to any extension when the dialed extension is busy or does not answer.

Client Room

Users can access Check-In, Check-Out, Room Change/Swap, and Maid status functions. In addition, Client Room is required at consoles or telephones that are to receive message-waiting notification. Available only with Hospitality Services and a Property Management System interface.

Console Permissions

Multi-appearance telephone users can control the same features that the attendant controls.

Contact Closure Activation

A user can open and close a contact closure relay. Contact closures control electrical devices remotely. Users dial a Feature Access Code (FAC) on a telephone to activate electrical devices such as electrical door locks.

COS Group

To administer a COS group, Tenant Partitioning must be enabled for the system.

Valid Entry	Usage
1 to 100	The COS Group number.

Related links

[Tenant Partitioning](#) on page 738

COS Name

The identifying name for this COS group. To administer a COS group, Tenant Partitioning must be enabled for the system.

Related links

[Tenant Partitioning](#) on page 738

Data Privacy

If activated, a user can enter a Feature Access Code (FAC) that protects a data call from being interrupted by any of the system override or ringing features.

Extended Forwarding All

A user can administer call forwarding from a remote location for all calls. Available only if Extended Coverage and Forwarding Administration is enabled for the system.

Related links

[Extended Cvg/Fwd Admin](#) on page 732

Extended Forwarding B/DA

A user can administer call forwarding from a remote location when the dialed extension is busy or does not answer. Available only if Extended Coverage and Forwarding Administration is enabled for the system.

Related links

[Extended Cvg/Fwd Admin](#) on page 732

Off-Hook Alert

Enables or disables the requirement that the system send an emergency call to the attendant if the telephone remains off-hook for a prescribed time. Available only if Basic Hospitality or Emergency Access to Attendant is enabled for the system.

Class of Service field does not apply to SIP endpoints. For more information on Off-hook alert feature and its parameters for J100 Series SIP phones, see *Installing and Administering Avaya J100 Series SIP IP Phones in Avaya Aura®*.

Related links

[Hospitality \(Basic\)](#) on page 733

[Emergency Access to Attendant](#) on page 732

Personal Station Access (PSA)

Users can associate a telephone with their extension using their programmed services using a Feature Access Code (FAC). Deactivate for virtual telephones. Required for a user's home station in order for that user to use the Enterprise Mobility User (EMU) feature at other stations. Available only if Personal Station Access (PSA) is enabled for the system.

Related links

[Personal Station Access \(PSA\)](#) on page 736

Priority Calling

If activated, a user can dial a Feature Access Code (FAC) to originate a priority call. Priority calls ring differently and override send all calls.

QSIG Call Offer Originations

If activated, a user can invoke QSIG Call Offer services.

Restrict Call Fwd-Off Net

Valid Entry	Usage
y	A user cannot forward calls to the public network. ! Security alert: For security reasons, this should be enabled for all classes of service except the ones used for special circumstances.
n	A user can forward calls to the public network.

Trk-to-Trk Restriction Override

Use this field to allow or deny user permission to override the system or COR-to-COR calling party restrictions that prohibit the trunk-to-trunk transfer operation for users with this COS.

! Security alert:

Use this COS capability with caution. The ability to perform trunk-to-trunk transfers greatly increases the risk of toll fraud.

Class of service: page 2

change cos-group 1	Page 2 of 2															
	CLASS OF SERVICE															
	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
VIP Caller	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>
Masking CPN/Name Override	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>
Call Forwarding Enhanced	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>
Priority Ip Video	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>
Ad-hoc Video Conferencing	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>
MOC Control:	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>
Match BCA Display To Principal	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>
DCC Activation/Deactivation	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>
Bridging Exclusion Override	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>	<u>n</u>

Ad hoc Video Conferencing

Enables or disables Ad-hoc Video Conferencing, so that up to six users can participate in a video conference call.

Bridging Exclusion Override

When enabled, allows a station to bridge onto a call that had exclusion activated.

Call Forwarding Enhanced

If activated, users can designate different preferred destinations for forwarding calls that originate from internal and external callers.

DCC Activation/Deactivation

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Valid Entry	Usage
y	Activate the Destination Code Control feature.
n	Deactivate the Destination Code Control feature. The default value is n.

Masking CPN/Name Override

Users can be assigned or denied users permission to override the MCSNIC capability. MCSNIC masks the display of calling party information and replaces it with a hard-coded system-wide text string, **Info Restricted**.

Match BCA Display to Principal

Valid entry	Usage
y	The display format of incoming calls on the bridged call appearances of a COS group is <calling name/number>.
n	The display format of incoming calls on the bridged call appearances of a COS group is <calling name/number> to <principal station>.

MOC Control

The **MOC Control** field indicates that the station is under Microsoft Office Communicator (MOC) control.

If a station uses the MOC client and also wants to use an office telephone and an off-PBX telephone, such as a mobile phone simultaneously, set the **MOC Control** field to y.

Valid entry	Usage
y	Station is under MOC control.
n	Station is beyond MOC control. This is the default value.

Priority Ip Video

Activate or deactivate priority video calling. Video calls have an increased likelihood of receiving bandwidth and can also be allocated a larger maximum bandwidth per call.

VIP Caller

Enables or disables automatic priority calling when assigned to the originator of a call. A call from a VIP telephone is always a priority call without the use of a feature button or FAC.

Chapter 27: Cluster Preferred CM List

Use this screen to administer lists of nodes. Each list item on the Cluster Preferred CM List screen is the name of a Communication Manager node. A cluster consists of several nodes.

You can configure this list using `change cluster preferred-CM-list <list#>`. Each node in the list must match the node name that you configured using the System Manager web console. This configuration is necessary for a Kafka integration.

*** Note:**

The Kafka integration feature is reserved for future use.

change cluster preferred-CM-list 3

Page 1 of 1

CLUSTER PREFERRED-CM-LIST

Pref-CM-list Number: 3

Pref-CM-list Name: _____

1: Jay _____

2: _____

3: _____

4: _____

5: _____

Basant

Jay

Ramlal

Sambhaji

(MORE)

ESC-x=Cancel Esc-e=Submit Esc-p=Prev Pg Esc-n=Next Pg Esc-h=Help Esc-r=Refresh

Related links

[Pref-CM-List Number](#) on page 138

[Pref-CM-List Name](#) on page 138

[1-5](#) on page 138

Pref-CM-List Number

A numeric that identifies this node list or cluster.

Related links

[Cluster Preferred CM List](#) on page 137

Pref-CM-List Name

An identifying name for this node list or cluster.

Related links

[Cluster Preferred CM List](#) on page 137

1-5

Type the name of a node in each field. Each node in the list must match the node name that you configured using the System Manager web console.

Related links

[Cluster Preferred CM List](#) on page 137

Chapter 28: Cluster Session Manager

Use this screen to administer Session Managers. You can administer up to 10 clusters of Session Managers by using the Communication Manager CLI. The node names administered on this screen can be either of IPv4 or IPv6 address. For example, if the primary Session Manager points to an IPv4 type address, then the secondary Session Manager must be IPv4 address type. The node names cannot be a mix of IPv4 and IPv6 address types on the same cluster.

To administer cluster Session Manager screen, use the following command:

```
change cluster session-manager
```

135.27.152.160 - PuTTY

change cluster session-manager 1 Page 1 of 1

CLUSTER SESSION MANAGER

Cluster Number: 1

Cluster Name: Default cluster

1: SM-Primary	15: _____
2: SM-Secondary	16: _____
3: _____	17: _____
4:	18: _____
5: _____	19: _____
6: _____	20: _____
7: _____	21: _____
8: _____	22: _____
9: _____	23: _____
10: _____	24: _____
11: _____	25: _____
12: _____	26: _____
13: _____	27: _____
14: _____	28: _____

ESC-x=Cancel Esc-e=Submit Esc-p=Prev Pg Esc-n=Next Pg Esc-h=Help Esc-r=Refresh

Cluster Number

By default, there are 10 clusters.

Cluster Name

Use this field to add Session Manager node names.

Chapter 29: Code Calling IDs

On systems with chime paging, assigns a unique series of chimes to extensions using a chime code. The chime code assigned to an extension plays over the speakers whenever that extension is paged. Assigns chime codes to up to 125 extensions.

Example command: `change paging code-calling-ids`

Ext

Assigns extensions to chime codes. Only one extension can be assigned to each chime code. The extension cannot be assigned to a code that is a Vector Directory Number (VDN).

Chapter 30: Configuration Set

Use the Configuration Set screen to define call treatment options for Extension to Cellular calls. With the Extension to Cellular feature, you can use up to 99 configuration sets that are already defined in the system with default values.

*** Note:**

In the One-X mobile environment, you can edit the values of only the **Cellular voice mail detection** field and the **Call log notify** field. All other fields are read-only.

To modify the values of the fields on the Configuration Set screen, type the following command:

`change off-pbx-telephone configuration-set n`, where *n* is the Configuration Set number.

```
change off-pbx-telephone configuration-set 1 Page 1 of 1

CONFIGURATION SET: 1

Configuration Set Description: _____
Calling Number Style: network
CDR for Origination: phone-number
CDR for Calls to EC500 Destination? y
Fast Connect on Origination? n
Post Connect Dialing Options: dtmf
Cellular Voice Mail Detection: timed (seconds): 4
Barge-in Tone? n
Calling Number Verification? n
Call Appearance Selection for Origination: primary-first
Confirmed Answer? n

Use Shared Voice Connections for Second Call Answered? n Initiated? n
Provide Forced Local Ringback for EC500? n EC500Delay Deactivation State? y
Apply Ringback upon Receipt of: Call-Proceeding

Location to Route Incoming Overlap Calls: station-location-if-set
Feature Invocation by In-Call DTMF Code? y
```

Apply Ringback upon Receipt of

Use the Apply ringback Upon Receipt of field to configure when Communication Manager must send the ringback tone to the caller.

Valid entry	Usage
Call-Proceeding	Communication Manager sends the ringback tone to the caller after receiving the Call-Proceeding message from the EC500 network. Call-Proceeding is the default value.
Alert	Communication Manager sends the ringback tone to the caller after receiving the Alert message from the EC500 network.

Barge-In Tone

Enables or disables a barge-in tone used to add security to Extension to Cellular calls. If a user is on an active Extension to Cellular call and another person joins the call from an Extension to Cellular enabled office telephone, all parties on the call hear the barge-in tone.

Call Appearance Selection for Origination

Specifies how the system selects a Call Appearance for call origination. To use this feature, bridged calls must be enabled for the system.

Valid Entry	Usage
first-available	The system searches for the first available regular or bridged Call Appearance.
primary-first	The system first searches for a regular Call Appearance for call origination. • If all Call Appearances are busy at the primary station, the call cannot be initiated. • If the primary station is logged-off, a second search is made that includes both regular and bridged Call Appearances. This is the default.

Related links

[Bridged Calls](#) on page 693

Call Log Notify

Use this field to update the Avaya one-X® Client Enablement Services logs with actual outpulsed digits and not the digits that the user dials.

Valid entry	Usage
dialed-digits	When you enter dialed-digits, the endpoint displays the dialed digits.
outpulsed-digits	When you enter outpulsed-digits, Communication Manager analyzes the dialed digits and sends the digits to Client Enablement Services for call logging.

Calling Number Style

Determines the format of the caller ID for calls from a local Communication Manager extension to an Extension to Cellular telephone.

Valid Entry	Usage
network	Provides a display of only 10-digit numbers. For internal calls, the ISDN numbering tables are used to create the calling number and DCS calls use the ISDN calling number if provided. The externally provided calling number is used when available for externally originated calls.
pbx	Provides a display of less than 10-digits. Extensions sent as the calling number for all internally- and DCS network-originated calls.

Calling Number Verification

Use this field to impose restrictions on calls originating from an Extension to Cellular-enabled cell phone to some other extension.

Valid entry	Usage
y	If you set the field to y and the network service provider has not marked the screening indicator as network provided or user provided verified and passed, Communication Manager treats the incoming call as an ordinary incoming trunk call and not as a call from an extension.
n	If you set the field to n, Communication Manager treats the incoming call as a call from the corresponding desk phone.

CDR for Calls to EC500 Destination

Determines whether a Call Detail Record (CDR) is generated for any call to the cell telephone. This field is available only if CDR reports are enabled for the trunk group.

Valid Entry	Usage
y	Treats call to the XMOBILE station as trunk call and generates a CDR.
n	Treats call to the XMOBILE station as internal call, without generating a CDR.

Related links

[CDR Reports](#) on page 435

CDR for Origination

Determines the CDR report format when CDR records are generated for a call originating from an Extension to Cellular phone. To generate this CDR, you must enable the Incoming Trunk CDR. The CDR report excludes dialed Feature Name Extensions (FNEs).

Valid Entry	Usage
phone-number	The calling party on the CDR report is the 10-digit cell phone number. This is the default.
extension	The calling party on the CDR report is the internal office telephone extension associated with the Extension to Cellular cell phone.
none	The system does not generate an originating CDR report.

Cellular Voice Mail Detection

You can use this field if you do not want an Extension to Cellular call to cover the cellular voice mail. When the call server detects that the call is covered to the cellular voice mail, the call server brings the call back to the server.

Valid entry	Usage
None	The call covers to the cellular voice mail. This entry is the default value.
Timed	The system displays the Seconds field. You can enter any value from 1 to 9. If the user answers the call on the cellular phone within the configured number of seconds, Communication Manager disconnects the cellular leg of the call. The call continues to ring at the desk phone. You can use this configuration for GSM, CDMA, and ISDN networks.

Table continues...

Valid entry	Usage
Message	The message option works with carriers who use non-ISDN voice mail systems. You must not use this option with ISDN-based voice mail systems.

Configuration Set Description

A description of the purpose of the configuration set. Accepts up to 20 alphanumeric characters.

Example

“Extension to Cellular handsets”

Confirmed Answer

Enables or disables Confirmed Answer on Extension to Cellular calls for this station. If enabled, requires the user to input a digit to confirm receipt of a call sent to a cellular telephone by the Extension to Cellular feature. Upon answering the incoming call on the cellular telephone, the user hears a dial tone. The user must then press any one of the digits on the telephone keypad. Until the system receives a digit, the system treats the call as unanswered. The length of time to wait for the digit can be administered from 5 to 20 seconds, with a default of 10 seconds. The system plays a recall dial-tone to indicate that input is expected. During the response interval, the original call continues to alert at the desk set and any stations bridged to the call. If the user does not enter a digit before the time-out interval expires, the call is pulled back from the cell telephone.

EC500Delay Deactivation State

Use this field to control sharing of the current EC500Delay state to IX Clients.

IX clients use the EC500Delay feature to delay the EC500 termination on the same device on which it has registered the Avaya Aura® application in VoIP mode.

Valid entry	Usage
y	Enables sharing the current EC500Delay state, if EC500Delay feature is deactivated due to any feature interaction. This is the default value.
n	Disables sharing the current EC500Delay state.

Fast Connect on Origination

Use this field to specify whether Communication Manager must perform additional processing before connecting an H.323 or a SIP call.

Valid entry	Usage
y	Communication Manager sends a CONNECT or 200 OK message immediately in response to the SETUP or the INVITE message. The call is connected immediately to the destination.
n	Communication Manager does not send the CONNECT or 200 OK message immediately in response to the SETUP or the INVITE message. The call is connected only when the digits in the SETUP or the INVITE message resolve to an extension or a feature access code.

Feature Invocation by In-Call DTMF Code

Use this field to enable EC500 users to use the in-call features, such as Hold and consult, Transfer, and Conference.

Valid entry	Usage
y	Enables EC500 in-call feature invocation.
n	Disables EC500 in-call feature invocation.

Location for routing incoming overlap calls

With the Location for routing incoming overlap calls feature, the administrator can configure the location to route incoming calls that are mapped to EC500, CSP, and Avaya one-X applications over ISDN-PRI, ISDN-BRI, and H.323 overlap trunks.

The overlap trunk refers to an ISDN trunk group configured for overlap receiving. The **Digit Handling (in/out)** field on the ISDN Trunk Group screen is set to `overlap/overlap` or `overlap/enbloc`.

Valid entry	Usage
Station-location-if-set	Communication Manager uses the station location to process all calls coming from an EC500-mapped phone. Communication Manager uses the station location only if the location is configured on the EC500-mapped station screen. This entry is the default value.

Table continues...

Valid entry	Usage
Trunk	Communication Manager uses the incoming trunk location to process all calls coming from an EC500-mapped phone.

Post Connect Dialing Options

Determines whether additional capabilities, beyond standard ISDN dialing, are available for those incoming ISDN trunk calls that are mapped into XMOBILE stations. These options come into effect after the call has entered the active state (Communication Manager has sent a CONNECT message back to the network).

Valid Entry	Usage
dtmf	Expect digits from either in-band or out-of-band, but not simultaneously. The server allocates a DTMF receiver whenever it needs to collect digits. This option normally would be used for Extension to Cellular XMOBILE station calls.
out-of-band	Expect all digits delivered by out-of-band signaling only. Communication Manager collects digits that it needs from the out-of-band channel (no touch-tone receiver). In addition, any digits received when the server is not collecting digits are converted to DTMF and broadcast to all parties on the call. This option is in force for DECT XMOBILE station calls.
both	Expect all subsequent digits delivered by simultaneous in-band and out-of-band signaling. Out-of-band signaling consists of digits embedded in ISDN INFO messages while the in-band signaling consists of DTMF in the voice path. Communication Manager collects all digits that it needs from the out-of-band channel. No touch tone receive is allocated to prevent collecting double digits. End-to-end signaling occurs transparently to the server via in-band transmission of DTMF. This option is in force for PHS XMOBILE station calls.

Provide Forced Local Ringback for EC500

When a caller calls an enterprise user, the caller expects a standard ringback tone. If the EC500 feature is activated on the telephone of the enterprise user and if the user uses custom ringback tones on the cell phone, the caller hears the custom ringback tone. If you set the **Provide Forced Local Ringback for EC500** field to y, Communication Manager provides the local ringback tone and ignores the custom ringback tone of the cell phone.

Chapter 31: Console parameters

Administers attendant console group parameters. This includes basic parameters for Centralized Attendant Service (CAS) and Inter-PBX Attendant Service (IAS). A list of the administered attendant consoles also displays.

Example command: `change console-parameters`

Console parameters: page 1

AAR/ARS Access Code

An optional AAR/ARS access code used to route to the main switch. Accepts an up to four-digit access code. Available only for a QSIG branch CAS configuration.

Related links

[CAS](#) on page 150

Allow Night Service Calls To Forward

Enables or disables forwarding of DID calls to station users. When Station A receives a DID call and Station A does not answer the call, DID times out and the system routes the call to an attendant. When the attendant is in Night Service mode, the system routes the call to a night service station. If Call Forward is active on the night service station, the system routes the call to Station B. When Call Forward is inactive, the night service station will continue to ring.

Valid Entry	Usage
y	Enables forwarding of DID calls from the night service station.
n	Disables the forwarding of DID calls to station users. This is the default value.

Alternate FRL Station

The extension of the alternate facility restriction level (FRL) activation station.

Attendant Group Name

A name for the attendant group. Accepts up to 27 alphanumeric characters.

Attendant Lockout

Activates or deactivates Privacy — Attendant Lockout. Privacy — Attendant Lockout prohibits an attendant from reentering a conference call that has been placed on hold unless recalled by a telephone user on the call. This feature provides privacy for parties on a multiple-party call held on the console. The held parties can hold a private conversation without interruption by the attendant.

Attendant Vectoring VDN

Assigns the VDN extension for Attendant Vectoring to a console. Available only if Attendant Vectoring is enabled for the system and Tenant Partitioning is disabled.

Related links

[Attendant Vectoring](#) on page 730

[Tenant Partitioning](#) on page 738

Backup Alerting

Used to assign or remove permissions for system users to pick up alerting calls if the attendant queue has reached its warning state.

Calls In Queue Warning

The number of incoming calls that can be in the attendant queue before the console's second Call Waiting lamp lights. The queue maximum depends on system capacities.

CAS

Using the Centralized Attendant Service (CAS), users at separate locations can concentrate attendant positions at one location. Incoming trunk calls to unattended branch locations are routed to the main attendant.

Valid Entry	Usage
main	The main Communication Manager server where the attendant group is located. Uses non-ISDN signaling. CAS Main must be enabled for the system.
branch	A branch Communication Manager server. There are no local attendants, so attendant-seeking calls route to the main Communication Manager server. Uses non-ISDN signaling. CAS Branch must be enabled for the system
none	Disables CAS.
QSIG-main	Same as main, but with QSIG signaling among the Communication Manager servers. Centralized Attendant must be enabled for the system.
QSIG-branch	Same as branch, but with QSIG signaling among the Communication Manager servers. Centralized Attendant must be enabled for the system.

Related links

[Centralized Attendant](#) on page 744

[RLT Trunk Group No.](#) on page 153

CAS Back-Up Ext

Valid Entry	Usage
Extension number Individual attendant console Hunt group TEG	Extension in the dial plan that handles attendant-seeking calls if the RLT trunk group to the CAS Main server is out of service or if CAS Back-Up is activated. Neither a prefixed extension nor a VDN extension is supported. Available only for the CAS Branch feature.

Related links

[CAS Branch](#) on page 730

[CAS Main](#) on page 730

[Centralized Attendant](#) on page 744

COR

Valid Entry	Usage
0 to 995	The class of restriction (COR) number for all attendant consoles. The COR for the individual Attendant Console overrides this value.

COS

Valid Entry	Usage
1 to 15	The class of service (COS) number that reflects the required features for all your attendant consoles. The COS for the individual Attendant Console overrides this value.

DID-LDN Only to LDN Night Ext

Valid Entry	Usage
y	Only listed directory number (LDN) calls go to the listed directory night service extension.
n	All attendant seeking calls route to the LDN night service extension.

Ext Alert Port (TAAS)

The seven-digit port number assigned to the external alerting device. This supports the Night Service — Trunk Answer From Any Station feature.

An x indicates that there is no hardware associated with this port assignment. If an x is used, there should also be a value for **Ext Alert (TAAS) Extension**.

Related links

[Ext Alert \(TAAS\) Extension](#) on page 152

Ext Alert (TAAS) Extension

This extension is used by the Terminal Translation Feature (TTI) to assign a port to the Ext Alert Port from a station on the Ext Alert port during system installation or provisioning. Once a port is assigned either through TTI or by entering the Ext Alert Port, the extension is automatically removed and treated as unassigned. Available only when the Ext Alert Port (TAAS) has associated hardware.

Related links

[Ext Alert Port \(TAAS\)](#) on page 151

IAS Att. Access Code

The extension number of the attendant group at the main server running Communication Manager. Required when IAS Branch is enabled. Unavailable if the Centralized Attendant feature is enabled.

Related links

[IAS \(Branch\)](#) on page 152

[Centralized Attendant](#) on page 744

IAS (Branch)

Enables or disables the Inter-PBX Attendant Service (IAS) Branch feature. Unavailable if Centralized Attendant is enabled.

Related links

[Centralized Attendant](#) on page 744

IAS Tie Trunk Group No.

Unavailable if the Centralized Attendant feature is enabled.

Valid Entry	Usage
1 to 2000	The number of the tie trunk group to the main for the IAS (Branch). Required when IAS Branch is enabled.

Related links

[Centralized Attendant](#) on page 744

Night Service Act. Ext.

The extension of the current night service activation station, if any. The station is administered by assigning it a **night-serv** button.

QSIG CAS Number

Contains the complete number of the attendant group at the main server running Communication Manager, or a Vector Directory Number (VDN) local to the branch server. Accepts up to 20 digits. Cannot be left blank. Available only for an QSIG-branch CAS configuration.

Related links

[CAS](#) on page 150

RLT Trunk Group No.

The trunk group number corresponding to the Release Link Trunk (RLT) trunk group to the main location when supporting CAS Branch service. Available only for a branch CAS configuration.

Related links

[CAS](#) on page 150

SIP

Use this field to configure SIP support on attendant consoles.

* Note:

If native attendant like type “302” or “console” is already configured in the same Attendant Group, then the Communication Manager will not allow you to enable SIP on Console Parameters form. Also, if SIP is enabled on the Console Parameters form, Communication Manager will not allow you to add a new attendant user of type “302” or “console” in the same Attendant Group. Before SIP is enabled, Attendant Vectoring VDN field must be blank.

Valid Entry	Usage
n	Disables SIP attendant. This is the default value.
y	Enabling the SIP Attendant field will cause attendant seeking calls to route to the designated number assigned in the associated field. When you change the value to y, then the following values appear: • Routing Digits • Attendant Server • Return Call timeout on page 2 When you set the SIP value to y, ensure that: • The Tenant Partitioning feature is disabled on the system parameters customer options screen. • All native attendants are removed, if available. • Communication Manager blocks new attendant if SIP is enabled on console parameters screen.

Routing Digits

Access code to route calls to SIP attendant server. You can enter up to four digits only.

The Routing Digits field appears on the Console Parameters form, only if the Tenant Partition is disabled.

Valid Entry	Usage
digits from 0 to 9	Digits comprising the AAR or ARS access code. The default value is blank.
blank	Call routing to attendant server is not specified.

Attendant Server

Attendant server number. This field cannot be blank.

The Attendant Server field appears on the Console Parameters form, only if the Tenant Partition is disabled.

Valid Entry	Usage
digits from 0 to 9	You can enter up to 17 digits.

Console parameters: page 2

ABBREVIATED DIALING

List1, List2, List3

Assigns up to three abbreviated dialing lists to each attendant. A personal list cannot be assigned to an attendant.

Valid Entry	Usage
enhanced	The attendant can access the enhanced system abbreviated dialing list.
group	The attendant can access the specified group abbreviated dialing list. A group number is required.
system	The attendant can access the system abbreviated dialing list.

SAC Notification

Enables or disables Enhanced Attendant Notification for Send All Calls.

COMMON SHARED EXTENSIONS

Busy Indicator for Call Parked on Analog Station Without Hardware?

Enables or disables the Busy Indicator lamp that lights for incoming calls parked on Administration Without Hardware (AWOH) stations.

Count

Valid Entry	Usage
1 to 1182 blank	The number of consecutive extensions, beginning with the Starting Extension used as common, shared extensions

Example

If you enter a starting extension of 4300 and a count of 3, the system provides three consecutive extension numbers (4300, 4301, and 4302) for parking calls.

Related links

[Starting Extension](#) on page 155

Starting Extension

The first extension number in a group of consecutive extensions that can be used by the attendant to park calls.

Related links

[Count](#) on page 155

INCOMING CALL REMINDERS

Alerting (sec)

The number of seconds after which a held or unanswered call is disconnected from an attendant loop and routed to another attendant or night service.

No Answer Timeout (sec)

Valid Entry	Usage
10 to 1024 blank	The number of seconds a call to the attendant can remain unanswered without invoking a more insistent sounding tone. With this option, the call rings for 5 seconds at all points in a coverage path to ensure that the entire path is completed before the call returns to the console.

Secondary Alert on Held Reminder Calls?

Valid Entry	Usage
y	Begin attendant alerting for Held Reminder Calls with secondary alerting.
n	Have held reminder calls alert the attendant the same as normal calls. Normal calls start with primary alerting and then switch to secondary alerting when an administered timeout expires.

Related links

[No Answer Timeout \(sec\)](#) on page 155

TIMING

Overview timer to Group Queue (sec)

Valid Entry	Usage
10 to 1024	The number of seconds a returning call queues to the individual attendant before overflowing to the group. The value applies if the attendant who previously handled the call is busy or unavailable.
blank	The call immediately goes to the group.

Return Call Timeout (sec)

Valid Entry	Usage
10 to 1024 blank	The time in seconds before a split away call (call extended and ringing a station or otherwise split away from the console) returns to the console. With this option, the call rings for 5 seconds at all points in a coverage path to ensure that the entire path is completed before the call returns to the console. This field applies even when SIP is set to 'y' on Console Parameters: page 1.

Time In Queue Warning (sec)

Valid Entry	Usage
9 to 999 blank	The number of seconds a call can remain in the attendant queue before activating an alert.

Time Reminder on Hold (sec)

Valid Entry	Usage
10 to 1024	The number of seconds a call can remain on Hold.

Console parameters: page 3

Call-Type Ordering Within Priority Levels?

Groups calls to the attendant in the following order:

1. Queue priority level
2. Call type
3. Order received

Call type	Description
Type 1 call	Outgoing public-network calls receiving answer supervision when the Answer Supervision Timer of the trunk group expires, even if the trunk is actually still ringing. Also, incoming calls when answered by the attendant.
Type 2 call	Incoming external public-network calls before they receive answer supervision or before the Answer Supervision Timer of the trunk group expires.
Type 3 call	All other calls (internal calls, conference calls, and tie-trunk calls of any type). Note that external public-network calls have priority over all other calls including conference calls. Answered public-network calls have priority over unanswered calls.

Valid Entry	Usage
y	Orders calls by call type within each queue priority category. A type-disp button is assigned to the Attendant Console. The attendant can review the call type for the active call.
n	Calls are queued in chronological order by queue priority level.

QUEUE PRIORITIES

Assigns a priority level from 1 through 13 to each call category when the call cannot be immediately terminated to an attendant. The calling party hears ringback until an attendant answers the call.

The same priority level can be assigned to more than one call.

Assistance Call

Valid Entry	Usage
1 to 13	Assigns a priority level for assistance calls. A call from a telephone user who dials the attendant-group access code, or from a telephone that has the Manual Originating Line Service feature activated. Priority 1 is the highest priority.

DID to Attendant

Valid Entry	Usage
1 to 13	Assigns a priority level for DID to Attendant calls that are incoming DID trunk calls to an attendant group. This excludes trunk calls that return to the attendant group after a timeout or deferred attendant recall. Priority 1 is the highest priority.

Emergency Access

Valid Entry	Usage
1 to 13	Assigns a priority level for emergency access calls that are calls from a telephone user who dials the emergency access code. The default is 1. Priority 1 is the highest priority.

Individual Attendant Access

Valid Entry	Usage
1 to 13	Assigns a priority level for Individual Attendant Access calls that are calls from a telephone user, incoming trunk call, or a system feature to the Individual Attendant Access (IAA) extension of a specific attendant. If the attendant is busy, the call queues until the attendant is available. Priority 1 is the highest priority.

Interposition

Valid Entry	Usage
1 to 13	Assigns a priority level for Interposition calls that are calls from one attendant to the Individual Attendant Access (IAA) extension of another attendant. Priority 1 is the highest priority.

Miscellaneous Call

Valid Entry	Usage
1 to 13	Assigns a priority level for any unlisted calls. Priority 1 is the highest priority.

Redirected Call

Valid Entry	Usage
1 to 13	Assigns a priority level for any calls assigned to one attendant, but redirected to the attendant group because the attendant is now busy. Priority 1 is the highest priority.

Redirected DID Call

Valid Entry	Usage
1 to 13	Assigns a priority level for any DID or ACD calls that time out due to ring/no-answer, busy condition (if applicable), or Number Unobtainable. Calls are rerouted to the attendant group. Priority 1 is the highest priority.

Return Call

Valid Entry	Usage
1 to 13	Assigns a priority level for any calls returned to the attendant after timing out. If the attendant is now busy, the call redirects to the attendant group. Priority 1 is the highest priority.

Serial Call

Valid Entry	Usage
1 to 13	Assigns a priority level for any calls from the Attendant Serial Call feature when an outside trunk call (designated as a serial call by an attendant) is extended to and completed at a telephone, and then the telephone user goes on-hook. If the attendant who extended the call is busy, the call redirects to the attendant group. Priority 1 is the highest priority.

Tie Call

Valid Entry	Usage
1 to 13	Assigns a priority level for incoming TIE trunk calls (dial-repeating or direct types) to an attendant group. This excludes trunk calls that return to the attendant group after a timeout or deferred attendant recall. Priority 1 is the highest priority.

VIP Wakeup Reminder Call

Valid Entry	Usage
1 to 13	Assigns a priority level for VIP Wakeup Reminder Calls from the Hospitality feature that send a wake-up reminder to the attendant to call a hotel or motel room. Priority 1 is the highest priority.

Console parameters: page 4

QUEUE PRIORITIES

Flash

Valid Entry	Usage
1 to 17	The queue priority for Flash precedence level calls.

Flash Override

Valid Entry	Usage
1 to 17	The queue priority for Flash Override precedence level calls.

Immediate

Valid Entry	Usage
1 to 17	The queue priority for Immediate precedence level calls.

Priority

Valid Entry	Usage
1 to 17	The queue priority for Priority precedence level calls.

Console parameters: page 5

**Note:**

If MLPP is disabled, the system does not display the MLPP Queues page, but displays page with the following message:

Use the 'list attendant' command to see all administered attendants.

Chapter 32: Coverage Answer Group

Establishes Call Coverage Answer Groups.

An answer group contains up to 100 members who act as a coverage point for another user. For example, if several secretaries are responsible for answering a department's redirected calls, all the secretaries could be assigned to an answer group. The answer group is assigned a group number, and the department's coverage path displays that group number. All telephones in an answer group ring (alert) simultaneously. Any member of the group can answer the call.

Each coverage answer group is identified by a number from 1 through the maximum number allowed by your system configuration. The members of the group are identified by their extension number. Any telephone, including those administered without hardware (but not attendants) can be assigned to a coverage answer group.

Example command: `change coverage answer-group n`, where *n* is the assigned group number.

add coverage answer-group 1

Page 1 of 1

COVERAGE ANSWER GROUP

Group Number: 1

Group Name: COVERAGE GROUP

GROUP MEMBER ASSIGNMENTS

Extension	Name
1: _____	
2: _____	
3: _____	
4: _____	
5: _____	
6: _____	
7: _____	
8: _____	

Ext

The extension number for each member of this coverage answer group. This number cannot be a Vector Directory Number (VDN) extension.

Group Name

The group name used to identify this group. Accepts up to 27 characters. Use the extension numbers of group members as the group name to help when determining which stations are involved in call coverage and trunk coverage paths.

Example

“typing pool”, “room 12”, “secy”

Group Number

The number associated with the Cover Answer Group.

Name

The name assigned when the member’s telephone was administered.

Chapter 33: Coverage Path

Implements Call Coverage Paths by providing the means to specify the call coverage criteria, the points in the coverage path used to redirect calls, and the number of times a principal's telephone rings before the call redirects to coverage.

Example command: `change coverage path n`, where *n* is the assigned coverage path number.

change coverage path 1

Page 1 of 1

COVERAGE PATH

Coverage Path Number: 1

Description: [Admin12345](#)

Cvg Enabled for VDN Route-To Party? n

Hunt after Coverage? n

Next Path Number:

Linkage

COVERAGE CRITERIA

Station/Group Status	Inside Call	Outside Call	
Active?	<u>n</u>	<u>n</u>	
Busy?	<u>y</u>	<u>y</u>	
Don't Answer?	<u>y</u>	<u>y</u>	Number of Rings: <u>1</u>
All?	<u>n</u>	<u>n</u>	
DND/SAC/Goto Cover?	<u>y</u>	<u>y</u>	
Holiday Coverage?	<u>n</u>	<u>n</u>	

COVERAGE POINTS

Terminate to Coverage Pts. with Bridged Appearances? n

Point1:

Point2:

Point3:

Point4:

Point5:

Point6:

Coverage Path Number

The coverage path being administered.

Coverage path description

The name for the coverage path. Accepts alphanumeric characters and must contain at least one alphabetical and one numeric character.

Cvg Enabled for VDN Route-To Party

Enables or disables the route-to party coverage path after a covered call hits a VDN vector route-to step. By default, the value is n.

Holiday Coverage

Use the **Holiday Coverage** field to redirect all calls during a holiday to a coverage path. For **Holiday Coverage** to function, set the **Don't Answer** field to y.

You must set the **Holiday Coverage** field separately for internal and external calls.

Valid Entry	Usage
y	Communication Manager checks the Holiday Table screen for a specific holiday entry. If an entry on the Holiday Table screen matches with the current date and time, Communication Manager forwards the call to the first point that is defined in the coverage path. If there is no entry that matches with the current date and time, Communication Manager forwards the call to the subsequent point that is defined in the coverage path.
n	Communication Manager forwards the call to the subsequent point in the coverage path.

Holiday Table

Available only when **Holiday Coverage** is set to y for inside or outside calls.

The number of the holiday table used for holiday coverage.

Hunt After Coverage

Valid Entry	Usage
y	Coverage treatment continues by searching for an available station in a hunt chain that begins with the hunt-to-station assigned to the station of the last coverage point.
n	Coverage treatment is terminated. The call is left at the last available location, the principal or coverage point.

Linkage

One or two additional coverage paths in the coverage path chain.

Next Path Number

Valid Entry	Usage
1 to 9999	The number of the next coverage path in a coverage path chain. If the coverage criteria of the current coverage path is dissatisfied, the system checks in this chain until it finds a coverage path with redirection criteria that matches the call status. If the chain is exhausted before the system finds a match, the call stays out of coverage.
blank	The only path for the principal.

Criteria

Active

Use this field to assign a coverage criteria. When the coverage criteria is met, the system redirects the call to coverage.

Valid Entry	Usage
y	The system redirects the call if at least one call appearance is busy.
n	The system does not redirect the call.

Busy

Use this field to assign a coverage criteria. When the coverage criteria is met, the system redirects the call to coverage.

Valid Entry	Usage
y	The system redirects the call if all call appearances that accept incoming calls are busy.
n	The system does not redirect the call.

Don't Answer

Use this field to assign a coverage criteria. When the coverage criteria is met, the system redirects the call to coverage.

Valid Entry	Usage
y	The system redirects the call when the specified number of rings have been exceeded.
n	The system does not redirect the call.

All

Use this field to assign a coverage criteria. When the coverage criteria is met, the system redirects the call to coverage.

Valid Entry	Usage
y	The system redirects all calls to coverage. This option overrides any other criteria. Calls redirect immediately to coverage. Overrides any other criteria administered for this field.
n	The system does not redirect the call.

DND/SAC/Go to Cover

Use this field to assign a coverage criteria. When the coverage criteria is met, the system redirects the call to coverage.

Valid entry	Usage
y	With this option, a calling user, when calling to another internal extension, can redirect a call immediately to coverage by pressing the Go to Cover button. A principal user can temporarily direct all incoming calls to coverage, regardless of the other assigned coverage criteria by pressing the Send All Calls or Do Not Disturb button. With the Send All Calls button, covering users can temporarily remove their telephones from the coverage path.  Note: You must assign this criteria before a user can activate Do Not Disturb (Hospitality Services), Send All Calls (SAC), or Go to Cover features.
n	The system does not redirect the call.

Logged off/PSA/TTI

Use this field to assign a coverage criteria. When the coverage criteria is met, the system redirects the call to coverage.

The system displays this field only when you set the **Criteria for Logged Off/PSA/TTI Stations** field to y.

Valid Entry	Usage
y	The system redirects the call after the number of rings exceeds the value specified in the Number of Rings field. The system displays the associated Number of Rings field only when the Logged off/PSA/TTI field is set to y.
n	The system does not redirect the call.

Number of Rings

Valid Entry	Usage
1 to 99	The number of times a telephone rings before the system redirects the call to the first point in the coverage path. By default, the value is 2.

Coverage points

Point1, Point2, Point3, Point4, Point5, Point6

The alternate destinations that comprise a coverage path. Coverage points must be assigned sequentially without steps beginning with Point 1. Each path can have up to six coverage points.

Subsequent coverage points should be unlisted if calls are redirected to:

- Message Center, a special Uniform Call Distribution hunt group
- Voice messaging

- The attendant

These calls normally queue and never redirect to another coverage point. Calls to hunt group queue if possible. Calls redirect from a hunt group only if all hunt group members are busy and either the queue is full, or is nonexistent.

If the Coverage of Calls Redirected Off-Net feature is not supported, a remote coverage point functions as the last point in the coverage path because the system can no longer control calls once they redirect off-net. However, if the Coverage of Calls Redirected Off-Net feature is enabled, calls redirected off-net can be monitored by the system and brought back for call coverage processing.

Valid Entry	Usage
extension	Redirects the call to an internal extension or announcement. * Note: If you enter a shortened extension of the multilocation dial plan, the system does not perform certain administration and validation tasks. Therefore, the system might not display the resultant warnings or submittal denials.
attd	Redirects the call to the attendant or attendant group. If the system has Centralized Attendant Service (CAS), the call goes to the CAS attendant.
h1 to h8000	Redirects the call to the corresponding hunt-group, for example, h32 routes to hunt group 32.
c1 to c1500	Redirects the call to the corresponding coverage answer group, for example, c20 routes to call coverage answer group 20.
r1 to r10000	Redirects the call to the corresponding remote coverage point number, for example, r27 routes to remote coverage point 27.
v + extension	Redirects the call to the corresponding Vector Directory Number (VDN) extension, for example, v12345 routes to the VDN associated with extension 12345. * Note: A VDN can be used only as the last administered point in a coverage plan.
y + extension	Redirects the call to an internal extension, announcement, or the corresponding Vector Directory Number (VDN) extension as per the current date and time set in Holiday Table.

Rng

Valid Entry	Usage
1 to 99 blank	The number of rings at this coverage point before the system redirects the call to the next point in the coverage path.

Terminate to Coverage Pts. with Bridged Appearances

Valid Entry	Usage
y	If activated, a call can alert as both a bridged call and a redirected call.
n	The call skips the coverage point if it has already alerted as a bridged call.

Chapter 34: Crisis Alert System Parameters

Defines the system parameters associated with sending crisis alert messages.

Example command: `change system-parameters crisis-alert`

```
change system-parameters crisis-alert Page 1 of 1 ^
CRISIS ALERT SYSTEM PARAMETERS

ALERT STATION
  Every User Responds? y

ALERT PAGER
  Alert Pager? y
  Originating Extension:   Retries: 0
  Crisis Alert Code:   Retry Interval(sec): 30
  Main Number:  

  ** Do not include ARS Access Code in Pager Number **
  Pager Number      Pin Number
  1:              1:  
  2:              2:  
  3:              3:  

  DTMF Duration - Tone (msec): 100 Pause (msec): 100

SNMP NOTIFICATION
  SNMP Inform to Notify Adjunct When DCP and H.323 Stations Go In-Service? y
  SNMP Inform to Notify Adjunct When SIP Station Dials Emergency Call? n
```

* Note:

From Release 10.1.0.2 onwards, if the **Notification using Crisis Alert** field on the Feature-Related System Parameters screen, page 5 is enabled, you cannot edit **Every User Responds?** and **Alert Pager** fields, and the Crisis Alert System Parameter screen displays the following text:

Crisis Alert is disabled for Emergency calls. To enable it, disable "Notification using Crisis Alert" on page 5 of the Feature-Related System Parameters screen.

ALERT STATION

Every User Responds

Controls who needs to respond to a crisis alert.

Valid Entries	Usage
y	All users who have a crisis alert button are notified and must clear the alert for every emergency alert. Crisis alert buttons should be assigned only to attendant consoles and stations that must be notified of an emergency call.
n	All users are notified, but only one user needs to acknowledge an alert. This user might be the attendant or any other digital telephone with a crisis alert button. When the alert is acknowledged by one user, the alert is cleared at all stations except the one that acknowledged the alert.

ALERT PAGER

Alert Pager

Supports or restricts use of the Crisis Alert to a Digital Pager.

Crisis Alert Code

The first three digits in the crisis alert pager message. This should be the numbers used to call the local emergency service or any digits used for an emergency situation (for example, 911).

Required when Crisis Alert to a Digital Pager is enabled.

Related links

[Alert Pager](#) on page 171

DTMF Duration - Tone (msec)

Available only when Crisis Alert to a Digital Pager is enabled.

Valid Entry	Usage
20 to 2550 (in increments of 10)	The length of time the Dual-Tone Multi-Frequency (DTMF) tone is heard for each digit.

Related links

[Alert Pager](#) on page 171

Main Number

Identifies the location where the crisis alert call originated. It can be the main number to the location or a numerical identification. Any dashes are for display purposes only and excluded from the message sent to the pager. This entry is the last group of digits displayed in the pager message. Accepts up to 15 digits. Available only when Crisis Alert to a Digital Pager is enabled.

Related links

[Alert Pager](#) on page 171

Originating Extension

Required when Crisis Alert to a Digital Pager is enabled.

Valid Entry	Usage
1 to 16	Requires a valid unassigned extension according to the dial plan

Related links

[Alert Pager](#) on page 171

Pager Number

The number which Communication Manager dials to make the call to the pager. The ARS access code is automatically pre-pended to the **Pager Number** before Communication Manager dials the call.

Pause (msec)

Available when Crisis Alert to a Digital Pager is enabled.

Valid Entry	Usage
20 to 2550 (in increments of 10)	The length of time between DTMF tones for each digit.

Related links

[Alert Pager](#) on page 171

Pin Number

If the pager service requires you to submit a PIN number, type the number in this field. The PIN number can be up to 15 digits. You can also enter the p (pause), # and * characters. A pause of up to 2 seconds can be used for the timing of the message. For instance, a pause might be necessary to allow time for the pager service to set up the correct pager message box.

The system displays this field only if Crisis Alert to a Digital Pager is enabled.

Related links

[Alert Pager](#) on page 171

Retries

Available when Crisis Alert to A Digital Pager is enabled.

Valid Entry	Usage
0 to 10	The number of times the system tries to send out the alert message in case of an unsuccessful attempt. This increases the chances that the pager receives a crisis alert message.

Related links

[Alert Pager](#) on page 171

Retry Interval (sec)

Available when the Crisis Alert to a Digital Pager is enabled and **Retries** is set to a value from 1 to 10.

Valid Entry	Usage
30 to 60	The time in seconds between retries. If an attempt to call the pager fails, the retry call attempts after the retry interval period.

Related links

[Alert Pager](#) on page 171

[Retries](#) on page 173

SNMP NOTIFICATION

SNMP Inform to Notify Adjunct When DCP and H.323 Stations Go In-Service

Enables or disables sending SNMP informs to a trap receiver when a DCP station comes into service, or an H.323 station is registered.

Valid Entries	Usage
y	Communication Manager sends SNMP informs to a trap receiver when a DCP station comes into service, or an H.323 station is registered.
n	Communication Manager does not send SNMP informs to a trap receiver when a DCP station comes into service, or an H.323 station is registered. This is the default value.

SNMP Inform to Notify Adjunct When SIP Station Dials Emergency Call

Controls sending inform traps to SNMP receiver when a SIP station places an emergency call.

 Note:

Ensure that, in the FP Traps screen of the Communication Manager System Management Interface (SMI), you set the **Notification** field to **Inform** for SNMP v3

Valid Entries	Usage
y	Communication Manager sends inform traps to SNMP receiver when a SIP station places an emergency call.
n	Communication Manager does not send inform traps to SNMP receiver when a SIP station places an emergency call. This is the default value.

Chapter 35: CTI Link

Available if either ASAI Link Core Capabilities or Computer Telephony Adjunct Links are enabled on the system.

Example command: `add cti-link n`, where *n* is the CTI link number.

Related links

[ASAI Link Core Capabilities](#) on page 729

[Computer Telephony Adjunct Links](#) on page 731

CTI link: page 1

The screenshot shows a terminal window with the command `change cti-link 1` at the top left. The top right corner displays `Page 1 of 3`. The main content area is titled `CTI LINK` and contains the following fields: `CTI Link: 1`, `Extension: 531-9090`, `Type: ADJ-IP` (with a horizontal line for editing), and `Name: _____` (with a horizontal line for editing). On the right side, there is a `COR: 1` field with a horizontal line for editing. The bottom of the window has a thick black bar.

If the solution uses clustering, the screenshot below is displayed.

```

- PuTTY
add cti-link 1 Page 1 of 3
CTI LINK
CTI Link: 1
Extension:
Type: ADJ-IP
Name:
Unicode Name? n
AES host:
COR: 1

```

*** Note:**

The **AES host** field is reserved for future use.

AES host

This is the hostname of Application Enablement Services (AES), without the domain. It should be fewer than 15 characters. AES does not connect directly with the Avaya Aura® CM in Avaya Aura® Distributed Architecture. Instead it connects via the Kafka cluster and uses the AES Host name to connect to the correct Kafka topic.

*** Note:**

The **AES host** field is reserved for future use.

COR

Class of Restriction (COR) number with the required restriction.

CTI Link

The Computer Telephony Integration (CTI) link number.

Extension

The extension for this link.

Name

The name associated with this CTI link.

Type

Use this field to specify the type of Computer Telephony Integration (CTI) link.

Valid entry	Usage
ADJ-IP	CTI links administered without hardware.
ASAI-IP	CTI links administered without hardware. The associated adjunct controls agent logins, and the logged-in agents can use the data terminal keyboards to perform telephone functions.

CTI link: page 2

change cti-link 1

Page 2 of 3

CTI LINK

FEATURE OPTIONS

Event Minimization? n

IC Adjunct Routing? n

Special Character for Restricted Number? n

Send Disconnect Event for Bridged Appearance? n

Two-Digit Aux Work Reason Codes? n

Block CMS Move Agent Events? n

Remove '+' from SIP Numbers? y

Block CMS Move Agent Events

Enables or disables the blocking of certain event report messages involved with the move of agents while staffed. If the Call Management System sends an agent-move-while-staffed message

(MVAGSFD8), ASAI does not send the associated agent Logout Event Report (C_Logout), Login Event Report (C_login) and Agent Work Mode Change event report messages.

Event Minimization

Enables or disables event minimization for this link that limits the number of event reports sent to an adjunct. This option can be used when event reports normally would be sent on multiple associations, but the adjunct can see only a single report. Typically, these event reports are identical except for the association they are sent over (for example, call control, domain control, or active notification). Some applications discard duplicate events, so in this case, there is no point in sending them across the ASAI CTI link. When activated, only a single such event can be sent. The selection of the association on which the event is sent is based on association precedence as follows: active notification (if enabled), call control (if enabled), or domain control (if enabled).

IC Adjunct Routing

IC Adjunct Routing enables or disables the Enhanced Adjunct Routing wait sequence. When disabled, the wait sequence consists of only one wait or announcement step following one or more adjunct-route steps. When enabled, the wait sequence can contain multiple wait steps, announcement steps, goto steps, and the collect step. If the adjunct route step is successful, the adjunct must respond before the wait sequence completes. Otherwise, the adjunct route step is canceled. If the adjunct route step fails, the vector steps in the wait sequence are skipped.

Remove '+' from SIP Numbers

If you set the **Remove '+' from SIP Numbers** field on the CTI Link screen to y and if a leading plus (+) sign is present in a calling or called number sent over a CTI link, Communication Manager removes the leading plus (+) sign from the number. The default value is y.

Send Disconnect Event for Bridged Appearance

Indicates whether or not an event report is sent when a bridged appearance disconnects.

Special Character for Restricted Number

Enables or disables an ASAI CTI link that indicates the calling number restricted presentation within an event report. When enabled, a calling number received in a SETUP message has the presentation indicator set (octet 3a in the calling number), then "*" is appended to the calling party number in the ASAI message.

Two-Digit Aux Work Reason Codes

Use this field to send a two-digit Reason code over the ASAI link. All messages that include the Aux-work Reason code allow codes from 1 to 99.

Related links

[Two-Digit Aux Work Reason Codes](#) on page 297

CTI link: page 3

change cti-link 1 Page 3 of 3

CTI LINK

Bridged Appearance Origination Restriction? n

SAC/CF Override: n

Bridged Appearance Origination Restriction

Restricts or allows call origination on the bridged appearance.

Valid Entry	Usage
y	Call origination on the bridged appearance is restricted.
n	Call origination on the bridged appearance is allowed. This is normal behavior, and is the default.

SAC/CF Override

With **SAC/CF Override**, the user of the calling station can override the redirection set by the called station.

Valid entry	Usage
ask	The system prompts the user of the calling station whether the call must follow the redirection path or override the redirection path. The user can type y or n.
no	The user of the calling station cannot override the redirection path of the call. The call follows the redirection path.
yes	The user of the calling station can override the redirection path of the call, provided the called station has at least one idle call appearance.

Chapter 36: Date and Time

Displays the system date and time, selects the daylight saving plan number, if any, and shows whether the current time is standard time or daylight saving. Settings on this screen affect the internal clock and timestamp of the server running Communication Manager. Update the date and time for a leap year or a system restart after a power failure. The correct date and time assure that CDR records are correct. CDR does not work until you enter the date and time.

Example command: `set time`

```
set time                                     Page 1 of 1
                                     DATE AND TIME

DATE

    Day of the Week: Friday           Month: December
    Day of the Month: 21             Year: 2012

TIME

    Hour: 13   Minute: 29   Second: 7   Type: Standard
                                     Daylight Saving Rule: 0

WARNING: Changing the date or time may impact BCMS, CDR, SCHEDULED EVENTS,
and MEASUREMENTS
```

Day of the Month

Valid entry	Usage
1 to 31	The current day of the month. The system clock uses this as the current date.

Day of the Week

Valid Entry	Usage
Sunday through Saturday	The current day of the week. The system clock uses this as the current day.

Daylight Saving Rule

Valid Entry	Usage
0 to 15	The daylight saving rule in use for the system. The system clock uses this as the current daylight saving rule.

Related links

[Daylight Saving Rules](#) on page 183

Hour

Valid Entry	Usage
0 to 23	The current hour to be used by the system clock. The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.

Minute

Valid Entry	Usage
0 to 59	The current minute. The system clock uses this as the current minute.

Month

Valid Entry	Usage
January to December	The current month. The system clock uses this as the current month.

Second

Displays the seconds and cannot be modified. Resets to zero when saved.

Type

Valid Entry	Usage
daylight-savings	Indicates daylight saving time is in effect.
standard	Indicates standard time is in effect.

Year

The current year in 20XX format. The system clock uses this as the current year.

Chapter 37: Daylight Saving Rules

Administers up to 15 customized daylight saving rules. Specifies the exact date and time each daylight saving rule goes into effect and when it stops. Rule 0 makes no adjustment to the system clock for daylight saving and cannot be modified.

In this example, Rule 1 applies to all time zones in the U.S. and begins on the first Sunday on or after March 8 at 2:00 a.m. with a 01:00 increment. Daylight Saving Time stops on the first Sunday on or after November 1 at 2:00 a.m., also with a 01:00 increment used as a decrement when switching back to Standard time. Telephone displays reflect these settings.

```
1: Start: first _ Sunday_ on or after _March 8_ at 2:00_ 01:00__  
Stop: first _ Sunday_ on or after November 1_ at 2:00_
```

Example command: `change daylight-savings-rules`

change daylight-savings-rules						Page 1 of 2
DAYLIGHT SAVING RULES						
Rule	Change Day	Month	Date	Time	Increment	
0: No Daylight Saving						
1:	Start: first Sunday	on or after	March 8	at 02:00	01:00	
	Stop: first Sunday	on or after	November 1	at 02:00		
2:	Start: first	on or after		at	:	:
	Stop: first	on or after		at	:	:
3:	Start: first	on or after		at	:	:
	Stop: first	on or after		at	:	:
4:	Start: first	on or after		at	:	:
	Stop: first	on or after		at	:	:
5:	Start: first	on or after		at	:	:
	Stop: first	on or after		at	:	:
6:	Start: first	on or after		at	:	:
	Stop: first	on or after		at	:	:
7:	Start: first	on or after		at	:	:
	Stop: first	on or after		at	:	:

Change Day (Start)

Valid Entry	Usage
Sunday to Saturday	The day of the week the clock moves ahead to begin daylight saving.
Day	The clock changes on the exact date entered for Month (Start) and Date (Start) values.

Related links

[Month \(Start\)](#) on page 185

Change Day (Stop)

Valid Entry	Usage
Sunday to Saturday	The day of the week the clock moves back to standard time
Day	The clock changes on the exact date entered for the Month (Stop) and Date (Stop) values

Related links

[Month \(Stop\)](#) on page 185

Date (Start)

Valid Entry	Usage
0 to 31	The day of the month the clock moves ahead to begin daylight saving.

Date (Stop)

Valid Entry	Usage
0 to 31	The day of the month the clock moves back to standard time

Increment (Start)

Valid Entry	Usage
0 to 23	The number of hours the clock moves ahead for daylight saving and moves back to return to standard time.
0 to 59	The number of minutes you want the clock to move ahead for daylight saving and to move back to return to standard time.

Month (Start)

Valid Entry	Usage
January to December	The month the clock moves ahead to begin daylight savings

Month (Stop)

Valid Entry	Usage
January to December	The month the clock moves back to standard time

Rule

The daylight saving rule number.

Time (Start)

Valid Entry	Usage
0 to 23	The hour the clock moves ahead to begin daylight saving. The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.
0 to 59	The minute the clock moves ahead to begin daylight saving.

Time (Stop)

Valid Entry	Usage
0 to 23	The hour the clock moves back to standard time. The system uses a 24-hour clock. For example, 14:00 is the same as 2:00 p.m.
0 to 59	The minute the clock moves back to standard time

Chapter 38: DCS to QSIG TSC Gateway

Determines when and how to convert messages from an administered AUDIX NCA-TSC to a QSIG NCA-TSC. Maps the AUDIX NCA-TSC to the appropriate machine ID index to find the QSIG subscriber entry in the QSIG MWI-Prefix screen. Assigns the voice mail number used when a DCS served-user node interrogates a QSIG message center.

Available only if **Interworking with DCS** is enabled for the system.

Example command: `change isdn dcs-qsig-tsc-gateway`

Related links

[Interworking with DCS](#) on page 744

AAR/ARS Access Code

The AAR/ARS Access Code. Accepts up to four digits, including characters * and #.

Mach ID

Valid Entry	Usage
digits	A unique machine ID. Do not repeat a machine ID if it is already associated with a processor channel on an Ethernet link. For the maximum allowed entry, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .

Sig Grp

Valid Entry	Usage
digits	The assigned signaling group number for each machine ID. For the maximum allowed entry, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .

TSC Index

The TSC Index for each machine ID.

Valid Entry	Usage
1 to 64	The assigned signaling group number for the qsig-mwi application type.

Voice Mail Number

The complete Voice Mail Dial Up number. Accepts up to 17 digits.

Chapter 39: Dial Plan Analysis Table

The Dial Plan Analysis Table is the system's guide to translating the digits dialed by users. It determines the beginning digits and total length for each type of call that Communication Manager needs to interpret. The Dial Plan Analysis Table and the Dial Plan Parameters screen work together to define the system's dial plan.

Example command: `change dialplan analysis`

Call Type

Valid Entry	Usage
aar	Automatic Alternate Routing — Routes calls within a company's own private network. Requires that ARS/ AAR Dialing without FAC is enabled.
ars	Automatic Route Selection — Routes calls that go outside a company over public networks. ARS also routes calls to remote company locations without a private network. Requires that ARS/ AAR Dialing without FAC is enabled.
attd	Attendant — Defines how users call an attendant. If a telephone's COR restricts the user from originating calls, this user cannot access the attendant using this code. The attendant access code can also be administered by entering an fac or dac.
dac	Dial access code — Using this feature, you can use trunk access codes (TAC) and feature access codes (FAC) in the same range.
enb-ext	Enbloc extension — Defines a block of extensions that must be dialed using a prefix when the caller dials from a keypad. These extensions can be dialed without a prefix if the caller dials enbloc, for example, from a station call log.
ext	Primary extension — Defines extension ranges that can be used on the system. Extension can have a first digit of 0 through 9 and can have up to 10 digits. Extension cannot have the same first digit as a 1-digit ARS or AAR feature access code (FAC).
fac	Feature access code only — Users dial an FAC instead of programming a button.
pext	Prefixed extension — Identifies the call type as an extension. After digit collection, the prefix digit is removed from the string of dialed digits. The remaining digits make up the extension number and are then processed. With a prefixed extension, extensions numbers with any dialed string can be used. The extension length must be specified on the table.
udp	Uniform Dial Plan — Shares a common dial plan among a group of servers.

Related links

[ARS/AAR Dialing without FAC](#) on page 729

[Dial Access](#) on page 436

[UDP Extension Search Order](#) on page 196

Dialed String

The digits that Communication Manager analyzes to determine how to process the call. Two Dial Plan entries can use the same Dialed String only if the Dialed String consists of one digit. Longer Dialed Strings must all be unique. A new entry cannot be administered if it causes an existing extension, feature access code, or trunk access code to become inaccessible.

Call Type	Valid Entry	Usage
aar	0 to 9	Dialed String entry can be one to six digits long.
ars	0 to 9	Dialed String entry can be one to six digits long.
attd	0 to 9	Dialed String entry can be one or two digits long.
dac	0 to 9 * #	Dialed String can be one to four characters long. The characters "*" and "#" can only be used as a first digit.
enb-ext	0 to 9	Dialed String can be one to ten digits long. The corresponding Total Length can be up to sixteen digits.
ext	0 to 9	Dialed String can be one to ten digits long. The extension cannot have the same first digit as a one-digit ARS or AAR feature access code (FAC). The corresponding Total Length can be up to sixteen digits.
fac	0 to 9 * #	Dialed String can be one to four characters long. The characters "*" and "#" can only be used as a first digit. A FAC must have the longest total length for a given dialed string when using mixed numbering. Otherwise, problems might occur when, for example, three-digit FACs and four-digit extensions begin with the same first digit and the FAC is an abbreviated dialing list access code. However, if the entry in the dial plan that defines the FAC is used to define the AAR or ARS access code, then it must have the longest total length in the dial plan.

Table continues...

Call Type	Valid Entry	Usage
pext	0 to 9	The prefix is the first digit and an extension number of up to five digits in length. The maximum length of a prefix and extension combination is six digits long. A prefixed extension cannot have the same first digit as a dial access code. A prefixed extension cannot have the same dialed string as the ARS or AAR FAC.
udp	0 to 9	Dialed String can be one to ten digits long. The corresponding Total Length can be up to eighteen digits.

Location

This field displays the location of the telephone that is used to dial the number. For more information about assigning a location and for a list of features that use the location, see *Avaya Aura® Communication Manager Feature Description and Implementation*.

Valid entry	Usage
1 to 250 or 1 to 2000	If there are matching entries in the location of the telephone, the system displays the matched entry.
all	If there are no matching entries in the location of the telephone, the system displays all .

Related links

[Multiple Locations](#) on page 736

Percent Full

Valid Entry	Usage
0 to 100	The percentage of system memory resources that have been allocated for the dial plan currently used. For details on the system memory resources, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .

Total Length

The number of digits for this call type. Must be greater than or equal to the number of digits in the Dialed String.

Valid Entry	Usage
1 to 2	attd
1 to 4	dac
1 to 4	fac
2 to 6	pext
1 to 16	enb-ext
1 to 18	udp

Related links

[Dialed String](#) on page 190

Chapter 40: Dial Plan Parameters

The Dial Plan Parameters screen works with the Dial Plan Analysis Table to define the system dial plan.

It also controls the appearance of digit extensions on station displays. These multi-digit extensions can be hard to read as a block. Using Communication Manager you can select the display format for 6- to 16-digit extensions.

AAR/ARS Internal Call Prefix

The digits entered here get concatenated with the calling or called extension. Accepts up to eight digits that excludes * or #. Available only when **ARS/AAR Dialing Without FAC** is enabled for the system. Requires administration of **AAR/ARS Internal Call Total Length**.

Related links

[AAR/ARS Internal Call Total Length](#) on page 193

AAR/ARS Internal Call Total Length

Available only if **ARS/AAR Dialing Without FAC** is enabled for the system.

Valid Entry	Usage
6 to 10 blank	The total length of the internal call digit string, including the Internal Call Prefix and the calling or called extension. Requires administration of an AAR/ARS Internal Call Prefix .



Note:

The longest extension length on the Dial Plan Analysis table, plus the length of the **ARS/AAR Internal Call Prefix**, must be equal to or greater than the **ARS/AAR Internal Call Total Length** value.

Call Type Analysis Behavior When Extension Not Assigned

Use this field to specify how Communication Manager must route a call when:

- The dialed string matches an **ext** or **udp** entry in the Call Type Analysis table.
- The resulting extension is valid, but unassigned.

*** Note:**

You can define valid extensions by assigning the network extensions to a block of numbers in the Uniform Dial Plan table.

Valid entry	Usage
intercept-treatment	Communication Manager stops processing the digits and routes the call to the appropriate intercept treatment: • For a local call, the intercept treatment is defined in the Invalid Number Dialed Intercept Treatment field. • For a trunk call, the intercept treatment is defined in the DID/Tie/ISDN/SIP Intercept Treatment field.
try-next-choice	Communication Manager processes the next administered choice. If the next choice is not administered, Communication Manager routes the call using the original digits. The default option is try-next-choice.

ETA Node Number

Valid Entry	Usage
1 to 999 blank	The number of the destination server for Extended Trunk Access (ETA) calls. ETA calls are unrecognized numbers you can send to another switch for analysis and routing. Such numbers can be Facility Access Codes, Trunk Access Codes, or extensions outside the UDP table.

ETA Routing Pattern

Valid Entry	Usage
1 to 999	The number of the routing pattern to reach the destination server.

Local Node Number

Valid Entry	Usage
1 to 63	Identifies a specific node in a server network. Must match the DCS switch node number and the CDR node number if they are specified.
blank	Leave blank if automatic restoration, DCS, and CDR are unused.

Retry ARS/AAR Analysis if All-Location Entry Inaccessible

Available with Communication Manager Release 4.0.x or later.

Valid Entry	Usage
y	The system finds and uses the best possible entry in the per-location ARS table, if the all-location table points to a trunk group that cannot be accessed because the network has fragmented.
n	The system stops retry of ARS analysis when a trunk group cannot be accessed as a result of the network being fragmented.

UDP-ARS Calls Considered Offnet

Valid entry	Usage
y	If you set the Net field to ARS and the UDP-ARS Calls Considered Offnet field to y, the call is offnet. If UDP-ARS calls are off-net then: • Users with the COS setting Restrict Call Fwd-Off Net enabled cannot forward calls to a UDP-ARS number. • Stations that are outward-restricted (through the COR) cannot place calls to a UDP-ARS number. • Toll analysis is applied when a call is placed by a toll-restricted user to a UDP-ARS number. • COR-COR checks are inapplicable when a user dials a call to a UDP-ARS number.
n	If you set the Net field to ARS and the UDP-ARS Calls Considered Offnet field to n, the call is onnet.

For more information about the **Net** field, see Uniform Dial Plan Table section of the *Avaya Aura® Communication Manager Screen Reference*.

UDP Extension Search Order

Specifies the first table to search to match a dialed extension. If the dialed extension is not found in the specified place, the system searches for the dialed extension in the alternate place. Available if **Uniform Dialing Plan** is enabled for the system.

Valid Entry	Usage
local-extensions-first	Searches the local server first to match a dialed extension, and if not found, then uses the UDP tables to route the call.
udp-table-first	Searches the UDP tables for an off-switch conversion, and if not found, then searches the local server for the dialed extension.

EXTENSION DISPLAY FORMATS

Extension display format

Specifies how the system punctuates extensions for display. The punctuation field is divided into two columns, one for Inter-Location/SAT displays, and one for Intra-Location displays.

In Communication Manager 6.0, the Inter-Location/SAT column is divided into two columns. You can insert a dial prefix in the Inter-Location column, if appropriate, before an enbloc extension. With this, called stations can store a dialable number in their call logs.

 Note:

The maximum length of a displayed extension (including punctuation) varies based on the extension length — for example, for a 10–digit extension, the maximum punctuated length is 13 characters. You must trade off a punctuation mark so that you can insert a prefix digit. You cannot insert a prefix digit or a punctuation mark into the inter-location format for an extension of length 13, 14, 15, or 16.

Blank spaces are sometimes used in telephone extensions, especially outside of the U.S. Dots (.) are used on SAT screens in place of blanks. The following table gives the maximum number of punctuation marks permitted for each extension length.

The number of punctuation marks that the system supports is determined by the number of “x”s in the format.

- If the format contains fewer than six x's, no punctuation marks can be entered.
- If the format contains six or more x's, the maximum number of punctuation marks is determined by the following table.

Extension Length	Maximum Punctuation Marks	Maximum Total Length
6	2	8
7	1	8
8	3	11
9	3	12
10	3	13
11	2	13
12	1	13
13	0	13
14	0	14
15	0	15
16	0	16

Valid Entry	Usage
xx.xx.xx	Can contain all "x" characters (no punctuation) or a combination of "x" characters and 0 to 2 hyphens (-), spaces, or periods (.) to depict how extensions display. A format must be specified or the default takes effect. This field cannot be left blank. The default values for the 8-, 9-, 10-, 11-, 12-, 13- and 16-digit fields are shown in the following example.

EXTENSION DISPLAY FORMATS

	SAT	Inter-Location	Intra-Location
6-Digit Extension:	xx.xx.xx	xx.xx.xx	xx.xx.xx
7-Digit Extension:	xxx-xxxx	xxx-xxxx	xxx-xxxx
8-Digit Extension:	xx.xx.xx.xx	xx.xx.xx.xx	xx.xx.xx.xx
9-Digit Extension:	xxx-xxx-xxx	xxx-xxx-xxx	xxx-xxx-xxx
10-Digit Extension:	xxx-xxx-xxxx	xxx-xxx-xxxx	xxx-xxx-xxxx
11-Digit Extension:	xxxx-xxx-xxxx	xxxx-xxx-xxxx	xxxx-xxx-xxxx
12-Digit Extension:	xxxxxx-xxxxxx	xxxxxx-xxxxxx	xxxxxx-xxxxxx
13-Digit Extension:	xxxxxxxxxxxxxx	xxxxxxxxxxxxxx	xxxxxxxxxxxxxx
14-Digit Extension:	xxxxxxxxxxxxxxxx	xxxxxxxxxxxxxxxx	xxxxxxxxxxxxxxxx
15-Digit Extension:	xxxxxxxxxxxxxxxx	xxxxxxxxxxxxxxxx	xxxxxxxxxxxxxxxx
16-Digit Extension:	xxxxxxxxxxxxxxxx	xxxxxxxxxxxxxxxx	xxxxxxxxxxxxxxxx

Location-Based Call Type Analysis

Use this field to enable the user to make a call to an internal contact using a short or a prefixed extension.

When this field is enabled and the user dials the short digit extension of an internal contact using the dialpad or call logs or contact from a supported endpoint, Communication Manager routes the call using the Call Type Digit Analysis table, taking the location of the caller into account. This field also applies to incoming trunk calls. For example, users in an organization can use five-digit extensions to make calls to internal contacts. If a user dials 8-2249, Communication Manager

maps the dialed number to 1212-848-2249 using the Call Type Digit Analysis table before routing the call.

Chapter 41: Digit Absorption

Implements up to five-digit absorption lists. Digit Absorption is required for each local telephone company central office (CO) and for a Foreign eXchange (FX) trunk group connected to a CO. Each outgoing digit string from the server running Communication Manager to the CO is treated according to entries in the Absorption Treatment Assignment section of the screen.

Available only if Digit Absorption is administered for the trunk group.

Example command: `change digit-absorption n`, where *n* is the absorption digit.

Related links

[Digits](#) on page 809

[Digit Absorption List](#) on page 792

Absorption Treatment Assignment

Valid Entry	Usage
A to F	The chosen treatment letter. All choices for the digits 0 through 9 must be taken from the same group (Group I or Group II).

Absorption Treatment Information

Shows how Digit Absorption treats each digit, 0 through 9, depending on the assignment of A through C for Group I, and A, D, E, and F for Group II.

List Number

Valid Entry	Usage
0 to 4	The Digit Absorption List number that is referenced from the associated trunk group.

Chapter 42: Display Parameters

Establishes how extensions of 6 to 13 digits are punctuated.

Example command: `change display-parameters`

Related links

[Extension display format](#) on page 196

EXTENSION DISPLAY FORMATS

The fields in this section of the screen override similar fields on the Dial Plan Parameters screen. If you leave these fields blank, the values on the Dial Plan Parameters screen apply.

Default Call Appearance Display Format

Affects call appearances only on telephones that support downloadable call appearance buttons, such as the 2420 and 4620 telephones. Bridged call appearances remain unaffected.

Valid Entry	Usage
inter-location	The complete extension on downloadable call appearance buttons. This is the default.
intra-location	A shortened version of the extension on downloadable call appearance buttons.

Display between Locations with these Display Parameters

The system displays this field only when you enable the **Multiple Locations** field. This field cannot be left blank.

Valid Entry	Usage
inter-location	Communication Manager uses the inter-location display format for calls between designated locations. This is the default value.

Table continues...

Valid Entry	Usage
intra-location	Communication Manager uses the intra-location display format for calls within a location. You can use the intra-location format within a branch to display short extensions within that location. This Display Parameter lets you display short extensions between two or more closely associated locations. For example, if you assign display parameter 3 to locations 7 and 17, and set the field to intra-location, Communication Manager uses intra-location format for calls between locations 7 and 17, however uses inter-location calls from 7 to any other location other than 17, and for calls from 17 to any other location other than 7.

Extension Display Format

Use this field to specify how the system must punctuate extensions for inter-location and intra-location displays. Blank spaces are sometimes used in telephone extensions, especially outside of the U.S. Dots (.) are used on SAT screens in place of blanks.

The number of punctuation marks that you can use is determined by the number of xs in the format:

- If the format contains fewer than six xs, there can be no punctuation marks.
- If the format contains six or more xs, the maximum number of punctuation marks is determined by the following table.

Extension length	Maximum punctuation marks	Maximum total length
6	2	8
7	1	8
8	3	11
9	3	12
10	3	13
11	2	13
12	1	13
13	0	13
14	0	14
15	0	15
16	0	16

Valid entry	Usage
xx.xx.xx blank	For six or more xs, characters can contain: • All x characters without punctuation • A combination of x characters and up to two hyphens • Spaces • Periods

Inter-Location

Specifies punctuation for calls between locations. This is the default.

Intra-Location

Specifies punctuation for calls within a location.

Chapter 44: Duplicate Vector

Duplicate vectors from an existing vector and edit the duplicate vectors to create vectors that are similar to the existing vector. You can use this functionality to configure one vector as a template that can be reused when creating similar vectors.

Example command: `duplicate vector n`, where n is the master vector.

[illegible]

Assigned VDN

The first assigned VDN if a VDN was assigned to the master vector.

More VDN's

Displays * if there is more than one VDN assigned to the same vector. For example, if 5555 displays in **VDN Assigned** and * displays in **More?**, this means that the master vector selected is already assigned to VDN 5555 as well as to other VDNs.

Name

The first row displays the vector name for the master vector if assigned. The following lines define the vector names for the duplications. The use of vector names is optional.

Vector

The first row displays the vector number for the master vector. The following lines define the vector numbers for the duplications. Any unassigned vector number is valid.

Chapter 45: Enable Session

Enables Secure SHell (SSH) instead of Telnet.

Example command: `enable session`

Login

The login ID. Accepts from three to six alphanumeric characters.

Password

Seven to 11 characters used as a password. The password must contain at least one number.

Secure

Enables or disables SSH instead of Telnet.

Time to Login

Available only if the board is a TN2302.

Valid Entry	Usage
0 to 255	The number of minutes allowed for login before the session times out.

Chapter 46: Extended Pick-Up Group

Organizes pickup groups into extended pickup groups. The extended group is a collection of pickup groups that can answer calls from other pickup groups in the same extended group. With Extended Pick-Up Group, users can answer calls outside their immediate group. The maximum number of groups that can be added to an extended pickup group is 25.

Example command: `change extended-pickup-group n`, where *n* is the group number.

Extended Group Number

The number associated with a collection of pickup groups.

Pickup Group Number

Valid Entry	Usage
1 to 5000 blank	The number for call pickup groups that can answer calls in the extended pickup group.

Pickup Number

The pickup number assigned to the pickup group. Users dial the pickup number after the feature access code (FAC) to pick up calls in their extended pickup group.

Chapter 47: Extensions administered to have an MCT-Control button

Lists the extensions that can take control of a Malicious Call Trace (MCT) request. To give a user the ability to take control of such requests, you must add their extension to this list and assign them a **mct-control** feature button.

Example command: `display mct-group-extensions`

change mct-group-extensions

Page 1 of 2

Extensions Administered to have an MCT-Control Button:

1: _____	19: _____	37: _____	55: _____
2: _____	20: _____	38: _____	56: _____
3: _____	21: _____	39: _____	57: _____
4: _____	22: _____	40: _____	58: _____
5: _____	23: _____	41: _____	59: _____
6: _____	24: _____	42: _____	60: _____
7: _____	25: _____	43: _____	61: _____
8: _____	26: _____	44: _____	62: _____
9: _____	27: _____	45: _____	63: _____
10: _____	28: _____	46: _____	64: _____
11: _____	29: _____	47: _____	65: _____
12: _____	30: _____	48: _____	66: _____
13: _____	31: _____	49: _____	67: _____
14: _____	32: _____	50: _____	68: _____
15: _____	33: _____	51: _____	69: _____
16: _____	34: _____	52: _____	70: _____
17: _____	35: _____	53: _____	71: _____
18: _____	36: _____	54: _____	72: _____

1 to 100

The extension of a telephone or attendant console that can take control of a Malicious Call Trace (MCT). A **mct-control** button must be assigned to the extension station or attendant console.

Chapter 48: Extensions to Call Which Activate Features by Name

Assigns a dialed extension to a feature within Communication Manager. This extension is called a feature name extension (FNE). The FNE mapping must be administered and all extensions must fit the dial plan. When a user calls the extension, Communication Manager activates the feature. For more information on the extensions that are paired with Feature Access Codes (FACs), see Chapter 51. For more information on the extensions that are not paired with Feature Access Codes (FACs), see Avaya Extension to Cellular User Guide.

The Transfer to Voice Mail FNE is used when a user is active on a call and wants to transfer the other party to voice mail, or to the principal's voice mail, if this is a covered call. This FNE can also be used when a user goes off-hook for the first time and dials the Transfer to Voice Mail FNE to be connected to the voice mail administered in his coverage path. This is identical to dialing a Transfer to Voice Mail feature access code (FAC).

Example command: `change off-pbx-telephone feature-name-extensions`

Extension

Any valid and assigned extension number. Users call it from Cellular telephones to access Communication Manager features. Dialing the extension from their Cellular telephone is equivalent to dialing the FAC administered for the associated feature, or activating the associated feature.

Chapter 49: Failover Group Domain Mapping

A Failover Group is a group of two active Session Manager instances that are interconnected to ensure high availability of Session Manager services. Route pattern and Failover Group domain mapping are used for call preservation during a network outage. You can enter up to 9 domains in the Failover Group Domain Mapping table.

For more information on call preservation administration, see *Call Preservation Feature Description and Administration Guide*.

Example command: `display failover-grp-domain-map`

Domain

Valid Entry	Usage
A fully qualified domain name	You must type the same primary and the inverse failover group domain names, which you have already set on the Session Manager Failover Group screen. For example, if you have set the Failover Group Domain names on the Session Manager Failover Group screen as SM1-SM2.company.com (primary domain name) and SM1-SM2-inverse. company.com (inverse domain name), must set the Failover Group Domain names on the Communication Manager Failover Group screen as the same. The maximum length of the field is 64 characters.  Note: If you enter a domain name, specify Route Pattern. If you set the Domain field to blank, set Route Pattern to blank.
blank	This is the default.

Related links

[Route Pattern](#) on page 211

Route Pattern

Valid Entry	Usage
1–2000	If you are using the Avaya supported server, you can enter 1999 possible route patterns. You must set the same route pattern in the order you defined on the Session Manager Failover Group screen. For example, if you have set the SM2 as first Session Manager and SM1 as secondary Session Manager in the route pattern list, on the Communication Manager Failover Group screen, the route pattern must have the trunk group that is associated to SM2 listed first and the trunk for SM1 listed second. You must set the Group Type field on the Trunk Group and the Signaling Group screens to sip and set the Peer Server field on the Signaling Group screen to SM for the referenced route number. For the Avaya supported servers, see <i>Avaya Aura® Communication Manager Hardware Description and Reference</i>.  Note: If you enter Route Pattern, do not set the Domain field to blank.
blank	This is the default.

Related links

[Domain](#) on page 210

[Group Type](#) on page 614

[Group Type](#) on page 437

Chapter 50: Feature Access Code (FAC)

Assigns feature access codes (FACs) that, when dialed, activate or cancel system features.

Example command: `change feature-access-codes`

Feature Access Code: page 1

```
change feature-access-codes                                     Page 1 of 11
FEATURE ACCESS CODE (FAC)
Abbreviated Dialing List1 Access Code: 
Abbreviated Dialing List2 Access Code: 
Abbreviated Dialing List3 Access Code: 
Abbreviated Dial - Prgm Group List Access Code: 
Announcement Access Code: *777
Answer Back Access Code: *112
Attendant Access Code: 
Auto Alternate Routing (AAR) Access Code: *111
Auto Route Selection (ARS) - Access Code 1:  Access Code 2: 
Automatic Callback Activation: *113 Deactivation: *114
Call Forwarding Activation Busy/DA:  All:  Deactivation: 
Call Forwarding Enhanced Status:  Act:  Deactivation: 
Call Park Access Code: *001
Call Pickup Access Code: 
CAS Remote Hold/Answer Hold-Unhold Access Code: 
CDR Account Code Access Code: 
Change COR Access Code: 
Change Coverage Access Code: 
Conditional Call Extend Activation:  Deactivation: 
Contact Closure Open Code:  Close Code: 
```

Abbreviated Dialing List1 Access Code

A feature access code (FAC) used to access AD list 1.

This value must conform to the FACs or dial access codes defined by the dial plan.

Abbreviated Dialing List2 Access Code

A feature access code (FAC) used to access AD list 2.

This value must conform to the FACs or dial access codes defined by the dial plan.

Abbreviated Dialing List3 Access Code

A feature access code (FAC) used to access AD list 3.

This value must conform to the FACs or dial access codes defined by the dial plan.

Abbreviated Dial - Prgm Group List Access Code

FAC used to enter a group list from a telephone. The user's extension must be administered with permission to program the group list.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Program Ext](#) on page 29

Announcement Access Code

FAC used to record announcements.

This value must conform to the FACs or dial access codes defined by the dial plan.

Answer Back Access Code

FAC used to retrieve parked calls. If no one answers the call before a system-wide expiration interval expires, the system redirects the call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Auto Route Selection (ARS) Access Code 2

Additional FAC used to access ARS.

This value must conform to the FACs or dial access codes defined by the dial plan.

Attendant Access Code

FAC used to call the attendant. While only one attendant can be administered for the dial plan, more than one attendant FAC can be administered in a single distributed network. Attendant access numbers can start with any number from 0 to 9 and contain one or two digits. Available only if an attendant call type is not administered for the dial plan.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Call Type](#) on page 189

Auto Alternate Routing (AAR) Access Code

FAC used to access AAR. AAR routes calls to a different route than the first-choice route when facilities are unavailable.

This value must conform to the FACs or dial access codes defined by the dial plan.

Automatic Callback Activation/Deactivation

FAC used to activate or cancel Automatic Callback. Automatic Callback enables internal callers, upon reaching a busy extension, to have the system automatically connect and ring both originating and receiving parties when the receiving party becomes available.

This value must conform to the FACs or dial access codes defined by the dial plan.

Auto Route Selection (ARS) Access Code 1

FAC used to access ARS. The system can automatically choose the least-expensive way to send a toll call. You can have one ARS access code for local and one for long distance, and route accordingly.

This value must conform to the FACs or dial access codes defined by the dial plan.

Call Forwarding Activation Busy/DA

FAC used to forward calls to an administered number if the user is busy or does not answer.

This value must conform to the FACs or dial access codes defined by the dial plan.

Call Forwarding Enhanced Activation/Deactivation

FAC numbers used to activate and deactivate Enhanced Call Forwarding. Enhanced Call Forwarding forwards incoming calls to different destinations depending on whether they are from internal or external sources. The FACs for activation and deactivation must be administered together.

This value must conform to the FACs or dial access codes defined by the dial plan.

Call Forwarding Enhanced Status

FAC used to display the status of Enhanced Call Forwarding.

This value must conform to the FACs or dial access codes defined by the dial plan.

Call Park Access Code

FAC used to park an active call that can then be retrieved from a different station using the answer back access code. The call park access code cannot have the same first digit as another feature access code that is longer in length.

This value must conform to the FACs or dial access codes defined by the dial plan.

Call Pickup Access Code

FAC used to answer a call directed to a pickup group.

This value must conform to the FACs or dial access codes defined by the dial plan.

CAS Remote Hold/Answer Hold-Unhold Access Code

FAC used by a Centralized Attendant Service (CAS) attendant to place calls on hold and answer calls held at a remote server running Communication Manager. This FAC can also be used by an analog station. Flashing the switch-hook for the proper interval (between 200 and 1000 ms) while talking on an existing call causes the existing call to be placed on soft hold, using which the analog user can dial the Answer Hold-Unhold FAC to Hard hold the call.

This value must conform to the FACs or dial access codes defined by the dial plan.

CDR Account Code

FAC used prior to entering an account code for Call Detail Recording (CDR) purposes. CDR is a feature that uses software and hardware to record call data.

This value must conform to the FACs or dial access codes defined by the dial plan.

Change COR Access Code

Using FAC, users can change their class of restriction (COR) from a telephone. Available only if **Change COR by FAC** is enabled for the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Change COR by FAC](#) on page 731

Change Coverage Access Code

FAC used to change a coverage path from a telephone or remote station. The coverage path is the order in which calls are redirected to alternate answering positions.

An extension must have station **Security Codes** administered to use this FAC.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Security Code](#) on page 656

Contact Closure Close Code

FAC used to close a contact closure relay. Contact closures control electrical devices remotely. Users use an FAC to activate electrical devices such as electrical door locks. If **Contact Closure Open Code** is administered, then **Contact Closure Close Code** must also be administered.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Contact Closure Open Code](#) on page 216

Contact Closure Open Code

FAC used to open a contact closure relay. Contact closures control electrical devices remotely. Users use an FAC to activate electrical devices such as electrical door locks. If **Contact Closure Close Code** is administered, then **Contact Closure Open Code** must also be administered.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Contact Closure Close Code](#) on page 215

Feature Access Code: page 2

The screenshot displays a terminal window titled 'change feature-access-codes' with 'Page 2 of 11' in the top right corner. The main heading is 'FEATURE ACCESS CODE (FAC)'. Below this, a list of settings is shown, each with a label and a corresponding input field. The 'Contact Closure Pulse Code' field is highlighted with a cursor. Other settings include Data Origination Access Code, Data Privacy Access Code, Directed Call Pickup Access Code (set to *102), Directed Group Call Pickup Access Code, Emergency Access to Attendant Access Code, EC500 Self-Administration Access Codes, Enhanced EC500 Activation, Enterprise Mobility User Activation, Extended Call Fwd Activate Busy D/A (set to All), Extended Group Call Pickup Access Code, Facility Test Calls Access Code, Flash Access Code, Group Control Restrict Activation, Hunt Group Busy Activation, ISDN Access Code, Last Number Dialed Access Code, Leave Word Calling Message Retrieval Lock, and Leave Word Calling Message Retrieval Unlock. Some settings have 'Deactivation' fields to their right.

Feature Access Code (FAC)	Value	Deactivation
Contact Closure Pulse Code:		
Data Origination Access Code:		
Data Privacy Access Code:		
Directed Call Pickup Access Code:	*102	
Directed Group Call Pickup Access Code:		
Emergency Access to Attendant Access Code:		
EC500 Self-Administration Access Codes:		
Enhanced EC500 Activation:		
Enterprise Mobility User Activation:		
Extended Call Fwd Activate Busy D/A All:		
Extended Group Call Pickup Access Code:		
Facility Test Calls Access Code:		
Flash Access Code:		
Group Control Restrict Activation:		
Hunt Group Busy Activation:		
ISDN Access Code:		
Last Number Dialed Access Code:		
Leave Word Calling Message Retrieval Lock:		
Leave Word Calling Message Retrieval Unlock:		

Contact Closure Pulse Code

FAC used to pulse a contact closure relay.

This value must conform to the FACs or dial access codes defined by the dial plan.

Data Origination Access Code

FAC used to originate a data call from a voice station.

This value must conform to the FACs or dial access codes defined by the dial plan.

Data Privacy Access Code

FAC used to isolate a data call from call waiting or other interruptions.

This value must conform to the FACs or dial access codes defined by the dial plan.

Directed Call Pickup Access Code

The FAC used with the extension number for which the user wants to pick up a call. To pick up a call for an extension, the user dials the FAC followed by the extension number. The called extension number stops ringing and the user connects to the call for pickup.

The FAC value must conform to the FACs or dial access codes defined in the dial plan.

Directed Group Call Pickup Access Code

The FAC used to pickup a call from any pickup group.

The FAC value must conform to the FACs or dial access codes defined in the dial plan.

EC500 Self Administration Access Code

Using FAC, users can self-administer their cell phone number for the Extension to Cellular feature. Users can add or change their cell phone number through this feature access code. An administrator can still enter or change cell phone numbers. The user calls the Self Administration Access Code access code and enters their cell phone number.

This value must conform to the FACs or dial access codes defined by the dial plan.

Emergency Access To Attendant Access Code

FAC used to gain access to the attendant in an emergency. Such calls alert as emergency calls.

Available only if **Emergency Access to Attendant** is enabled for the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Emergency Access to Attendant](#) on page 732

Enhanced EC500 Activation

Using FAC, users can activate Extension to Cellular remotely.

This value must conform to the FACs or dial access codes defined by the dial plan.

Enhanced EC500 Deactivation

Using FAC, users can deactivate Extension to Cellular remotely.

This value must conform to the FACs or dial access codes defined by the dial plan.

Enterprise Mobility User Activation

FAC used to activate the Enterprise Mobility User feature for a particular user, associating the features and permissions of their primary telephone to a telephone of the same type anywhere within the customer's enterprise.

This value must conform to the FACs or dial access codes defined by the dial plan.

Enterprise Mobility User Deactivation

FAC used to deactivate the Enterprise Mobility User feature.

This value must conform to the FACs or dial access codes defined by the dial plan.

Extended Call Fwd Activate All

FAC used to activate call forwarding from a telephone or remote location.

An extension must have station **Security Codes** administered to use this FAC.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Security Code](#) on page 656

Extended Call Fwd Activate Busy D/A

FAC used to activate call forwarding from a telephone or remote location.

An extension must have station **Security Codes** administered to use this FAC.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Security Code](#) on page 656

Extended Call Fwd Deactivation

FAC used to deactivate call forwarding from a telephone or remote location.

An extension must have station **Security Codes** administered to use this FAC.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Security Code](#) on page 656

Extended Group Call Pickup Access Code

FAC used to answer a call directed to another pickup group. Users must enter a valid pickup number following this field to complete the operation.

This value must conform to the FACs or dial access codes defined by the dial plan.

Facility Test Calls Access Code

FAC used to place a facility test call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Security alert:

To ensure the security of your system, leave this field blank except when actually testing trunks.

Flash Access Code

FAC used to generate trunk flash. This code ensures that the flash signal is interpreted by the local telephone company central office switch, rather than by Communication Manager.

This value must conform to the FACs or dial access codes defined by the dial plan.

Group Control Restrict Activation/Deactivation

FAC used to change the restriction level for all users with a given class of restriction. Requires console permissions.

This value must conform to the FACs or dial access codes defined by the dial plan.

Hunt Group Busy Activation/Deactivation

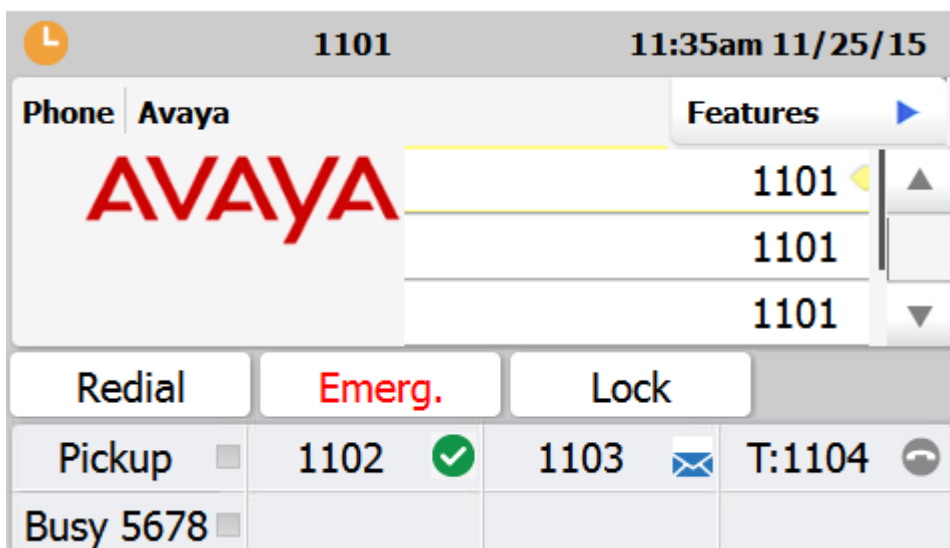
Hunt group members use the **Hunt Group Busy Activation** FAC to make the extension unavailable and the **Hunt Group Busy Deactivation** FAC to make the extension available.

The hunt group member must dial:

- a two-digit hunt group number for a hunt group supporting up to 99 extensions (small hunt group) or
- a four-digit hunt group number for a hunt group supporting up to 8000 extensions (large hunt group)

If needed, the hunt group member must add zeros before the hunt group number to ensure that the small hunt group has a two-digit number and the large hunt group has a four-digit number.

This value must conform to the FACs or dial access codes defined by the dial plan.



ISDN Access Code

FAC used to place an ISDN call without using ARS, AAR, or UDP.

This value must conform to the FACs or dial access codes defined by the dial plan.

Last Number Dialed Access Code

FAC used to redial the last number dialed from this station.

This value must conform to the FACs or dial access codes defined by the dial plan.

Leave Word Calling Message Retrieval Lock

FAC used to lock the display module on telephones. The lock function activates at a telephone by dialing this system-wide lock access code. This prevents unauthorized users from displaying, canceling, or deleting messages associated with the telephone. Available only if **Lock Messages** is administered for the station.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Lock Messages](#) on page 650

Leave Word Calling Message Retrieval Unlock

FAC used to unlock a telephones display module. The lock function is canceled at the telephone by dialing this unlock FAC followed by the SCC.

This value must conform to the FACs or dial access codes defined by the dial plan.

Feature Access Code: page 3

change feature-access-codes		Page 3 of 11
FEATURE ACCESS CODE (FAC)		
Leave Word Calling Send A Message:	_____	
Leave Word Calling Cancel A Message:	_____	
Limit Number of Concurrent Calls Activation:	_____	Deactivation: _____
Malicious Call Trace Activation:	_____	Deactivation: _____
Meet-me Conference Access Code Change:	_____	
Message Sequence Trace (MST) Disable:	_____	
PASTE (Display PBX data on Phone) Access Code:	_____	
Personal Station Access (PSA) Associate Code:	_____	Dissociate Code: _____
Per Call CPN Blocking Code Access Code:	_____	
Per Call CPN Unblocking Code Access Code:	_____	
Posted Messages Activation:	_____	Deactivation: _____
Priority Calling Access Code:	_____	
Program Access Code:	_____	
Refresh Terminal Parameters Access Code:	_____	
Remote Send All Calls Activation:	_____	Deactivation: _____
Self Station Display Activation:	_____	
Send All Calls Activation:	<u>*100</u>	Deactivation: <u>*101</u>
Station Firmware Download Access Code:	_____	

Leave Word Calling Cancel A Message

FAC used to cancel a leave word calling message.

This value must conform to the FACs or dial access codes defined by the dial plan.

Leave Word Calling Send A Message

Using FAC, users can send a leave word calling message that allows internal system users to leave a short pre-programmed message for other internal users.

This value must conform to the FACs or dial access codes defined by the dial plan.

Limit Number of Concurrent Calls Activation/Deactivation

Use this field to configure the feature access codes to enable or disable the Limit Number of Concurrent Calls feature. If a user dials the activation FAC, Communication Manager restricts the number of incoming calls on the station to one call at a time. If the station user is busy, the subsequent incoming calls receive a busy tone or are transferred to voice mail.

*** Note:**

Communication Manager Release 6.2 and earlier supported the Limit Number of Concurrent Calls feature on H.323 and DCP stations. Communication Manager Release 6.3.6 extends this support to SIP stations.

Malicious Call Trace Activation

FAC used to activate a trace request on a malicious call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Meet-me conference access code change

Using FAC, the controlling user of a Meet-me Conference VDN can change the access code. The Meet-me Conference feature is used to set up a dial-in conference of up to 6 parties.

 Note:

For 12 parties to participate in a conference, you must enable the **12-party Conferences** field in the Feature-Related System-Parameters screen.

This value must conform to the FACs or dial access codes defined by the dial plan.

Message Sequence Trace (MST) Disable

Provides the ability to disable the MST traces using a Feature Access Code.

PASTE (Display PBX data on telephone) Access Code

FAC used to view call center data on display telephones. PASTE is used in conjunction with Avaya IP Agent.

This value must conform to the FACs or dial access codes defined by the dial plan.

Per Call CPN Blocking Code Access Code

Use this FAC to activate Calling Party Number (CPN) blocking of a station. When users dial the code, the calling party information is not sent to the public network.

This value must conform to the FACs or dial access codes defined by the dial plan.

Per Call CPN Unblocking Code Access Code

Use this FAC to deactivate Calling Party Number (CPN) blocking of a station. When users dial the code, the calling party information is sent to the public network.

This value must conform to the FACs or dial access codes defined by the dial plan.

Personal Station Access (PSA) Associate Code

FAC used to associate a telephone with the telephone features assigned to a users extension.

Available only if **Personal Station Access (PSA)** is enabled for the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Personal Station Access \(PSA\)](#) on page 736

Personal Station Access (PSA) Dissociate Code

FAC used to remove the association between a physical telephone and an extension number.

Available only if **Personal Station Access (PSA)** is enabled for the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Personal Station Access \(PSA\)](#) on page 736

Posted Messages

FAC used to access the Posted Messages feature. The Posted Messages feature provide callers with a displayed message on the telephone that states why the user is unavailable to take a call.

Available only if the Posted Messages feature is enabled for the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Posted Messages](#) on page 736

Priority Calling Access Code

FAC used to enable priority calling, a special type of call alerting between internal telephone users, including the attendant. The called party hears a distinctive ringing when the calling party uses Priority Calling.

This value must conform to the FACs or dial access codes defined by the dial plan.

Program Access Code

FAC used to program abbreviated dial buttons on an individual telephone.

This value must conform to the FACs or dial access codes defined by the dial plan.

Refresh Terminal Parameters Access Code

Feature Access Code (FAC) is used to request a refresh of the terminal parameters on a telephone that supports downloadable parameters. This FAC is used after a DCP telephone is installed or replaced to ensure that all the terminal parameters, including button labels, are sent to the telephone.

Remote Send All Calls Activation/Deactivation

FAC used to activate or deactivate the Send All Calls feature. Requires console permissions.

This value must conform to the FACs or dial access codes defined by the dial plan.

Self Station Display Activation

FAC used on a digital station to display its primary extension number when the FAC is entered.

This value must conform to the FACs or dial access codes defined by the dial plan.

Send All Calls Activation/Deactivation

FAC used to activate or deactivate sending all calls to coverage with minimal or no alerting at the station.

This value must conform to the FACs or dial access codes defined by the dial plan.

Station Firmware Download Access Code

FAC used for 2420/2410 DCP station firmware downloads.

This value must conform to the FACs or dial access codes defined by the dial plan.

Feature Access Code: page 4

```
change feature-access-codes                                     Page 4 of 11
FEATURE ACCESS CODE (FAC)
  Station Lock Activation: _____ Deactivation: _____
  Station Security Code Change Access Code: _____
  Station User Admin of FBI Assign: _____ Remove: _____
  Station User Button Ring Control Access Code: _____
  Terminal Dial-Up Test Access Code: _____
  Terminal Translation Initialization Merge Code: _____ Separation Code: _____
  Transfer to Voice Mail Access Code: _____
  Trunk Answer Any Station Access Code: _____
  User Control Restrict Activation: _____ Deactivation: _____
  Voice Coverage Message Retrieval Access Code: _____
  Voice Principal Message Retrieval Access Code: _____
  Whisper Page Activation Access Code: _____
  3PCC H323 Override SIP Station Activation: _____ Deactivation: _____
  PIN Checking for Private Calls Access Code: _____
  PIN Checking for Private Calls Using ARS Access Code: _____
  PIN Checking for Private Calls Using AAR Access Code: _____
```

3PCC H323 Override SIP Station Activation/Deactivation

FAC used to activate and deactivate third party call control (3PCC) redirect actions from a SIP desktop station to a remote softphone or virtual private network H.323 telephone.

This value must conform to the FACs or dial access codes defined by the dial plan.

PIN Checking for Private Calls Access Code

This FAC enables the PIN Checking for Private Calls feature. Use this feature to allow users to make calls that are restricted and that prompt users to enter a Personal Identification Number (PIN) after dialing the FAC. The system displays this field only if **PIN Checking for Private Calls** is enabled.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[PIN Checking for Private Calls](#) on page 314

PIN Checking for Private Calls Using AAR Access Code

This FAC enables the PIN Checking for Private Calls feature. Use this feature to allow users to make calls that are restricted and prompt users to enter a Personal Identification Number (PIN) after dialing the FAC. The system displays this field only if **PIN Checking for Private Calls** is enabled.

This value must conform to the FACs or dial access codes defined by the dial plan.

If a user dials and assigns this FAC, the Call Processing feature treats the call as if the user has dialed the FAC for AAR explicitly after entering the PIN code.

Related links

[PIN Checking for Private Calls](#) on page 314

PIN Checking for Private Calls Using ARS Access Code

This FAC enables the PIN Checking for Private Calls feature. Use this feature to allow users to make calls that are restricted and prompt users to enter a Personal Identification Number (PIN) after dialing the FAC. The system displays this field only if **PIN Checking for Private Calls** is enabled.

This value must conform to the FACs or dial access codes defined by the dial plan.

If a user dials and assigns this FAC, the Call Processing feature treats this call as if the user has dialed the FAC for ARS explicitly after entering the PIN code.

Related links

[PIN Checking for Private Calls](#) on page 314

Station Lock Activation/Deactivation

FAC used to activate or deactivate Station Lock. The Station Lock feature locks a telephone to prevent others from placing outgoing calls from the telephone.

This value must conform to the FACs or dial access codes defined by the dial plan.

Station Security Code Change Access Code

FAC used to change a Station Security Code (SSC). The SSC feature is used to deny other users access to the functions that are associated with the station. Each station user can change their own SSC if they know the current settings for the station. The SSC must be administered before the user can change it using this FAC.

This value must conform to the FACs or dial access codes defined by the dial plan.

Station User Admin of FBI Assign

FAC used to activate or deactivate Facility Busy Indicators that provide visual indicators of the busy or idle status of any particular trunk group, hunt group member, or station user.

This value must conform to the FACs or dial access codes defined by the dial plan.

Station User Button Ring Control Access Code

FAC used to control the ring behavior for each line appearance and bridged appearance from the station. Users can have their telephones ring either silently or audibly.

This value must conform to the FACs or dial access codes defined by the dial plan.

Terminal Dial-Up Test Access Code

FAC used to perform tests on digital telephones to make sure that the telephone and the buttons are communicating properly with the server running Communication Manager. The Terminal Dial-Up test ensures that the terminal and each of its buttons can communicate with the server. This test is initiated by a user entering this feature access code. This test is mostly for use by terminal service personnel, but can be used by any station user.

This value must conform to the FACs or dial access codes defined by the dial plan.

Terminal Translation Initialization Merge Code

FAC used to install, or merge, a station without losing any of its previous feature settings. The Terminal Translation Initialization Separation Code must already have been activated or the station administered without hardware, when the telephone was removed from its former location in order for the Terminal Translation Initialization Merge Code to be effective.

This value must conform to the FACs or dial access codes defined by the dial plan.

Terminal Translation Initialization Separation Code

FAC used to remove, or separate, a station from a location without losing any of its feature settings.

This value must conform to the FACs or dial access codes defined by the dial plan.

Transfer to Voice Mail Access Code

FAC used to activate coverage to transfer the caller to the original call recipient's voice mail where the caller can leave a message. This FAC cannot have the same first digit as another FAC that is longer in length.

This value must conform to the FACs or dial access codes defined by the dial plan.

Trunk Answer Any Station Access Code

The FAC used to answer calls alerting on night bells.

This value must conform to the FACs or dial access codes defined by the dial plan.

User Control Restrict Activation/Deactivation

FAC used to activate and deactivate specific restrictions for an individual user or an attendant. Requires console permissions.

This value must conform to the FACs or dial access codes defined by the dial plan.

Voice Coverage Message Retrieval Access Code

FAC used to retrieve voice messages for another user used as a coverage point, using a digital display module.

This value must conform to the FACs or dial access codes defined by the dial plan.

Voice Principal Message Retrieval Access Code

FAC used by a user to retrieve their own voice messages for another user using a digital display module.

This value must conform to the FACs or dial access codes defined by the dial plan.

Whisper Page Activation Access Code

FAC used to place a page to another user's telephone when active on a call. Only the paged user hears the page, not the other parties on the call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Feature Access Code: page 5

change feature-access-codes		Page 5 of 11
FEATURE ACCESS CODE (FAC)		
Call Center Features		
AGENT WORK MODES	After Call Work Access Code: _____ Assist Access Code: _____ Auto-In Access Code: _____ Aux Work Access Code: _____ Login Access Code: _____ Logout Access Code: _____ Manual-in Access Code: _____	
SERVICE OBSERVING	Service Observing Listen Only Access Code: _____ Service Observing Listen/Talk Access Code: _____ Service Observing No Talk Access Code: _____ Service Observing Next Call Listen Only Access Code: _____ Service Observing by Location Listen Only Access Code: _____ Service Observing by Location Listen/Talk Access Code: _____	
AACC CONFERENCE MODES	Restrict First Consult Activation: #31 Deactivation: #32 Restrict Second Consult Activation: #33 Deactivation: #34	

AACC Conference Modes

Avaya Aura® Contact Center uses the Agent Mode feature access codes to restrict the call transfers or conferences that an agent can initiate. In the Restrict First Consult mode, an agent can initiate only one conference or one transfer sequence at a time. When an agent initiates a call transfer or a conference, the system matches the held call appearance and the initiated call appearance. Before initiating a new conference or a transfer sequence, the agent must either complete the transfer with one of the two appearances or drop one of the calls.

 Note:

The AACC Conference modes are not supported on ISDN-BRI telephones.

Restrict First Consult Activation/Deactivation

Use this field to enable the Restrict Call Joining feature. The Restrict Call Joining feature is supported only on Avaya Aura® Contact Center. With this feature, Communication Manager restricts the agents from initiating a transfer or a conference operation.

Restrict Second Consult Activation/Deactivation

Use this field to enable the Restrict Second Agent Consult feature. This feature is supported only on Avaya Aura® Contact Center. With this feature, Communication Manager allows the agents to use only one consult operation at a time. The consult operations include transfer and conference.

 Note:

If you set the **Restrict Second Call Consult** field to n, after you dial the **Restrict Second Consult** feature access code, the feature access code overrides the setting of the **Restrict Second Call Consult** field on the COR group of the station.

Add Agent Skill Access Code

FAC dialed by an agent to add a skill to their current skill set.

This value must conform to the FACs or dial access codes defined by the dial plan.

After Call Work Access Code

FAC dialed by an agent when the agent performs work-related Automatic Call Distribution (ACD) activities.

This value must conform to the FACs or dial access codes defined by the dial plan.

Assist Access Code

FAC dialed by an agent to request assistance from the split supervisor. The split supervisor is someone working in a call center that uses various Communication Manager and Call Management System features to check split and agent performance and to provide assistance if necessary.

This value must conform to the FACs or dial access codes defined by the dial plan.

Auto-In Access Code

FAC dialed when an agent is ready to process another call as soon as the current call is completed.

This value must conform to the FACs or dial access codes defined by the dial plan.

Aux Work Access Code

FAC used when an agent is unavailable to receive ACD, or work-related, calls. Agents use this FAC for activities such as taking a break, going to lunch, or placing an outgoing call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Login Access Code

FAC dialed by an agent to gain access to the ACD functions. This is a system-wide code for all ACD agents.

This value must conform to the FACs or dial access codes defined by the dial plan.

Logout Access Code

FAC dialed by the agent to exit ACD. This is a system-wide logout code for all ACD agents.

This value must conform to the FACs or dial access codes defined by the dial plan.

Manual-In Access Code

FAC dialed when an agent is ready to process another call manually.

This value must conform to the FACs or dial access codes defined by the dial plan.

Remote Logout of Agent Access Code

FAC typically used by a supervisor to logout an idle agent without being physically present at the agent station. The supervisor can be locally or remotely located. Available only if **Service Observing (Remote/By FAC)**, **Vectoring (Basic)**, and **Vectoring (Prompting)** are enabled on the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Service Observing \(Remote/By FAC\)](#) on page 741

[Vectoring \(Basic\)](#) on page 741

[Vectoring \(Prompting\)](#) on page 742

Remove Agent Skill Access Code

FAC dialed by an agent to remove a skill from their current skill set. Available only if **Service Observing (Remote/By FAC)** is enabled on the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Service Observing \(Remote/By FAC\)](#) on page 741

Service Observing Listen Only Access Code

Dial FAC for a station with Service Observing permission to listen to other agent ACD calls without being heard on the ACD call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Service Observing Listen/Talk Access Code

Dial FAC for a station with Service Observing permission to both listen and be heard on an ACD call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Service Observing Next Call Listen Only Access Code

You can assign a Feature Access Code (FAC) to the Service Observing Next Call Listen Only Access Code feature. When you dial the Service Observing Next Call Listen Only Access Code, Communication Manager waits until the current call ends that the extension to be service observed is on. Communication Manager then starts service observing at the beginning of that station's next call. You can listen to the other call, but not speak into it.

Service Observing No Talk Access Code

Dial FAC for a station with Service Observing permission to listen only. Any attempt to toggle between listening and talking using the Service Observing button is denied. Available only if **Expert Agent Selection (EAS) Enabled** is administered for the system.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Expert Agent Selection \(EAS\) Enabled](#) on page 282

Feature Access Code: page 6

```

change feature-access-codes                                     Page 6 of 11
FEATURE ACCESS CODE (FAC)

Call Center Features

FORCED LOGOUT/AUX
  Forced Agent Logout by Location Access Code: ____
  Forced Agent Logout by Skill Access Code: ____
  Forced Agent Aux Work by Location Access Code: ____
  Forced Agent Aux Work by Skill Access Code: ____

MISCELLANEOUS
  Add Agent Skill Access Code: ____
  Remove Agent Skill Access Code: ____
  Remote Logout of Agent Access Code: ____
  
```

Converse Data Return Code

FAC used to pass values between a Voice Response Unit (VRU) and Communication Manager to play an announcement, collect digits from the caller, and so on.

Vector Variable x

FAC (# can be used as the first digit), used to change the value of defined variables.

Related links

[Variables for Vectors](#) on page 868

Feature Access Code: page 7

```
change feature-access-codes                                     Page 7 of 11
                        FEATURE ACCESS CODE (FAC)
                        Call Vectoring/Prompting Features

Converse Data Return Code: _____

Vector Variable 1 (VV1) Code: _____
Vector Variable 2 (VV2) Code: _____
Vector Variable 3 (VV3) Code: _____
Vector Variable 4 (VV4) Code: _____
Vector Variable 5 (VV5) Code: _____
Vector Variable 6 (VV6) Code: _____
Vector Variable 7 (VV7) Code: _____
Vector Variable 8 (VV8) Code: _____
Vector Variable 9 (VV9) Code: _____
```

Automatic Wakeup Call Access Code

FAC dialed to schedule or cancel a wakeup call.

This value must conform to the FACs or dial access codes defined by the dial plan.

Housekeeping Status (Client Room) Access Code

FAC the housekeeper dials from a client room to provide room status. These codes are transmitted to the Property Management System (PMS) for processing.

This value must conform to the FACs or dial access codes defined by the dial plan.

Housekeeping Status (Station) Access Code

FAC the housekeeper dials to provide room status. This access code must be dialed from designated telephones.

This value must conform to the FACs or dial access codes defined by the dial plan.

Verify Wakeup Announcement Access Code

FAC dialed to verify a wakeup announcement.

This value must conform to the FACs or dial access codes defined by the dial plan.

Voice Do Not Disturb Access Code

FAC dialed to enter or cancel a “do not disturb” request using voice prompting.

This value must conform to the FACs or dial access codes defined by the dial plan.

Feature Access Code: page 8

```

change feature-access-codes                                     Page 8 of 11
FEATURE ACCESS CODE (FAC)
Hospitality Features

Automatic Wakeup Call Access Code: _____
Housekeeping Status (Client Room) Access Code: _____
Housekeeping Status (Client Room) Access Code: _____
Housekeeping Status (Client Room) Access Code: _____
Housekeeping Status (Client Room) Access Code: _____
Housekeeping Status (Client Room) Access Code: _____
Housekeeping Status (Client Room) Access Code: _____
Housekeeping Status (Station) Access Code: _____
Housekeeping Status (Station) Access Code: _____
Housekeeping Status (Station) Access Code: _____
Housekeeping Status (Station) Access Code: _____
Verify Wakeup Announcement Access Code: _____
Voice Do Not Disturb Access Code: _____
  
```

Basic Mode Activation

FAC dialed to revert an Enhanced multimedia complex to a Basic multimedia complex.

This value must conform to the FACs or dial access codes defined by the dial plan.

Enhanced Mode Activation

FAC used to convert a Basic multimedia complex to an Enhanced multimedia complex.

This value must conform to the FACs or dial access codes defined by the dial plan.

Multimedia Call Access Code

FAC that indicates to Communication Manager that an Enhanced mode multimedia call is being made. This FAC originates a multimedia call according to system-wide default settings.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Default Multimedia Outgoing Trunk Parameter Selection](#) on page 244

Multimedia Data Conference Activation

FAC that when entered from any voice station that is participating in a multimedia call, alerts Communication Manager to enable data collaboration with the other parties on the call. If this FAC is entered a second time, multimedia data conference activation is denied since it is already active. This FAC only applies to voice stations on servers equipped with ESM adjuncts.

This value must conform to the FACs or dial access codes defined by the dial plan.

Multimedia Data Conference Deactivation

FAC that when entered from the telephone deactivates the data session and reverts to a voice and video call. This telephone supports data collaboration on a multimedia mode call. If a user enters this FAC while participating in a data collaboration multimedia call that the user did not initiate, the system denies the deactivation.

This value must conform to the FACs or dial access codes defined by the dial plan.

Multimedia Multi-Address Access Code

FAC that supports origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2B-channels.

For example, ISDN-BRI provided by a local telephone company central office is provisioned with separate listed directory numbers for each B-channel. To make a 2B multimedia call to such a device, two sets of addresses must be entered.

Originating a multimedia call with the multimedia multi-address access code originates a call according to system-wide default settings.

This value must conform to the FACs or dial access codes defined by the dial plan.

Related links

[Default Multimedia Outgoing Trunk Parameter Selection](#) on page 244

Multimedia Parameter Access Code

FAC that prompts Avaya Communication Manager to initiate a multimedia mode call with a specific bearer capability. This FAC would be followed by a 1 or 2 to indicate the following parameter selections respectively: 2x64 (unrestricted initial system default), 2x56 (restricted).

This value must conform to the FACs or dial access codes defined by the dial plan.

Feature Access Code: page 9

The screenshot shows a terminal window titled 'change feature-access-codes' with 'Page 9 of 11' in the top right corner. The main content is titled 'FEATURE ACCESS CODE (FAC)' and 'Multimedia Features'. It lists several configuration options, each followed by a blank line for input:

- Basic Mode Activation: _____
- Enhanced Mode Activation: _____
- Multimedia Call Access Code: _____
- Multimedia Data Conference Activation: _____ Deactivation: _____
- Multimedia Multi-Address Access Code: _____
- Multimedia Parameter Access Code: _____

Precedence Calling Access Code

FAC used to access the Multiple Level Precedence and Preemption (MLPP) feature using which users can request priority processing of calls during critical situations.

This value must conform to the FACs or dial access codes defined by the dial plan.

WNDP PRECEDENCE ACCESS CODES

FAC used to determine the precedence level for a call when the Worldwide Numbering Dial Plan (WNDP) feature is active. The WNDP feature is compatible with the standard numbering system that the Defense Communications Agency (DCA) established. Different feature access codes are assigned for each PRECEDENCE level.

This value must conform to the FACs or dial access codes defined by the dial plan.

Flash Override Access Code

FAC that corresponds to the Flash Override preemption level.

This value must conform to the FACs or dial access codes defined by the dial plan.

This field is visible only if **Multi level precedence preemption** is set to **Y** on the system-features customer options form.

Flash Access Code

FAC that corresponds to the Flash preemption level.

This value must conform to the FACs or dial access codes defined by the dial plan.

Immediate Access Code

FAC that corresponds to the Immediate preemption level.

This value must conform to the FACs or dial access codes defined by the dial plan.

Priority Access Code

FAC that corresponds to the Priority preemption level.

This value must conform to the FACs or dial access codes defined by the dial plan.

Routine Access Code

FAC that corresponds to the Routine preemption level.

This value must conform to the FACs or dial access codes defined by the dial plan.

Destination Code Control

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Control DCC activation and deactivation by dialing the Activation code and the Deactivation code. DCC activates and deactivates the functionality systemwide.

Activation Code

 Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

The FAC used to activate the DCC feature.

This value must conform to the FACs defined by the dial plan.

Deactivation Code

 Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

The FAC used to deactivate the DCC feature.

This value must conform to the FACs defined by the dial plan.

Chapter 51: Feature-related system parameters

This screen implements system parameters associated with various system features.

*** Note:**

Call Coverage and Call Forwarding parameters are located on the System Parameters Call Coverage / Call Forwarding screen.

Feature-related system parameters: page 1

```
change system-parameters features                                     Page 1 of 19
FEATURE-RELATED SYSTEM PARAMETERS
  Self Station Display Enabled? n
    Trunk-to-Trunk Transfer: none
  Automatic Callback with Called Party Queuing? n
Automatic Callback - No Answer Timeout Interval (rings): 3
    Call Park Timeout Interval (minutes): 10
  Off-Premises Tone Detect Timeout Interval (seconds): 20
    AAR/ARS Dial Tone Required? y

  Music (or Silence) on Transferred Trunk Calls? no
  DID/Tie/ISDN/SIP Intercept Treatment: attendant
Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
  Automatic Circuit Assurance (ACA) Enabled? n

  Abbreviated Dial Programming by Assigned Lists? n
  Auto Abbreviated/Delayed Transition Interval (rings): 2
    Protocol for Caller ID Analog Terminals: Bellcore
Display Calling Number for Room to Room Caller ID Calls? n
```

AAR/ARS Dial Tone Required

Enables or disables a second dial tone that tells the user that additional dialing can occur. The second dial tone is provided on a incoming tie or DID trunk call routed through AAR/ARS.

Abbreviated Dial Programming by Assigned Lists

Valid Entry	Usage
y	Programming is done by the station-assigned list.
n	Indicates that a Program Access code is being used to indicate which personal list is to be programmed.

ACA Long Holding Time Originating Extension

The extension number that the ACA feature uses when sending a long holding time referral call. This extension number must be different from the extension number used for **ACA Short Holding Time Originating Extension**. Available for local or primary ACA Referral Calls. Available only when **Automatic Circuit Assurance (ACA) Enabled** is enabled for the system.

ACA Referral Calls

Indicates where Automatic Circuit Assurance (ACA) referral calls generate. The Automatic Circuit Assurance (ACA) feature is used to identify possible trunk malfunctions. If a possible trunk malfunction is identified, Communication Manager sends the attendant a referral call that consists of a display message or a voice-synthesized message indicating the problem and where the problem is located. Available only when **Automatic Circuit Assurance (ACA) Enabled** is enabled for the system.

Valid Entry	Usage
local	Generate on and for the local switch.
primary	Generate on the local switch for remote servers and switches as well as the local switch.
remote	Generate at another server in a DCS network. In this case, the remote node number must also be entered. The remote node number is the same node number administered for the dial plan. Also, ACA button status transmits to other servers and switches when in a DCS network.

ACA Referral Destination

The specified extension or attendant must be equipped with a display module. Available only for local or primary ACA Referral Calls. Available only when **Automatic Circuit Assurance (ACA) Enabled** is enabled for the system.

Valid Entry	Usage
<i>extension number</i>	The extension on a local server running Communication Manager receives the ACA referral call.
attd	The attendant receives the ACA referral call.

ACA Remote PBX Identification

Available for remote ACA Referral Calls. Available only when **Automatic Circuit Assurance (ACA) Enabled** is enabled for the system.

Valid Entry	Usage
1 to 63	Identifies the switch in a DCS network that makes the referral call. The switch identified here cannot be the remote server or switch identified as local on the system Dial Plan.

ACA Short Holding Time Originating Extension

The extension number that the ACA feature uses when sending a short holding time referral call. This extension number must be different from the extension number used for **ACA Long Holding Time Originating Extension**.

Available for local or primary ACA Referral Calls. Available only when **Automatic Circuit Assurance (ACA) Enabled** is enabled for the system.

Auto Abbreviated/Delayed Transition Interval (rings)

Valid Entry	Usage
1 to 16	The number of rings before an automatic abbreviated or delayed transition is triggered for a call.

Automatic Callback — No Answer Timeout Interval (rings)

Valid Entry	Usage
2 to 9	The number of times the callback rings at the calling station before the callback is canceled. Using Automatic Callback, internal users who place a call to a busy or an unanswered internal telephone can be called back when the called telephone becomes available.

Automatic Circuit Assurance (ACA) Enabled

Enables or disables the ACA feature to measure the holding time of each trunk call to identify possible trunk malfunctions. Long duration thresholds are designed to detect trunks that have been connected beyond the limit. Short duration thresholds detect trunks that might be dropping. Requires that an **aca-halt** button is administered on the user station.

Call Park Timeout Interval (minutes)

Valid Entry	Usage
1 to 90	The number of minutes a call remains parked before it cancels. Using the Call Park, users can retrieve a call on hold from any other telephone within the system.

DID/Tie/ISDN/SIP Intercept Treatment

Valid Entry	Usage
announcement	Routes intercepted calls to a recorded announcement that indicates that the number reached is inoperative or restricted.  Note: This intercept treatment is provided as a system option. Only one message is given to the user.
attendant	Routes intercept calls to the attendant so that the attendant knows when a problem exists.
vdn	Routes intercepted calls to a vector directory number.

Display Calling Number for Room to Room Caller ID Calls

Enables or disables displaying the calling number for room to room hospitality calls.

Internal Auto-Answer of Attd-Extended/Transferred Calls

Using Internal Automatic Answer (IAA), users can answer internal calls on digital telephones (except BRI) using a speakerphone or a headset.

Valid entry	Usage
attd-extended	Enables IAA for only attendant-extended calls.
both	Enables IAA for station-transferred and attendant-extended calls.
none	Disables IAA for all calls.
transferred	Enables IAA for only station-transferred calls.

Music (or Silence) On Transferred Trunk Calls

Valid Entry	Usage
all	All transferred trunk calls receive music until the call is answered if the Music-on-Hold feature is available.
no	When the Trunk callers are waiting to be answered, they hear music or nothing if Music-on-Hold is not administered. When the transfer is completed, the callers then hear ringback till the call is answered.
call-wait	Trunk calls are transferred to stations that require the call to wait and hear music, if administered. All other transferred trunk calls receive ringback tone.

Music/Tone on Hold

Indicates what a caller hears while on hold. Available only if **Tenant Partitioning** is disabled for the system.

Valid Entry	Usage
music	The caller hears music while on hold.
tone	The caller hears a tone while on hold.
none	The caller hears silence while on hold.

Off-Premises Tone Detect Timeout Interval (seconds)

Valid Entry	Usage
5 to 25	The number of seconds a call progress tone receiver (CPTR) tries to detect dial tone from a trunk during dialing. Once the time-out interval occurs, the call either outpulses on the trunk or gets intercept treatment depending on the Outpulse Without Tone settings administered for the system.

Port

Available only when the music on hold type is port.

Valid Entry	Usage
1 to 250	Gateway
V1 to V9	Module
01 to 31	Circuit

Protocol for Caller ID Analog Terminals

Determines the protocol or tones sent to a Caller ID telephone. The provider should have the correct protocol for each country.

Valid Entry	Usage
Bellcore	Telcordia Technologies protocol with 212 modem protocol tones. Typically used in the U.S.
V23–Bell	Telcordia Technologies protocol with V.23 modem tones. Typically used in Bahrain.

Self Station Display Enabled

Enables or disables the displaying of the primary extension associated with a digital display telephone when the **inspect** button is pressed or when a feature access code (FAC) is entered.

- When using an FAC, the display continues until the user picks up the telephone or receives an incoming call.
- When using the **inspect** button, the display continues until the user presses the **normal** or **exit** button, or until the user picks up the telephone or receives an incoming call.

Trunk-to-Trunk Transfer

Regulations in some countries control the settings for this field. For more information, go to the Avaya Support website at <http://support.avaya.com/> for current documentation, product notices, knowledge articles.

Valid Entry	Usage
all	Enables all trunk-to-trunk transfers. With this, telephone users can set up trunk-to-trunk transfer, go on-hook without disconnecting the call, and forward the call to a remote location. This value is required for SIP support.
restricted	Restricts all public trunks (CO, WATS, FX, CPE, DID, and DIOD) from being transferred.
none	Restricts all trunks (except CAS and DCS) from being transferred.

Type

Indicates whether the source for Music on Hold is an announcement extension, an audio group, or a port on a VAL board.

Available only if music is administered as the treatment for calls placed on hold.

Valid Entry	Usage
ext	Extension. Requires entry of the corresponding extension number of the announcement or audio source.
group	Analog group. Requires entry of the corresponding Music-on-Hold analog group number.
port	Analog/aux-trunk. Requires entry of the corresponding location of the Music-on-Hold analog or aux-trunk source.  Note: A source identifier (extension number, audio group number, or port number) must be administered with the source type.

Feature-related system parameters: page 2

```

change system-parameters features                                     Page 2 of 19
FEATURE-RELATED SYSTEM PARAMETERS
LEAVE WORD CALLING PARAMETERS
Maximum Number of Messages Per Station: ~0_
Maximum Number of External Calls Logged Per Station: 0_
Message Waiting Indication for External Calls? n_
Stations with System-wide Retrieval Permission (enter extension)
1: _____ 9: _____ 17: _____ 25: _____
2: _____ 10: _____ 18: _____ 26: _____
3: _____ 11: _____ 19: _____ 27: _____
4: _____ 12: _____ 20: _____ 28: _____
5: _____ 13: _____ 21: _____ 29: _____
6: _____ 14: _____ 22: _____ 30: _____
7: _____ 15: _____ 23: _____
8: _____ 16: _____ 24: _____

Prohibit Bridging Onto Calls With Data Privacy? n_
Enhanced Abbreviated Dial Length (3 or 4): 3_
Default Multimedia Outgoing Trunk Parameter Selection: 2x64

```

LEAVE WORD CALLING PARAMETERS

Default Multimedia Outgoing Trunk Parameter Selection

Valid Entry	Usage
2x56	Sets the default parameter for bandwidth and bearer for all video calls.
2x64	

Related links

[Multimedia Call Access Code](#) on page 234

Enhanced Abbreviated Dial Length (3 or 4)

Enhanced abbreviated dial lists are for users who need to store a vast quantity of speed-dialed numbers. The administrator might not be able to use all entry slots because of system capacity constraints.

Valid Entry	Usage
3	Administrator can use 1000 enhanced list entries.
4	Administrator can use 10,000 entries available.

Maximum Number of External Calls Logged Per Station

When an external call is unanswered, the server running Communication Manager keeps a record of up to 15 calls, provided information on the caller identification is available, and the telephone message lamp lights. The telephone set displays the names and numbers of unsuccessful callers.

Valid Entry	Usage
0 to 15	The maximum number of calls that can be logged for each user. The assigned number cannot be larger than the entry in Maximum Number of Messages Per Station .

Maximum Number of Messages Per Station

Valid Entry	Usage
0 to 125	The maximum number of Leave Word Calling (LWC) messages that can be stored by the system for a telephone at a given time. With LWC, internal system users can leave a short pre-programmed message for other internal users. When the message is stored, the Automatic Message Waiting lamp lights on the called telephone.

Message Waiting Indication for External Calls

Enables or disables a station's ability to receive a message waiting indication when external calls are logged.

Prohibit Bridging Onto Calls with Data Privacy

Valid Entry	Usage
y	Prohibits calls from getting bridged by any party, including Service Observing, Intrusion, Verify, and Bridging.
n	Allows calls to be bridged.

Stations With System-wide Retrieval Permission (enter extension)

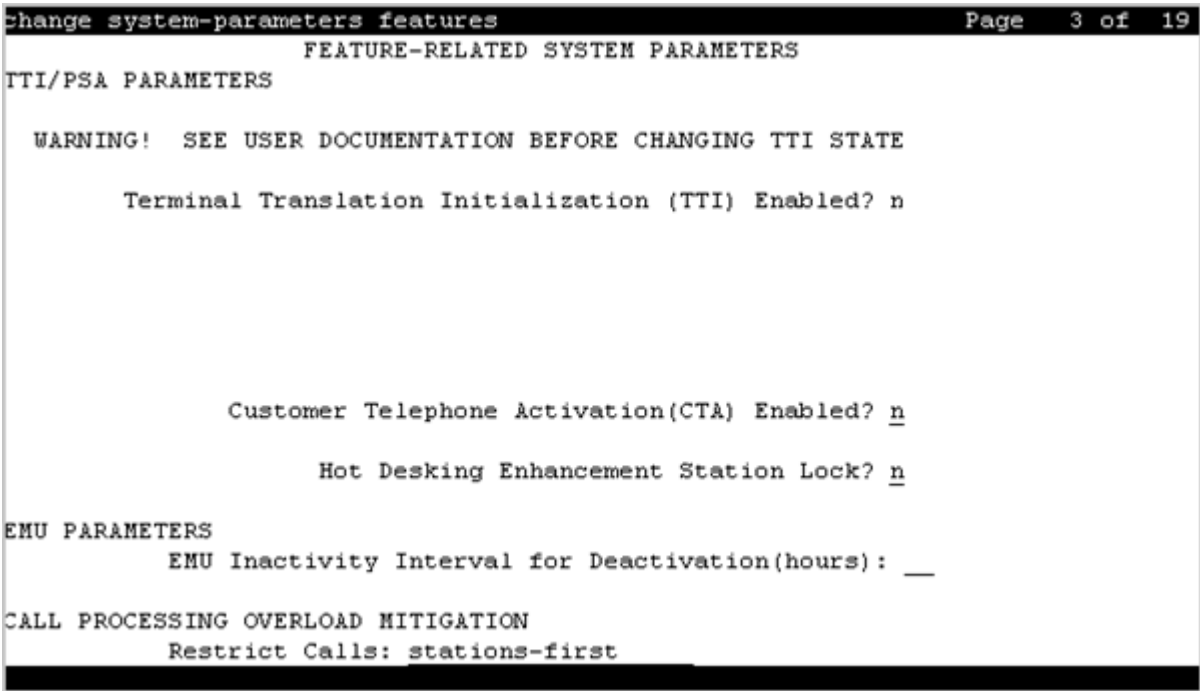
The server running Communication Manager refers to extensions with system-wide retrieval permission as “super-retrievers”.

Valid Entry	Usage
<i>Assigned extension</i>	Up to ten telephone extension numbers that have permission to retrieve LWC Messages or External Call Log records for all other telephones. A VDN extension is disallowed.
attd	All attendants have retrieval permission.

Note:

An extension must be removed from this list before the station can be removed from the system.

Feature-related system parameters: page 3



TTI/PSA PARAMETERS

CPN, ANI for Dissociated Sets

Specifies the ISDN calling party number (CPN), R2-MFC ANI, and CAMA CESID applied to calls made from PSA dissociated sets if no system-wide calling party information has been administered for those protocols. A dissociated set is a telephone without an extension. Available only if a default COR has been administered for dissociated sets. Accepts up to 20 digits.

Customer Telephone Activation (CTA) Enabled

Enables or disables the Customer Telephone Activation (CTA) feature that associates a physical telephone with a station extension. CTA applies only to DCP and analog telephones.

Default COR for Dissociated Sets

Available only when Terminal Translation Initialization is enabled for the system.

Valid Entry	Usage
0 to 995 blank	The Class of Restriction (COR) that the system uses for calls made from dissociated telephones. A dissociated set is a telephone without an extension.

Enhanced PSA Location/Display Information Enabled

Enables or disables the display of Personal Station Access (PSA) information used to disassociate and associate telephones.

When enabled, displays:

- PSA login and associated station information when a station is PSA associated
- PSA logout and the port when a station is PSA dissociated

Available only when Terminal Translation Initialization (TTI) is enabled for the system.

Hot Desking Enhancement Station Lock

Administers the Hot Desking Enhancement (HDE) feature. "Hot desking" is a feature that can lock and unlock telephones or move a fully customized station profile to another compatible telephone.

Valid Entry	Usage
y	Enables the Hot Desking Enhancement feature.
n	Disables the Hot Desking Enhancement feature.

Terminal Translation Initialization (TTI) Enabled

Technicians use the **Terminal Translation Initialization (TTI) Enabled** field to associate and disassociate telephones. This field is available only if TTI is enabled.

Caution:

Before making changes to the TTI settings, go to the Avaya Support website at <http://support.avaya.com/> for current documentation, product notices, knowledge articles.

Valid entry	Usage
y	Starts ACTR, TTI, and PSA transactions for extension and telephone moves between ports.
n	Removes existing TTI port translations and prevents the generation of new TTI port translations.

TTI Security Code

Number that a TTI user uses to access TTI from a telephone or data terminal. Accepts up to seven digits. Available only if Terminal Translation Initialization (TTI) is enabled.

TTI State

The type of port translation used for unadministered digital ports. Available only when Terminal Translation Initialization (TTI) is enabled.

Valid Entry	Usage
data	A stand-alone data module is the TTI port translation for the system. The activation and deactivation sequence is entered at the data terminal.
resume	TTI is available after TTI has been manually suspended. The state of TTI returns to the state that it was in before TTI was manually suspended.
suspend	TTI voice or TTI data translations are temporarily unavailable. The system does not remove existing TTI translations.
voice	A voice or voice/data terminal is the TTI port translation for the system. The activation and deactivation sequence is entered from a telephone.

Unnamed Registrations and PSA for IP Telephones

Use this field to allow IP telephones access to the Personal Station Access (PSA) feature. If you set this field to y, IP telephones can register into the PSA dissociated state. The PSA dissociated state is also known as TTI unmerged state, TTI state, and Unnamed Registered for H.323 standards state.

EMU PARAMETERS

EMU Inactivity Interval for Deactivation (hours)

A system-wide administrable interval for EMU de-registration at the visited switch. This timer is applicable to inter and intra-Communication Manager EMU registrations.

Valid Entry	Usage
1 to 24	The interval in hours, after which a visiting user is dropped due to inactivity. Default is 1. An entry of 1 means that after 1 hour of inactivity, the telephone is dropped from the visited home server.
blank	The timer is not used and the visited station remains active until de-registration by another means occurs.

CALL PROCESSING OVERLOAD MITIGATION

Restrict Calls

Indicates the type of calls to block first during overload traffic conditions on the system.

Valid Entry	Usage
stations-first	Deny new traffic generated by internal stations, allowing inbound calls only. This works best in call center environments.
all-trunks-first	Deny all incoming trunk calls and does not block station off-hook.
public-trunks-first	Deny incoming calls on ISDN public trunks and analog CO trunks.

Feature-related system parameters: page 4

```

change system-parameters features                                     Page 4 of 19
      FEATURE-RELATED SYSTEM PARAMETERS
Reserved Slots for Attendant Priority Queue: 5
      Time before Off-hook Alert: 10
      Emergency Access Redirection Extension:
Number of Emergency Calls Allowed in Attendant Queue: 5
      Drop Parking User From the Call After Timeout? n
      Deluxe Paging and Call Park Timeout to Originator? n
      Controlled Outward Restriction Intercept Treatment: tone
      Controlled Termination Restriction (Do Not Disturb): tone
      Controlled Station to Station Restriction: tone

AUTHORIZATION CODE PARAMETERS
      Authorization Codes Enabled? n

      Controlled Toll Restriction Replaces: none

```

Call Pickup on Intercom Calls

Provides or prohibits the system-wide use of Call Pickup or Directed Call Pickup features on intercom calls.

Controlled Outward Restriction Intercept Treatment

The type of intercept treatment the caller receives when the call is outward restricted.

Valid Entry	Usage
announcement	Provides a recorded announcement to calls that cannot be completed as dialed. The calling party receives indication that the call is receiving Intercept Treatment. Requires an extension number for the announcement in the associated field.  Note: If you enter a shortened extension of the multilocation dial plan, the system does not perform certain administration and validation tasks. Therefore, the system might not display the resultant warnings or submittal denials.
attendant	Allows attendants to provide information and assistance to outgoing calls that cannot be completed as dialed or that are transferred to incomplete or restricted stations.
extension	Routes to an extension. Requires an extension number for the extension in an associated field. Cannot be a VDN extension.
tone	Provides a siren-type tone to internal calls that cannot be completed as dialed.

Controlled Station-to-Station Restriction

The type of intercept treatment the caller receives when the call is placed to a restricted telephone.

Valid Entry	Usage
announcement	Provides an announcement. Requires entry of the announcement extension.
attendant	Intercepted calls are redirected to the attendant.
extension	Routes the call to an alternate extension. Requires entry of the station or individual attendant. This cannot be a VDN extension.
tone	Intercepted calls receive intercept (siren) tone.

Controlled Termination Restriction (Do Not Disturb)

The type of intercept treatment the caller receives when the call is placed to a termination restricted telephone.

Valid Entry	Usage
announcement	Redirects intercepted calls to an announcement. Requires an associated extension number.
attendant	Redirects intercepted calls to the attendant.
coverage	Redirects intercepted calls to coverage.
extension	Redirects intercepted calls to an extension. Requires an associated extension number. Cannot be a VDN extension.
tone	Provides a siren-type tone to calls that cannot be completed as dialed.

Deluxe Paging and Call Park Timeout to Originator

Enables or disables the Call Park feature so that a user can retrieve a call that is on hold from any other telephone within the system. For example, a user can answer a call at one extension, put the call on hold, and then retrieve the call at another extension. Or the user can answer a call at any telephone after an attendant or another user pages the user. Paged calls that are to be parked require separate activation of the Call Park feature.

Valid Entry	Usage
y	The system redirects a parked call to the user who parked the call.
n	The system redirects a parked call to the attendant.

Drop Parking User From the Call After Timeout

Valid Entry	Usage
y	Drops the parking user from the call automatically after the timeout.
n	Does not drop the parking user from the call automatically after the timeout.

Emergency Access Redirection Extension

The assigned extension number or Vector Directory Number (VDN) where emergency queue overflow redirects.

Extended Group Call Pickup

Enables call pickup groups to answer calls directed to another call pickup group.

Valid Entry	Usage
flexible	Flexible feature version supporting a one-to-n (pickup group-to-extended pickup group) mapping.
simple	Simple feature version with a one-to-one pickup group-to-extended pickup group mapping supported.
none	Extended group call pickup is ignored.

Number of Emergency Calls Allowed in Attendant Queue

Valid Entry	Usage
0 to 75	The number of emergency calls allowed in the attendant queue before additional calls are routed to the backup extension.

Reserved Slots for Attendant Priority Queue

Valid Entry	Usage
2 to 342	The number of calls that can go in to the emergency queue.

Time Before Off-Hook Alert

Valid Entry	Usage
1 to 3000	The time in seconds that a telephone with an Off-Hook Alert Class of Service can remain off-hook (after intercept tone has started) before an emergency call is sent to the attendant.

AUTHORIZATION CODE PARAMETERS

Attendant Time Out Flag

Applies only to remote users or incoming calls over trunks requiring an authorization code. Available only if authorization codes are enabled for the system.

Valid Entry	Usage
y	A call is routed to the attendant if the caller does not dial an authorization code within 10 seconds or dials an invalid authorization code.
n	The caller receives intercept tone.

Related links

[Authorization Codes Enabled](#) on page 252

Authorization Code Cancellation Symbol

The symbol a caller must dial to cancel the 10-second wait period during which the user can enter an authorization code. Available only if authorization codes are enabled for the system.

Valid Entry	Usage
(#)	The main and tandem servers or switches are both of the same type.
(1)	An Avaya server or switch is part of the complex or network.

Authorization Code Length

Available only if authorization codes are enabled for the system.

! Security alert:

Enhance system security by using the maximum length for your authorization code.

Valid Entry	Usage
4 to 13 digits	The required length of the authorization code. All administered authorization codes must be this length.

Authorization Codes Enabled

Enables or disables authorization codes on a system-wide basis.

Available only if authorization codes are enabled.

! Security alert:

To maintain system security, use authorization codes.

Controlled Toll Restriction Intercept Treatment

Applies an intercept treatment to a toll call during the call processing. Available only if the Controlled Toll Restriction feature is activated.

Valid Entry	Usage
announcement	Intercepted calls are redirected to an announcement. Requires an assigned announcement extension.
attendant	Intercepted calls are redirected to the attendant.
extension	Intercepted calls are redirected to an extension. Requires an extension assigned to a station or individual attendant.
tone	Intercepted calls receive intercept (siren) tone.

SYSTEM PRINTER PARAMETERS

The system printer is the printer dedicated to support scheduled reports.

Endpoint

Valid Entry	Usage
SYS_PRNT	If the system printer is connected over a TCP/IP link, and the link is defined as SYS_PRNT for IP Services
blank	Not administered

Lines Per Page

Valid Entry	Usage
24 to 132	The number of lines per page required for the report.

SYSTEM-WIDE PARAMETERS

COR to Use for DPT

The Class of Restriction used for the Dial Plan Transparency (DPT) feature.

Valid Entry	Usage
station	The Facility Restriction Level (FRL) of the calling station determines whether that station is permitted to make a trunk call and if so, which trunks it is eligible to access. This is the default.
unrestricted	The first available trunk preference pointed to by ARS routing is used.

Emergency Extension Forwarding (min)

If an emergency call disconnects, public safety personnel always attempt to call back. If the ELIN that was sent differs from the caller's extension number, the return call rings a set other than the one that dialed 911. To overcome this limitation, you can automatically forward that return call to the set that placed the emergency call for an administered period of time.

Emergency Extension Forwarding only applies if the emergency location extension number is an extension on the same switch as the extension that dialed 911. Customers who have several switches in a campus should assign emergency location extensions accordingly.

Sets the Emergency Extension Forwarding timer for all incoming trunk calls if an emergency call gets cut off.

Valid Entry	Usage
0 to 999	The time in minutes that an incoming trunk call forwards to the extension that made the initial 911 call. The default value for both new installs and upgrades is 10.

 Note:

If a user at the emergency location extension (the extension that made the initial 911 call) manually turns off the Call Forwarding feature, the feature is off no matter how many minutes remain on the timer.

Enable Dial Plan Transparency in Survivable Mode

Enables or disables Dial Plan Transparency (DPT) without changing or removing other feature administration associated with DPT. DPT is enabled if a gateway registers with a Survivable Remote Server (Local survivable processor).

Enable Inter-Gateway Alternate Routing

Enables or disables the Inter-Gateway Alternate Routing (IGAR) feature that provides a means of alternately using the public network when the IP-WAN is incapable of carrying the bearer connection.

IGAR Over IP Trunks

The system displays the field when you set the **Enable Inter-Gateway Alternate Routing** field to y.

Valid Entry	Usage
allow	The IGAR feature uses H.323 or SIP trunks when selecting a trunk from a Route Pattern.
skip	The IGAR feature skips H.323 and SIP trunks when selecting a trunk from a Route Pattern. By default, the value is skip.

Switch Name

A name used for identifying the switch. Accepts up to 20 alphanumeric characters.

MALICIOUS CALL TRACE PARAMETERS

Apply MCT Warning Tone

Enables or disables an audible tone to the controlling station when a Malicious Call Trace (MCT) recorder is actively recording a malicious call.

Delay Sending Release (seconds)

Available only if Malicious Call Trace is enabled for the system.

Valid Entry	Usage
0 to 30	The time in seconds the system waits before sending an ISDN release message in response to receiving an ISDN disconnect message. Available in increments of 10.

MCT Voice Recorder Trunk Group

Valid Entry	Usage
1 to 2000	Assigns the trunk group number for Malicious Call Trace (MCT) voice recorders.

Notification using Crisis Alert

From Release 10.1.0.2, you can enable the **Notification using Crisis Alert** field. To enable the **Notification using Crisis Alert** field, do the following:

- You must first empty the **mct-group-extensions** table on the **change mct-group-extensions** page.
- Then, disable the **Every User Responds** and **Alert Pager** fields on the **change system-parameters crisis-alert** page.

Valid Entry	Usage
y	If this field is enabled, all SIP users with a crisis alert button are notified of the malicious call.
n	If this field is disabled, SIP users with a crisis alert button are notified of the emergency call.

SEND ALL CALL OPTIONS

Auto Inspect on Send All Calls

Valid Entry	Usage
y	The system displays Calling Party information for calls that are silently alerting their station because of the Send-All-Calls feature.
n	Calling Party display for calls sent directly to Coverage by the Send-All-Calls feature is not guaranteed.

Send All Calls Applies to

Valid Entry	Usage
station	Any call to a station, regardless of the number dialed, causes calls to that station's own extension to be sent immediately to Coverage, or causes calls to different extensions assigned to the station as bridged appearances to have Ring-Ping notification if redirect notification is enabled.
extension	Only the calls sent to that extension are placed to coverage.

Send All Calls on Ringing Bridge Leaves Call Ringing on Other Bridges

Valid Entry	Usage
y	From Release 10.1.0.2 onwards, if a call rings on a bridging group and anyone in the group has the SAC enabled or presses the SAC button, then the call completes the total number of rings. However, if the principal presses the SAC button, then the call goes to the next coverage point quickly without completing the total number of rings.
n	From Release 10.1.0.2 onwards, if a call rings on a bridging group, then the call completes the total number of rings before moving on to the next coverage point.

UNIVERSAL CALL ID

Create Universal Call ID (UCID)

Enables or disables the generation of a UCID for each call when necessary.

UCID Network Node ID

Valid Entry	Usage
1 to 327 blank	A number unique to this server or switch in a network of switches. This number is an important part of the UCID tag and must be unique to the server or switch.

Feature-related system parameters: page 6

change system-parameters features		Page 6 of 19
FEATURE-RELATED SYSTEM PARAMETERS		
Public Network Trunks on Conference Call:	<u>5</u>	Auto Start? <u>n</u>
Conference Parties with Public Network Trunks:	<u>6</u>	Auto Hold? <u>n</u>
Conference Parties without Public Network Trunks:	<u>6</u>	Attendant Tone? <u>y</u>
Night Service Disconnect Timer (seconds):	<u>180</u>	Bridging Tone? <u>n</u>
Short Interdigit Timer (seconds):	<u>3</u>	Conference Tone? <u>n</u>
Unanswered DID Call Timer (seconds):	<u> </u>	Intrusion Tone? <u>n</u>
Line Intercept Tone Timer (seconds):	<u>30</u>	Mode Code Interface? <u>y</u>
Long Hold Recall Timer (seconds):	<u>0</u>	
Reset Shift Timer (seconds):	<u>0</u>	
Station Call Transfer Recall Timer (seconds):	<u>0</u>	Recall from VDN? <u>n</u>
Trunk Alerting Tone Interval (seconds):	<u>15</u>	
DID Busy Treatment:	<u>tone</u>	
Allow AAR/ARS Access from DID/DIOD?	<u>n</u>	
Allow ANI Restriction on AAR/ARS?	<u>n</u>	
Use Trunk COR for Outgoing Trunk Disconnect/Alert?	<u>n</u>	
7405ND Numeric Terminal Display?	<u>n</u>	7434ND? <u>n</u>
DTMF Tone Feedback Signal to VRU - Connection:		Disconnection:

7405ND Numeric Terminal Display

Enables or disables allowing a 7405ND type of station. This is not an actual telephone type, but it can be used to define ports for certain types of voice messaging systems. This numeric display setting sends only numbers, and not names, to the messaging system.

Related links

[Type](#) on page 660

7434ND

Enables or disables allowing a 7434ND type of station. This is not an actual telephone type, but it can be used to define ports for certain types of messaging systems. Used if the voice messaging system operates in Bridged Mode.

Related links

[Type](#) on page 660

Allow AAR/ARS Access from DID/DIOD

Enables or disables allowing calls for DID and DIOD type trunk groups to complete calls using ARS or AAR.

Allow ANI Restriction on AAR/ARS

(For Russia only). Enables or disables allowing a call placed over a Russian shuttle trunk or a Russian rotary trunk using AAR or ARS to have the ANI requirement administered as restricted. In this case, when ANI is requested, if the request fails, the call immediately drops.

Related links

[ANI Reqd](#) on page 18

[ANI Reqd](#) on page 23

[Request Incoming ANI \(non-AAR/ARS\)](#) on page 519

Attendant Tone

Enables or disables providing call progress tones to the attendants.

Auto Hold

Enables or disables the Automatic Hold feature on a system-wide basis.

Auto Start

If enabled, the Start buttons on all attendant consoles are disabled and the Automatic Start feature is enabled.

Bridging Tone

Enables or disables providing a bridging tone when calls are bridged on primary extensions.

Conference Parties without Public Network Trunks

Valid entry	Usage
3 to 6, or 12	The maximum number of parties allowed in a conference call involving no public network trunks.  Note: For 12 parties to participate in a conference, you must enable the 12-party Conferences field in the Feature-Related System-Parameters screen.

Conference parties with public network trunks

Available only if public network trunks are allowed on a conference call.

Valid Entry	Usage
3 to 6, or 12	The maximum number of parties allowed in a conference call involving a public network subscriber. * Note: For 12 parties to participate in a conference, you must enable the 12-party Conferences field in the Feature-Related System-Parameters screen.

Related links

[Public network trunks on conference call](#) on page 261

Conference Tone

Enables or disables providing a conference tone as long as three or more calls are in a conference call.

*** Note:**

Bridging and Conference Tones are not supported by all countries. If these tones are enabled for countries other than Italy, Belgium, United Kingdom, or Australia, the tones are equivalent to no tone (silence) unless the tone is independently administered or customized.

Related links

[Tone Generation](#) on page 783

DID Busy Treatment

Specifies how to handle a direct inward dialing (DID) call to a busy station.

Valid Entry	Usage
attendant	Call is routed to attendant.
tone	Caller hears a busy tone.

DTMF Tone Feedback Signal to VRU - Connection, Disconnection

Available only if DTMF Feedback Signals for the Voice Response Unit (VRU) are enabled for the system.

Valid Entry	Usage
0 to 9, *, #, A, B, C, D	The code used to connect or disconnect the VRU. This can be a single digit, or a combination such as *99 to connect, #99 to disconnect. The tones must be programmed at the VRU as well.
blank	No tone is sent to the VRU.

Related links

[DTMF Feedback Signals For VRU](#) on page 739

Intrusion Tone

Enables or disables applying an intrusion tone (executive override) when an attendant intrudes on the call.

Line Intercept Tone Timer (seconds)

Valid Entry	Usage
0 to 60	How long an analog station user can wait after hearing warning tone without going on hook, before the station is placed in the lockout state.

Long Hold Recall Timer (seconds)

Valid Entry	Usage
0 to 999	The number of seconds a call can be on hold before the system re-alerts the user to remind them of the call.

Mode Code Interface

Activates or deactivates the use of the Mode Code Voice Mail System Interface to connect the server running Communication Manager over a DTMF interface to other voice mail systems.

 Note:

After making changes, log off and log back on to access the **Mode Code Related System Parameters**.

Night Service Disconnect Timer (seconds)

Valid Entry	Usage
10 to 1024 blank	The number of seconds a trunk call can be unanswered during night service before being disconnected. The trunk must not have Disconnect Supervision for this timer to apply.

Public network trunks on conference call

Valid Entry	Usage
0 to 5, or 11	The number of public network trunks allowed on a conference call.  Note: For 12 parties to participate in a conference, you must enable the 12-party Conferences field in the Feature-Related System-Parameters screen.

Recall from VDN

Indicates whether or not a call that is transferred to a VDN and then routed to a station is recalled to the originating station after the Station Call Transfer Recall Timer expires. If enabled, calls are recalled from a VDN when the Station Call Transfer Recall Timer expires.

Available only if Basic Vectoring and Vectoring (Prompting) are enabled for the system.

Related links

[Vectoring \(Basic\)](#) on page 741

[Vectoring \(Prompting\)](#) on page 742

[Station Call Transfer Recall Timer \(seconds\)](#) on page 263

Reset Shift Timer (seconds)

Valid Entry	Usage
0 to 255	Specifies the number of seconds that reset shift dial tone is audible before busy tone is heard. With reset shift dial tone, the user can dial a new extension by dialing one new digit that replaces the last digit of the extension previously dialed. The new digit replaces the last digit of the extension previously dialed. An entry of 0 disables this feature. Used only for station-to-station calls or private network calls using ISDN trunks.

Short Interdigit Timer (seconds)

Valid Entry	Usage
3 to 9	The time limit that digit analysis will wait for the next digit when it has predicted that all the digits have already been collected.

Special Dial Tone

If enabled, Special Dial Tone can be used.

Special dial tone notifies an analog-telephone user if certain features are still active when the user goes off-hook. These features include:

- Call Forwarding
- Send All Calls
- Do Not Disturb

Related links

[Special Dial Tone](#) on page 301

Station Call Transfer Recall Timer (seconds)

A user-transferred call (station-to-station, a trunk call, or a DCS call) can re-terminate with priority ringing back to the station user who initiates the transfer operation if the transfer-to party does not answer the call within the administered Station Call Transfer Recall timer.

Valid Entry	Usage
0 to 999	The time in seconds before a call redirects back to the station user who initiated the transfer operation. The entry 0 disables this feature.

Trunk Alerting Tone Interval (seconds)

Valid Entry	Usage
1 to 60	Specifies the interval at which the alerting tone is repeated on the call. By default, the value is 15 seconds.

Related links

[Outgoing Trunk Alerting Timer \(minutes\)](#) on page 125

Unanswered DID Call Timer (seconds)

Valid Entry	Usage
10 to 1024	Limits how long a DID call can remain unanswered before routing to the DID/TIE/ISDN Intercept Treatment. This timer interacts with the nonadministrable 50-second Wait for Answer Supervision Timer (WAST). The WAST timer overrides this field. Therefore, if this field is set to a value equal to or greater than 50 seconds, the caller receives intercept tone instead of the normal attendant or announcement treatment that is given when the Unanswered DID Call Timer expires before the WAST. If the Unanswered DID Call Timer expires while the DID call is being processed by call vectoring, the timer is ignored.
blank	Disables the timer.

Related links

[Wait Answer Supervision Timer](#) on page 280

Use Trunk COR for Outgoing Trunk Disconnect/Alert

Indicates whether the outgoing trunk disconnect timer or the outgoing trunk alerting timer is set based on the COR of the originating station or the trunk group. If enabled, the timer is based on the COR of the trunk, not the originating station. By default, this field is disabled.

Important:

You must disable outgoing trunk disconnect and outgoing trunk alerting timers at the same time. If you try to administer both timers, the system displays the following message:

```
Cannot enable both Outgoing Trunk Disc and Outgoing Trunk Alert
timers
```

Related links

[Outgoing Trunk Disconnect Timer \(minutes\)](#) on page 121

[Outgoing Trunk Alerting Timer \(minutes\)](#) on page 125

DISTINCTIVE AUDIBLE ALERTING

Available only if **Tenant Partitioning** is disabled for the system.

Attendant Originated Calls

Indicates which type of ringing applies to attendant-originated calls.

Valid Entry	Usage
internal	Internal ringing applies to attendant-originated calls.
external	External ringing applies to attendant-originated calls. Default is external.
priority	Priority ringing applies to attendant-originated calls.

Distinctive Audible Alerting (Internal, External, Priority)

The number of rings for Internal, External, and Priority calls. For virtual stations, this applies to the mapped-to physical telephone.

Valid Entry	Usage
1	One burst of ringing signal per period. Default for internal calls.
2	Two bursts of ringing signal per period. Default for external and attendant calls.
3	Three bursts of ringing signal per period. Default for priority calls.

Related links

[Distinctive Audible Alerting \(Internal, External, Priority\)](#) on page 761

Feature-related system parameters: page 7

change system-parameters features		Page 7 of 19
FEATURE-RELATED SYSTEM PARAMETERS		
CONFERENCE/TRANSFER		
Abort Transfer? <u>n</u>	No Dial Tone Conferencing? <u>n</u>	
Transfer Upon Hang-Up? <u>n</u>	Select Line Appearance Conferencing? <u>n</u>	
Abort Conference? <u>n</u>	Unhold? <u>n</u>	
No Hold Conference Timeout: <u>60</u>	Maximum Ports per Expanded Meet-me Conf: <u>7</u>	
	12-party Conferences? <u>n</u>	
External Ringing for Calls with Trunks? <u>remote-only</u>		
ANALOG BUSY AUTO CALLBACK		
Without Flash? <u>n</u>		
AUDIX ONE-STEP RECORDING		
	Recording Delay Timer (msec): <u>500</u>	
Apply Ready Indication Tone To Which Parties In The Call? <u>all</u>		
Interval For Applying Periodic Alerting Tone (seconds): <u>15</u>		
	Audix Recording Display? <u>n</u>	
POSTED MESSAGE		
Require Security Code? <u>n</u>		

! Important:

To enable 12-party conference, you must disable the DCS feature. **DCS** field is available in the change system-parameters customer-options page.

CONFERENCE/TRANSFER

12-party Conferences

If enabled, 12 parties can participate in ad-hoc and meet-me conferences.

If disabled, the maximum number of participants in the conference remains 6.

! Important:

For 12 parties to participate in a call, you must configure the following fields in page 6 of the Feature-related system parameters screen:

- Public Network Trunks on conference call
- Conference parties without Public Network Trunks
- Conference parties with Public Network Trunks

For more information on the valid entries for these fields, see the [Feature-related system parameters: page 6](#) on page 258 section.

Abort Conference

DCP, hybrid, IP, wireless, or ISDN-BRI telephone users can abort the conference operation when they are disconnected. If enabled, this field changes a call placed on soft-hold in the conference-pending status to hard-held status if the user disconnects.

Abort Transfer

If enabled, stops the transfer operation whenever users press a non-idle call appearance button in the middle of the transfer operation, or when they disconnect. If the system is configured to transfer calls upon disconnecting the call, users can press the **Transfer** button, dial the complete transfer-to number, and disconnect to transfer the call. Users must select another non-idle call appearance to abort the transfer. Requires DCP, Hybrid, IP, ISDN-BRI or wireless telephones.

External Ringing for Calls with Trunks

Specifies ringing behavior on external trunk calls that are transferred or conferenced by stations or attendants, or extended by the attendant to an on-switch extension.

Valid Entry	Usage
all-calls	All external trunk calls that are transferred or conferenced (either locally or remotely) receive external ringing.
local-only	External trunk calls that are transferred or conferenced locally receive external ringing.
none	External ringing does not apply to external trunk calls that are transferred or conferenced.
remote-only	External trunk calls that are transferred or conferenced remotely receive external ringing. This is the default value.

Maximum Ports per Expanded Meet-me Conf

Available only if ports are administered for the Expanded Meet-me Conference feature.

Valid Entry	Usage
3 to 300	The maximum number of conferees in an Expanded Meet-me Conference. This is a system-wide limit (that is, not administrable on a per Expanded-Meet-me VDN basis).

No Dial Tone Conferencing

If enabled, eliminates dial tone while setting up a conference when another line is on hold or is alerting.

No Hold Conference Timeout

Valid Entry	Usage
20 to 120	The number of seconds No Hold Conference call setup times out. The system Answer Supervision timer should be set to a value less than this.

Select Line Appearance Conferencing

User can use the line appearance rather than the **Conference** button to include a call in a conference. If a user is on a call, and another line is on hold or an incoming call alerts on another line, the user can press the **Conference** button to bridge the calls together. Using the select line appearance capability, the user can press a **line appearance** button to complete a conference instead of pressing the **Conference** button a second time.

Enabling this field activates Select Line Appearance Conferencing.

Transfer Upon Hang-Up

With this feature, users can transfer a call by pressing the **Transfer** button, dialing the required extension, and then disconnecting the active call.

Users using the **Call Park** FAC can park a call without pressing the **Transfer** button a second time.

Note:

Communication Manager Release 6.2 and earlier supported this feature on H.323 and DCP stations. Communication Manager Release 6.3.6 extends this support to 96x1 SIP stations. However, for SIP stations, this field works only when the Team button feature is enabled.

Unhold

With unhold, the user can press the hold button on a telephone to release a hold (if no other line appearance is on hold or alerting). This does not apply to BRI telephones or attendant consoles.

Enabling this field activates this unhold capability.

ANALOG BUSY AUTO CALLBACK

If the Analog Busy Auto Callback Without Flash (ACB) feature is enabled, when a caller places a call through an analog station, and the called station is busy and has no coverage path or forwarding, then an announcement plays, announcing that the station is busy and prompting the caller to enter 1 for ACB or 2 to cover to a voice mail hunt group extension.

Announcement

The extension of the announcement that plays for the Analog Busy Auto Callback Without Flash feature. This field cannot be left blank. Available only if Auto Callback Without Flash is enabled.

Voice Mail Hunt Group Ext

A voice mail hunt group extension where the call is forwarded if the user enters 2 at the ACB announcement prompt. Available only if Auto Callback Without Flash is enabled.

Without Flash

If enabled, provides automatic callback for analog stations without flashing the hook. It is applied only when the called station is busy and has no other coverage path or call forwarding. The caller can enable the automatic callback without flashing the hook or entering the feature access code.

AUDIX ONE-STEP RECORDING

On stations administered with this feature button, users can activate and deactivate the recording of active calls to their voice messaging system with the press of one button.

Apply Ready Indication Tone To Which Parties In The Call

Use this field to administer who hears the ready indication tone.

Valid entry	Usage
all	All parties in the call hear the tone. The default value is all.
initiator	Only the initiator hears the tone. The initiator is the user who activates the Audix one-step recording feature.
none	None of the parties in the call hear the tone.

Note:

When the **Apply Ready Indication Tone To Which Parties In The Call** field is set to `initiator`, all parties in the call hear the ready indication tone. When the **Apply Ready Indication Tone To Which Parties In The Call** field is set to `all` and the **Interval For Applying Periodic Alerting Tone (seconds)** field is set to `0`, the parties in the call do not hear the ready indication tone and the alerting tone.

Interval For Applying Periodic Alerting Tone (seconds)

Available only if all parties on a call hear the voice messaging recording ready tone.

Valid Entry	Usage
0 to 60	The number of seconds between alerting tones. The default value is a 15 second interval. Zero disables the tone.

Recording Delay Timer (msecs)

Valid Entry	Usage
0 to 4000 in increments of 100	The delay interval before starting a voice messaging recording.

Feature-related system parameters: page 8

```

change system-parameters features                                     Page 8 of 19
FEATURE-RELATED SYSTEM PARAMETERS

ISDN PARAMETERS

Send Non-ISDN Trunk Group Name as Connected Name? n
Display Connected Name/Number for ISDN DCS Calls? ~
    Send ISDN Trunk Group Name on Tandem Calls? n
    Send Custom Messages Through QSIG? n
    QSIG/ETSI TSC Extension: _____
MWI - Number of Digits Per Voice Mail Subscriber: 7

PARAMETERS FOR CREATING QSIG SELECTION NUMBERS
    Network Level: _____
    Level 2 Code: _____
    Level 1 Code: _____

    National CPN Prefix: _____
    International CPN Prefix: _____
    Pass Prefixed CPN: ASAI? n    VDN/Vector? n
    Delay for USNI Calling Name for Analog Caller ID Phones (seconds): 0
    Unknown Numbers Considered Internal for AUDIX? n
    USNI Calling Name for Outgoing Calls? n
    Path Replacement with Measurements? y
    QSIG Path Replacement Extension: _____
    Send QSIG Path Replacement Conf. Event to ASAI? y
    Path Replace While in Queue/Vectoring? n
  
```

ISDN PARAMETERS

Delay for USNI Calling Name for Analog Caller ID Phones (seconds)

For analog caller ID telephones, Communication Manager waits for the FACILITY message (includes the caller information) before delivering an incoming call to the receiver.

Valid Entry	Usage
0 to 6	Communication Manager waits for the administered duration for the FACILITY message before delivering the incoming call to the receiver. The default value of the field is 0 which indicates that the feature is disabled. Use the timer value of 3 seconds.

*** Note:**

On page 3 of the ISDN Trunk Group screen, you must set **US NI Delayed Calling Name Update** field to **y** for this feature to work.

Display Connected Name/Number for ISDN DCS Calls

If enabled, displays the connected name/number (if received) for ISDN DCS calls.

Feature Plus Ext

An extension used for proper termination of Feature Plus signaling. For example, Message Waiting Indication (MWI) requires this extension to send the indication to the appropriate server running Communication Manager. Available only if ISDN Feature Plus signaling is enabled for the system.

International CPN Prefix

You can use this number to apply prefixes to international calling numbers for display at receiving telephones. Accepts up to five digits and includes the characters * and #. This number is useful for those telephones that use or implement call back features based on incoming call numbers.

When an ISDN-PRI call arrives, the incoming call setup is analyzed for:

- Whether the Type of Address (TOA) is national or international
- Whether the Numbering Plan Identifier (NPI) is Unknown or ISDN/Telephony

This administered prefix is applied to international calls.

Prefixing applies to any subsequent display on the same server when the call is transferred, covered, or forwarded. The same prefixing applies to outgoing ISDN-PRI calls when the connected number information is returned and meets the same TOA and NPI criteria. The prefix plus the calling/connected number digit string is limited to 15 digits, with truncation occurring at the least significant digits.

Maximum Length

Available only if unknown numbers are administered as internal for the voice messaging system.

Valid Entry	Usage
1 to 20	The maximum length of an unknown private number. Any unknown number longer than the administered value is external. This field cannot be blank.

MWI - Number of Digits Per Voice Mail Subscriber

The number of digits per voice messaging system subscriber. Available only if Basic Supplementary Services or ISDN Feature Plus signaling is enabled for the system.

Valid Entry	Usage
3 to 7	The digit string length of subscribers translated in the Message Center entity. The value in this field must match the value administered for the voice messaging system.

National CPN Prefix

Applies prefixes to national calling numbers for display at receiving telephones. This is useful for those telephones that use or implement call back features based on incoming call numbers. When an ISDN-PRI call arrives, the incoming call setup is analyzed for: (1) whether the Type of Address (TOA) is national or international, and (2) whether the Numbering Plan Identifier (NPI) is Unknown or ISDN/Telephony.

This administered prefix is applied to national calls. Prefixing applies to any subsequent display on the same server when the call is transferred, covered, or forwarded. The same prefixing applies to outgoing ISDN-PRI calls when the connected number information is returned and meets the same TOA and NPI criteria. The prefix plus the calling/connected number digit string is limited to 15 digits, with truncation occurring at the least significant digits.

Valid Entry	Usage
0 to 9 * or # blank	A five-digit number that you can apply as prefixes to national calling numbers for display.

Pass Prefixed CPN to ASAI

If enabled, passes Calling Party Number information (CPN) to ASAI. The prefixed number is not passed on to other adjuncts, Call Detail Recording, or servers/switches.

Path Replacement While in Queue/Vectoring

Use this field to allow Path Replacement after queue/vector processing has started. Depending on the version of CMS, some calls can go unrecorded if this capability is enabled. For more information, go to Avaya Support website at <http://support.avaya.com/>.

Path Replacement with Measurements

Use this field to allow QSIG path replacement or DCS with Reroute to be attempted on measured calls.

QSIG/ETSI TSC Extension

The phantom endpoint extension for QSIG Call Independent Signaling Connections (CISCs) that are similar to NCA Temporary Signaling Connections (TSCs) (both incoming and outgoing). ETSI protocol TSCs as well as QSIG TSCs are supported.

QSIG Path Replacement Extension

The extension for the system used as part of the complete number sent in the Path Replacement Propose message.

Send Custom Messages Through QSIG?

If enabled, provides appropriate display information over QSIG links. Display information can include the Posted Messages feature.

Send ISDN Trunk Group Name on Tandem Calls

If enabled, provides consistent display information regardless of trunk type. Also provides only trunk group name.

Send Non-ISDN Trunk Group Name as Connected Name

If enabled, sends a name of the non-ISDN trunk group as the connected name when a call routes from ISDN to non-ISDN and the call is answered.

Unknown Numbers Considered Internal for AUDIX

Controls the treatment of an ISDN number whose numbering plan identification is “unknown” in a QSIG centralized voice messaging system arrangement. Available only if ISDN trunks are enabled for the system and if the hunt group is administered to send the calling party number to the voice messaging system.

Valid Entry	Usage
y	The unknown number is “internal” and the voice messaging system tries to find a calling party name match for the digit string. If a name match is found, the voice messaging system provides the calling party’s name. If no name is found, the voice messaging system provides the calling party’s telephone number.
n	The unknown number is “external” and the voice messaging system provides the calling party’s telephone number.

USNI Calling Name for Outgoing Calls?

Valid Entry	Usage
y	Sends a name on outgoing calls over NI PRI trunks.
n	Prevents sending calling name information with outgoing calls over NI PRI trunks. Overrides the trunk group Send Name if enabled.

Caution:

The service provider’s local telephone company central office (CO) must be capable of accepting calling name information from Communication Manager in this way. For example, if the CO has a 5ESS, it must be a generic 5EXX or later. Failure to validate the CO capability might cause the CO to drop outgoing calls from the Avaya S8XXX Server. In this case, this field should be disabled.

PARAMETERS FOR CREATING QSIG SELECTION NUMBERS

Level 1 Code

The first level regional code of the Avaya S8XXX Server in the network.

Administer this field carefully. Communication Manager does not verify if you have entered a code that supports your entry in the **Network Level** field. Accepts up to five digits. Because blank regional codes are valid, an entry is not required if the **Network Level** field is 1 or 2.

In QSIG standards, this Level 1 code is called the Level 0 Regional Code.

Available only if the **Network Level** is set to 1 or 2.

Level 2 Code

The second level regional code of the Avaya S8XXX Server in the network.

Administer this field carefully. Communication Manager does not check to ensure you have entered a code that supports your entry in the **Network Level** field. Accepts up to five digits. Because blank regional codes are valid, an entry is not required if the **Network Level** field is 2.

Available only if the **Network Level** is set to 2.

In QSIG standards, this Level 2 code is called the Level 1 Regional Code.

Network Level

The value of the highest regional level employed by the PNP network. Use the following table to find the relationship between the network level and the Numbering Plan Identification/ Type of Number (NPI/TON) encoding used in the QSIG PartyNumber or the Calling Number and Connected Number IEs.

Valid Entry	Usage
0	NPI - PNP TON - local
1	NPI - PNP TON - Regional Level 1
2	NPI - PNP TON - Regional Level 2
blank	If this field is blank and the Send Calling Number and/or Send Connected Number field is y or r with private specified for the Numbering Format field on the ISDN Trunk Group screen, the Calling Number and/or Connected Number IEs is not sent. If the field is left blank but private has been specified in the Numbering Format field on the ISDN Trunk Group screen, the Identification Number (PartyNumber data type) is sent for QSIG PartyNumbers encoded in ASN.1-defined APDUs. In this case, the ASN.1 data type containing the PartyNumber (PresentedAddressScreened, PresentedAddressUnscreened, PresentedNumberScreened, or PresentedNumberUnscreened) is sent marked as PresentationRestricted with NULL for the associated digits.

Feature-related system parameters: page 9

```

change system-parameters features                                     Page 9 of 19
                                FEATURE-RELATED SYSTEM PARAMETERS

CPN/ANI/ICLID PARAMETERS
  CPN/ANI/ICLID Replacement for Restricted Calls: _____
  CPN/ANI/ICLID Replacement for Unavailable Calls: _____

DISPLAY TEXT
                                                    Identity When Bridging: principal
                                                    User Guidance Display? n
  Extension only label for Team button on 96xx H.323 terminals? n

INTERNATIONAL CALL ROUTING PARAMETERS
  Local Country Code: _____
  International Access Code: _____

SCCAN PARAMETERS
  Enable Enbloc Dialing without ARS FAC? n

CALLER ID ON CALL WAITING PARAMETERS
  Caller ID on Call Waiting Delay Timer (msec): 200

```

CPN/ANI/ICLID PARAMETERS

CPN/ANI/ICLID Replacement for Restricted Calls

Use this field to specify a text string to replace the restricted numbers on the telephone display. You can enter up to 15 characters.

 Note:

This field does not apply to SIP telephones.

CPN/ANI/ICLID Replacement for Unavailable Calls

Use this field to specify a text string to replace the unavailable numbers on the telephone display. You can enter up to 15 characters.

 Note:

This field does not apply to SIP telephones.

DISPLAY TEXT

Extension only label for Team button on 96xx H.323 terminals

Available only if the **Team Btn Display Name** for the Class of Restriction is enabled.

Valid entries	Usage
y	For 96xx H.323 telephones, displays the station extension without the team button label. 96xx H323 firmware version 2.0 or greater is recommended as it provides a special icon for team buttons. This field does not impact label customizations.
n	For 96xx H.323 telephones, displays the station extension with the team button label.

Identity When Bridging

Determines whether the telephone display shows the identity of the extension that the bridged appearance represents, or the identity of the physical telephone that the bridged appearance button is on. This applies when the bridged appearance is the caller, and when it is the answering party.

* Note:

When you choose the station option, you must update the Public/Unknown Numbering Format with the Extension Codes of the stations that display the caller's or answering party's assigned identification.

Valid Entry	Usage
principal	The extension that the bridged appearance button points to. This is the default.
station	The physical telephone that the bridged appearance button is on.

User Guidance Display

If enabled, the telephone display shows user guidance messages. This field is disabled by default.

INTERNATIONAL CALL ROUTING PARAMETERS

Local Country Code

A valid PSTN E.164 country code. For example, an SBS node in the United States uses a code of 1. For a list of country codes, see the International Telecommunications Union *List of ITU-T Recommendation E.164 Assigned Country Codes*. Accepts up to three digits.

Related links

[SBS](#) on page 456

[International Access Code](#) on page 275

[Group Type](#) on page 437

[Supplementary Service Protocol](#) on page 449

[Carrier Medium](#) on page 434

International Access Code

The access code required by the PSTN to route calls out of the country. This code is included with the telephone number received from the SBS terminating node if the Local Country Codes of the

originating and terminating nodes are different. For example, an SBS node in the United States uses an access code of 011. Accepts up to five digits.

 Note:

Once administered, this field cannot be cleared until all trunk groups administered for SBS signaling have been removed. If the international call routing parameters are not administered for the system and SBS is enabled for the trunk, a warning is displayed: `Must set INTERNATIONAL CALL ROUTING parameters on system-parameters features.`

Related links

[SBS](#) on page 456

[Local Country Code](#) on page 275

[Group Type](#) on page 437

[Supplementary Service Protocol](#) on page 449

[Carrier Medium](#) on page 434

SCCAN PARAMETERS

Enable Enbloc Dialing without ARS FAC

If the **Enable Enbloc Dialing without ARS FAC** field is enabled, Communication Manager inserts the ARS access code, when a dialed digit string has:

- **Minimum Digit Length** or more digits, and is sent from the telephone to Communication Manager in a single message
- a matching ARS entry

If the dialed digit string does not meet above points, the digit string is used as it is without prepending the ARS access code. The call is passed to ARS for digit analysis. By default, the **Enable Enbloc Dialing without ARS FAC** field is disabled.

Minimum Digit Length

Available only if Enbloc Dialing without an ARS FAC is enabled.

Valid Entry	Usage
1 to 20	The number of digits before Enbloc Calling Treatment is activated. Default is extension length plus 1.

CALLER ID ON CALL WAITING PARAMETERS

Caller ID on Call Waiting Delay Timer (msec)

Valid Entry	Usage
5 to 1275	The required delay in 5-millisecond intervals. Default is 200.

Feature-related system parameters: page 10

change system-parameters features

Page 10 of 19

FEATURE-RELATED SYSTEM PARAMETERS

```

          Pull Transfer: n
          Outpulse Without Tone? y
          Misoperation Alerting? n
          Allow Conference via Flash? y
          Vector Disconnect Timer (min):       
          Hear Zip Tone Following VOA? y
          Update Transferred Ring Pattern? n
          Wait Answer Supervision Timer? n
          Repetitive Call Waiting Tone? n
          Network Feedback During Tone Detection? y
          System Updates Time On Station Displays? n

          Station Tone Forward Disconnect: silence
          Level Of Tone Detection: precise
          Charge Display Update Frequency (seconds): 30
          Date Format on Terminals: mm/dd/yy
          Onhook Dialing on Terminals? n
          Edit Dialing on 96xx H.323 Terminals? n
          Allow Crisis Alert Across Tenants? n
          Block Service Observe Across Tenants? n
          Send DTMF Over Telecommuter Link? y
ITALIAN DCS PROTOCOL
  Italian Protocol Enabled? n

```

Allow Conference via Flash

If enabled, an analog station can use flash to conference calls.

Allow Crisis Alert Across Tenants

If enabled, allows crisis alert across different tenant partitions. Disabled by default.

Block Service Observe Across Tenants

If enabled, the service observer cannot monitor the agent if the service observer and agent are in different tenant partition groups. By default, this field is disabled.

Charge Display Update Frequency (seconds)

Available only if Advice of Charge or Periodic Pulse Metering with display functions is used.

Valid Entry	Usage
10 to 60 blank	The amount of time in seconds between charge-display updates. Frequent display updates might have considerable performance impact. If the duration of a call is less than the Charge Display Update Frequency, the display does not automatically show charge information.

Date Format on Terminals

The system displays the **Date Format on Terminals** field if the **Multinational Locations** field on the System Parameters Customer Options screen is set to n. This field affects the 607/2400/4600/6400 terminals. The date is in mm/dd/yy, dd/mm/yy, and yy/mm/dd formats.

Edit Dialing on 96xx H.323 Terminals

If enabled, an end-user can dial a number when the telephone is on-hook. Disabled by default.

Hear Zip Tone Following VOA?

Enables or disables zip tone alerts to a telephone user when the announcement has completed and a caller is now connected. CallMaster set and attendant console users hear double zip tone following the announcement. All other telephone users hear single zip tone.

 Note:

This field does not effect auto-answer zip tone heard prior to the VDN of Origin Announcement (VOA).

Intercept Treatment on Failed Trunk Transfers

Valid Entry	Usage
y	Provides intercept treatment to calls failing trunk transfers.
n	Drops calls failing trunk transfers.

Level of Tone Detection

Used only when users are having difficulty placing outgoing calls due to inaccurate detection of network dial tone.

Valid Entry	Usage
broadband	This is the least exact of the levels of tone detection. If Avaya Communication Manager detects any tone at all, it interprets this as dial tone.
medium	The server running Avaya Communication Manager interprets any tone which has a continuous "on" period of longer than 1 second as dial tone.
precise	Communication Manager accepts whatever the tone detector circuit pack reports.

Misoperation Alerting

Valid Entry	Usage
y	Enables misoperation recall alerting on multi-appearance stations, analog stations, and attendant consoles.
n	Uses standard misoperation handling without recall alerting.

Network Feedback During Tone Detection

If enabled, provides audible feedback to the user while the system attempts to detect dial tone.

On-hook Dialing on 607/2400/4600/6400/8400 Terminals

If enabled, users can dial without lifting the handset. Users hear dial tone when they press the **Speaker** button, even if the handset is on-hook.

For 6400/8400, 607, 2420, 2410, and 4600 telephone users with speakerphones.

Outpulse Without Tone

Valid Entry	Usage
y	The server outpulses digits even when a dial tone has not been received.
n	The calling party receives intercept tone if no dial tone is detected.

Pull Transfer

Enables or disables the Pull Transfer feature on a system-wide basis. With this, the transferring or transferred-to party can press the **Transfer** button to complete the transfer operation.

Repetitive Call Waiting Interval (sec)

Available only if the Repetitive Call Waiting Tone is enabled.

Valid Entry	Usage
1 to 99	The number of seconds between call waiting tones.

Repetitive Call Waiting Tone

If enabled, a repetitive call waiting tone is provided to the called party for all types of call waiting access.

Send DTMF Over Telecommuter Link

In the telecommuter mode, when a user is on a call and presses any digits from a deskphone or a softphone, the user hears loud DTMF tones over the telecommuter link. Use the **Send DTMF Over Telecommuter Link** field to send or prevent DTMF tones over the telecommuter link.

Valid entry	Usage
y	Communication Manager sends DTMF tones over the telecommuter link. The default value is y.
n	Communication Manager does not send DTMF tones over the telecommuter link.

Station Tone Forward Disconnect

Applies to any station other than one administered as a data endpoint, an attendant console, a BRI telephone, an auto answer, or as an Outgoing Call Management (OCM) agent.

Valid Entry	Usage
busy intercept silence	The treatment given to a station that is the last party remaining off-hook on a call, until that station is placed on-hook, or until the tone has played for 45 seconds and is followed by silence.

System Updates Time On Station Displays

If enabled, automatically updates the system time on two-line display telephones when you run background maintenance, for example, when you plug in the telephone set. The system sends the time update periodically to the telephones. This parameter does not apply to telephones such as the BRI telephones, where the user sets the time.

Update Transferred Ring Pattern

If enabled, changes the ringing pattern from internal to external when an internal station transfers an external call. Use this feature if most calls go through an attendant, so that users are able to distinguish an external call.

Vector Disconnect Timer (min)

Valid Entry	Usage
1 to 240	Determines when the trunk gets disconnected if incoming or outgoing disconnect supervision is disabled for the trunk group.
blank	Avaya Communication Manager does not initiate a disconnect.

Wait Answer Supervision Timer

Valid Entry	Usage
y	Calls to stations unanswered after 50 seconds are dropped.
n	Unanswered calls drop only when the calling party goes on-hook.

Related links

[Unanswered DID Call Timer \(seconds\)](#) on page 263

ITALIAN DCS PROTOCOL

The next three fields control the Italian DCS Protocol feature.

Apply Intercept Locally

If enabled, DID/CO intercept treatment is applied locally instead of on the originating server or switch. Available only if **Italian Protocol** is enabled.

Enforce PNT-to-PNT Restrictions

If enabled, restrictions and denial of PNT-to-PNT connections are enforced when the EDCS message is unavailable. Available only if **Italian Protocol** is enabled.

Italian Protocol Enabled

Enables or disables the Italian DCS feature on a system-wide basis.

Feature-related system parameters: page 11

change system-parameters features		Page 11 of 19
FEATURE-RELATED SYSTEM PARAMETERS		
CALL CENTER SYSTEM PARAMETERS		
EAS		
Expert Agent Selection (EAS) Enabled?	<u>y</u>	
Minimum Agent-LoginID Password Length:	<u> </u>	
Direct Agent Announcement Extension:	<u> </u>	Delay: <u> </u>
Message Waiting Lamp Indicates Status For:	<u>station</u>	
Work Mode On Login:	<u>aux</u>	
VECTORIZING		
Converse First Data Delay:	<u>0</u>	Second Data Delay: <u>2</u>
Converse Signaling Tone (msec):	<u>100</u>	Pause (msec): <u>70</u>
Prompting Timeout (secs):	<u>10</u>	
Interflow-qpos EWT Threshold:	<u>2</u>	
Reverse Star/Pound Digit For Collect Step?	<u>n</u>	
Available Agent Adjustments for BSR?	<u>n</u>	
BSR Tie Strategy:	<u>1st-found</u>	
Store VDN Name in Station's Local Call Log?	<u>n</u>	
SERVICE OBSERVING		
Service Observing: Warning Tone?	<u>y</u>	or Conference Tone? <u>n</u>
Allowed with Exclusion: Service Observing?	<u>n</u>	SSC? <u>n</u>
Allow Two Observers in Same Call?	<u>n</u>	Coach on SSC? <u>n</u>

CALL CENTER SYSTEM PARAMETERS EAS

Delay

Available only if Expert Agent Selection (EAS) or ASAI Link Core Capabilities are enabled for the system.

Valid Entry	Usage
0 to 99 blank	The number of seconds the caller hears ringback before the Direct Agent Announcement is heard by the calling party.

Direct Agent Announcement Extension

The extension of the direct agent announcement.

Expert Agent Selection (EAS) Enabled

Enables or disables Expert Agent Selection (EAS). Enable only if ACD or vectoring hunt groups exist or if existing ACD or vectoring hunt groups are skill-based. Available only if EAS is enabled for the system.

Message Waiting Lamp Indicates Status For

Available only if Expert Agent Selection (EAS) is enabled for the system.

Valid Entry	Usage
station	The message waiting lamp on a telephone indicates the message is for the telephone extension.
loginID	The message waiting lamp on a telephone indicates the message is for the agent login ID.

Minimum Agent-LoginID Password Length

The minimum number of digits that must be administered as an EAS Agent's LoginID password. Accepts up to nine digits. Available only if Expert Agent Selection (EAS) is enabled for the system.

VECTORIZING

Available Agent Adjustments for BSR

If activated, Best Service Routing (BSR) adjustments are made available to available agents. Available only if Vectoring (Best Service Routing) is enabled for the system.

BSR Tie Strategy

Available only if Vectoring (Best Service Routing) is enabled for the system.

Valid Entry	Usage
1st-found	BSR uses the first selection for routing. This is the default.
alternate	Alternating the BSR selection algorithm is possible when a tie in Expected Wait Time (EWT) or available agent criteria occurs. Every other time a tie occurs for calls from the same active Vector Directory Number (VDN), the selection from the step with the tie is used instead of the first selected split/skill or location to send the call. This helps balance the routing over the considered local splits/skills and remote locations when cost of routing remotely is of least concern.

Converse First Data Delay

Available only if Vectoring (Basic) is enabled for the system.

Valid Entry	Usage
0 to 9	The number of seconds data is prevented from being outputted as a result of a converse vector step from the system to Avaya IR before IR is ready. The delay commences when the IR port answers the call.

Converse Second Data Delay

Available only if Vectoring (Basic) is enabled for the system.

Valid Entry	Usage
0 to 9	Number of seconds used when two groups of digits are being outputted, as a result of a converse vector step, from the system to Avaya IR. Prevents the second set from being outputted before IR is ready. The delay commences when the first group of digits has been outputted.

Converse Signaling Pause

The length in milliseconds of the delay between digits being passed. The optimum timer settings for Avaya IR are 60 msec tone and 60 msec pause.

Available only if Vectoring (Basic) and DTMF feedback signals are enabled for the system.

Valid Entry	Usage
40 to 2550 (in increments of 10)	Values are rounded up or down depending upon the type of circuit pack used to output the digits. • TN742B or later suffix analog board — Rounds up or down to the nearest 25 msec. For example, a 130 msec tone rounds down to 125 msec, a 70 msec pause rounds up to 75 msec for a total of 200 msec per tone • TN464F, TN767E or later suffix DS1 boards — Rounds up to the nearest 20 msec. For example, a 130 msec tone rounds up to 140 msec, a 70 msec pause rounds up to 80 msec for a total of 220 msec per tone. If a circuit pack has been used for end-to-end signaling to IR, and has then been used to send digits to a different destination, IR timers might stay in effect. To reset the timers to the system default, pull and reseat the circuit pack.

Converse Signaling Tone

The length in milliseconds of the digit tone for digits being passed to Avaya IR. The optimum timer settings for IR are 60 msec tone and 60 msec pause.

Available only if Vectoring (Basic) and DTMF feedback signals are enabled for the system.

Valid Entry	Usage
40 to 2550 (in increments of 10)	Values are rounded up or down depending upon the type of circuit pack used to outpulse the digits. • TN742B or later suffix analog board — Rounds up or down to the nearest 25 msecs. For example, a 130 msec tone rounds down to 125 msecs, a 70 msec pause rounds up to 75 msec for a total of 200 msecs per tone • TN464F, TN767E or later suffix DS1 boards — Rounds up to the nearest 20 msecs. For example, a 130 msec tone rounds up to 140 msecs, a 70 msec pause rounds up to 80 msecs for a total of 220 msecs per tone. If a circuit pack has been used for end-to-end signaling to IR, and has then been used to send digits to a different destination, IR timers might stay in effect. To reset the timers to the system default, pull and reseal the circuit pack.

Interflow-qpos EWT Threshold

Available only if the Lookahead Interflow (LAI) is enabled for the system.

Valid Entry	Usage
0 to 9 blank	Number of seconds for this threshold. Any calls predicted to be answered before this threshold are not interflowed, therefore saving CPU resources.

Prompting Timeout (secs)

Available only if Vectoring (Prompting) is enabled for the system.

Valid Entry	Usage
4 to 10	The number of seconds before the Collect Digits command times out for callers using rotary dialing.

Reverse Star/Pound Digit for Collect Step

The “*” is interpreted as a caller end-of-dialing indicator and the “#” is an indicator to clear all digits previously entered by the caller for the current collect vector step.

Valid Entry	Usage
y	Reverses the star and pound digits for the collect vector step. This does not affect any other vector step or other non-ACD feature, such as ARS.
n	The “*” and “#” digit-processing is unchanged.

Store VDN Name in Station’s Local Call Log

Enables or disables the sending of a message from Communication Manager telling the telephone to store the VDN name or the calling party’s name in the station call log for any of the following telephones:

- 2420
- 4610

- 4620
- 4625

Service Observing field descriptions

Security alert:

The use of Service Observing features might be subject to federal, state, or local laws, rules or regulations, or require the consent of one or both parties in the conversation. You must familiarize yourself and comply with all the applicable laws, rules, and regulations before using the Service Observing features.

Field title	Field description
Allow Two Observers in Same Call	Controls whether two observers can monitor the same call. Valid entries are y and n (default). If you select y: • Two observers can monitor the same Expert Agent Selection (EAS) agent login ID or station extension. • Up to two service observers can be on the same two-party call or in a conference call with more than two parties. If you select n, only one service observer can monitor the EAS agent login ID or station extension.
Service Observing: Warning Tone	Controls whether the system plays a warning tone that the phone users and calling parties can hear when a supervisor monitors the call. Valid entries are y and n (default). The option is applicable if the Service Observing (Basic) field on the System-Parameters Customer-Options screen is active.
Service Observing: Conference Tone	Controls whether the system plays a conference tone that the phone users and calling parties can hear when a supervisor monitors the call. Valid entries are y and n (default). The option is applicable if the Service Observing (Basic) field on the System-Parameters Customer-Options screen is active. This option is not applicable if you set the Service Observing: Warning Tone option to y.

Table continues...

Field title	Field description
Allowed with Exclusion: Service Observing	Controls whether multi-appearance phone users can prevent other service observers from bridging on to an existing call. Valid entries are y and n (default). If you select y, Communication Manager allows a Service Observing connection towards a station with Exclusion active, using Class of Service (COS) or by manual activation of Exclusion. If you select n, Communication Manager denies a Service Observing connection towards a station with Exclusion active. If an agent activates Exclusion while being observed, Communication Manager drops all bridged parties including the observer from the call.
Allowed with Exclusion: SSC	Controls whether multi-appearance phone users can prevent ASAI Single Step Conference (SSC) parties from bridging on to an existing call. Valid entries are i, b, and n (default). If you select i, Communication Manager allows invisible SSC parties a connection towards a station with Exclusion active, using COS or by manual activation of Exclusion. If you select b, Communication Manager allows visible and invisible SSC parties a connection towards a station with Exclusion active, using COS or by manual activation of Exclusion. If you select n, Communication Manager denies a SSC connection towards a station with Exclusion active. If an agent activates Exclusion while being observed, Communication Manager drops all bridged parties including the observer from the call.
Coach on SSC	Controls whether a supervisor can provide whisper coaching on a call with an invisible SSC. When using whisper coaching, only the agent will hear the supervisor. Valid entries are y and n (default). If you select y, the supervisor can change to coaching mode to give advice to the agent without the customer hearing that advice. The supervisor is heard by the agent, but other parties on the call will not hear the supervisor. If you are using Avaya Call Recording, anything said by the supervisor is recorded except while in the coaching mode. If you select n, the supervisor is blocked from changing to the coaching mode while the invisible SSC is on the call.  Note: When using the sip-sobsrv feature button, coaching is available from the AXP On-Prem and Unified Communications SIP devices. For Unified Communications SIP devices, sip-sobsrv feature button is only supported on J1xx SIP phones.

Feature-related system parameters: page 12

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change system-parameters features                                     Page 12 of 19
      FEATURE-RELATED SYSTEM PARAMETERS
AGENT AND CALL SELECTION
      MIA Across Splits or Skills? n
      ACW Agents Considered Idle? y
      AUX Agents Considered Idle (MIA)? n
      AUX Agent Remains in LOA Queue? n
      Call Selection Measurement: current-wait-time
      Service Level Supervisor Call Selection Override? n
      Auto Reserve Agents: none
      Block Hang-up by Logged-in Auto-Answer Agents? n
CALL MANAGEMENT SYSTEM
      REPORTING ADJUNCT RELEASE (determines protocol used by appl link)
      CMS (appl mis): R19.1+
      AAPC/IQ (appl ccr):           
      BCMS/VuStats LoginIDs? y
      BCMS/VuStats Measurement Interval: hour
      BCMS/VuStats Abandon Call Timer (seconds):       
      Validate BCMS/VuStats Login IDs? n
      Clear VuStats Shift Data: on-login
      Remove Inactive BCMS/VuStats Agents? n
R16.1/R16.x/R17.0      R18      R18.1/R19.0      R19.1+

```

AGENT AND CALL SELECTION

ACW Agents Considered Idle

Valid Entry	Usage
y	Include agents who are in After Call Work (ACW) mode in the Most-Idle Agent queue. This means that ACW is counted as idle time.
n	Exclude ACW agents from the queue.

Auto Reserve Agents

When a critical skill is not meeting its service level, auto-reserve puts agents in standby for their other skills to ensure that there is an available agent when the next call arrives for the critical skill. When an agent becomes available, all of his or her assigned skills are checked to see if any auto-reserve skills are not meeting their target service level. If so, the agent is made available only in those skills.

Valid Entry	Usage
all	Puts an agent on standby for all skills.
none	Agent is not on standby for any additional skills.
secondary-only	Puts an agent on standby only for secondary skills.

Block Hang-up by Logged-In Auto Answer Agents

The system displays the field only if you set the **EAS** field to y. The enhancement is available for agents using H.323 IP and DCP telephones, but does not apply and is not required for agents using BRI, analog, CallMaster, or SIPCC telephones.

Valid Entry	Usage
n	If an auto-answer agent goes on-hook when on an ACD call: • Communication Manager drops the active call and logs the agent out. • The reporting adjunct fails to record the call and track the logged-in agent.
y	You, as an administrator, can prevent an agent in the auto-answer mode from accidentally logging out or dropping an active call. The agent can perform the following functions to drop calls: • Use the release button to disconnect the call and stay logged-in. • Use the drop button to release the last connected party.  Note: If you set the field to y and the logged-in agent appearance is on-hook, a new call is not delivered automatically. The new call rings at the station and the agent must manually answer the call. If you have administered RONA for the skill, the call is redirected if the agent fails to answer within the administered number of rings.

Call Selection Measurement

Determines how Avaya Communication Manager selects a call for an agent when the agent becomes available and there are calls in queue.

Valid Entry	Usage
current-wait-time	Selects the oldest call waiting for any of the agent's skills.
predicted-wait-time	Uses the time a call is predicted to wait in queue instead of the time the call has already waited. Available only if Business Advocate features are enabled for the system.

Copy ASAI UII During Conference/Transfer

Enables or disables copying user-to-user (UII) information during a conference call or during call transfers. Available only if ASAI Link Core Capabilities or Computer Telephony Adjunct Links are enabled for the system.

*** Note:**

The system copies all UUI information, not just ASAI UUI. Copying occurs only during a human-initiated conference or transfer. Communication Manager does *not* copy the UUI if the conference or transfer is initiated by ASAI.

MIA Across Splits or Skills

Enables or disables the removal of an agent from the Most Idle Agent (MIA) queue for all available splits/skills/hunt groups when answering a call from any of these groups.

Service Level Maximizer Algorithm

Selects an alternative algorithm for selecting agents and delivering calls to maximize service level targets. Available only if Service Level Maximizer is enabled for the system.

Valid Entry	Usage
actual	The Actual Service Level (ASL) is determined as a percentage on a hunt group basis using the number of accepted calls in the current interval divided by the total calls in the current interval. A call is counted as accepted if it is answered within the target service level time period.
weighted	The Weighted Service Level (WSL) is based on a weighting calculation that uses the difference between the target time and the estimated wait time.

Service Level Supervisor Call Selection Override

Determines whether Avaya Communication Manager changes agent call handling preferences when a skill using Service Level Supervisor exceeds its Level 1 threshold. Available only if Expert Agent Selection (EAS) and the Business Advocate features are enabled for the system.

Valid Entry	Usage
y	Overrides the normal call handling preferences of a skill's assigned agents in this situation.
n	Normal call handling preferences are in effect when the skill exceeds its Level 1 threshold.

ASAI

Call Classification After Answer Supervision?

For use with ASAI Outbound Call Management (OCM).

Valid Entry	Usage
y	Forces the server running Communication Manager to rely on the network to provide answer/busy/drop classification to the server. After the call has been answered, a call classifier can be added to perform answering machine, modem, and voice answering detection.
n	Always connects a classifier after call setup for determining call progress and answer. ISDN progress messages generally take precedence.

Send UCID to ASAI

Enables or disables the transmission of Universal Call ID (UCID) information to ASAI.

CALL MANAGEMENT SYSTEMS

REPORTING ADJUNCT RELEASE (determines protocol used by appl link)

CMS (appl mis)

Use this field to specify the release of the Call Management System (CMS) reporting adjunct. Management Information System (MIS) is a communication-interface link that connects CMS to Communication Manager.

Valid entry	Usage
R12	CMS Release 12 connects to Communication Manager through the mis1 link. CMS Release 12 is incompatible with Avaya IQ.
R13	CMS Release 13 connects to Communication Manager through the mis1 link. CMS Release 13 is incompatible with Avaya IQ.
R13.1	CMS Release 13.1 connects to Communication Manager through the mis1 link. CMS Release 13.1 and later is compatible with Avaya IQ.
R14.1	CMS Release 14.1 connects to Communication Manager through the mis1 link.
R15/R16	CMS Release 15 and 16 connects to Communication Manager through the mis1 link.
R16.1/R16.x	CMS Release 16.x and 17 connects to Communication Manager through the mis1 link.
R19.1+ (secured)	CMS Release 19.1 or higher connects to Communication Manager through the TLS link.
blank	CMS does not connect to Communication Manager. This is the default value.

In the case of two CMS adjuncts, one CMS adjunct connects to Communication Manager through the mis1 link and the other CMS adjunct connects to Communication Manager through the mis2 link.

For more information about **CMS (appl mis)**, see *Administering Avaya Aura® Call Center Elite*.

IQ (appl ccr)

Expert Agent Selection (EAS) and Universal Call ID (UCID) must be enabled before a connection can be established with Avaya IQ.

Valid Entry	Usage
4.0 5.0 5.1	The release of Avaya IQ connected to the ccr1 link, and to the ccr2 link for a second Avaya IQ.
blank	
blank	Avaya IQ is disconnected. This is the default.

OTHER CALL MANAGEMENT SYSTEM FIELDS

ACD Login Identification Length

Available only if Expert Agent Selection (EAS) is disabled and BCMS/VuStats Login IDs are enabled for the system.

Valid Entry	Usage
1 to 9	The number of digits for an ACD agent login ID that identifies an ACD agent to the Call Management System. This number must equal the number of characters in the agent's login ID and cannot be 0.

BCMS/VuStats Abandon Call Timer (seconds)

Valid Entry	Usage
1 to 10 blank	The number of seconds before calls are abandoned. Calls with talk time that is less than this number and are not held, are tracked by the Basic Call Management System (BCMS) and displayed by VuStats as ABAND calls.

BCMS/VuStats LoginIDs

Enables or disables valid agent login IDs to monitor call activity by an agent. Basic Call Management System (BCMS) and VuStats login IDs are available in addition to Expert Agent Select (EAS) login IDs if EAS is enabled on the system. Both BCMS and CMS use the same login ID for an agent.

BCMS/VuStats Measurement Interval

Available only if BCMS (Basic) or VuStats is enabled for the system.

Valid Entry	Usage
half-hour hour	Selects measurement intervals for polling and reporting data. There are a maximum of 25 time slots available for measurement intervals. If hour is specified, an entire day of traffic information will be available for history reports. Otherwise, only half a day will be available. This does not affect daily summaries as they always reflect traffic information for the entire day. The interval can be changed at any time, but remains until the current interval completes.

Clear VuStats Shift Data

Valid Entry	Usage
on-login	Clears shift data for an agent when the agent logs in.
at-midnight	Clears shift data for all agents at midnight.

Remove Inactive BCMS/VuStats Agents

Valid Entry	Usage
y	Agents are removed from reports when they have no staff time during the previous seven days.
n	Agents remain on the report even if they have no staff time for any period of time.

Validate BCMS/VuStats Login IDs

Valid Entry	Usage
y	Only login-IDs that have been administered for the Basic Call Management System (BCMS) can be entered.
n	Any ACD login of the proper length can be entered.

Feature-related system parameters: page 13

```
change system-parameters features Page 13 of 19
      FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER MISCELLANEOUS
    Callr-info Display Timer (sec): 10
      Clear Callr-info: next-call
    Allow Ringer-off with Auto-Answer? n

    Reporting for PC Non-Predictive Calls? n

    Agent/Caller Disconnect Tones? n

    Zip Tone Burst for Callmaster Endpoints: double

ASAI
    Copy ASAI UUI During Conference/Transfer? n
    Call Classification After Answer Supervision? n
      Send UCID to ASAI? n
    For ASAI Send DTMF Tone to Call Originator? y
    Send Connect Event to ASAI For Announcement Answer? n
```

CALL CENTER MISCELLANEOUS

Allow Ringer-off with Auto-Answer

If activated, an agent can use the ringer-off feature button that prevents ringing on Expert Agent Selection (EAS) auto-answer calls.

Callr-info Display Timer (sec)

Valid Entries	Usage
3 to 60	Administer the timer to display the information which caller has dialed on IP (H.323) telephones. The default value is 10 secs.

Clear Callr-info

Specifies when the collected digits Callr-Info display is removed from the agent or station display.

Valid Entry	Usage
leave-ACW	Leaves the display up while the agent is in After Call Work (ACW) mode.
next-call	Clears the display when the next call is received. This is the default.
on-call-release	Clears the display on the second line of a two-line display telephone or the display of a one-line display telephone with the Enhanced Callr-Info display for 1-line display phones option active when the call is released. The call is released if the receiving call is disconnect or if the agent presses the release button.

Interruptible Aux Notification Timer (sec)

Valid Entry	Usage
1 to 9	Specifies the number of seconds the endpoint interruptible aux notifications, the flashing lamp, display, or tone, are on before an auto-in-interrupt or manual-in-interrupt agent is made available. This delay ensures that an agent is not immediately made available when he presses an interruptible aux button. Also, this delay provides a brief period to an agent already in interruptible Aux mode before that agent is made available automatically. Default is 3.

PC Non-Predictive Reports Skill

Administers a skill hunt group used for reporting associated with Proactive Contact non-predictive switch-classified calls on a per-system basis. Reports are generated as though the agent were in the ACD-OUT state. Available only if reporting for PC Non-Predictive Calls is enabled.

Valid Entry	Usage
1 to 99	Skill number for Avaya Solutions Platform S8300 switches.
1 to 8000	Skill number for all other platforms

Prefer H.323 Over SIP For Dual-Reg Station 3PCC Make Call

The **Prefer H.323 Over SIP For Dual-Reg Station 3PCC Make Call** field is used to enable the Redirect 3PCC to H.323 station from SIP desktop station feature.

For more information about the Redirect 3PCC to H.323 station from SIP desktop station feature, see *Avaya Aura® Communication Manager Feature Description and Implementation*.

Valid entries	Usage
y	Redirects the 3PCC actions from a SIP desktop station to a remote soft phone or virtual private network H.323 telephone.
n	Does not redirect the 3PCC actions from a SIP desktop station to a remote soft phone or virtual private network H.323 telephone.

Reporting for PC Non-Predictive Calls

Activates or deactivates improved integration with Proactive Contact Outbound Calling for non switch-classified outbound calling. For example, this feature improves Call Management System tracking for switch-classified and non-switch classified (agent classified) outbound calls placed by the Proactive Contact soft dialer through ASAI. Default is n.

Send Connect Event to ASAI For Announcement Answer

Use this field to enable Communication Manager to send a Connected or Answered event to the calling party if the calling party is an ASAI application and the called party uses an announcement system. If you set this field to y, Communication Manager sends the Connected Answered event to the ASAI application when the announcement system at the called party answers the call. The announcement system answers the call by collecting the digits that the ASAI application provides. The system uses the **collect** command of the vector to collect the digits.

* Note:

Communication Manager does not send a Drop event when the call gets disconnected.

Valid entry	Usage
n	Communication Manager does not send the Connected or Answered event to the ASAI application. This is the default value.
y	Communication Manager sends the Connected or Answered event to the ASAI application.

* Note:

You should be careful while creating vectors that include announcements and other advanced features like route requests. The features can lead to:

- Call sequences in which Communication Manager sends a Connected event to the called party but do not send an Alerting event.
- Call sequences in which Communication Manager sends a Connected event to a calling party followed by an Alerting event to the same party but with different domains associated with the Connected event and the Alerting event.

Service Level Algorithm for SLM

Administers the algorithm used to determine service levels for Service Level Maximizer (SLM). Available only if Expert Agent Selection (EAS) has been enabled for the system.

Valid Entry	Usage
actual	The Actual Service Level (ASL) based on an algorithm that is determined as a percentage on a hunt group (skill) basis using the number of calls answered within the target service level time period in the current interval, divided by the total calls in the interval. This is the default.
weighted	The Weighted Service Level (WSL) algorithm based on a weighting calculation that uses the difference between the target time and the estimated wait time.

Zip Tone Burst For Callmaster Endpoints

Valid entry	Usage
double	When the station set type is a Callmaster series, Communication Manager retains existing operation which applies the following to ACD agents: • two bursts of zip tone for auto-answer ACD calls • two burst of ICI tweedle-dee tone for non-ACD auto-answer calls This is the default value.
single	Communication Manager eliminates the 2nd burst of zip/ICI tone reducing time for agent to start conversation with the caller and possibility of the agent and the caller hear open mike background noise between the first and second tones. Use with a Callmaster station type when the agent can always hear enough of the single burst auto-answer to recognize that a call is being delivered.

Feature-related system parameters: page 14

```

change system-parameters features                                     Page 14 of 19
      FEATURE-RELATED SYSTEM PARAMETERS

REASON CODES
      Aux Work Reason Code Type: none
      Logout Reason Code Type: none
      Two-Digit Aux Work Reason Codes? n
      Redirection on No Answer Aux Work Reason Code: 0
      Redirection on OPTIM Failure Aux Work Reason Code: 0

REDIRECTION ON IP CONNECTIVITY FAILURE
      Switch Hook Query Response Timeout:
      IP Failure Aux Work Reason Code: 0

MAXIMUM AGENT OCCUPANCY PARAMETERS
      Maximum Agent Occupancy Percentage: 100
      Maximum Agent Occupancy Aux Work Reason Code: 9
  
```

REASON CODES

Aux Work Reason Code Type

Valid Entry	Usage
none	An agent does not enter a Reason Code when entering AUX work.
requested	An agent can enter a Reason Code when entering AUX mode but is not forced to do so. Available only if Reason Codes and EAS are enabled for the system.
forced	An agent is forced to enter a Reason Code when entering AUX mode. Available only if Reason Codes and EAS are enabled for the system.

Logout Reason Code Type

Valid Entry	Usage
none	An agent does not enter a Reason Code when logging out.
requested	An agent can enter a Reason Code when logging out but do not want to force the agent to do so. Available only if Reason Codes and EAS are enabled for the system.
forced	An agent is forced to enter a Reason Code when logging out. Available only if Reason Codes and EAS are enabled for the system.

Redirection on No Answer Aux Work Reason Code

Redirection on No Answer (RONA) redirects a ringing ACD skill or direct agent call after an administered number of rings. When RONA is invoked for a call, the system places an agent in the Aux work mode, and the agent is unavailable to receive calls from other splits and skills. This field is used to define the Reason Code for reporting, instead of the default of 0, when the state of the agent is changed to Aux due to RONA.

Valid Entry	Usage
0 to 99	If the Two-Digit Aux Work Reason Code field is set to y.
0 to 9	If the Two Digit Aux Work Reason Code field is set to n.

ROOF Failure and Unreachable Aux Work Reason Code

This field provides redirection of calls back into queue or to the specified VDN. The agent is not aware that the line is in the Aux work mode during an IP connectivity failure. If connectivity is restored during the IP network TCP retry period, the lamp will indicate that the line is in the Aux work mode. The value in the field is used to define the Reason Code for reporting, instead of the default of 0, when the state of the agent is changed to Aux due to ROOF.

This field applies only to ACD agents using SIP hard or soft phones administered as Off-PBX Telephone Integration and Mobility (OPTIM) endpoints

Valid entry	Usage
0 to 99	If the Two-Digit Aux Work Reason Code field is set to y.
0 to 9	If the Two Digit Aux Work Reason Code field is set to n.

Two-Digit Aux Work Reason Codes

Enables or disables two-digit reason codes for agent state changes for AUX Work.

REDIRECTION ON IP CONNECTIVITY FAILURE

Auto-answer IP Failure AUX Reason Code

Valid Entry	Usage
0 to 99	The reason code assigned for auto-answer IP failure, as the reason the agent was put into AUX Work.

Switch Hook Query Response Timeout

Valid Entry	Usage
500 to 5000 (msec)	The time that call processing waits for a response from the switch hook query before Return on IP Connectivity Failure (ROIF) is triggered.
blank	ROIF is inactive.

MAXIMUM AGENT OCCUPANCY PARAMETERS

The Maximum Agent Occupancy (MAO) threshold is a system-administered value that is applied across all administered agents and is based on the total percentage of agent time in call service. MAO data is derived from the same calculations that are used to derive Least Occupied Agent (LOA).

When an agent who exceeds the specified MAO threshold attempts to become available, he or she is automatically placed in AUX mode for the reason code administered for this purpose. When the occupancy for such pending agents drops below the MAO, they are released from AUX mode and made available.

Maximum Agent Occupancy AUX Reason Code

Valid Entry	Usage
0 to 99	A reason code value. Do not use reason code 0. Default is 9.

Maximum Agent Occupancy Percentage

Valid Entry	Usage
0 to 100	The maximum percentage of time an agent can be taking calls. This time is based on Maximum Agent Occupancy (MAO) calculations. Default is 100.

SIP Transfer VDN Counted-Calls Options field descriptions

Field title	Field description
Suppress Target VDN counted-calls?	To suppress the counted-calls increment against a VDN which is the target of a SIP transfer, for example from AAEP. Valid entries are y and n . The default field option is n .

Feature-related system parameters: page 15

```

change system-parameters features                                     Page 15 of 19
      FEATURE-RELATED SYSTEM PARAMETERS

FORCED AGENT LOGOUT/AUX PARAMETERS
  Maximum Time Agent in ACW before Logout (sec): ____

      ACW Forced Logout Reason Code: 0
      Clock Time Forced Logout Reason Code: 0
  Forced Agent Logout by Location Reason Code: 0
      Forced Agent Logout by Skill Reason Code: 0
  Forced Agent Aux Work by Location Reason Code: 0
      Forced Agent Aux Work by Skill Reason Code: 0
  
```

FORCED AGENT LOGOUT PARAMETERS

ACW Forced Logout Reason Code

Available only if the Call Center Release is 3.0 or later and Expert Agent Selection (EAS) is enabled for the system.

Valid Entry	Usage
0 to 9	The reason for logging out the agent due to time-out in After Call Work (ACW).

Clock Time Forced Logout Reason Code

Valid Entry	Usage
0 to 9	The reason code for the Forced Agent Logout by Clock Time feature used by administrators to set a specific time when the system automatically logs out Expert Agent Selection (EAS) agents.

Maximum Time Agent in ACW before Logout (sec.)

Available only for Call Center Release 3.0 or later and if Expert Agent Selection (EAS) is enabled for the system.

Valid Entry	Usage
30 to 9999	A system-wide maximum time an agent can be in After Call Work (ACW). When this timer expires, the agent is logged out.
blank	There is no time-out. This is the default.

Forced Agent Aux Work by Location Reason Code

This field displays the reason that all agents in a location are put in the AUX work mode. The valid entries are from 0 to 9. The default value is 0.

Forced Agent Aux Work by Skill Reason Code

This field displays the reason that all agents of a skill are put in the AUX work mode. The valid entries are from 0 to 9. The default value is 0.

Forced Agent Logout by Location Reason Code

This field displays the reason that all agents in a location are logged out. The valid entries are from 0 to 9. The default value is 0.

Forced Agent Logout by Skill Reason Code

This field displays the reason that all agents of a skill are logged out. The valid entries are from 0 to 9. The default value is 0.

Forced SIP Agent Logout for Unreachable Reason Code

The field indicates the reason that an agent is logged out due to the final unreachable failure.

You can set a value from 0 to 9. By default, the value is set to 0.

SIP STATION REACHABILITY CHECKING OPTIONS

Enable Reachability for Domain Control SIP Stations

Using the Enable Reachability for Domain Control SIP Stations-y/n/disable-all field, you can enable or disable options polling for specific stations. The options are:

Valid entry	Usage
Y	To enable the polling feature for an individual SIP station.
N	To disable the polling feature for an individual SIP station.
disable-all	To apply systemwide values.

Enable SIP Agent Reachability?

Use the field to enable the AXP On-Prem agents to enable or disable **Options** polling.

SIP Reachability Polling Interval

The field defines the frequency of OPTIONS message sent by Communication Manager. The field has a range from 5 to 30 minutes. By default, the field is set to 5 minutes.

SIP Station Reachability Attempts

The field administers the number of times an **Options** poll fails before taking the final action with the polling during error response polling. This field has a range from 1 to 5, and 3 is set by default.

SIP Unreachable Polling Period

The field defines the period of time that the OPTIONS messaging continues after Communication Manager polls and declares the agent as unreachable. The field has a range from 0 to 1440 minutes. By default, the field is set to 60 minutes.

Feature-related system parameters: page 16

```
change system-parameters features Page 16 of 19
                                FEATURE-RELATED SYSTEM PARAMETERS

SPECIAL TONE
                                Special Dial Tone? n
                                Special Dial Tone for Digital/IP Stations: none
                                Busy/Reorder/Intercept Tone Timeout (Seconds): 10
REDIRECTION NOTIFICATION
                                Display Notification for Do Not Disturb? n
                                Display Notification for Send All Calls? n
                                Display Notification for Call Forward? n
                                Display Notification for Enhanced Call Forward? n
                                Display Notification for a locked Station? n
                                Display Notification for Limit Number of Concurrent Calls? n
                                Display Notification for Posted Messages? n
                                Scroll Status messages Timer(sec.):     

Chained Call Forwarding? n
```

SPECIAL TONE

Special Dial Tone

Enables or disables an audible tone indicating that the station is locked.

Related links

[Special Dial Tone](#) on page 262

Special Dial Tone for Digital/IP Stations

Valid Entry	Usage
all none non-display	System plays special dial tone for digital or IP stations, if the Limit Number of Concurrent Calls, Station Lock, or Call Forward feature is active.

Busy/Reorder/Intercept Tone Timeout (Seconds)

Valid Entry	Usage
3 to 45	Allows you to configure the time to play busy, reorder, or intercept tone. Communication Manager releases the call appearance after the specified time. This field is not applicable for SIP stations. This Timer is applicable only when Treatment type is tone. For other Treatment types like announcement, the timer will be ignored. This field is applicable to the following fields: • Controlled Outward Restriction Intercept Treatment • Controlled Termination Restriction (Do Not Disturb) • Controlled Station to Station Restriction • DID/Tie/ISDN/SIP Intercept Treatment • Invalid Number Dialed Intercept Treatment • Invalid Number Dialed Display • Restricted Number Dialed Intercept Treatment • Restricted Number Dialed Display

REDIRECTION NOTIFICATION

Chained Call Forwarding

Enables or disables Chained Call Forwarding. With Chained Call Forwarding, calls can be forwarded to as many as 10 forwarded-to stations using a preset forwarding path.

Display Notification Enhanced Call Forward

Enables or disables display notification for Enhanced Call Forward.

Display Notification for a locked Station

Enables or disables display notification for a locked station.

Display Notification for Call Forward

Enables or disables display notification for Call Forward.

Display Notification for Do Not Disturb

Enables or disables display notification for Do Not Disturb.

Display Notification for Limit Number of Concurrent Calls

Enables or disables display notification for Limit Number of Concurrent Calls.

Display Notification for Posted Messages

Enables or disables display notification for posted messages.

Display Notification for Send All Calls

Enables or disables display notification for Send All Calls.

Scroll Status messages Timer (sec.)

Valid Entry	Usage
5 to 10	The time in seconds for displaying scroll status messages. The status information shows the feature that has the highest priority and is enabled. The features in order of decreasing priority are: • Do Not Disturb • Send All Calls • Call Forward • Posted Messages • Limit Number of Concurrent Calls (LNCC) • Station Lock
blank	Deactivates the scrolling. This is the default.

Feature-related system parameters: page 17

```

change system-parameters features                                     Page 17 of 19
                                FEATURE-RELATED SYSTEM PARAMETERS

AUTOMATIC EXCLUSION PARAMETERS
                                Automatic Exclusion by COS? n

                                Recall Rotary Digit: 2

                                Duration of Call Timer Display (seconds): 3
WIRELESS PARAMETERS
Radio Controllers with Download Server Permission (enter board location)
  1:         2:         3:         4:         5:       

RUSSIAN MULTI-FREQUENCY PACKET SIGNALING
                                Re-try? n
                                T2 (Backward Signal) Activation Timer (secs): 20

```

AUTOMATIC EXCLUSION PARAMETERS

Automatic Exclusion by COS

Valid Entry	Usage
y	Enables automatic exclusion by a class of service when a user goes off-hook on a station with an assigned Exclusion button. This works only for stations on the local server running Communication Manager.
n	Exclusion operates normally.

Automatic Exclusion Coverage/Hold

Available only when Automatic Exclusion by COS is enabled. This field is applied only for "DND/SAC/Goto Cover" in Coverage path. This field is not applied for Bridge Call Appearance.

Valid Entry	Usage
y	The principal can bridge onto the call by pressing the appropriate bridged appearance button. And, if the coverage point places the exclusion call on hold, the principal can retrieve the call.
n	If a coverage point has answered a call and there is active exclusion on the call, the principal cannot bridge onto the call. And, if the coverage point places the exclusion call on hold, the principal cannot retrieve the call.

Automatic Exclusion with Whisper Page

Available only Automatic Exclusion by COS is enabled.

Valid Entry	Usage
y	The whisper page goes through to an excluded call.
n	The whisper page is denied when a station attempts to whisper page to a station that is on an excluded call.

Duration of Call Timer Display

Valid Entry	Usage
3 to 30	The length of time in 3 second increments that the call information remains on display after the call is terminated.

Password to Change COR by FAC

If this field contains a value, a password option is required. Accepts from four to eight digits.

Available only if **Change COR by FAC** is enabled for the system.

Recall Rotary Digit

Valid Entry	Usage
0 to 9	The digit used for rotary telephones to receive recall dial tone. Dialing this digit simulates switch-hook flash so that users of rotary telephones can use features such as conference and transfer. The telephone must also be administered to use the recall rotary. This should be a number that is not the first digit in normal dialing patterns.

IP PARAMETERS

Direct IP-IP Audio Connections

Use this field to enable direct audio connections between IP endpoints. Direct audio connections save bandwidth resources and improve the sound quality of voice over IP transmissions.

 Note:

This field has no impact on a SIP station (OPS) on shuffling. For SIP stations, this field is controlled by the "Direct IP-IP Audio Connections" field in the Signaling group form.

RUSSIAN MULTI-FREQUENCY PACKET SIGNALING

Re-try

Enables or disables the resending of address information on outgoing Russian MFP trunks. Specifically, the server running Communication Manager resends Russian MFP calling party number and dialed number information to the local telephone company central office (CO). The

server resends the information only once over another outgoing trunk port of the same trunk group if Communication Manager receives a message that the information was received incorrectly by the CO. The switch also sends Russian MFP information over another trunk port if Communication Manager does not receive a timely response for the information.

T2 (Backward Signal) Activation Timer (secs)

Valid Entry	Usage
5 to 20	The number of seconds Communication Manager waits to receive confirmation after sending calling party number and dialed number information on outgoing Russian MFP trunks.

Feature-related system parameters: page 18

```
change system-parameters features Page 18 of 19
FEATURE-RELATED SYSTEM PARAMETERS

INTERCEPT TREATMENT PARAMETERS
  Invalid Number Dialed Intercept Treatment: tone
    Invalid Number Dialed Display: _____
  Restricted Number Dialed Intercept Treatment: tone
    Restricted Number Dialed Display: _____
  Intercept Treatment On Failed Trunk Transfers? n

WHISPER PAGE
  Whisper Page Tone Given To: all

6400/8400/2420J LINE APPEARANCE LED SETTINGS
  Station Putting Call On Hold: green wink
    Station When Call Is Active: steady
  Other Stations When Call Is Put On Hold: green wink
    Other Stations When Call Is Active: green
      Ringing: green flash
      Idle: steady

Pickup On Transfer? y
```

INTERCEPT TREATMENT PARAMETERS

Intercept Treatment on Failed Trunk Transfers

Valid Entry	Usage
y	Provides intercept treatment to calls failing trunk transfers.
n	Drops calls failing trunk transfers.

Invalid Number Dialed Display

Used to display a name in either Latin or Asian characters for an invalid number calling in. This field supports both a NAME1 and a NAME2 value. A NAME1 value directs the system to use the table of names that contains Latin characters that can be displayed. A value of NAME2 directs the system to use the UTF-8 table of names that contains non-ASCII characters suitable for Asian language names. Accepts up to 15 alphanumeric characters.

*** Note:**

The system displays the message administered for the **Invalid Number Dialed Display** field on the telephone only if the **Invalid Number Dialed Intercept Treatment** field is set to `announcement`.

Invalid Number Dialed Intercept Treatment

The type of intercept treatment the end-user hears after dialing an invalid number.

Valid Entry	Usage
announcement	Provides a recorded announcement when the end-user dials an invalid number. You select and record the message. Requires entering the extension number for the announcement.
tone	Provides intercept tone when the end-user dials an invalid number. This is the default.

Restricted Number Dialed Display

The string of alphanumeric characters assigned for calls that are denied because of COS, COR, or FRL restrictions. This field supports both a NAME1 and a NAME2 value. A NAME1 value directs the system to use the table of names that contains Latin characters. A value of NAME2 directs the system to use the UTF-8 table of names that contains non-ASCII characters suitable for Asian language names. Accepts up to 15 alphanumeric characters.

*** Note:**

The system displays the message administered for the **Restricted Number Dialed Display** field on the telephone only if the **Restricted Number Dialed Intercept Treatment** field is set to `announcement`.

Restricted Number Dialed Intercept Treatment

The type of intercept treatment the caller hears after dialing a number restricted from them due to COS, COR, or FRL restrictions.

Valid Entry	Usage
tone	Provides intercept tone. This is the default.
announcement	Provides a recorded announcement. You select and record the message. Requires entering the extension number for the announcement.

WHISPER PAGE

Whisper Page Tone Given To

Determines who should hear a Whisper Page beep tone.

Valid Entry	Usage
all	All parties hear the whisper page beep tone.
paged	The whisper page feature sends a beep tone only to the paging and the paged party.

6400/8400/2420J LINE APPEARANCE LED SETTINGS

Warning:

The following fields change only the LED operation for 84xx and 64xx model telephones. When the LED operation is changed using any of these fields, then IP Agent and IP Softphone using a station type of 84xx or 64xx does not work. For station types other than 84xx or 64xx, a change to the LEDs using these fields does not affect either IP Agent or IP Softphone.

Note:

The system generates a warning if the default values of the LED Settings field are changed. The warning message states

WARNING: Avaya Softphone will not operate correctly if this value is changed

This warning message displays for Avaya Communication Manager Release 3.1 or later.

Display Information With Bridged Call

Controls whether or not name and number for a bridged call are displayed on the telephone of the called party. This field does *not* control the content of the display.

Idle

Note:

This field applies only to 8400 and 6400 series telephones. The 2400 series telephone uses icons rather than LEDs.

Valid Entry	Usage
steady	The LED flash rate for an idle station. Default is steady.
off	The correct value for the Japanese environment is off.

Other Stations When Call Is Active

Controls a DCP bridged appearance LED for those non-active parties with a bridged appearance that is active.

*** Note:**

This field applies only to 8400 and 6400 series telephones. The 2400 series telephone uses icons rather than LEDs.

Valid Entry	Usage
green	The LED color. Default is green.
red	Red is the correct value in the Japanese environment.

Other Stations When Call Is Put On-Hold

Controls LED options for the other stations with a Bridged Appearance that have been placed on hold, but the user of this station has not pressed the hold button.

*** Note:**

This field is for a DCP bridged appearance LED color and flash rate when a call on a bridged appearance is put on hold by another party on the DCP bridged appearance. Additionally, this field only applies to 8400 and 6400 series telephones. The 2400 series telephone uses icons rather than LEDs. Correct operation in the Japanese environment requires the administrator to select the values red and flash for this field.

Valid Entry	Usage
green red	The color of the LED. Default is green.
off wink inverse-wink flash flutter broken-flutter steady	The flash rate for the LED. Default is wink.

Pickup on Transfer

Valid Entry	Usage
y	A station with a bridged appearance can pick up a call on hold because of a transfer.
n	Another station with bridged appearances cannot pick up a call on hold because of a transfer.

Ringing

Controls the LED color and flash rate while a call is ringing.

*** Note:**

This field only applies to 8400 and 6400 series telephones. The 2400 series telephone uses icons rather than LEDs. Correct operation in the Japanese environment requires the administrator to select the values red and wink for this field.

Valid Entry	Usage
green red	The LED color. Default is green.
off wink inverse-wink flash flutter broken-flutter steady	The flash rate. Default is flash.

Station Putting Call On-Hold

Controls the LED color and flash rate on the 8400 and 6400 series telephones for a call held on a Primary or Bridged Appearance.

Valid Entry	Usage
green red	The color of the LED. The LED for the color not selected is turned OFF. Default is green.
off wink inverse-wink flash flutter broken-flutter steady	The flash rate for a call on hold. Default is wink.

Station When Call is Active

Controls the red LED on the 8400 and 6400 series telephones, for a station active on a call.

Valid Entry	Usage
steady	Communication Manager controls the red LED. This is the default.
off	The red LED is always OFF.

Feature-related system parameters: page 19

```

change system-parameters features                                     Page 19 of 19
                                FEATURE-RELATED SYSTEM PARAMETERS
IP PARAMETERS
    Direct IP-IP Audio Connections? y      IP Audio Hairpinning? n
    Synchronization over IP? n

    SIP Endpoint Managed Transfer? n

    Expand ISDN Numbers to International for 1XCES? n

CALL PICKUP
    Maximum Number of Digits for Directed Group Call Pickup: 4
    Call Pickup on Intercom Calls? y      Call Pickup Alerting? y
    Temporary Bridged Appearance on Call Pickup? y    Directed Call Pickup? n
    Extended Group Call Pickup: none
    Enhanced Call Pickup Alerting? n

    Call Pickup for Call to Coverage Answer Group? n
    Display Information With Bridged Call? n
    Keep Bridged Information on Multiline Displays During Calls? y
    PIN Checking for Private Calls? n

```

IP PARAMETERS

Allow Bridge DM Answer

Use this field to enable the Bridge call answer as Direct Media feature. To enable this feature, set the **SIP Endpoint Managed Transfer** and **Allow Bridge DM Answer** fields to **y**.

Valid entry	Usage
y	The calling party can have direct media path with the bridge call appearance of the called party station if all the involved stations support the Enhanced SIP Signaling feature.
n	This is the default value.

Direct IP-IP Audio Connections

Use this field to enable direct audio connections between IP endpoints. Direct audio connections save bandwidth resources and improve the sound quality of voice over IP transmissions.

 Note:

This field has no impact on a SIP station (OPS) on shuffling. For SIP stations, this field is controlled by the "Direct IP-IP Audio Connections" field in the Signaling group form.

Synchronization over IP

Enables or disables the Clock Synchronization over IP (CSolP) feature to provide IP sync to a group of members.

Valid entry	Usage
y	Enables the CSolP feature.
n	Disables the CSolP feature. This is the default value.

SDP Capability Negotiation for SRTP

The system displays SDP Capability Negotiation for SRTP when the **Media Encryption Over IP** field is set to y. Enables or disables SIP SRTP enhancements with Direct Media support in Communication Manager using Session Description Protocol (SDP) capability negotiation.

SIP Endpoint Managed Transfer

Use this field to enable the **SIP Endpoint Managed Transfer** feature. Before enabling the feature, you must ensure that the **Network Call Redirection** field is disabled for the SIP trunk group between Communication Manager and Session Manager.

Note:

The **SIP Endpoint Managed Transfer** field only controls the SIP transfer changes made in release 6.2. The field does not control some related Communication Manager SIP conference changes.

For more information on the **SIP Endpoint Managed Transfer** feature and on the related SIP conference changes, see *Avaya Aura® Communication Manager Feature Description and Implementation*.

Expand ISDN Numbers to International for 1XCES

Enable this field to convert ISDN calling numbers into the full international format when notifying Avaya one-X® Client Enablement Services about a call. This conversion is necessary in countries with complex numbering plans so that users can return calls from their one-X Mobile call log.

Valid entry	Usage
y	Convert ISDN calling numbers into the full international format when notifying an Avaya one-X® Client Enablement Services server about a call. If you enable this field, you must also configure the following: • The Local E.164 Country Code field on the Location Parameters form: To convert a number from national to the international format. • The City/Area column on the Locations form: To convert a number from local or subscriber format to the international format.
n	Do not convert the ISDN calling numbers to the full international format, when notifying an Avaya one-X® Client Enablement Services server about a call. This is the default value.

CALL PICKUP

A pickup group is a collection, or list, of individual extensions that have permission to answer calls originally dialed towards a different member of the group.

Call Pickup Alerting

Enables or disables Call Pickup Alerting on a system-wide basis. Call Pickup Alerting provides pickup group members with a visual indication on the **Call Pickup** status lamp for calls eligible to be answered using Call Pickup.

Call Pickup for call to Coverage Answer Group

Use this field to enable or disable interaction between Coverage Answer Group and Call Pickup group. Same extension can be configured to be part of coverage answer-group and pickup group.

Valid entries	Usage
y	When a call rings at a Coverage Answer Group member, Communication Manager triggers call pickup alerting if the Coverage Answer Group member is part of a Call Pickup group.
n	When a call rings at a Coverage Answer Group member, Communication Manager does not trigger call pickup alerting if the Coverage Answer Group member is part of a Call Pickup group. This is applicable for the H.323, DCP, Analog and SIP endpoints. The default value is n.

Call Pickup on Intercom Calls

System-wide users can attend or drop an intercom call using the Call Pickup or Directed Call Pickup features.

Directed Call Pickup

Activates or deactivates the use of Directed Call Pickup, where users can specify what ringing telephone they want to answer.

Enhanced Call Pickup Alerting

Valid entries	Usage
y	Apply the Enhanced Call Pickup Delay Timer.
n	Silently flash the pickup button.

Audible Notification

Valid Entry	Usage
1 to 15 seconds	Set the audio notification timer. When the timer is set and a call arrives to a group member, the other members of the group are informed with an audio notification when the timer expires.

Enhanced Call Pickup Delay Timer (sec.) Display

Valid Entry	Usage
1 to 15	When a call arrives at one of the group members the other members are informed with a display notification. Set the timer in seconds to delay the display notification. When the timer expires the display notification is sent.

Extended Group Call Pickup

Selects how call pickup groups can answer calls directed to another call pickup group.

Valid Entry	Usage
flexible	A one-to- <i>n</i> pickup group-to-extended pickup group mapping.
simple	A one-to-one pickup group-to-extended pickup group mapping.
none	Extended group call pickup is prohibited.

Keep Bridged Information on Multiline Displays During Calls

If this field is enabled, for Multiline Display sets (1603, 1608, 1616, 4610, 4620, 4620SIP, 4620SIPCC, 4622, 4630, 9620, 9630 and 9650), the system displays the incoming caller information for the duration of the call. The system clears the caller information on the display for the user who puts the call on hold but retains the information on the display for the other users on the bridge. For outgoing calls, if the user on the bridge puts the call on hold, the display for the other user on the bridge is populated with the caller information. The default is n.

Maximum Number of Digits for Directed Group Call Pickup

Valid Entry	Usage
1 to 4	The maximum number of digits accepted for the pickup group number. The pickup group number is complete when it is followed by a # symbol. Default is 4.

PIN Checking for Private Calls

Enables or disables the PIN Checking for Private Calls feature that restricts users from making internal or external private calls by forcing them to enter a PIN code after dialing a PIN Feature Access Code (FAC).

Related links

[PIN Checking for Private Calls Using AAR Access Code](#) on page 225

[PIN Checking for Private Calls Using ARS Access Code](#) on page 225

[PIN Checking for Private Calls Access Code](#) on page 225

Temporary Bridged Appearance on Call Pickup

Valid entries	Usage
y	Enter y to allow a temporary bridged appearance for calls answered with the Call Pickup or Directed Call Pickup features. This field controls this capability on a system-wide basis.
n	Enter n to prevent the temporary bridged appearance of calls answered with these features.

Commercial Cost Avoidance (CCA) Related System Parameters

```

change system-parameters features                                     Page 20 of 20
      COMMERCIAL COST AVOIDANCE (CCA) RELATED SYSTEM PARAMETERS

      CCA Enabled? y
      CTI Link Number: _
      LSC CCA-ID: Main
Primary MFSS CCA-ID:
Backup MFSS CCA-ID:

```

Backup MFSS CCA-ID

Use this field to search the PSTN or DSN mapping entries in the LDAP database. Enter a string without any white space.

CCA Enabled

Use this field to enable **Cost Commercial Avoidance (CCA)**.

Valid entry	Usage
y	The system displays the CCA-related fields.
n	The system does not display the CCA-related fields. The default value is n.

CTI Link Number

The CTI link number establishes a connection between Communication Manager and AES applications. Enter a CTI Link number of type ASAI-IP or ADJ-IP that is not used in any vector.

LSC CCA-ID

Use this field to search the PSTN or DSN mapping entries in the LDAP database. The default value of this field is the value populated in the **Call Control Agent Identification (CCA-ID)** field of MLPP. Though the value in the **LSC CCA-ID** field is from the **Call Control Agent Identification (CCA-ID)** field, you can change the value of the **LSC CCA-ID** field.

 Note:

You can confirm the value of **Call Control Agent Identification (CCA-ID)** field on page 2 of the change system-parameters mlpp screen.

Primary MFSS CCA-ID

Use this field to search the PSTN or DSN mapping entries in the LDAP database. Enter a string without any white space.

Chapter 52: Firmware Station Download

This screen downloads firmware to multiple stations of the same telephone type, either 2420 or 2410 DCP telephones. Downloads firmware to as many as 1000 stations per download schedule. You can schedule a specific time for the download, or you can administer the download to run immediately.

Example command: `change firmware station-download`

Beginning Station

The first extension number in the range of telephones used to download the firmware. Up to 1000 stations can be included in a scheduled download. Accepts up to 16 digits.

Continue Daily Until Completed

Enables or disables the execution of a firmware download each day at the scheduled time until all specified telephones have received the firmware.

Download Set Type

Valid Entry	Usage
2410 DCP 2420 DCP	The set type of DCP telephones to which firmware is to be downloaded.

Download Station Type

Type of station to which the firmware is to be downloaded.

Ending Station

The last extension number in the range of telephones used to download firmware. Up to 1000 stations can be included in a scheduled download. Accepts up to eight digits.

Schedule Download

Enables or disables a request to schedule a time for a firmware download to multiple DCP stations.

Source File

The name of the file used to retrieve the firmware download. Accepts up to 32 alphanumeric characters.

Related links

[File to Retrieve](#) on page 771

Start Date/Time

Available only if **Schedule Download** is enabled.

Valid Entry	Usage
mm, dd, yyyy; hh, mm	The month, day, year, and time for the firmware download to begin.

Related links

[Schedule Download](#) on page 318

Stop Date/Time

Available only if **Schedule Download** is enabled.

Valid Entry	Usage
mm, dd, yyyy; hh, mm	The month, day, year, and time for the firmware download to end.

Related links

[Schedule Download](#) on page 318

Chapter 53: Group Paging Using Speakerphone

Assigns digital speakerphones to a paging group. Users can page all the telephones in the group simultaneously by dialing the group extension.

Example command: `change group-page n`, where *n* is the assigned group number.

Alert

Set this field to `y` for telephones that require an alert message to ring for an inbound call from a group page number. For example, Spectralink wireless telephones. Communication Manager does not activate the speakerphone of the Spectralink wireless telephones for a group page number. Therefore, the users do not hear the call ring from a group page number until the users manually answer the call.

 Note:

Most of the Avaya telephones have the auto-answer capability, so if you set the **Alert** field to `y`, an Avaya telephone does not ring because the group paging initiates immediately.

Value	Usage
y	Telephones, such as Spectralink wireless, receive an alert to ring for an inbound call from a group page number.
n	Telephones, such as Spectralink wireless, do not receive an alert to ring for an inbound call from a group page number. This is a default value.

ASAI

Value	Usage
y	Enables ASAI events for paging groups. Enabling this functionality creates large spikes in the messaging traffic to the application.
n	ASAI events are not forwarded to the application for any call that has a paging group as parties on the call.

COR

Valid Entry	Usage
0 to 995	A class of restriction (COR). To page the group, users' class of restriction must give them calling permission for the group's class of restriction.

Ext

Assigns a telephone extension to the group.

Group Extension

The extension users dial to page the members of this group.

Group Name

A name for the group that is informative to users. The callers' telephone displays when the caller pages the group. Accepts up to 27 alphanumeric characters.

 Note:

Avaya BRI deskphones support only ASCII characters because non-ASCII characters, such as Eurofont and Kanafont, show up incorrectly.

Group Number

The identifying number the server running Communication Manager assigns to the group when it is created.

Group Timeout (secs)

Valid Entry	Usage
0 to 600	Administer a timeout in seconds for the group page. After timeout, the paging party will be disconnected. The default value is 0 for no timeout.

Name

The name assigned to each extension in the group.

**Note:**

Avaya BRI deskphones support only ASCII characters because non-ASCII characters, such as Eurofont and Kanafont, show up incorrectly.

TN

The tenant number for this paging group. With tenant number, group paging can be partitioned by tenant.

Zip Tone Timer (secs)

If you set the value of the **Zip Tone Timer (secs)** field to 2 seconds, Communication Manager times out after this time and plays a Zip tone.

If you set the value in the **Zip Tone Timer (secs)** field to greater than 2 seconds and less than or equal to 10 seconds, Communication Manager tries to terminate every member of the group page within 2 seconds. If the attempt is successful, Communication Manager plays the Zip tone. Otherwise, Communication Manager sets the timer for the remaining time. After the second timer times out, Communication Manager plays the Zip tone and does not wait for any member of the group page to answer the call.

For example, if you set the value as 6 seconds, Communication Manager tries to terminate every member of the group page within 2 seconds. If the attempt is successful, Communication Manager plays the Zip tone. Otherwise, Communication Manager sets the timer for the remaining 4 seconds. After the 4 seconds timer times out, Communication Manager plays the Zip tone.

The range for this field is 2 through 10 seconds, and the default value is 7 seconds. If you set the value of this field between 2 through 6 seconds, the system displays the following warning

message: Group page may not reach all members with timer value less than 7 sec.

Chapter 54: Holiday Table

Defines individual holidays or holiday ranges.

Example command: `change holiday-table n`, where *n* is the holiday table number.

change holiday-table 1

Page 1 of 1

HOLIDAY TABLE

Number: 1

Name: _____

START				END				Description
Month	Day	Hour	Min	Month	Day	Hour	Min	
__	__	__	__	__	__	__	__	_____
__	__	__	__	__	__	__	__	_____
__	__	__	__	__	__	__	__	_____
__	__	__	__	__	__	__	__	_____
__	__	__	__	__	__	__	__	_____
__	__	__	__	__	__	__	__	_____
__	__	__	__	__	__	__	__	_____
__	__	__	__	__	__	__	__	_____
__	__	__	__	__	__	__	__	_____
__	__	__	__	__	__	__	__	_____
__	__	__	__	__	__	__	__	_____
__	__	__	__	__	__	__	__	_____
__	__	__	__	__	__	__	__	_____
__	__	__	__	__	__	__	__	_____

Description

A phrase that describes the holiday. Accepts up to 27 characters.

End Day

Valid Entry	Usage
1 to 31	The ending day of the holiday.

End Hour

Valid Entry	Usage
0 to 23	The ending hour of the holiday using a 24-hour clock.

End Min

Valid Entry	Usage
0 to 59	The ending minute of the holiday.

End Month

Valid Entry	Usage
1 to 12	The ending month of the holiday.

Name

The name of the holiday table. Accepts up to 27 characters.

Number

Valid Entry	Usage
1 to 10	The holiday table number.

Start Day

Valid Entry	Usage
1 to 31	The starting day of the holiday.

Start Hour

Valid Entry	Usage
0 to 23	The starting hour of the holiday using a 24-hour clock.

Start Min

Valid Entry	Usage
0 to 59	The starting minute of the holiday.

Start Month

Valid Entry	Usage
1 to 12	The starting month of the holiday.

Chapter 55: Hospitality

Implements the system parameters associated with the hospitality features. Available only if Hospitality features are enabled for the system.

Example command: `change system-parameters hospitality`

Related links

[Hospitality \(Basic\)](#) on page 733

[Hospitality \(G3V3 Enhancements\)](#) on page 733

Hospitality: page 1

Client Room Coverage Path Configuration

Indicates the server and the Property Management System (PMS) exchange coverage path information for guest stations.

Valid Entry	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	The server and PMS exchange and accept coverage path information. This field does not apply to normal PMS Protocol mode.

Controlled Restrictions Configuration

Indicates whether controlled restriction information is exchanged between the server and the PMS.

Valid Entry	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	The server and the PMS exchange and accept controlled restriction information.

Default Coverage Path for Client Rooms

Applies only to stations with a client room class of service in the occupied mode. This field is used for transparent or ASCII mode. The value in this field is also used during a translation save as the coverage path for each station with client room class of service.

Valid Entry	Usage
1 to 9999 blank	The coverage path assigned when the server receives a check-out message for a valid extension or a new check-in.

Forward PMS Message to INTUITY Lodging

This field is used only in ASCII mode.

Valid Entry	Usage
y	PMS-to-INTUITY messages are sent through the server.
n	PMS-to-INTUITY messages are sent directly to the Avaya INTUITY Lodging system.

Housekeeper Information Configuration

Indicates whether housekeeper information is exchanged between the server and the PMS.

Valid Entry	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.
act-pms	The server and PMS exchange and accept housekeeper information.

Journal/Schedule Endpoint

Valid Entry	Usage
<i>Valid data extension number</i>	A valid data extension number that is assigned to the data module connected to the Journal/Schedule printer. Cannot be a VDN extension. This extension can be the same as the PMS/ Log printer and both sets of reports can be printed on the same printer. This extension is dialed by the server to send journal information or schedule reports to the printer.
PMS_LOG	The printer is connected over a TCP/IP link, and this link is administered with a PMS_LOG service type for IP Services.
PMS_JOURNAL	The printer is connected over a TCP/IP link, and this link is administered with a PMS_JOURNAL service type for IP Services.

Related links

[Service Type](#) on page 429

Message Waiting Configuration

Indicates whether message waiting notification requests and changes are being exchanged between the server and the PMS.

Valid Entry	Usage
act-nopms	The message is acknowledged (MESSAGE ACK), but no action is taken.

Table continues...

Valid Entry	Usage
act-pms	Message waiting is active on the server and information between the PMS and server is transmitted.

Number of Housekeeper ID Digits

Valid Entry	Usage
0 to 6	The number of digits that the housekeeper must dial for identification.

PMS Log Endpoint

Valid Entry	Usage
<i>Valid extension</i>	The data extension number the server dials to access PMS. Cannot be a VDN extension.
PMS	The PMS is connected over a TCP/IP link. This link is administered with PMS service type for IP Services.

Related links

[Service Type](#) on page 429

PMS LINK PARAMETERS

ASCII mode

Enables or disables ASCII-only mode used for the PMS message set. Available only with a transparent PMS protocol mode.

PMS Link Maximum Retransmission Requests

Valid Entry	Usage
1 to 5	The number of times that the PMS can request acknowledgment for a message sent to the server.

PMS Link Maximum Retransmissions

Valid Entry	Usage
1 to 5	The number of times that the server retransmits a message to the PMS in response to a negative acknowledgment, or sends an inquiry for acknowledgment from the PMS before giving up on the message.

Milliseconds Before PMS Link Acknowledgment Timeout

Regulates how quickly the system responds to a message from the PMS, also known as “pace timing”. This value is also used as the inquiry message (ENQ) time-out value. Should be kept as short as possible.

Valid Entry	Usage
100 to 1500	The time in milliseconds the system waits for an acknowledgment from the PMS indicating it correctly received a message.

PMS Endpoint

Valid Entry	Usage
<i>Valid extension</i>	The data extension number the server dials to access PMS. Cannot be a VDN extension.
PMS	The PMS is connected over a TCP/IP link. This link is administered with PMS service type for IP Services.

Related links

[Service Type](#) on page 431

PMS Protocol Mode

Valid Entry	Usage
normal	Indicates the message protocol mode used between the server and PMS. Coordinate this option with your PMS vendor.
transparent	

Seconds Before PMS Link Idle Timeout

Valid Entry	Usage
5 to 20	The idle time in seconds that the server waits for an acknowledgment from the PMS before the server enters link failure mode from the PMS transmission link.

Take Down Link for Lost Messages

Enables or disables taking down the PMS link if messages are getting lost. The PMS error log should be monitored if disabled.

Hospitality: page 2

Announcement Ports

Indicates the equipment location of two ports on the voice synthesizer circuit pack. Available only with a voice-synthesis announcement type.

Valid Entry	Usage
1 to 250	Gateway

Table continues...

Valid Entry	Usage
V1 to V9	Module
01 to 31	Port number

Related links

[Announcement Type](#) on page 331

Announcement Type

Indicates the type of automatic wake up announcement the hotel guest receives.

Valid Entry	Usage
external	Applicable when using an announcement adjunct. Requires entry of the circuit connection to the external announcement equipment.
integrated	Requires entry of an extension for the integrated announcement.
mult-integ	Automatic Wakeup can use integrated announcement board to play any one of multiple announcements to different extensions during a wake up call. Requires entry of an announcement extension.
voice-synthesis	A voice synthesis message is heard during the wake up announcement.
music-on-hold	Uses the Music-on-Hold feature to provide the wake up announcement.
silence	Silence is heard during the wake up announcement.

Related links

[Announcement Ports](#) on page 330

[Auxiliary Board for Announcement](#) on page 331

[Integrated Announcement Extension](#) on page 333

[Default Announcement Extension](#) on page 332

Automatic Selection of DID Numbers

Enables or disables the Automatic Selection of DID Numbers for Guest Rooms feature. This feature assigns a two- to five-digit extension from a predetermined list of numbers to a hotel room telephone number that is disassociated with the room number.

Auxiliary Board for Announcement

The equipment location of an auxiliary trunk circuit that connects to the external announcement equipment. Available only for an external announcement type.

Valid Entry	Usage
1 to 250	Gateway
V1 to V9	Module
01 to 31	Port number

Related links

[Announcement Type](#) on page 331
[Integrated Announcement Extension](#) on page 333
[Default Announcement Extension](#) on page 332

Custom Selection of VIP DID Numbers

The selection or removal of a DID number assigned to a room when a guest checks in. Available only if **Automatic Selection of DID Numbers** is enabled.

Related links

[Automatic Selection of DID Numbers](#) on page 331

Daily Wakeup

Each extension permission can request daily wake up calls.

Default Announcement Extension

The default wake up announcement extension when using the integrated announcement board.
Available only with a multi-integrated announcement type.

Related links

[Announcement Type](#) on page 331
[Auxiliary Board for Announcement](#) on page 331
[Integrated Announcement Extension](#) on page 333

Digit to Insert/Delete

Valid Entry	Usage
0 to 9	The current PMS message set uses the extension number as the room identifier. In many customer configurations, the leading digit of the extension number is dropped to screen the room number. To accommodate PMS devices that are based on room number and not extension. This leading digit can be deleted on messages from Avaya Communication Manager to the PMS, and then inserted back on messages from the PMS to Communication Manager.  Note: The PMS interface supports three-, four-, or five-digit extensions, but prefixed extensions do not send the entire number across the interface. Only the assigned extension number is sent. Therefore, do not use prefixed extensions for numbers that are also going to use the Digit to Insert/Delete function.

Display Room Information in Call Display

Indicates the type of guest room information displayed on telephone displays.

Valid Entry	Usage
y	Telephones display the name and room number. The extension number and room number are not always the same number.
n	Telephones display the name and extension number.

Dual Wakeup

Allows or disallows each extension permission to request two wake up calls within one 24-hour period.

Extension to Receive Failed Wakeup LWC Messages

An extension that indicates where unsuccessful wake up LWC messages are stored. This is usually administered to an unassigned extension or to the attendant (attd). This extension cannot be a VDN extension. In addition, a LWC lamp for that extension is usually assigned to the attendant console as an indication of failed wake up calls.

Integrated Announcement Extension

The wake up announcement extension when using the integrated announcement board.

Available only for integrated announcement types.

Related links

[Announcement Type](#) on page 331

[Auxiliary Board for Announcement](#) on page 331

[Default Announcement Extension](#) on page 332

Length of Time to Remain Connected to Announcement

Valid Entry	Usage
0 to 300	The length of time in seconds that a hotel guest is connected to an announcement. Applies only after the guest has heard the announcement completely one time, but continues to listen.

Number of Digits from PMS

The number of digits being sent from the PMS to the server to identify room numbers.

Valid Entry	Usage
1 to 4	For normal mode
1 to 5	For transparent or ASCII mode
blank	For mixed numbering in the server.

Number of Digits in PMS Coverage Path

Valid Entry	Usage
3 or 4	The number of digits in the coverage path.

PMS Sends Prefix

Enables or disables PMS Sends Prefix to Indicate if the PMS sends a prefix digit to the server as part of the room numbering plan.

Room Activated Wakeup with Tones

Enables or disables the activation of wake up calls with tones. Wake up calls can be activated with tones that prompt users for the time they want to wake up. This allows room activated wake up calls without the use of a speech synthesizer or a display telephone.

Routing Extension on Unavailable Voice Synthesis

Valid Entry	Usage
<i>Assigned extension</i>	A call is placed to this extension or to the attendant if a voice synthesis port is unavailable during voice synthesis entry of wakeup requests. This extension cannot be a VDN extension.
attd	An attendant group code.

Time of Scheduled Emergency Access Summary Report

Valid Entry	Usage
hh:mm:am/pm	The time of day that the Emergency Access Summary Report gets printed on the Journal/ Schedule printer.

Caution:

Set the report for a time other than when the system does its scheduled maintenance tests.

Time of Scheduled Wakeup Activity Report

Valid Entry	Usage
hh:mm:am/pm	The time of day that the Wakeup Activity Report gets printed on the Journal/ Schedule Printer. This report summarizes the wake up activity for each extension that had wake up activity for the past 24 hours.

Caution:

Set the report for a time other than when the system does its scheduled maintenance tests.

Time of Scheduled Wakeup Summary Report

Valid Entry	Usage
hh:mm:am/pm	The time of day that the Wakeup Summary Report gets printed on the Journal/Schedule printer. This report gives an hour-by-hour summary of the number of scheduled wake up calls and a list of extensions to which wake up calls were attempted but were left incomplete during the hour.

Caution:

Set the report for a time other than when the system does its scheduled maintenance tests.

VIP Wakeup

Allows or disallows permission for each extension to request VIP wake up calls.

VIP Wakeups Per 5 Minutes

Available only if **VIP Wakeup** is enabled for each extension.

Valid Entry	Usage
1 to 50	The number of VIP Wakeup calls allowed in a 5-minute interval.

Related links

[VIP Wakeup](#) on page 335

Hospitality: page 3

ROOM STATES

Definition for Rooms in State 1 through 6

A definition for each room status. These definitions are only for **Attendant Room Status**. Accepts up to 30 characters.

Example

State 1 could be “clean, ready to use” and state 2 could be “occupied, needs cleaning”.

HOSPITALITY FEATURES

Cancel Do-Not-Disturb for Wakeup Calls

Activates and deactivates whether a wakeup call will cancel Do-not-disturb.

Suite Check-in

Allows or disallows attendant's permission to have the system automatically check-in several related extensions with one check-in command.

Chapter 56: Hunt Group

With Hunt groups, users or agents can answer calls at a predefined group of telephones or devices.

Use the Hunt Group screen to create a hunt group, identified by a hunt group number, and to assign hunt group member users by their extension numbers. This screen can also be used to implement associated features such as Automatic Call Distribution (ACD) and Hunt Group Queuing.

When a call comes into a hunt group, the system checks for the busy or idle status of extension numbers in the hunt group when answering. A Uniform Call Distribution (UCD) type hunt group selects the “most idle” extension in the group when answering a new call. A Direct Department Calling (DDC) type hunt group selects the first available extension (in the administered sequence) when answering a new call. Expert Agent Distribution (EAD), used only with Expert Agent Selection (EAS), selects the “most idle” agent or the “least occupied” agent with the highest skill level for the call’s skill.

From Release 8.1.3 onwards, if you enable **(SA9142) - Redirected Display for SIP Hunt Group Members** field on the system-parameters special-applications screen, SIP station displays the caller’s name and the hunt group name when a call is received on that SIP station. For more information, see *Avaya Aura® Communication Manager Special Application Features* document.

Note:

Vector controlled splits/skills can be called directly through the split/skill extension instead of calling a VDN mapped to a vector that terminates the call to a vector controlled split/skill. However, the calls do not receive any announcements, be forwarded, redirect to coverage, or intraflow/interflow to another hunt group.

Example command: `change hunt-group n`, where *n* is the assigned hunt group number.

Hunt group: page 1

```

change hunt-group 1                                     Page 1 of 62
HUNT GROUP

Group Number: 1                                         ACD? y
Group Name: Ravi Test                                   Queue? n
Group Extension: 369-9999                               Vector? n
Group Type: ucd-mia                                     Coverage Path:
TN: 1                                                    Night Service Destination:
COR: 1                                                    MM Early Answer? n
Security Code:                                           Local Agent Preference? n
ISDN/SIP Caller Display: mbr-name

SIP URI:

```

ACD

Indicates whether or not to use Automatic Call Distribution (ACD) for this hunt group. Available only if ACD is enabled for the system.

Valid Entry	Usage
y	The hunt group functions as an ACD split/skill. Hunt groups used for voice messaging can function as ACD splits/skills.
n	The hunt group does not function as an ACD split/skill. This option should be used if this hunt group is on a remote server running Communication Manager and using voice messaging in a Distributed Communications System (DCS).

Related links

[ACD](#) on page 739

(Calls Warning) Extension

Extension used by the Terminal Translation Initialization (TTI) feature to assign a port to this extension from the port itself. Once **Calls Warning Port** is assigned a valid port, then the extension is removed and unassigned.

This field cannot be blank.

Available only if a queue has been enabled for this hunt group, and if a port number is not administered for the calls warning and time warning ports.

Related links

[\(Calls Warning\) Port](#) on page 339

[Queue](#) on page 344

[\(Time Warning\) Port](#) on page 345

(Calls Warning) Port

The seven-character port number assigned to connect the optional external **Auxiliary Queue Call Warning Threshold** lamp that flashes when the number of calls in queue has exceeded the queue warning threshold (assigned in **Calls Warning Threshold**). Available only if a queue has been enabled for this hunt group.

Related links

[Calls Warning Threshold](#) on page 339

[Queue](#) on page 344

Calls Warning Threshold

Available only if a queue has been enabled for this hunt group.

Valid Entry	Usage
1 to 999	The number of calls that can be queued before the system flashes the queue status (feature buttons assigned on agents telephones) and the optional Auxiliary Queue Call Warning Threshold lamp assigned to the split/skill. These lamps are lighted steadily when at least one call is in queue and within the threshold. This value must be less than or equal to the queue length or left blank. If you administer the Calls Warning Port field, do not leave this field blank.

Related links

[Queue](#) on page 344

[\(Calls Warning\) Port](#) on page 339

COR

Valid Entry	Usage
0 to 995	The class of restriction (COR) number that reflects the required restriction for the hunt group. If this is a hunt group supporting voice messaging in a Distributed Communications System (DCS), the CORs on this screen for each server running Communication Manager must be the same.

Coverage Path

The coverage path for the hunt group. Available only if the hunt group is not vector controlled.

Valid Entry	Usage
digits	A coverage path number. For the maximum allowed entry, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .
t1 to t999	Time of day table.

Related links

[Vector](#) on page 346

Group Extension

An unused extension number assigned to the hunt group. This field cannot be blank.

Group Name

A character string that uniquely identifies the hunt group. Accepts up to 27 characters.

*** Note:**

This field is supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW telephones.

For more information on Unicode language display, see *Administering Avaya Aura® Communication Manager*.

Example

“parts dept”, “purchasing”, or “sales dept”

Related links

[Display Character Set](#) on page 722

Group Number

The hunt group number.

Group Type

The group types of the hunt group depend on the system configuration. For more information about the availability of the group types, see the following table:

Hunt Group and Call Center options	circ	ddc	ucd-mia	ead-mia	ucd-loa	ead-loa	pad	sim
ACD = n, Queue = n, Vector = n	x	x						

Table continues...

Hunt Group and Call Center options	circ	ddc	ucd-mia	ead-mia	ucd-loa	ead-loa	pad	sim
ACD = y, Vector = n or y, Skill = n		x	x					
ACD = y, Vector = y, Skill = y			x	x				
ACD = y, Vector = y, Skill = y, LOA = y			x	x	x	x		
ACD = y, Vector = y, Skill = y, Business Advocate = y			x	x	x	x		
ACD = y, Vector = y, Skill = y, Dynamic Advocate = y			x	x	x	x	x	
ACD = y, Vector = y, Skill = y, Service Level Maximize = y								x

For a configuration that is not listed, the **LOA** field and the **Business Advocate** or the **Dynamic Advocate** customer options are disabled and therefore, the configuration does not apply.

Valid entry	Usage
circ	With the Circular call distribution, Communication Manager routes the calls to the administered extensions in a round-robin order. Communication Manager keeps track of the last extension in the hunt group to which a call was routed and routes the next call to the next extension in the circular list, independent of how long the extension has been idle. You cannot use this option with ACD, Queues, or Vectors enabled. The Circular option is available only with ACD disabled.
ddc	With the Direct Department Call (DDC) distribution, Communication Manager routes the calls to the first extension or ACD agent assigned in the ACD split. This option is also known as hot seat distribution. Communication Manager always routes an incoming call to the first available agent extension in the administered sequence. If the agent is unavailable, the call routes to the next available agent. Incoming calls are always routed to the first agent in the sequence, so calls are not evenly distributed among agents. You cannot use this option when the hunt group is administered as a skill. The DDC option is available only with the following configurations: • ACD disabled, or • ACD enabled, Split enabled or disabled, Vector enabled or disabled
ucd-mia	With the Uniform Call Distribution-Most Idle Agent (UCD-MIA) call distribution, Communication Manager routes calls to the most-idle agent, based on the agent occupancy, and adds the agent who finishes the most recent call to the end of the idle list. You can use this option if the hunt group is used for voice messaging. Enter the <code>ucd-mia</code> or the <code>ucd-loa</code> group types when the system supports the Outbound Call Management feature and Controlling Adjunct type is <code>asai-ip</code> or <code>adj-ip</code>. The UCD-MIA option is available only with ACD enabled.
ucd-loa	With the Uniform Call Distribution-Least Occupied Agent (UCD-LOA) call distribution, Communication Manager routes calls to the least occupied agent, based on agent occupancy. You can use this option if the hunt group is used for voice messaging. Enter the <code>ucd-mia</code> or the <code>ucd-loa</code> group types when the system supports the Outbound Call Management feature and when the Controlling Adjunct type is <code>asai-ip</code> or <code>adj-ip</code>. The UCD-LOA option is available only with LOA or Business Advocate enabled.
ead-mia	With the Expert Agent Distribution-Most Idle Agent (EAD-MIA) call distribution, Communication Manager routes calls to the agent with the highest skill level available for the call. If two or more agents with equal skill levels are available, Communication Manager routes the call to the most idle agent, based on when the agent finished the most recent call. The call is distributed to the agent best able to handle the call. The EAD-MIA option is available only with EAS and Skill enabled.

Table continues...

Valid entry	Usage
ead-loa	With the Expert Agent Distribution-Least Occupied Agent (EAD-LOA) call distribution, Communication Manager routes calls to the agent available with the highest skill level for the call. If two or more agents are available with equal skill levels, Communication Manager routes the call to the least occupied agent, based on agent occupancy. The call is distributed to an agent best able to handle the call. The EAD-LOA option is available only with EAS, LOA, or Business Advocate enabled.
pad	With the Percentage Allocation Distribution (PAD) call distribution, Communication Manager selects an agent from a group of available agents, based on a comparison of the work time, and the target allocation for the skill of an agent. The PAD option is available only with Dynamic Advocate enabled.
slm	With the Service Level Maximizer (SLM) call distribution, Communication Manager compares the current service level for each SLM-administered skill to a user-defined call service level target and identifies the skills that are most in need of agent resources to meet their target service level. Communication Manager then identifies the available agents, assesses the overall opportunity cost of the agents, and selects only those agents whose other skills have the least need for their service at present. The SLM option is available only with Service Level Maximizer enabled and Business Advocate disabled.

Related links

[ACD](#) on page 739

[Business Advocate](#) on page 739

[Controlling Adjunct](#) on page 347

[Expert Agent Selection \(EAS\)](#) on page 740

ISDN/SIP Caller Disp

This field is required for **ISDN-PRI**, **ISDN-BRI**, and **SIP Trunks**.

Valid Entry	Usage
grp-name	The hunt group name is sent to the originating user.
mbr-name	The member name is sent to the originating user.
blank	Displays the VDN name to the originating user.

Note:

Avaya BRI deskphones support only ASCII characters because non-ASCII characters, such as Eurofont and Kanafont, show up incorrectly.

Related links

[ISDN-BRI Trunks](#) on page 734

[ISDN-PRI](#) on page 734

Local Agent Preference

Enables or disables Local Agent Preference that routes an incoming ACD call to an idle agent by matching the location number of the incoming caller's station or trunk to the location number of an idle agent. Available only with Call Center Release 3.0 or later when Expert Agent Selection (EAS) and multiple locations are enabled for the system. Also, the hunt group must be administered as a skill hunt group.

Related links

[Call Center Release](#) on page 739
[Expert Agent Selection \(EAS\)](#) on page 740
[Multiple Locations](#) on page 736
[Skill](#) on page 353

MM Early Answer

Enables or disables MM Early Answer. The system begins to answer an H.320 call and establish an audio channel before offering the conversion call to the hunt group. This starts billing for the call when the call is first put into queue. This field applies only for systems using Multimedia Call Handling.

Related links

[Multimedia Call Handling \(Basic\)](#) on page 735

Night Service Destination

Unavailable for vector-controlled hunt group.

Valid Entry	Usage
<i>An assigned extension number</i>	The destination where calls to this split redirect when the split is in the night service mode. This extension can be a VDN extension. Must be a local extension for all features to work correctly.
attd	An attendant group code.

Related links

[Vector](#) on page 346

Queue

Enables or disables a queue for the hunt group.

Queue Limit

Available only if a queue is enabled for the hunt group.

Valid Entry	Usage
1 to 999	The limit to the number of calls that will queue.
unlimited	The system dynamically allocates the queue slots from a common pool on an as-needed basis.

Related links

[Queue](#) on page 344

Security Code

Use this field to enter the four-digit station security code for the **Demand Print messages** feature.

SIP URI

Use this field to place a call using the URI. URI is an alphanumeric string such as letters, numbers, and special characters. You can enter up to 70 characters, including handle and domain name. For example:

help.desk@example.com

helpdesk123@example.com

helpdesk@example.com

Following are not supported:

- Unicode characters
- Eurofont characters
- Using same URIs across hunt groups, VDNs, and stations within the same Communication Manager.

Time Warning Extension

Extension used by the Terminal Translation Initialization (TTI) feature to assign a port to this extension from the port itself. Once **Time Warning Port** is assigned a valid port, then the extension is removed and considered unassigned.

Available only if a queue is enabled for the hunt group. Required if a port number is not administered for the time warning port.

Related links

[Queue](#) on page 344

[\(Time Warning\) Port](#) on page 345

(Time Warning) Port

The port number assigned to the **Auxiliary Queue Time Warning** lamp that flashes when the Time Warning Threshold has been reached by a call in queue. Available only if a queue has been enabled for the hunt group.

Related links

[Queue](#) on page 344

[Time Warning Threshold](#) on page 346

Time Warning Threshold

Available only if a queue is enabled for the hunt group and if a port number is not administered for the call warning and time warning ports.

Valid Entry	Usage
0 to 999	The time in seconds that a call can remain in the queue before the system flashes the Queue status lamps (feature buttons assigned members telephones) and the Auxiliary Queue Time Warning lamp assigned to this split/skill. An entry of 0 provides a warning whenever a call is queued.

Related links

[Queue](#) on page 344

[\(Time Warning\) Port](#) on page 345

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Vector

Enables or disables this hunt group as vector controlled. Available only if Basic Vectoring is enabled for the system.

Related links

[Vectoring \(Basic\)](#) on page 741

Hunt group: page 2

This screen can vary according to values for particular fields on the previous page.

If the **ACD** is disabled for the system, this page is omitted.

change hunt-group 1		Page 2 of 62
HUNT GROUP		
Skill? <u>n</u>	Expected Call Handling Time (sec): <u>180</u>	
AAS? <u>n</u>		
Measured: <u>none</u>		
Supervisor Extension: _____		
Priority On Intraflow? <u>n</u>		
Controlling Adjunct: <u>none</u>		
Multiple Call Handling: <u>none</u>		
Timed ACW Interval (sec): _____	After Xfer or Held Call Drops? <u>n</u>	

AAS

Enables or disables this hunt group serving as an Auto-Available Split (AAS). With AAS, members of an ACD split or skill can be in auto-in work mode continuously. An agent in auto-in work mode becomes available for another ACD call immediately after disconnecting from an ACD call. Available only if ACD is enabled for this hunt group.

Related links

[ACD](#) on page 338

Adjunct CTI Link

Use this field to assign a CTI link to the hunt group. You must not leave this field blank. The system displays this field only if ACD is enabled for the hunt group and the **Controlling Adjunct** field shows asai-ip or adj-ip.

Related links

[ACD](#) on page 338

[Controlling Adjunct](#) on page 347

Controlling Adjunct

Use this field to specify the controlling adjunct. The system displays this field only if ACD is enabled for the hunt group.

Valid entry	Usage
none	Members of the hunt group are not controlled by an adjunct processor.
asai-ip or adj-ip	Members of the hunt group are controlled by an ASAI-IP or ADJ-IP link.

Related links

[ACD](#) on page 338

[ASAI Link Core Capabilities](#) on page 729

[Computer Telephony Adjunct Links](#) on page 731

Dynamic Percentage Adjustment

Enables or disables automatic adjustments to agents' target allocations as needed to help meet the administered service level targets. Available only if ACD is enabled for the hunt group and this is a Percent Allocation Distribution (PAD) hunt group. Requires Business Advocate software.

Related links

[ACD](#) on page 338

[Business Advocate](#) on page 739

[Group Type](#) on page 340

Dynamic Queue Position

Enables or disables dynamic queue operation to the calls queued to the skill. Dynamic Queue Position is a Business Advocate feature for calls to queue from multiple VDNs to a single skill, while maintaining different service objectives for those VDNs. Available only if ACD, Expert Agent Selection (EAS), and Skill are enabled for the hunt group. Requires Business Advocate software.

Related links

[ACD](#) on page 338

[Business Advocate](#) on page 739

[Skill](#) on page 353

Dynamic Threshold Adjustment

Enables or disables automatic adjustments to overload thresholds to engage reserve agents a bit sooner or a bit later to meet the administered service levels. Available only if ACD and Service Level Supervisor are enabled for the hunt group. Requires Business Advocate software.

Related links

[ACD](#) on page 338

[Business Advocate](#) on page 739

[Service Level Supervisor](#) on page 352

Expected Call Handling Time (sec)

Available only if ACD is enabled for the system and if either Vectoring (Advanced Routing) or Business Advocate is enabled for the system.

Valid Entry	Usage
1 to 9999 in increments of 1	Establishes the number of seconds for expected call handling. This value is used to initialize Expected Wait Time and is also used by the Business Advocate Percent Allocation feature.

Related links

[ACD](#) on page 338

[Business Advocate](#) on page 739

[Vectoring \(G3V4 Advanced Routing\)](#) on page 742

Inflow Threshold (sec)

Available only if ACD and a queue are enabled for the system. unavailable for a vector-controlled hunt group.

Valid Entry	Usage
0 to 999	The number of seconds that a call can remain in the queue before no more calls are accepted by the queue. If 0 is entered, a call is redirected to this split/skill only if there is an available agent.

Related links

[ACD](#) on page 338

[Queue](#) on page 344

[Vector](#) on page 346

Level 1 Threshold (sec)

Available only if ACD and Service Level Supervisor are enabled for the hunt group.

Valid Entry	Usage
0 to 60	The number of seconds for the first Expected Wait Time (EWT) threshold.

Example

If there are 45 calls whose EWT exceeds 45 seconds, threshold 1 will have been exceeded.

Related links

[Service Level Supervisor](#) on page 352

Level 2 Threshold (sec)

Available only if ACD and Service Level Supervisor are enabled for the hunt group.

Valid Entry	Usage
0 to 60	The number of seconds for the second Expected Wait Time (EWT) threshold.

Example

If there are 60 calls whose EWT exceeds 60 seconds, threshold 2 will have been exceeded.

Related links

[ACD](#) on page 338

[Service Level Supervisor](#) on page 352

Maximum Auto Reserve Agents

Available only if ACD is enabled for the hunt group and it is a Service Level Maximizer (SLM) type hunt group.

Valid Entry	Usage
0 to 9	The maximum number of Auto Reserve Agents available for this skill (hunt group). Any time an auto-reserve skill is in danger of falling below its target service level percent, some of this skill's agents are auto-reserved (kept idle in other skills) so that they are available when a new call arrives for this skill. Default is 0.

Related links

[Group Type](#) on page 340

Measured

Provides measurement data for the ACD split/skill to VuStats or BCMS. Available only if ACD is enabled for the hunt group and VuStats or BCMS is enabled for the system.

Valid Entry	Usage
internal	Provides measurements made by the Call Management System that are internal to the server running Communication Manager.
external	Provides measurements made by the Call Management System that are external to the server running Communication Manager.
both	Provides measurements collected both internally and externally.
none	Measurement reports for this hunt group are not required.

Related links

[ACD](#) on page 338

[BCMS \(Basic\)](#) on page 739

[VuStats](#) on page 743

Multiple Call Handling

Defines whether the hunt group can have multiple call handling capabilities, and if so, what type. Available only if ACD is enabled for the hunt group and Multiple Call Handling is enabled for the system.

Valid Entry	Usage
none	Agents who are members of that split/skill can only receive an ACD call from that split/skill when the telephone is idle.
on-request	Agents in the Multiple Call Handling split/skill can place a non-ACD or an ACD call on hold and select an available work mode. A queued ACD split/ skill or direct agent call then is routed to the agent.
many-forced	An ACD call is delivered automatically to an idle line appearance if the agent is in the Auto-In/Manual-In (MI/AI) work mode and an unrestricted line appearance is available.
one-forced	An ACD call is delivered automatically to an idle line appearance if the agent has no other ACD call on the station, is in the Auto-In/Manual-In (MI/AI) work mode, and an unrestricted line appearance is available.
one-per-skill	An ACD call is delivered automatically to an idle line appearance if the agent has no other ACD call for that skill on the station, is in the Auto-In/ Manual-In (MI/AI) work mode, and an unrestricted line appearance is available. Valid in an EAS environment for a skill-enabled hunt group.

Related links

[ACD](#) on page 338

[Multiple Call Handling \(Forced\)](#) on page 740

[Skill](#) on page 353

Priority On Intraflow

Enables or disables having calls intraflowing from this split to a covering split and given priority over other calls waiting in the covering split queue. Available only if ACD field is enabled for the hunt group. Unavailable for a vector-controlled hunt group.

Related links

[ACD](#) on page 338

[Vector](#) on page 346

Service Level Interval

The time interval when Actual Service Level (ASL) calculations run. ASL is one of the Service Level Maximizer (SLM) algorithms used for most situations, particularly for low staff or low traffic. The interval can be set to the same interval used when specifying the target objectives for the application. Available only if Actual is administered for the SLM algorithm feature and this is an SLM-type hunt group.

Valid Entry	Usage
hourly	ASL algorithm calculations for accepted call and total call components are set to 0 at hourly intervals.
daily	ASL algorithm calculations for accepted call and total call components are set to 0 at daily intervals. This is the default.
weekly	ASL algorithm calculations for accepted call and total call components are set to 0 at weekly intervals. The weekly interval starts as 00:00 hours on Sunday.

Related links

[Service Level Maximizer Algorithm](#) on page 289

[Group Type](#) on page 340

Service Level Supervisor

Enables or disables Service Level Supervisor for this skill. Service Level Supervisor is a Business Advocate feature that alleviates the need to move agents from skill to skill during emergencies or unanticipated peaks in call volume. Available only if ACD is enabled for the hunt group and this is an Expert Agent Selection (EAS) skill hunt group. Requires Business Advocate software.

Related links

[ACD](#) on page 338

[Skill](#) on page 353

[Business Advocate](#) on page 739

Service Level Target (% in sec)

The system displays Service Level Target (% in sec) when the **ACD** field and the **Measured** field is not blank, and when one or more of the following features are active:

- **BCMS/VuStats Service Level** field on the System Parameters Customer-Options screen is active and the **Measured** field is set to internal or both. The service level target in seconds is used as the acceptable level for reporting the percentage of calls answered within the specified time. The percentage can be set to the default of 80%.
- Business Advocate on the System Parameters Customer-Options screen is active. The service level target in seconds is used for the Business Advocate Service Level Supervisor service level objective. This service level target can also be used for the dynamic percentage adjustment when the **Dynamic Threshold Adjustment** field on the Hunt Group screen is y and for the dynamic percent adjustment when the **Group Type** field on the Hunt Group screen is pad and the **Dynamic Percent Adjustment** field on the Hunt Group screen is set to y.
- The system displays the **Service Level Target** field when the **Group Type** field on the Hunt Group screen is slm, and on the System Parameters Customer-Options screen, the Service Level Maximizer is active, and the Business Advocate customer option license is inactive. In this case the setting is also used as the service level target to trigger SLM.

- **Interruptible Aux Threshold** field on the Hunt Group screen is set to service-level-target. The Interrupt Aux feature is triggered if the service level drops below the administered percentage of calls in the specified seconds.

Valid Entry	Usage
1 to 99 (percentage)	The percentage component of the service level target. The default value is 80%.
1 to 9999 (time in seconds)	The time component of the service level target. The default value is 20 seconds.

Related links

[ACD](#) on page 338
[BCMS \(Basic\)](#) on page 739
[Business Advocate](#) on page 739
[Dynamic Percentage Adjustment](#) on page 348
[Dynamic Threshold Adjustment](#) on page 348
[Group Type](#) on page 340
[Interruptible Aux Threshold](#) on page 356
[Service Level Maximizer](#) on page 741
[VuStats](#) on page 743

Service Objective

Available only if ACD is enabled for the hunt group and this is an Expert Agent Selection (EAS) skill hunt group. Requires Business Advocate software.

Valid Entry	Usage
1 to 9999	Sets a service objective for a specific skill as the number of seconds the call gets answered. The default value is 20.

Related links

[ACD](#) on page 338
[Business Advocate](#) on page 739
[Skill](#) on page 353

Skill

Enables or disables this hunt group as an Expert Agent Selection (EAS) skill. Available only if ACD is enabled for the hunt group and EAS is enabled for the system.

Related links

[ACD](#) on page 338
[Group Type](#) on page 340
[Expert Agent Selection \(EAS\)](#) on page 740

SLM Count Abandoned Calls

Available only if Actual is the administered Service Level Maximizer (SLM) algorithm for the feature and this is an SLM-type hunt group.

Valid Entry	Usage
y	Abandoned calls are included in the Actual Service Level (ASL) algorithm calculations for SLM.
n	Abandoned calls are excluded in the ASL algorithm calculations for SLM. This option is best used when reporting for this application does not account for calls that are abandoned while in skill queues.

Related links

[Group Type](#) on page 340

[Service Level Algorithm for SLM](#) on page 295

Supervisor Extension

The extension number of the ACD split/ skill supervisor that agents reach when using the Supervisor Assist feature. The extension number cannot be a Vector Directory Number (VDN).

Available only if ACD field is enabled for the system.

Related links

[ACD](#) on page 338

Timed ACW Interval (sec)

Available only if ACD is enabled for the hunt group and Timed ACW is enabled for the system.

* Note:

This field can be overridden by the settings administered for a vector. Coordinate the settings when setting up delays.

Valid Entry	Usage
1 to 9999	The number of seconds an agent in auto-in work mode remains in After Call Work (ACW) mode after a call drops. After this time interval expires, the agent automatically becomes available. Timed ACW cannot be administered if the hunt group is adjunct controlled, is an AUDIX Message Center, or is an auto-available split.

Related links

[ACD](#) on page 338

[Timed ACW](#) on page 741

VuStats Objective

Available only if ACD is enabled for the hunt group and VuStats is enabled for the system. Also, the hunt group must be administered to collect internal or both internal and external measurement data for VuStats.

Valid Entry	Usage
0 to 99999	A numerical objective. An objective is a split or skill goal for the call. This could be an agent objective such as a specific number of calls handled or an average talk time. The objective could also be a percent within the service level. The VuStats screen displays the objective. The agents and supervisors can compare the current performance against the value of the objective for the split or skill. This value applies to customized VuStats display formats.

Related links

[ACD](#) on page 338

[Measured](#) on page 350

[VuStats](#) on page 743

Hunt group: page 3

This screen can vary according to values for particular fields on the previous page.

If the **ACD** is disabled for the system, this page is omitted.

change hunt-group 1

Page 3 of 62

HUNT GROUP

Redirect on No Answer (rings):

Redirect on No Answer to VDN:

Redirect on IP/OPTIM Failure to VDN:

Forced Entry of Stroke Counts or Call Work Codes? n

Forced Entry of Stroke Counts or Call Work Codes

Enables or disables the requirement that either a Stroke Count or Call Work Code must be entered for each call answered by an agent when in the Manual-In mode. Available only if ACD is enabled for the hunt group and if the hunt group does *not* have a Controlling Adjunct.

Related links

[ACD](#) on page 338

[Controlling Adjunct](#) on page 347

Interruptible Aux Deactivation Threshold

This field specifies the maximum level of service above which the system deactivates the Interruptible Aux feature.

Use the **Interruptible Aux threshold** and the **Interruptible Aux Deactivation Threshold** fields to maintain a buffer between the two levels of service that trigger the **Interruptible Aux** feature. The values of the **Interruptible Aux threshold** and the **Interruptible Aux Deactivation Threshold** fields must have a difference of one unit.

The valid entries for this field depend on the value of the **Interruptible Aux threshold** field:

- If you set the value of the **Interruptible Aux threshold** field to calls-warning-threshold, the valid entries for the **Interruptible Aux Deactivation Threshold** field are from 0 to 998.
- If you set the value of the **Interruptible Aux threshold** field to service-level-target, the valid entries for the **Interruptible Aux Deactivation Threshold** field are from 0 to 100.
- If you set the value of the **Interruptible Aux threshold** field to time-warning-threshold, the valid entries for the **Interruptible Aux Deactivation Threshold** field are from 0 to 998.
- If you set the value of the **Interruptible Aux threshold** field to none, the system does not display the **Interruptible Aux Deactivation Threshold** field.

Interruptible Aux Threshold

This field specifies the minimum level of service below which the system enables the Interruptible Aux feature. Use the Interruptible Aux feature to make the Expert Agent Selection (EAS) agents in the AUX work mode available to receive calls. This feature works only if the agents have an interruptible reason code.

Valid entry	Usage
service-level-target	Use this option to enable the Interruptible Aux feature when the service level drops below the administered percentage of calls within the specified period. For example, if you set the Service Level Target (% in sec) field to 90% of calls in 30 seconds, the system enables the Interruptible Aux feature if the service drops to 89% of calls in 30 seconds.

Table continues...

Valid entry	Usage
calls-warning-threshold	Use this option to enable the Interruptible Aux feature when the number of calls in the queue for a hunt group exceeds the specified number of calls. For example, if you set the Calls Warning Threshold field to 20, the system enables the Interruptible Aux feature if the number of calls in the queue exceeds 20.
time-warning-threshold	Use this option to enable the Interruptible Aux feature when the oldest call is in the queue for longer than the specified number of seconds. For example, if you set the Time Warning Threshold field to 60 seconds, the system enables the Interruptible Aux feature if the oldest call is in the queue for more than 60 seconds.
none	Use this option if you do not want to administer the Interruptible Aux feature.

Redirect on IP/OPTIM Failure to VDN

A blank in the field redirects the call back to the hunt group. VDN extension redirects to the specified VDN.

If **Redirect on IP/OPTIM Failure to VDN** is not assigned, the call is re-queued to the same skill at a high priority. If there are no queue slots available, the caller will hear a busy signal. If all fails, the caller receives ring back until the system receives a caller disconnect.

Related links

[ACD](#) on page 338

Redirect on No Answer (rings)

Available only if ACD is enabled for the hunt group.

Valid Entry	Usage
1 to 20	The maximum number of rings before a call redirects back to the split/skill, or to the administered VDN.
blank	Deactivates Redirect on No Answer.

Related links

[ACD](#) on page 338

Redirect on No Answer to VDN

The extension number of the VDN used to redirect a Redirect On No Answer (RONA) call to a VDN instead of to the split/skill. The administered VDN must be on-premises and must be administered on the system. The VDN can specify a vector that routes to an off-premises VDN.

Direct Agent calls go to the agent's coverage path if it is administered. If not, the calls go to a VDN.

Available only if ACD is enabled for the hunt group. Requires administration of the number of rings before a call will redirect.

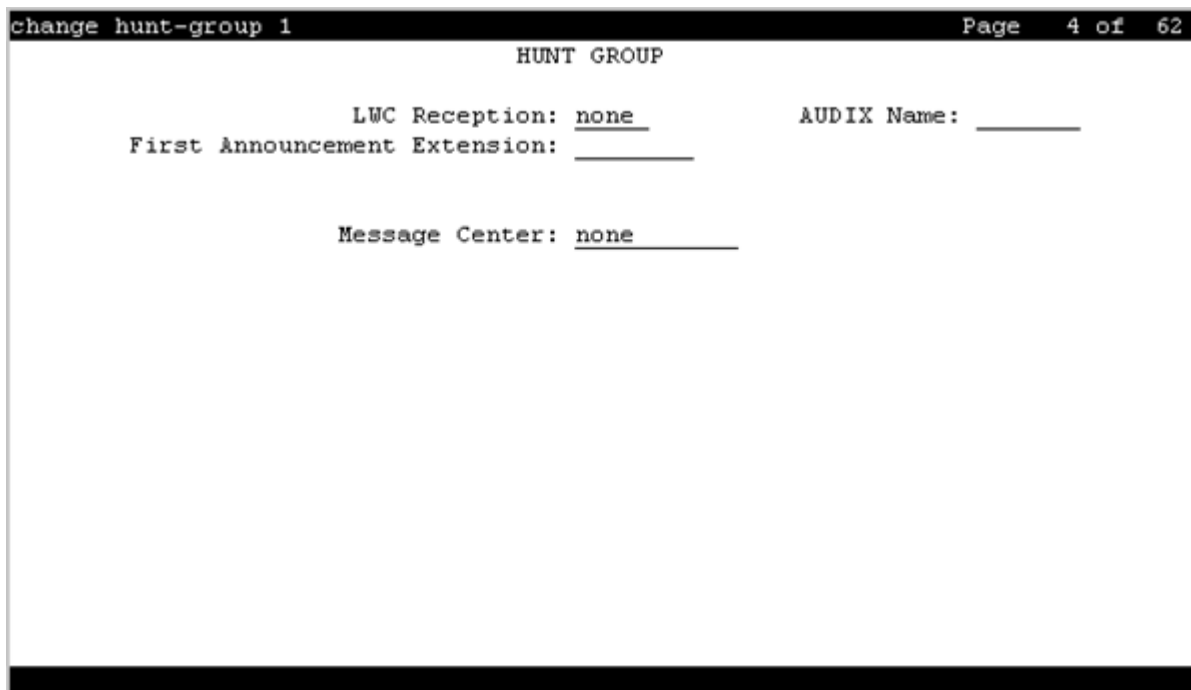
Related links

[ACD](#) on page 338

[Redirect on No Answer \(rings\)](#) on page 357

Hunt Group: page 4

The Hunt Group Message Center screen can vary according to system configuration and values populating particular fields.



The screenshot shows a terminal-style interface for the 'change hunt-group 1' screen. The title bar at the top indicates 'Page 4 of 62'. The main content area is titled 'HUNT GROUP' and contains the following fields:

LWC Reception: <u>none</u>	AUDIX Name: _____
First Announcement Extension: _____	
Message Center: <u>none</u>	

AUDIX Name

The name of the AUDIX machine. Must be the same name as the IP Node name and administered *after* the IP Node is configured.

Related links

[IP Node Names](#) on page 413

[IP Node Names](#) on page 413

Calling Party Number to INTUITY AUDIX

The system displays Calling Party Number to INTUITY AUDIX when the **Message Center** field is audix or rem-vm.

Valid entries	Usage
y/n	Enter y to send the calling party's number to INTUITY AUDIX®

Related links

[Message Center](#) on page 360

First Announcement Delay (sec)

Available only if a queue is administered for the hunt group. Unavailable if the hunt group is vector controlled.

Valid Entry	Usage
0 to 99	The number of seconds that a call remains in queue before the associated first announcement is given the calling party. The call retains its place in the queue while the caller is listening to the recorded announcement. If the call has not been answered after the announcement, the caller hears music for first announcement only if Music-on-Hold is provided, or ringing for as long as the call remains in queue. When 0 is entered, the first announcement is provided immediately to the caller. This value is set automatically to 0 if there is no queue.
blank	There is no first announcement.

Related links

[Queue](#) on page 344

[Vector](#) on page 346

First Announcement Extension

The recorded announcement extension number the caller receives after being in the queue for the time interval specified in First Announcement Delay. If the call hasn't been answered after the announcement, the caller hears music only after the first announcement if Music-on-Hold is provided, or ringing for as long as it remains in the queue. If this is the forced first announcement, the caller always hears ringback after the announcement. Otherwise, the caller hears music, if provided.

Available only if ACD and a queue is administered for the hunt group. Unavailable if the hunt group is vector controlled.

Note:

If you enter a shortened extension of the multilocation dial plan, the system does not perform certain administration and validation tasks. Therefore, the system might not display the resultant warnings or submittal denials.

Related links

[ACD](#) on page 338

LWC Reception

Use this field to specify the location where the system must store the LWC messages.

Valid entry	Usage
spe	Use this option to store the LWC messages on Switch Processor Element (SPE).
none	Use this option if you do not want to store the LWC messages.
audix	Use this option to store the LWC messages on the voice messaging system.

Related links

[AUDIX Name](#) on page 358

Message Center

The type of messaging adjunct for the hunt group. For information on system capacities, see *Avaya Aura® Communication Manager System Capacities Table*.

Valid Entry	Usage
audix	AUDIX located on this server running Communication Manager
fp-mwi	Public network allowing AUDIX to be located on another switch. Available only if ISDN Feature Plus is administered for the system.
msa	Messaging Server Adjunct
msa-vm	A voice mail system integrated using Mode Codes or Digital Station Emulation
rem-vm	DCS feature allowing voice mail to be located on another server
qsig-mwi	QSIG network allowing voice mail to be located on another server
sip-adjunct	SIP message center server
none	The hunt group does not serve as a message hunt group.

Related links

[ISDN Feature Plus](#) on page 734

Message Center AUDIX Name

The name of the Message Center AUDIX. Available only if the messaging type is audix or rem-vm.

Related links

[Message Center](#) on page 360

Message Center MSA Name

The name of the Message Center Messaging Server Adjunct (MSA). Available only if the messaging type is msa.

Related links

[Message Center](#) on page 360

[Message Center AUDIX Name](#) on page 360

Primary

Available only if the **Message Center** field is audix or rem-audix.

Valid entries	Usage
y/n	Enter y to indicate that the specified messaging server is the primary adjunct.

Related links

[Message Center](#) on page 360

Provide Ringback

Enables or disables ringback to the calling party until a **Connect** is received for the call to the Messaging system. Ringback is discontinued upon receipt of the **Connect** indication. Used for an SBS trunk for the QSIG MWI hunt group. A call covering to the message center provides ringback to the caller during the coverage interval. Available only if the messaging type is fp-mwi or qsig-mwi.

Related links

[Message Center](#) on page 360

Routing Digits (e.g. AAR/ARS Access Code)

A one- to four-digit AAR (qsig-mwi) or ARS (fp-mwi) access code. This access code is prepended to the AUDIX Complete Number to define a route to the Message Center switch hunt group containing the line ports to the AUDIX. Accepts characters * and #.

Available only if the messaging type is qsig-mwi or fp-mwi.

Related links

[Message Center](#) on page 360

Second Announcement Delay (sec)

Available only if ACD and queues are enabled for the hunt group. Unavailable if the hunt group is vector controlled.

Valid Entry	Usage
1 to 99	The time in seconds before the call in the queue receives a second recorded announcement or that the second announcement is repeated. If this split/skill or hunt group is a coverage point for another split/skill, this delay should not be more than 15 seconds.
blank	There is no second announcement.

Related links

[ACD](#) on page 338

[Queue](#) on page 344

[Vector](#) on page 346

Second Announcement Extension

The extension number assigned to a second recorded announcement. Left blank if there is no second announcement.

Available only if ACD and queues are enabled for the hunt group. Unavailable if the hunt group is vector controlled.

Note:

If you enter a shortened extension of the multilocation dial plan, the system does not perform certain administration and validation tasks. Therefore, the system might not display the resultant warnings or submittal denials.

Related links

[ACD](#) on page 338

[Queue](#) on page 344

[Vector](#) on page 346

Second Announcement Recurring

Allows or disallows repeating the second announcement. Available only if ACD and queues are enabled for the hunt group. Unavailable if the hunt group is vector controlled.

Related links

[ACD](#) on page 338

[Queue](#) on page 344

[Vector](#) on page 346

Send Reroute Request

Allows or disallows rerouting getting invoked when a call covers through a qsig-mwi hunt group. Available only if the messaging type is qsig-mwi and Supplementary Services with Rerouting is enabled for the system.

Related links

[Message Center](#) on page 360

[Supplementary Services with Rerouting](#) on page 744

TSC per MWI Interrogation

Controls Temporary Signaling Connections (TSCs) used for message waiting interrogations for users that are “local” to the system in which the hunt group is administered. Available only if the messaging type is qsig-mwi.

Valid Entry	Usage
y	Communication Manager brings the TSC up, executes the Interrogate operation, and then tears the TSC down.
n	Communication Manager uses the existing TSC sending FACILITY messages to request MWI status if the TSC is already set up, or sets up a TSC. When the interrogation operation is complete, leaves the TSC up, subject to the existing timer. This is the default.

Related links

[Message Center](#) on page 360

Voice Mail Extension

The UDP extension of the voice mail hunt group on the host server running Communication Manager.

Available only if the messaging type is rem-vm.

Related links

[Message Center](#) on page 360

Voice Mail Handle

The Session Manager (SM) handle that can receive voice mail. Can be left blank if a Voice Mail Number has been assigned.

Voice Mail Number

The 1- to 17-digit voice mail dial-up number. The qsig-mwi selection shows the complete number of the AUDIX hunt group on the Message Center server for QSIG MWI. The fp-mwi selection shows the public network number of the AUDIX hunt group on the Message Center server.

Available only if Basic Call Setup and Basic Supplementary Services are enabled for the system, and the messaging type is qsig-mwi or fp-mwi.

Related links

[Basic Call Setup](#) on page 743

[Basic Supplementary Services](#) on page 743

[Message Center](#) on page 360

Hunt Group: page 5 through X

change hunt-group 1		Page 5 of 62	
HUNT GROUP			
Group Number: 1	Group Extension: 369-9999	Group Type: ucd-mia	
Member Range Allowed: 1 - 1500	Administered Members (min/max): 1 /1		
Total Administered Members: 1			
GROUP MEMBER ASSIGNMENTS			
Ext	Name (19 characters)	Ext	Name (19 characters)
1: 360-5003	Kaliya 3605003	14: _____	_____
2: _____	_____	15: _____	_____
3: _____	_____	16: _____	_____
4: _____	_____	17: _____	_____
5: _____	_____	18: _____	_____
6: _____	_____	19: _____	_____
7: _____	_____	20: _____	_____
8: _____	_____	21: _____	_____
9: _____	_____	22: _____	_____
10: _____	_____	23: _____	_____
11: _____	_____	24: _____	_____
12: _____	_____	25: _____	_____
13: _____	_____	26: _____	_____
At End of Member List			

Administered Members (min/max)

The minimum and maximum member number administered for this hunt group. Available for all member pages.

At End of Member List

Displays the current page as also the last page.

Group Extension

The extension of the hunt group.

Group Number

Displays the number of the hunt group.

Group Type

Displays the type of hunt group.

Member Range Allowed

The range of allowed members. These values vary depending on the system or configuration.

More Members Exist

Statement that there are more members and more pages than currently displayed.

Total Administered Members

The total number of members administered for the hunt group.

GROUP MEMBER ASSIGNMENTS

Ext

The assigned station or attendant console extension. This extension cannot be a Vector Directory Number (VDN). The data module cannot be a member of an ACD split/skill. Administers the assigned station or attendant console extension only if the controlling adjunct is administered as none. Displays the assigned station or attendant console extension if the controlling adjunct is administered as asai-ip or adj-ip.

Note:

Avaya BRI deskphones support only ASCII characters because non-ASCII characters, such as Eurofont and Kanafont, show up incorrectly.

Related links

[Controlling Adjunct](#) on page 347

Name

The name assigned to the extension number when it is administered in the system.

Related links

[Ext](#) on page 365

Chapter 57: Incoming Call Handling Treatment

Incoming Call Handling Treatment (ICHT) specifies a unique call treatment for incoming calls on ISDN and SIP trunk groups. This screen is available only if digit handling is administered as overlap, and the direction is set to outgoing.

*** Note:**

ICHT is also used to modify the Calling Party number when Communication Manager is used as a feature server.

The Incoming Call Handling Treatment Table screen displays entries with a predefined service or feature before entries with a user-defined service or feature.

The user-defined entries are always listed in the reverse order on Incoming Call Handling Treatment Table as compared to the entries that show on Network Facilities. Therefore, for two user-defined services or features, ABC and XYZ, you can force XYZ to show before ABC on Incoming Call Handling Treatment Table by administering XYZ after ABC on Network Facilities.

*** Note:**

DCS features that use the **remote-tgs** button on the remote server do not work when the local trunk group deletes or inserts digits in the incoming call.

To change the value of the fields on the Incoming Call Handling Treatment Table, use the following command:

`change inc-call-handling-trmt trunk-group n`, where *n* is an ISDN or a SIP trunk group.

Called Len

Valid Entry	Usage
0 to 21	The number of digits received for an incoming call. Zero is used when the Public Switched Telephone Network (PSTN) provider does not provide any "Number Digits" within the received Called Party IE , such as in Japan.
blank	When Called Number has also been set to blank, so that any length of digits associated with the Called Party IE of the Incoming SETUP message matches this field.

Called Number

Valid Entry	Usage
1 to 16	Specifies the leading digits received for an incoming call.
blank	Matches any number associated with the specified service or feature.

Del

Valid Entry	Usage
1 to 21	The number of leading digits to be deleted from the incoming Called Party Number . Calls of a particular type can be administered to be routed to a single destination by deleting all incoming digits and then administering the Insert field with the required extension.
blank	

Insert

Valid Entry	Usage
1 to 16	The number of digits prepended to the front of the remaining digits after any optional digit deletions have been performed. The resultant number formed from digit deletion and insertion is used to route the call, provided night service is not in effect.
*	
#	

Per Call CPN/BN

Specifies when and how to request **Calling Party Number** (CPN) or **Billing Number** (BN) for calls of this type. Available only with ISDN trunk groups.

Valid Entry	Usage
cpn-only	Calling party number only
bn-only	Billing number only
bn-pref	Prefers billing number, but accepts calling party number
cpn-pref	Prefers calling party number, but accepts billing number
none	Communication Manager will not request either CPN or BN for any incoming calls of this type.

Table continues...

Valid Entry	Usage
blank	Leave blank when connected to another media server or switch, or when connected to a public network outside North America. Within North America, leave blank when connected to a public network that prohibits customer equipment to request CPN or BN for individual incoming calls. The AT&T Switched Network offers this service under the titles "CPN/BN to Terminating End on a Per-Call Basis" and "ANI (BN) on Request".

*** Note:**

A 4-second delay occurs in terminating the call to the far-end station if the connecting server or switch stops responding to the request.

Night Serv

Available only for ISDN trunk groups.

Valid Entry	Usage
<i>Assigned extension</i> <i>Attendant group access code</i> attd blank	Specifies a night service extension per Service/Feature. A Vector Directory Number (VDN) is acceptable. This entry is overridden by the night service administered for an individual trunk or the trunk group.

Related links

[Night Service](#) on page 797

[Night](#) on page 472

Service/Feature

Displays the number that corresponds to the server type administered for the trunk group or network facility. Administration is required for a call-by-call service type.

Valid Entry	Usage
Valid pre-defined Service/Feature values for cbc trunk groups	These values are administered for Network Facilities.

Table continues...

Valid Entry	Usage
User-defined facility type administered for the service type	• 0 = feature • 1 = service • 2 = incoming • 3 = outgoing

Related links

[Network Facilities](#) on page 540

[Facility Type](#) on page 540

Chapter 58: Integrated Announcement Boards

Moves integrated announcement boards that have been previously administered on the Announcements/Audio Sources screen to a new board location. Displays a list of all administered integrated announcement board.

Example command: `display integrated-annc-boards`

display integrated-annc-boards

Page1 of 13

INTEGRATED ANNOUNCEMENT SOURCES													
	Src	Sfx	Tim Rem	Rt	Num Annc	Chk Sum		Src	Sfx	Tim Rem	Rt	Num Annc	Chk Sum
1:	M1		NA	NA	1	NA	16:						
2:	001V9	AP	1372	64	1	NA	17:						
3:							18:						
4:							19:						
5:							20:						
6:							21:						
7:							22:						
8:							23:						
9:							24:						
10:							25:						
11:							26:						
12:							27:						
13:							28:						
14:							29:						
15:							30:						

Board Location

The physical location of the integrated announcement board (UUCSS).

Number of Recordings

The number of non-zero-length announcement recordings or files on the board.

Rate

The announcement's compression rate.

Sfx

The board suffix letters.

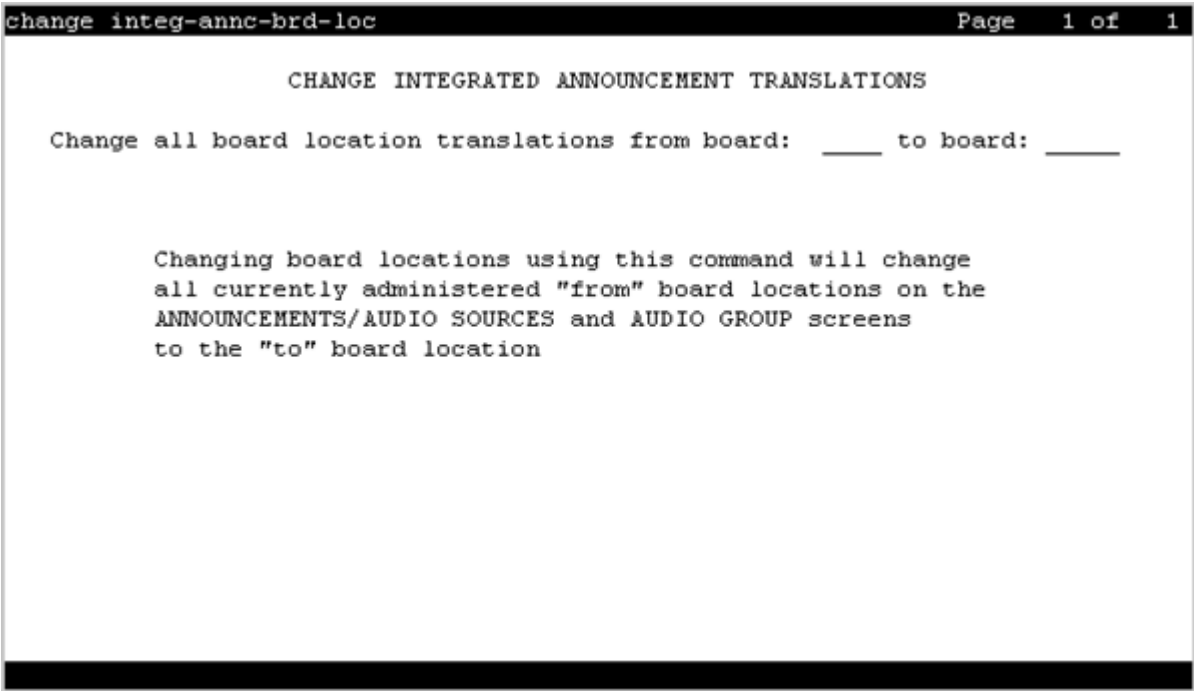
Time Remaining

The amount of recording time in seconds remaining on the board at the 64Kb rate.

Chapter 59: Integrated Announcement Translations

Changes board locations currently administered on the Announcements/ Audio Sources screen to a new board location.

Example command: `change integ-annc-brd-loc`



Change all board location translations from board

Valid Entry	Usage
gateway 1 to 999 module V1 to V9	The VAL board that is currently administered.

to board

Use this field to specify the VAL board for announcement translations.

You can use one of the following ways to enter a value in this field:

- gateway 1 to 999: module V1 to V9

Chapter 60: Intercom Group

Assigns extensions to intercom groups.

Example command: `change intercom-group n`, where *n* is the assigned intercom group number.

DC

Assigns a dial code to an extension. Accepts up to two digits. The dial code is the code users must dial to make intercom calls to the corresponding extension. The number of digits entered must exactly match the number administered for the length of the dial code. This field cannot be blank.

Example

If the length of the dial code is set to 2, type 1 as 01 in the **DC** field.

Related links

[Length of Dial Code](#) on page 375

Ext

Assigns an extension to the group. A Vector Directory Number (VDN) cannot be used as an extension.

Group Number

The group ID number.

Length of Dial Code

Sets the number of digits that users must dial to access an extension in the group.

Valid Entry	Usage
1	For nine or fewer members.
2	For 10 or more members.

Name

The name associated with the extension that has been administered to join the group.

**Note:**

Avaya BRI deskphones support only ASCII characters because non-ASCII characters, such as Eurofont and Kanafont, show up incorrectly.

Related links

[Name](#) on page 653

Chapter 61: Inter-Exchange Carrier (IXC) Codes

Identifies the IXC in the Call Detail Recording (CDR).

Example command: `change ixc-codes`

Inter-Exchange Carrier (IXC) Codes: page 1

IXC Access Number

The digits dialed or inserted by AAR/ARS into the outpulsed digit string to access the interexchange carrier. You cannot use duplicate access numbers in the table. Accepts from 2 to 11 digits and the * character.

IXC Name

A description that identifies the IXC. Accepts up to 15 characters.

Inter-Exchange Carrier (IXC) Codes: page 2

IXC Code Format

A one- to four-digit IXC code format. Includes *, x, X, xxxx (for line 1), and xxx (for line 2).

IXC Prefix

A one- to three-digit prefix. Includes *, 101 (for line 1) and 10 (for line 2).

Chapter 62: Intra-Switch CDR

Administers extensions for which Intra-Switch Call Detail Recording (CDR) is enabled.

 Note:

Attendants are disallowed to be optioned for the Intra-Switch CDR feature.

Example command: `change intra-switch-cdr n`, where *n* is the assigned extension number.

Assigned Members

The number of extensions currently administered for Intra-switch CDR.

Extension

The local extensions used to track with Intra-Switch CDR. The number of tracked extensions can vary by system.

Chapter 63: IP Address Mapping

Defines feature characteristics that depend on the IP address.

Example command: `change ip-network-map`

Emergency Location Extension

The emergency location extension for this station. Accepts up to sixteen digits. Allows the system to properly identify the location of a caller who dials a 911 emergency call from this station. An entry in this field must be of an extension type included in the dial plan, but does not have to be an extension on the local system. It can be a Uniform Dial Plan (UDP) extension. A blank entry is typically used for an IP softphone dialing in through PPP from somewhere outside the network.

For administered emergency numbers, the feature functions as follows:

- If the emergency location extension administered on the station screen is the same as the emergency location extension administered on the IP Address Mapping screen, the feature sends the station's own extension to the Public Safety Answering Point (PSAP).
- If the emergency location extension for the station is different from the emergency location extension administered here, the feature sends the extension administered here to the PSAP.

 Caution:

On the ARS Digit Analysis Table, administer 911 to be call type emer or alrt in order for the E911 Emergency feature to work properly.

Related links

[Emergency Location Ext](#) on page 672

From IP Address

The starting IP address. IPv6 format is supported.

Network Region

Valid Entry	Usage
1 to 2000	The network region for the IP address range. For SIP, the value for this field must correlate with the configured network region for this range of addresses. This field must contain a non-blank value if the corresponding From IP Address contains a non-blank value.

Subnet Bits

Valid Entry	Usage
0 to 64 blank	The number of bits of the subnet mask. Used in conjunction with the From IP Address to specify the end of the ID address range if a To IP Address is not administered.

To IP Address

The terminating IP address. IPv6 format is supported.

VLAN

Sends VLAN instructions to IP endpoints such as IP telephones and softphones. This field does not send VLAN instructions to the PROCR (Avaya Solutions Platform S8300 Server).

Valid Entry	Usage
0 to 4094	The virtual LAN value.
n	Disabled

Chapter 64: IP Codec Set

Use this screen to specify the type of codec used for voice encoding and companding (compression/decompression).

The default codec is set for G711. The G711 provides the highest voice quality, but G711 uses the maximum bandwidth. If G711 does not meet the required voice-quality or bandwidth trade-off specification, you can change the G711 default setting.

To modify the values of the fields on this screen, use the following command:

`change ip-codec-set n`, where *n* is the IP codec set number.

 Note:

The command `change ip-media-parameters n` works similar to the command `change ip-codec-set n`. Both the commands are valid.

IP Codec Set: page 1

Defines the allowed codecs and packet sizes used between VoIP resources. Enables silence suppression on a per-codec basis and dynamically displays the packet size in milliseconds for each codec in the set, based on the number of frames administered per packet.

Audio Codec

Specifies the audio codec used for this codec set.

- G.711A (a-law)
- G.711MU (mu-law)
- G.722-64k
- G.722.1-24k
- G.722.1-32k
- G.722.2
- G.723-5.3
- G.723-6.3
- G.726A-32K

- G.729
- G.729A
- G.729B
- G.729AB
- SIREN14-24k
- SIREN14-32k
- SIREN14-48k
- SIREN14-S48k
- SIREN14-S56k
- SIREN14-S64k
- SIREN14-S96k
- OPUS-NB12K
- OPUS-NB16K
- OPUS-WB20K
- OPUS-SWB24K

! Important:

Include at least two codecs for every telephone to avoid incompatible codecs. Only one OPUS codec is allowed on ip-codec form. Use the codecs specified in the following table for the telephones shown.

Telephone	Codec to use
All Avaya IP Telephones	G.711, G.729B
4601 4602 4602SW 4620SW 4621SW 4622SW	add G.726A (requires firmware R2.2)

Codec Set

The number assigned to this Codec Set.

Encrypted SRTCP

Use the Encrypted SRTCP field to select a policy option to activate SRTCP for all call connections within IP network regions governed by this IP codec set. This field works only if the associated

RTP channel is operating in encrypted mode, that is, srtcp-aes128 or srtcp-aes256. The system displays this field only when you enable the **Media Encryption Over IP** field on the change system-parameters customer-options screen.

Valid entry	Usage
enforce-enc-srtcp	To permit only encrypted SRTCP calls. Use this option to achieve high security standards.
best-effort	If you want Communication Manager to facilitate negotiation of the encrypted SRTCP parameter on call connections that traverse the IP network regions that are governed by this IP codec set.
enforce-unenc-srtcp	To permit only calls that do not use media encryption. For networks that have Communication Manager Release 7.0 with older endpoints. This option is the default value for this field.

Frames Per Pkt

Specifies the number of frames per packet up to a packet size of 60 milliseconds (ms).

Valid Entry	Usage
1 to 6	Default frame sizes for codecs: • G.711 and G.729: 2 frames (20 ms) • G.723: 1 frame (30 ms) • G.726A: 2 frames (20 ms) • OPUS-NB12K: 1 frame (20 ms) • OPUS-NB16K: 1 frame (20 ms) • OPUS-WB20K: 1 frame (20 ms) • OPUS-SWB24K: 1 frame (20 ms)

Media Encryption

This field specifies a priority listing of five options for the negotiation of encryption. Communication Manager provides bearer encryption according to the administered priority order. The selected option for an IP codec set applies to all codecs defined in that set. The ip-codec-set screen displays the **Media Encryption** field only if you enable the **Media Encryption over IP** field.

Note:

SRTP is a media encryption standard defined in RFC 3711 as a profile of RTP. Communication Manager supports the following functionality as specified in RFC 3711.

- Encryption of RTP (optional but recommended)
- Authentication of RTCP streams (mandatory)
- Authentication of RTP streams (optional but recommended)

- Protection against replay

*** Note:**

AES and AEA are Avaya proprietary techniques and not recommended for use. The SRTCP options are recommended. The following four choices are the recommended options for use:

- 10-srtp-aescm256-hmac80
- 11-srtp-aescm256-hmac32
- 1-srtp-aescm128-hmac80
- 2-srtp-aescm128-hmac32

Valid entry	Usage
aes	Advanced Encryption Standard (AES), a standard cryptographic algorithm for U.S. government organizations to protect sensitive (unclassified) information. Use this option to encrypt the following links: • Server-to-gateway (H.248) • Gateway-to-endpoint (H.323)
aea	Avaya Encryption Algorithm. Use this option as an alternative to AES encryption when: • All endpoints within a network region using this codec set must be encrypted. • All endpoints communicating between two network regions and administered to use this codec set must be encrypted.
1-srtp-aescm128-hmac80	1-Encrypted/Authenticated RTP with 80-bit authentication tag
2-srtp-aescm128-hmac32	2-Encrypted/Authenticated RTP with 32-bit authentication tag
3-srtp-aescm128-hmac80-unauth	3-Encrypted RTP but not authenticated
4-srtp-aescm128-hmac32-unauth	4-Encrypted RTP but not authenticated
5-srtp-aescm128-hmac80-unenc	5-Authenticated RTP with 80-bit authentication tag but not encrypted
6-srtp-aescm128-hmac32-unenc	6-Authenticated RTP with 32-bit authentication tag but not encrypted
7-srtp-aescm128-hmac80-unenc-unauth	7-Unencrypted/Unauthenticated RTP

Table continues...

Valid entry	Usage
8-srtp-aescm128-hmac32-unenc-unauth	8-Unencrypted/Unauthenticated RTP  Note: For stations, the only value supported is srtp-aescm128-hmac80. H.323 IP trunks support all eight listed algorithms.
10-srtp-aescm256-hmac80	10-AES-256 Encrypted/Authenticated RTP with 80-bit authentication tag
11-srtp-aescm256-hmac32	11-AES-256 Encrypted/Authenticated RTP with 32-bit authentication tag
none	Unencrypted media stream. This entry is the default.

Packet Size (ms)

The packet size in milliseconds.

Silence Suppression

Enables or disables RTP-level silence suppression on the audio stream.

IP Codec Set: page 2

You can configure the following capabilities on the codec set:

- Enable or disable Direct-IP Multimedia for videophone transmissions.
- Enable or disable endpoints in the assigned network region to send fax, modem, or TTY calls over IP trunks.
- Define the mode that the system uses to route the fax, modem, or TTY calls.
- Determine whether redundancy must be added to the transmission for higher reliability and quality.

You must assign these capabilities to the codec set, and assign the codec set to a network region for endpoints in that region to be able to use the capabilities established on this screen.

Using V.150.1, you can configure fax and modem so that both use a relay mode together in an IP codec set. You can set the **FAX Mode** field to t.38—standard for fax calls to use t.38. However, modem calls use V.150.1.

 Note:

Transporting modem signals over IP between Avaya Communication Manager systems and media gateways is a proprietary implementation. The selection of the V.150.1 Modem Relay codec choice is not proprietary. For modem transport interoperability with third-party gateways, v150mr is the recommended mode of operation.

⚠ Caution:

If users are using Super G3 fax machines and modems, do not assign these fax machines to a network region with an IP codec set that is modem-enabled as well as fax-enabled. If its **Codec set** is enabled for both modem and fax signaling, a Super G3 fax machine incorrectly tries to use the modem transmission instead of the fax transmission.

Therefore, assign modem endpoints to a network region that uses a modem-enabled IP Codec set, and the Super G3 fax machines to a network region that uses a fax-enabled IP Codec set.

Allow Direct-IP Multimedia

To enable or disable direct multimedia, use the following codecs:

- H.261
- H.263
- H.264 (video)
- H.224
- H.224.1 (data, far-end camera control)

Clear-channel Mode

Enables or disables supporting this codec set for BRI data calls.

ECM

With **Error Correction Mode (ECM)**, fax endpoints correct the fax transmission error without retransmitting multiple pages.

The **ECM** field is available only when **FAX** is t.38-standard or t.38fallback.

Valid entry	Usage
y	ECM-capable gateways use the T.38 protocol to relay T.30 ECM signaling and ECM frames between a local Group 3 fax machine and a far-end T.38 receiver. The default value is y.
n	ECM-capable gateways do not use the T.38 protocol to relay T.30 ECM signaling and ECM frames.

FAX

Valid entry	Usage
off	For users using this mode, Communication Manager treats a fax call as an ordinary voice call.

Table continues...

Valid entry	Usage
relay	For users using this mode, Communication Manager uses the Avaya proprietary relay mode for fax transmissions over IP network facilities only between Avaya devices. This mode can be coupled with Redundancy.
pass-through	For users using this mode, Communication Manager uses the Avaya proprietary pass-through mode for in-band fax transmissions over IP network facilities only between Avaya devices. This mode can be coupled with Redundancy.
T.38-standard	For users using this mode, Communication Manager uses the T.38 protocol for fax transmission over IP network facilities. Communication Manager supports the transition of an existing audio call to a fax call, for H.323 and SIP calls.
t38fallback	For users using this mode, Communication Manager uses the T.38 protocol for fax transmission only if the protocol can be successfully negotiated with the peer SIP entity. Otherwise, Communication Manager falls back to G.711 for fax transmission. This mode requires a G.711 codec to be administered on the ip-codec-set screen.

 Note:

If you have a telephone on an IP trunk that is close to a fax machine, the telephone can change into the fax mode by picking up the tones from the fax machine. To prevent the telephone from changing into fax mode, set the **FAX** to off and put the fax machines in an Automatic Route Selection (ARS) partition that uses only circuit switched trunks, even for intergateway fax calls.

FB Timer

The system displays this field if the mode is T38Fallback. The valid range is 1-20. The default value is 4.

XMT

XMT field is used to define the transport type. There are two supported modes, **udptl** and **rtp**. To support the backward compatibility, use the default mode **udptl**. The XMT field is available only when FAX is t.38fallback.

Maximum Call Rate for Priority Direct-IP Multimedia

The system displays this field only if **Allow Direct-IP Multimedia** is set to y.

Maximum Call Rate for Direct-IP Multimedia

Use this field to limit the size of bandwidth used for video calls. The system displays this field if **Allow Direct-IP Multimedia** is set to y. The default value is 384 kilobits. You can enter a value between 64 and 15360 kilobits, with an increment of 64 kilobits. A typical video endpoint does not require more than 2048 kilobits.

Media Connection IP Address Type Preferences

Use this field to specify the type of IP address and the preference of the IP address type to be used for media. The following are the valid entries:

- IPv4/IPv6
- IPv6/IPv4
- IPv4/none
- IPv6/none

Modem Mode

Valid entry	Usage
off	In the off mode, a modem call is treated as an ordinary voice call. The default mode for new installations and upgrades is off.
relay	In the Avaya proprietary relay mode, Avaya devices detect the tones of the call and use the appropriate modulation protocol or TTY to terminate or originate the modem calls over the IP network.
pass-through	In the Avaya proprietary pass-through mode, Avaya devices detect the tones of the modem call and use G.711 encoding to process the call over the IP network.
v150mr	In the v150mr mode, Avaya devices detect the tones of the modem call and use V.150.1 protocol to transmit V-series modem signals between secure terminals and third-party SIP gateways.  Note: V.150.1 protocol is a standard recommended by International Telecommunication Union (ITU) to use a modem over IP networks that support dialup modem calls.

Packet Size

 Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

The **Packet Size** column supports choices of 10, 20, 30, 40, 50, or 60 millisecond IP packet size to be used in the RFC 4040 transport.

The default choice is 20 milliseconds.

If the packet size is of value other than 20 milliseconds, then the SDP Offer must include an Attribute format string.

For example, for an administered Packet Size of 10 milliseconds, Communication Manager sends an SDP Offer message that includes the attribute string of a=ptime 10.

Redundancy

Use this field to assign the number of duplicate or redundant packets to be sent with the primary packet. You can enter a value from 1 to 3. The default value is 0.

 Note:

- For the **pass-through** and the **clear-channel** modes, you can use the default value or assign 1. This field does not apply to the v150mr mode. If you set the **Modem Mode** field to v150mr, the system sets the **Redundancy** field to display-only with the default value.
- This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

The **Redundancy** field supports choices of 0 and 1 for the **SIP64K** field. 0 designates that no redundant media packets are transmitted. 1 designates that a single redundant packet should be transmitted for each media stream packet that is sent.

 Note:

- If the Fax field is set to t38fallback then, Avaya recommends to leave this field to the default of “0”. With the use of the “Error-Correction Mode”, the use of sending redundant packets is not recommended by Avaya.
- If the Communication Manager is administered to send two redundant PDUs and the peer side of the SIP trunk can only handle one redundant PDU, then the second redundant packet is ignored at the peer side receiver. Alternatively, when the peer side of the SIP trunk transmits information, it encode this information with 1 redundant payload in the RTP packet body. Thus, both sides of the SIP trunk can claim redundancy. They do not negotiate on the number of redundant PDUs to be encoded into the RTP packet. Communication Manager always signals to the local gateway to inform it about the administered redundancy setting that is configured on the SAT IP-Codec-Set screen.

SIP 64K Data

With SIP 64K Data, Communication Manager controls the mechanism to enable the support of the RFC 4040 media service.

Valid Entry	Usage
y	Communication Manager searches for an RFC 4040 capable gateway, G450 or G430. If the given gateway does not support the RFC 4040 feature, Communication Manager searches for an available gateway in the connected network region. If Communication Manager cannot detect a suitable gateway within the network region, the call request is dropped.
n	The default value is n.

 Note:

G430 and G450 gateways support the RFC 4040 feature. The gateway in the network region must support the RFC 4040 feature for Communication Manager to connect RFC 4040 calls.

Communication Manager uses RFC 4040 Clear Mode data transport to support the media transport of ISDN traffic. The ISDN traffic is directed to a destination that is reached through a SIP trunk.

Associated with the **SIP 64K Data** field are two other fields, **Redundancy** and **Packet Size (ms)**. Communication Manager communicates the values of the **Redundancy** and **Packet Size (ms)** fields to the media gateway so that the gateway properly operates with the DSP conversion of the TDM media into an IP media stream.

TDD/TTY Mode

Valid entry	Usage
off	Turn off special TTY handling when using this codec set. In this case, the TTY transmission is treated like an ordinary voice call. With a codec set that uses G.711, this setting must send TTY calls to non-Avaya systems. However, there might be errors in character transmissions.
US	For users in regions using this codec, use U.S. Baudot 45.45 mode for TTY transmissions over IP network facilities. This is the default value for new installations and upgrades.
UK	For users in regions using this codec, use U.K. Baudot 50 mode for TTY transmissions over IP network facilities.
pass-through	For users in regions using this codec, use the Avaya proprietary pass-through mode for TTY transmissions over IP network facilities.

IP Codec Set: page 3

The system displays this page only when you set the **Modem Mode** field to v150mr.

Modem Relay Preferred (NoAudio)

If you set this field to y, Communication Manager includes the NoAudio codec in the SIP SDP offer/answer message. Enable this field when a gateway interoperates with the devices that support offer/answer NoAudio codec. The default value is n.

Modulation Mode

Avaya V.150.1 supports the V.32, the V.34, the V.90, and the V.92 modem modulations. For users who need other modulations, you must set the **Modem Mode** field to pass-through. The following fields show whether V.150.1 is enabled to transport the respective modulations as Modem Relay:

V.32

Valid entry	Usage
y	Use this entry to enable V.32 modem modulation. The default value is y.
n	Use this entry to disable V.32 modem modulation.

V.34

Valid entry	Usage
y	Use this entry to enable V.34 modem modulation. The default value is y.
n	Use this entry to disable V.34 modem modulation.

V.90

Valid entry	Usage
y	Use this entry to enable the V.90 digital modem modulation. The default value is y.  Note: The system sets the V.34 field to display-only with the default value y.
n	Use this entry to disable the V.90 digital modem modulation.

V.92

Valid entry	Usage
y	Use this entry to enable the V.92 digital modem modulation.  Note: The system sets the V.34 field to display-only with the default value y.
n	Use this entry to disable the V.92 digital modem modulation. The default value is n.

SPRT Retransmissions

Use this field to assign the maximum number of data retransmissions to Transport Channel 1 (TC1) and Transport Channel 2 (TC2). You can assign a value from 1 to 32. The default value is 3.

SPRT TC0 Payload Size

Use this field to assign the maximum payload size to SPRT Transport Channel 0. You can enter a value from 140 to 256. The default value is 140 bytes.

 Note:

Maximum SPRT payload sizes might vary across different gateways depending on the firmware and the hardware version.

SPRT TC1 Payload Size

Use this field to assign the maximum payload size for SPRT Transport Channel 1. You can enter a value between 132 to 256. The default value is 132 bytes.

 Note:

Maximum SPRT payload sizes might vary across different gateways depending on the firmware and the hardware versions.

SPRT TC1 Window Size

Use this field to assign the window size to SPRT Transport Channel 1. You can enter a value from 32 to 96. The default value is 32 bytes.

SPRT TC2 Payload Size

Use this field to assign the maximum payload size to SPRT Transport Channel 2. You can enter a value from 132 to 256. The default value is 132 bytes.

 Note:

Maximum SPRT payload sizes might vary across different gateways depending on the firmware and the hardware versions.

SPRT TC2 Window Size

Use this field to assign the window size to SPRT Transport Channel 2. You can enter a value from 8 to 32. The default value is 8 bytes.

SPRT TC3 Payload Size

Use this field to assign the maximum payload size to SPRT Transport Channel 3. You can enter a value from 140 to 256. The default value is 140 bytes.

 Note:

Maximum SPRT payload sizes might vary across different gateways depending on the firmware and the hardware versions.

SPRT Timer TC1-TA01

Use this field to set the value of the acknowledgement timer of Transport Channel 1. This field accepts values between 50 and 1000 but only in multiples of 10. The default value is 90.

SPRT Timer TC1-TA02

Use this field to set the value of the acknowledgement update timer of Transport Channel 2. This field accepts values between 50 and 1000 but only in multiples of 10. The default value is 130.

SPRT Timer TC1-TR03

Use this field to set the value of the retransmit timer of Transport Channel 1. This field accepts values between 50 and 1000 but only in multiples of 10. The default value is 500.

SPRT Timer TC2-TA01

Use this field to set the value of the acknowledgement timer of Transport Channel 2. This field accepts values between 50 and 1000 but only in multiples of 10. The default value is 90.

SPRT Timer TC2-TA02

Use this field to set the value of the acknowledgement update timer of Transport Channel 2. This field accepts values between 50 and 1000 but only in multiples of 10. The default value is 500.

SPRT Timer TC2-TR03

Use this field to set the value of the retransmit timer of Transport Channel 2. This field accepts values between 50 and 1000 but only in multiples of 10. The default value is 500.

SSE Inter-Packet Interval

Use this field to assign an interval between the State Signaling Events (SSE) packets. You can enter a value between 10 to 40 with the increments of 10. The default value is 20 mSec.

SSE Repetition Redundancy (Packets)

Use this field to assign the number of redundant SSE packets for transmission. The redundancy option provides robust transport of data packets across an IP-based network. You can enter a value from 0 to 3. The default value is 3.

V.42 Error Correction

V.42 is an error correction protocol for V.150.1 media transport between modems. If you set this field to y, the gateways use the **V.42 Error Correction** protocol. The default value is n.

Chapter 65: IP Interfaces

Assigns a network region to an IP interface device, or administers Ethernet options.

The appearance of the IP Interfaces screen can vary according to the interface type you are administering, and your system's configuration.

Example command: `add ip-interface n`, where *n* is the procr.

IP Interface: page 1

Allow H.248 Gateways

Controls whether branch gateways (G700, G450, G430) can or cannot register on the interface.

Valid Entry	Usage
y	On a simplex main server, enables H.248 endpoint connectivity to the Processor Ethernet (PE) interface. Used for a Survivable Remote Server (Local Survivable Processor).
n	Disables H.248 endpoint connectivity to the PE interface. H.248 endpoint connectivity using the PE interface on a Survivable Core Server (Enterprise Survivable Server) is not supported.

Allow H.323 Endpoints

Controls whether H.323 endpoints can or cannot register on the interface.

Valid Entry	Usage
y	On a simplex main server, enables H.323 endpoint connectivity to the Processor Ethernet (PE) interface. Used for a Survivable Remote Server (Local Survivable Processor).
n	Disables H.323 endpoint connectivity to the PE interface. H.323 endpoint connectivity using the PE interface on a Survivable Core Server (Enterprise Survivable Server) is not supported.

Enable Ethernet Interface

Valid Entry	Usage
y	Enables the Ethernet port.
n	Disables when there is no standby, or when the standby has been disabled. Should be disabled before administering the IP interface.

Ethernet Link

The administered link number for an Ethernet link.

Gatekeeper Priority

Available only if H.323 endpoints are enabled and the Communication Manager server is a main server or a Survivable Remote Server. Unavailable on a Survivable Core Server.

Valid Entry	Usage
1 to 9	Sets a priority on the interface that affects where the interface is displayed on the gatekeeper list. The value in this field is used on the alternate gatekeeper list. The lower the number, the higher the priority. Default is 5.

Related links

[Allow H.323 Endpoints](#) on page 393

Network Region

Valid Entry	Usage
1 to 2000	The value of the network region.

Network uses 1's for Broadnet Addresses

Allows or denies using a broadcast address to send the same message to all systems or clients on a local area network.

Node Name

The node name associated with the IP address of the procr. Accepts up to 15 characters.

Related links

[Near-end Node Name](#) on page 620

[Name](#) on page 413

[Group Type](#) on page 614

Subnet Mask

A 32-bit binary number that divides the network ID and the host ID in an IP address.

Target socket load

The maximum number of sockets targeted for this interface. The default is 80% of the platform maximum. Used for load balancing endpoint traffic across multiple IP interfaces. Controls the percentage of sockets allocated to each IP interface within the same Gatekeeper Priority. When all the IP interfaces within the same Gatekeeper Priority exceeds the target number allocated, the system continues to add sockets until the interface is at its maximum capacity. Available only with a procr type IP interface.

Valid Entry	Usage
1 to 3500	Dell™ PowerEdge™ R630 R640 and Duplex servers

- From Avaya Aura® Release 10.1, HP ProLiant DL360p G8 (CSR2), HP ProLiant DL360 G9 (CSR3), Dell™ PowerEdge™ R620 (CSR2), Dell™ PowerEdge™ R630 (CSR3), and Avaya Solutions Platform 120 servers are not supported.

However, in Release 10.1, Avaya Solutions Platform 120 can be upgraded to Avaya Solutions Platform 130 Release 5.x, and S8300E can be upgraded to Avaya Solutions Platform S8300 R5.1.x.

- From Avaya Aura® Release 10.1, Appliance Virtualization Platform is not available for deploying or upgrading the Avaya Aura® applications. To upgrade the Avaya Aura® applications, migrate the Appliance Virtualization Platform to Avaya Solutions Platform 130 (Avaya-Supplied ESXi 7.0) Release 5.x.

Related links

[Type](#) on page 395

Target socket load and Warning level

Controls the percentage of sockets allocated to each IP interface within the same Gatekeeper Priority. When all the IP interfaces within the same Gatekeeper Priority exceeds the target number allocated, the system continues to add sockets until the interface is at its maximum capacity.

Valid Entry	Usage
1 to 499	The maximum number of sockets targeted for this interface. If the number of sockets exceeds the targeted number, a warning alarm is generated. The default is 400.

Related links

[Type](#) on page 395

Type

The type of IP interface.

Valid Entry	Usage
procr	Processor — Avaya Solutions Platform S8300 and duplex media servers

VLAN

Sends Virtual Local Area Network (VLAN) instructions to the PROCR (8300E/Duplex Media Servers), CLAN, and Media Processor boards. It does not send VLAN instructions to IP endpoints such as IP telephones and softphones. Unavailable for Voice Announcement over LAN (VAL) boards.

Valid Entry	Usage
0 to 4095	Specifies the virtual LAN value.
n	Disabled. This is the default.

IP Interface: page 2

ETHERNET OPTIONS

The Ethernet port associated with the TN2602AP must be disabled before any changes can be made to these fields.

Auto?

Valid Entry	Usage
y	Enables auto-negotiation. Default.
n	Applies manual speed and duplex settings.

Duplex

Available only if auto-negotiation is disabled.

Valid Entry	Usage
Full	The full duplex setting for this IP board. Default when speed is set to 100 Mbps.
Half	The half duplex setting for this IP board. Default.

Related links

[Auto?](#) on page 396

IPV6 PARAMETERS

Enable Ethernet Interface

Valid Entry	Usage
y	Enables the Ethernet port.

Table continues...

Valid Entry	Usage
n	Disables when there is no standby, or when the standby has been disabled. Should be disabled before administering the IP interface.

Ethernet Link

The administered link number for an Ethernet link.

Node Name

The node name associated with the IP address of the procr. Accepts up to 15 characters.

Related links

[Near-end Node Name](#) on page 620

[Name](#) on page 413

[Group Type](#) on page 614

Subnet Mask

A 64-bit binary number that divides the network ID and the host ID in an IP address.

Chapter 66: IP network region

Configures within-region and between-region connectivity settings for all VoIP resources and endpoints within a given IP region. The first page is used to modify the audio and QoS settings. The **Codec Set** field on this page reflects the CODEC set that must be used for connections between telephones within this region or between telephones and MedPro/Prowler boards and gateways within this region. The ability to do NAT shuffling for direct IP-to-IP audio connections is also supported.

Example command: `change ip-network-region n`, where *n* is the network region number.

IP network region: page 1

NR Group

Use this field to assign a network region group to the network region. You can enter a value from:

- 1 to 2000 for large systems.
- 1 to 250 for small systems.

Do not leave the field blank.

You can assign multiple network regions to the same network region group. During translations upgrade, the system assigns a network region group to each network region. For example, network region 45 will be assigned to network region group 45. The network region group names will also be assigned. If the network region has a name, the name is used for the associated network region group name. If the network region name is blank, a unique network region group name is created by appending the network region group number to the characters NRG. For example, NRG 16.

Authoritative Domain

The domain for which this network region is responsible. The system displays the Authoritative Domain in the **From** header of any SIP messages. Accepts a name or IP address consisting of up to 20 characters.

Location

Note:

If the Multinational Locations feature is enabled, and IP telephones derive their network region from the IP Network Map, administer this field with a valid value (1 to 2000). With this, the IP endpoints can use the right VoIP resources.

Valid Entry	Usage
1 to 250 or 1 to 2000 Depending on the server configuration. For more information, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .	Assigns the location number to the IP network region. Correct date and time information, and trunk routing based on IP network region can be configured. The IP endpoint uses this as its location number. See the Location sections in <i>Avaya Aura® Communication Manager Feature Description and Implementation</i> , for the other ways to assign a location, and for a list of features that use location.
blank	Obtains the location from the gateway where the endpoint is registered. By default, the value is blank.

Name

A description of the region. Accepts up to 20 characters.

Region

The number of the network region being administered.

Stub Network Region

Using this field, you can create a core network region or a stub network region.

Valid entry	Usage
y	Creates a stub network region. In the dst rgn field, you must enter the number of the destination network region that directly connects with this stub network region.  Note: If you are converting a core network region to a stub network region, you must ensure that the core network region is directly connected with only one core network region. A stub network must have only one direct connection with a core network region.
n	Creates a core network region. For network regions 1 to 250, the default value of this field is n. For network regions 251 to 2000, this field is noneditable and displays a default of y.

MEDIA PARAMETERS

Codec Set

Valid Entry	Usage
1 to 7	The number for the codec set for the region.

Intra-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints within a network region.

Valid Entry	Usage
y	Saves on bandwidth resources and improves sound quality of voice over IP transmissions.
n	Might be used if, for example, the IP telephones within the region are behind two or more firewalls.
native(NAT)	The IP address from which audio is to be received for direct IP-to-IP connections within the region is that of the telephone/ softphone itself (without being translated by NAT). IP telephones must be configured behind a NAT device before this entry is enabled.
translated(NAT)	The IP address from which audio is to be received for direct IP-to-IP connections within the region is to be the one with which a NAT device replaces the native address. IP telephones must be configured behind a NAT device before this entry is enabled.

Inter-region IP-IP Direct Audio

Allows direct audio connections between IP endpoints in different regions

Valid Entry	Usage
y	Saves on bandwidth resources and improves sound quality of voice over IP transmissions.
n	Might be used if, for example, the IP telephones within the region are behind two or more firewalls.
native(NAT)	The IP address from which audio is to be received for direct IP-to-IP connections between regions is that of the telephone itself (without being translated by NAT). IP telephones must be configured behind a NAT device before this entry is enabled.
translated(NAT)	The IP address from which audio is to be received for direct IP-to-IP connections between regions is to be the one with which a NAT device replaces the native address. IP telephones must be configured behind a NAT device before this entry is enabled.

UDP PORT RANGE

UDP Port Range Min

Valid Entry	Usage
1024 to 65534	The minimum range of the UDP port number used for audio transport. Defaults to 2048 to 3028.

UDP Port Range Max

Valid Entry	Usage
1025 to 65535	The maximum range of the UDP port number used for audio transport. Defaults to 2048 to 3028.

DIFFSERV/TOS PARAMETERS

Audio PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop.

Valid Entry	Usage
0 to 63	The decimal equivalent of the DiffServ Audio PHB value. Default is 46.

Call Control PHB Value

Provides scalable service discrimination in the Internet without per-flow state and signaling at every hop.

Valid Entry	Usage
0 to 63	The decimal equivalent of the Call Control PHB value. Default is 34.

Video PHB Value

Valid Entry	Usage
0 to 63	The decimal equivalent of the DiffServ Video PHB value. Default is 26.

802.1P/Q PARAMETERS

Audio 802.1p Priority

Valid Entry	Usage
0 to 7	Provides Layer 2 priority for Layer 2 switches. Changes take effect after telephone reboot, or system reset.

Call Control 802.1p Priority

Valid Entry	Usage
0 to 7	Provides Layer 2 priority for Layer 2 switches. Changes take effect after telephone reboot, or system reset.

Video 802.1p Priority

Valid Entry	Usage
0 to 7	The Video 802.1p priority value. Changes take effect after telephone reboot, or system reset.

AUDIO RESOURCE RESERVATION PARAMETERS

Retry upon RSVP Failure Enabled

Enables or disables retries when RSVP fails.

Available only if RSVP is enabled.

Related links

[RSVP Enabled](#) on page 402

RSVP Enabled

Enables or disables RSVP.

RSVP Profile

Available only if RSVP is enabled.

Set this field to what you have configured on your network.

Valid Entry	Usage
guaranteed-service	Limits end-to-end queuing delay from sender to receiver. This setting is best for VoIP applications.
controlled-load	This subset of guaranteed-service provides for a traffic specifier, but not end-to-end queuing delay.

RSVP Refresh Rate (secs)

Available only if RSVP is enabled.

The system displays this field only if the **RSVP Enabled** field is set to y.

Valid Entry	Usage
1 to 99	The RSVP refresh rate in seconds.

Related links

[RSVP Enabled](#) on page 402

RSVP unreserved (BBE) PHB Value

Valid Entry	Usage
0 to 63	The BBE codepoint is used whenever an RSVP reservation is being obtained (pending), or has failed in some way, to provide efficient service to the voice stream.

H.323 IP ENDPOINTS**H.323 Link Bounce Recovery**

Enables or disables the H.323 Link Bounce Recovery feature for this network region. The default is enabled.

Idle Traffic Interval (seconds)

Valid Entry	Usage
5 to 7200	The maximum traffic idle time in seconds after which a TCP Keep-Alive (KA) signal is sent from the endpoint. Default is 20.

Keep-Alive Interval (seconds)

Valid Entry	Usage
1 to 120	Sets the interval between TCP Keep-Alive re-transmissions. When no ACK is received for all retry attempts, the local TCP stack ends the TCP session and the associated socket is closed. Default is 5.

Keep-Alive Count

Valid Entry	Usage
1 to 20	Sets the number of times the Keep-Alive message is transmitted if no ACK is received from the peer. Default is 5.

IP network region: page 2**ALTERNATIVE NETWORK ADDRESS TYPES**

To control the call processing behavior to send Alternative Network Address Types (ANAT) offer for the IP network region.

ANAT Enabled

Use this field to control the call processing behavior of the Alternative Network Address Types (ANAT) for the IP network region.

Valid entry	Usage
y	Communication Manager sends ANAT offer to SIP elements in the IP network region when you set this field to y as well as system-wide ANAT Enabled field to y.
n	Communication Manager does not send ANAT offer to SIP elements in the IP network region.

RTCP Reporting to Monitor Server Enabled

If enabled, sends RTCP Reports to a special server, such as for the VMON tool.

 Note:

Regardless of how this field is administered, RTCP packets are always sent peer-to-peer.

RTCP MONITOR SERVER PARAMETERS

IPv4 Server Port

Available only if RTCP Reporting is enabled and if Default Server Parameters are disabled.

Valid Entry	Usage
1 to 65535	The port for the RTCP Monitor server. Default is 5005.

IPv6 Server Port

Available only if RTCP Reporting is enabled and if Default Server Parameters are disabled.

Valid Entry	Usage
1 to 65535	The port for the RTCP Monitor server. Default is 5005.

RTCP Report Period (secs)

Available only if RTCP Reporting is enabled and if Default Server Parameters are disabled.

Valid Entry	Usage
5 to 30	The report period for the RTCP Monitor server in seconds.

Related links

[Use Default Server Parameters](#) on page 405

Server IPV4 Address

The IPv4 address for the RTCP Monitor server.

Available only if RTCP Reporting is enabled and if Default Server Parameters are disabled.

Server IPV6 Address

The IPv6 address for the RTCP Monitor server.

Available only if RTCP Reporting is enabled and if Default Server Parameters are disabled.

Use Default Server Parameters

If enabled, uses the system-wide default RTCP Monitor server parameters. Available only if RTCP Reporting is enabled.

IP network region: page 3

This page covers the information for Inter-Gateway Alternate Routing (IGAR), backup server names in priority order, and security procedures.

INTER-GATEWAY ALTERNATE ROUTING/DIAL PLAN TRANSPARENCY

If **Inter-Gateway Alternate Routing** (IGAR) is enabled for any row on subsequent pages, the following fields for each network region must be administered to route the bearer portion of an IGAR call.

Conversion to Full Public Number - Delete

Valid Entry	Usage
0 to 7	The digits to delete

Conversion to Full Public Number - Insert

Valid Entry	Usage
0 to 16 blank	The number of digits to insert. International numbers should begin with plus (+). The Inter-Gateway Alternate Routing (IGAR) and Dial Plan Transparency (DPT) features convert the plus (+) digit to appropriate international access code when starting the trunk call.  Note: The optional plus (+) at the beginning of the inserted digits is an international convention indicating that the local international access code must be dialed before the number.

Dial Plan Transparency in Survivable Mode

Valid Entry	Usage
y	Enables the Dial Plan Transparency feature when a gateway registers with a Survivable Remote Server (Local survivable processor).
n	Default is n.

Incoming LDN Extension

An extension used to assign an unused Listed Directory Number for incoming IGAR calls.

Maximum Number of Trunks to Use for IGAR

It is necessary to impose a limit on the trunk usage in a particular port network in a network region when Inter-Gateway Alternate Routing (IGAR) is active. The limit is required because if there is a major IP WAN network failure, it is possible to use all trunks in the network region(s) for IGAR calls.

Valid Entry	Usage
1 to 999, or blank	The maximum number of trunks to be used for Inter-gateway alternate routing (IGAR).

BACKUP SERVERS IN PRIORITY ORDER

Lists the backup server names in priority order. Backup server names should include Survivable Remote Server names and Survivable Core Server names. If you are using the Processor Ethernet, the backup servers list must include the survivable core PE address else the phones will not register to the survivable core during a failure. Any valid node name is a valid entry. Valid node names can include names of Customer LANs, ICCs, Survivable Core Servers, and Survivable Remote Servers.

H.323 SECURITY PROFILES

Permitted security profiles for endpoint registration in the network region. You must enter at least one security profile. Otherwise, no endpoint will be permitted to register from the region.

Valid Entry	Usage
challenge	Includes the various methods of PIN-based challenge and response schemes in current use. This is a relatively weak security profile.
pin-eke	The H.235 Annex H SP1.
strong	Permits the use of any strong security profile. The H323TLS profile is the strongest security profile in Communication Manager.
any-auth	Includes any of the security profiles.

Table continues...

Valid Entry	Usage
H323TLS	Communication Manager applies this security profile when the network region of an H.323 phone is administered with H323TLS or Strong security profiles. Also, Communication Manager and the endpoint negotiate by using the H323 TLS profile. H323TLS profile sends H.323 signaling messages through a TLS-encrypted channel.

Allow SIP URI Conversion

Administers whether or not a SIP URI should be permitted to change. Degrading the URI from sips//: to sip//: might result in a less secure call. This is required when SIP SRTP endpoints are allowed to make and receive calls from endpoints that do not support SRTP.

Valid Entry	Usage
y	Allows conversion of SIP URIs. Default is y.
n	No URI conversion. Calls from SIP endpoints that support SRTP made to other SIP endpoints that do not support SRTP will fail. However, if you enter y for the Enforce SIPS URI for SRTP field on the signaling group screen, URI conversion takes place independent of the value set for the Allow SIP URI conversion field on the IP Network Region screen.

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS

Near End Establishes TCP Signaling Socket

Indicates whether Communication Manager (the near end) can establish the TCP socket for H.323 IP endpoints in this network region.

Valid Entry	Usage
y	Communication Manager determines when to establish the TCP socket with the IP endpoints, assuming the endpoints support this capability. This is the default.
n	The IP endpoints always attempt to set up the TCP socket immediately after registration. This field should be disabled only in network regions where a non-standard H.323 proxy device or a non-supported network address translation (NAT) device would prevent the server from establishing TCP sockets with H.323 IP endpoints.

Near End TCP Port Min

Valid Entry	Usage
1024 to 65531	The minimum port value used by the processor Ethernet when establishing the TCP signaling socket to the H.323 IP endpoint. The range of port number must be at least 5 (Max-Min+1). Default is 61440.

Related links

[Near End TCP Port Max](#) on page 408

Near End TCP Port Max

Valid Entry	Usage
1028 to 65535	The maximum port value to be used by the processor Ethernet when establishing the TCP signaling socket to the H.323 IP endpoint. The range of port number must be at least 5 (Max-Min+1). Default is 61444.

Related links

[Near End TCP Port Min](#) on page 407

IP network region: page 4

Each subsequent page shows the inter-region connectivity for 15 region pairs.

AGL

The maximum number of destination region IP interfaces included in alternate gatekeeper lists (AGL).

Valid Entry	Usage
0 to 16	Communication Manager uses the numeric value of gatekeeper addresses.
all	Communication Manager includes all possible gatekeeper addresses in the endpoint's own network region and in any regions to which the endpoint's region is directly connected.
blank	The administration field is ignored.

codec-set

Valid Entry	Usage
1 to 7 pstn blank	The codec set used between the two regions. This field cannot be blank if this route through two regions is being used by some non-adjacent pair of regions. If the two regions are disconnected at all, this field should be blank.

direct-WAN

Indicates whether the two regions (source and destination) are directly connected by a WAN link. The default value is enabled if a **codec-set** is administered.

Related links

[codec-set](#) on page 408

dst rgn

Valid Entry	Usage
1 to 250	The destination region for this inter-network connection.

Dyn CAC

Use this field to configure a gateway for Call Admission Control (CAC). The system displays this field only if the **WAN-BW-limits (Units)** field is set to *Dynamic*.

Valid Entry	Usage
1 to 250	The gateway that reports the bandwidth-limit for this link. Default is blank.
blank	

* Note:

If you set the **BW Management Option** field to *shared-SM*, you cannot view this field.

Related links

[WAN-BW-limits \(Units\)](#) on page 412

IGAR

Allows pair-wise configuration of Inter-Gateway Alternate Routing (IGAR) between network regions.

Valid Entry	Usage
y	Enables IGAR capability between this network region pair. Default for a pstn codec set.
n	Disable IGAR capability between this network region pair. Default, except for a pstn codec set.
f	Forced. Moves all traffic onto the PSTN. This option can be used during initial installation to verify the alternative PSTN facility selected for a network region pair. This option can also be used to temporarily move traffic off of the IP WAN if an edge router is having problems or an edge router needs to be replaced between a network region pair.

Intervening-regions

Allows entry of intervening region numbers between the two indirectly-connected regions.

Valid Entry	Usage
1 to 250	Up to four intervening region numbers between the two indirectly-connected regions.  Note: Indirect region paths cannot be entered until all direct region paths have been entered. In addition, the order of the path through the regions must be specified starting from the source region to the destination region.

Mtce

Valid Entry	Usage
t	This is a test-only option. Inter-region connectivity testing is performed for the network region pair by using a simple PING sent between entities in each network region. If a test fails, only an error is added to the system error log. IP media connections between the region pair are never blocked. The testing is done at the rate of not more than once per 5 minutes.
m	This is a measurement based option. Inter-region connectivity testing is performed by a continuous set of PINGs sent between entities in each network region. The Ping Test Interval (sec) and Number of Pings Per Measurement Interval fields control the rate of testing. The Roundtrip Propagation Delay (ms) and Packet Loss (%) thresholds control success or failure. If the, test measurements exceed the administered thresholds; future IP media connections between the network region pair will be blocked.
d	No testing is performed for the network region pair.

Related links

[Ping Test Interval \(sec\)](#) on page 416

[Number of Pings Per Measurement Interval](#) on page 416

[Roundtrip Propagation Delay \(ms\)](#) on page 417

[Packet Loss \(%\)](#) on page 416

src rgn

Valid Entry	Usage
1 to 250	The source region for this inter-network connection.

Sync

The system displays Sync when the **Synchronization over IP** field is enabled.

Valid Entry	Usage
y	Timing IGC streams are allowed between the region pair that is being administered. The default value is y.
n	Do not allow timing IGC streams between the region pair that is being administered.

Video (Norm)

Use this field to specify the amount of bandwidth to allocate for the normal video pool to each IP network region. You can enter:

- 0 to 9999 for Kbits.
- 0 to 65 for Mbits.
- blank for NoLimit.

 Note:

If you set the **BW Management Option** field to `shared-SM`, you cannot view this field.

Video (Prio)

Use this field to specify the amount of bandwidth to allocate for the priority video pool to each IP network region. You can enter:

- 0 to 9999 for Kbits.
- 0 to 65 for Mbits.
- blank for NoLimit.

 Note:

If you set the **BW Management Option** field to `shared-SM`, you cannot view this field.

Video (Shr)

Use this field to specify whether the normal video pool can be shared for each link between IP network regions.

 Note:

If you set the **BW Management Option** field to `shared-SM`, you cannot view this field.

WAN-BW limits (Total)

Use this field to specify the bandwidth limit for direct WAN links. You can enter:

- 0 to 9999 for Kbits
- 0 to 65 for Mbits
- blank for NoLimit

 Note:

If you set the **BW Management Option** field to `shared-SM`, you cannot view this field.

WAN-BW-limits (Units)

Use this field to specify the unit of measurement corresponding to the value entered for bandwidth limitation.

- Calls
- Dynamic
- Kbits
- Mbits
- NoLimit

 Note:

If you set the **BW Management Option** field to `shared-SM`, you cannot view this field.

Chapter 67: IP Node Names

Administers node names and IP addresses for the switch and the terminal server media processors administered on the IP Interfaces screen.

 Note:

The system displays the Processor Ethernet interface node name (procr or procr6) on the IP Node Names screen. The PE interface node name cannot be added to the IP Node Names screen. The line containing the keywords procr or procr6 displays the IP address.

Example command: `change node-names ip`

Name

The name of the adjunct, server, or switch node used as a label for the associated IP address. The node names must be unique for each server or switch. Uses up to 15 alpha-numeric characters.

IP Address

A unique IP address assigned to each port on any IP device that is used for a connection. Also supports IPv6 address format.

Chapter 68: IP options system parameters

Example command: `change system-parameters ip-options`

SYSTEM PARAMETERS IP-OPTIONS: page 1

AUTOMATIC TRACE ROUTE ON

Link Failure

Enables or disables the automatic trace route command. If enabled, to diagnose network problems, especially to determine where a network outage exists, Communication Manager initiates an automatic trace-route command when the connectivity between a server and its gateways, or IP trunks is lost.

 Note:

If disabled, any automatic trace-route currently in progress finishes, and no subsequent trace-route commands are started or logged. In other words, the link failure buffer is cleared.

H.248 MEDIA GATEWAY

Link Loss Delay Timeout (minutes)

Valid Entry	Usage
1 to 30	The number of minutes to delay the reaction of the call controller to a link bounce. Assists with the H.248 link bounce recovery mechanism of the Avaya G700 branch gateway. Specifically, prevents the call controller from removing all boards and ports prematurely in response to a link bounce. Default is 5.

H.323 IP ENDPOINT

Link Loss Delay Timer (minutes)

Valid Entry	Usage
1 to 60	The number of minutes to delay the reaction of the call controller to a link bounce. Specifies how long the Communication Manager server preserves registration and any stable calls that might exist on the endpoint after it has lost the call signaling channel to the endpoint. If the endpoint does not re-establish connection within this period, the system cancels the registration and any calls of the endpoint. This timer does not apply to soft IP endpoints operating in telecommuter mode. Default is 5.

Periodic Registration Timer (min)

This timer is started when an IP telephone registration is taken over by another IP endpoint. When the timer expires, the telephone tries to reregister with the server. Default timer value is dependent on the number of unsuccessful periodic registration attempts. Sample field values apply unless the endpoint is interrupted, such as by power loss, or the user takes manual action to override this automatic process:

- 20 means once every 20 minutes for two hours, then once an hour for 24 hours, then once every 24 hours continually.
- 60 means once an hour for two hours, then once an hour for 24 hours, then once every 24 hours continually.

Valid Entry	Usage
1 to 60	The number of minutes before an IP telephone registration is taken over by another IP endpoint attempts to re-register with the server. Default is 60.

Primary Search Time (seconds)

Valid Entry	Usage
15 to 3600	The maximum number of seconds the IP endpoint attempts to register with its current Communication Manager server while the telephone is disconnected before going to a Survivable Remote Server. With this timer, the customer can specify the maximum time that an IP endpoint spends on trying to connect to the PROCR before going to a Survivable Remote Server. When the IP telephone's receiver is lifted, the endpoint continues trying to re-establish connection with the current server until the call ends. Default is 75.

Short/Prefixed Registration Allowed

The system displays Short/Prefixed Registration Allowed if the **IP Stations** field on the System Parameters Customer Options screen is set to y.

Valid Entry	Usage
y	Call Processing allows an IP endpoint to register using a short extension, for the extensions that have Short/Prefixed Registration Allowed field set to default on the Station screen. The default value is y.
n	Call Processing does not allow an IP endpoint to register using a short extension, for the extensions that have Short/Prefixed Registration Allowed field set to default on the Station screen.

Related links

[Short/Prefixed Registration Allowed](#) on page 656

IP MEDIA PACKET PERFORMANCE THRESHOLDS**Enable Voice/Network Stats**

Enables or disables the recording of voice and network statistics at a system level for all TN2302/ TN2602 media processor boards in the network. The default value is disabled.

Number of Pings Per Measurement Interval

Valid Entry	Usage
10 to 100	the number of test pings that comprise a measurement from which the performance values (delay and loss) are calculated. Default is 10.

Packet Loss (%)

Specifies thresholds to be applied to packet loss rates (as measured by ping) for determining activation or deactivation of signaling group bypass.

High:

Valid Entry	Usage
0 to 100	The high value cannot be less than the minimum value. Default is 40.

Low:

Valid Entry	Usage
0 to 100	The low value cannot be more than the maximum value. Default is 15.

Ping Test Interval (sec)

Valid Entry	Usage
10 to 999	The time between performance test pings for each testable signaling group. Default is 20.

Roundtrip Propagation Delay (ms)

Specifies thresholds to be applied to roundtrip packet propagation delays as measured by ping, for use in activating or clearing signaling group bypass.

High:

Valid Entry	Usage
10 to 9999	The high value cannot be less than the minimum value. Default is 800.

Low:

Valid Entry	Usage
10 to 9999	The low value cannot be more than the maximum value. Default is 400.

MEDIA GATEWAY ANNOUNCEMENT SERVER PARAMETERS

Announcement Server IP Address

The IP address of the Announcement Server that is a unique IP address assigned to each port on any IP device that is used for a connection.

Announcement Storage Path Name

The directory path name on the Announcement Server where the announcements are stored. Accepts up to 40 characters.

Login

The login used by the gateway to access the announcement server. Accepts up to 10 characters.

Password

The password used by the gateway to access the announcement server. Accepts up to 10 characters.

RTCP MONITOR SERVER

The RTCP monitor is a separate computer that receives RTCP packets from many devices. Communication Manager pushes these values to IP telephones, IP softphones and VoIP media modules, such that they know where to send the data.

Default RTCP Report Period (secs)

Valid Entry	Usage
5 to 99	The number of seconds IP telephones, IP softphones, and VoIP media modules send RTCP packets to the RTCP server.

Default Server IP Address

The default IP address of the RTCP server used for each administered region. A unique IP address is assigned to each port on any IP device that is used for a connection.

Related links

[Server IPV4 Address](#) on page 405

Default Server Port

Valid Entry	Usage
1 to 65535	The default TCP/IP port of the RTCP server. Default is 5005.

Link Failure

Enables or disables the automatic trace route command. If enabled, to diagnose network problems, especially to determine where a network outage exists, Communication Manager initiates an automatic trace-route command when the connectivity between a server and its gateways, or IP trunks is lost.

* Note:

If disabled, any automatic trace-route currently in progress finishes, and no subsequent trace-route commands are started or logged. In other words, the link failure buffer is cleared.

SYSTEM PARAMETERS IP-OPTIONS: page 2

Prefer use of G.711 by Music Sources

Enables or disables using G.711 for intra-switch Music-On-Hold. The default value is n.

Force Phones and Gateways to Active LSPs

Enables or disables forcing telephones and gateways to active LSPs. The default is disabled.

HYPERACTIVE MEDIA GATEWAY REGISTRATIONS

Enable Detection and Alarms

Enables or disables the hyperactive gateway registration feature. Default is disabled.

Number of Registrations within the Window

Valid Entry	Usage
1 to 19	The number of registrations that occur within the hyperactivity window for generating a Gateway alarm. Default is 3. Available only if detection and alarming is enabled.

Related links

[Enable Detection and Alarms](#) on page 418

Parameters for Media Gateway Alarms: Hyperactive Registration Window (minutes)

Available only if detection and alarming is enabled.

Valid Entry	Usage
1 to 15	The time in minutes for checking hyperactive gateway registrations. Default is 4 minutes.

Related links

[Enable Detection and Alarms](#) on page 418

Parameters for Network Region Registration (NR-REG) Alarms: % of Gateways in Network Region with Hyperactive Registration Alarms

Available only if detection and alarming is enabled.

Valid Entry	Usage
1 to 99	The percent of Gateways within an ip-network region that should be alarmed before an IP-Registration alarm is generated. Default is 80%.

Related links

[Enable Detection and Alarms](#) on page 418

IP DTMF TRANSMISSION MODE

Inter-System IP DTMF Transmission Mode

Specifies the touchtone signals that are used for dual-tone multifrequency (DTMF) telephone signaling.

Valid Entry	Usage
in-band	All G711 and G729 calls pass DTMF in-band. DTMF digits encoded within existing RTP media stream for G.711/G.729 calls. G.723 is sent out-of-band. It also supports the SIP trunks.
in-band-g711	Only G711 calls pass DTMF in-band. The system displays the in-band-g711 option if the Group Type field is set to h.323.

Table continues...

Valid Entry	Usage
out-of-band	All IP calls pass DTMF out-of-band. For IP trunks, the digits are done with either Keypad IEs or H245 indications. This is the default for newly added H.323 signaling groups. The out-of-band option enables out of band signaling for the SIP Signaling group. When you select the out-of-band option, Communication Manager sends all outgoing DTMF messages as SIP INFO messages over a SIP signaling group. This option is interoperable with H.323 networks. You can connect an Avaya non-SIP endpoint or trunk to a voice mail system on the H.323 network that is integrated with Session Manager and Communication Manager through SIP. Examples of Avaya non-SIP endpoints or trunks are H.323, Analog, DCP, and Digital trunks.
rtp-payload	This is the method specified by RFC 2833. This is the default value for newly added SIP signaling groups. Support for SIP trunks requires the default entry of rtp-payload.

Intra-System IP DTMF Transmission Mode

The IP transmission mode.

Valid Entry	Usage
in-band	DTMF digits encoded within existing RTP media stream for G.711/G.729 calls. G.723 is sent out-of-band.
rtp-payload	Support for SIP trunks requires the entry of rtp-payload.

SYSTEM PARAMETERS IP-OPTIONS: page 3

ALTERNATIVE NETWORK ADDRESS TYPES

To control the call processing behavior to send Alternative Network Address Types (ANAT) offer system wide.

ANAT Enabled

Use this field to control the call processing behavior to send Alternative Network Address Types (ANAT) offer system wide.

Valid entry	Usage
y	Communication Manager sends ANAT offer to the SIP elements system wide.
n	Communication Manager does not send ANAT offer to the SIP elements system wide.

SERVICES DIALPAD PARAMETERS

To disable the services dialpad parameters on the telephones, set the **Download Flag** field to y and leave the **Password** field blank.

Download Flag

If you set this field to y, Communication Manager downloads the services dialpad parameters and the associated IP addresses to the telephones. The default value is n.

Password

Use this field to enter the password that a technician uses to administer craft procedures. During the installation or after the successful installation of an IP telephone, the technician performs craft procedures to customize the telephone installation for specific operating environment. You can provide a password of length up to seven digits or leave the field blank. The default value is 27238.

SNMP STATION PARAMETERS

To disable the SNMP parameters on the telephones, set the **Download Flag** field to y and leave the **Community String** field blank.

Community String

Use this field to enter a string that an IP telephone uses to determine whether the telephone must receive SNMP messages. You can enter up to 32 characters.

The telephone does not respond to the incoming SNMP message if you:

- leave the **Community String** field blank and the source address of the SNMP message does not match with an address in the **SNMP Source Address list**
- leave the **Community String** field blank and the source address of the SNMP message matches with an address in the **SNMP Source Address list**
- set the **Community String** field to a value and the source address of the SNMP message does not match with an address in the **SNMP Source Address list**
-

The telephone responds to the incoming SNMP message only if you set the **Community String** field to a value and the source address of the SNMP message matches with an address in the **SNMP Source Address list**.

Download Flag

If you set this field to y, Communication Manager downloads the SNMP parameters and the associated IP addresses to the telephones. The default value is n.

SOURCE ADDRESSES

Use this field to enter node names, mapping to proper IP address, that an IP telephone uses to validate the source address of an SNMP message. The telephone does not respond to the incoming SNMP message if you:

- leave the **Community String** field blank and the source address of the SNMP message does not match with an address in the **SNMP Source Address list**
- leave the **Community String** field blank and the source address of the SNMP message matches with an address in the **SNMP Source Address list**
- set the **Community String** field to a value and the source address of the SNMP message does not match with an address in the **SNMP Source Address list**
-

The telephone responds to the incoming SNMP message only if you set the **Community String** field to a value and the source address of the SNMP message matches with an address in the **SNMP Source Address list**.

You can enter up to six node names. Use the IP Node Names screen to administer node names.

Related links

[IP Address](#) on page 413

SYSTEM PARAMETERS IP-OPTIONS: page 4

Dest # 1, 2, or 3 IP address

The valid destination IPv4 address. The default destination address is 0.0.0.0.

Local Facility

Valid Entry	Usage
local0 to local7	Displays the help message upon acceptable values for local use. The default value is local4.

Port

Valid Entry	Usage
1 to 65535	The valid port number associated with the destination IPv4 address. The default port number is 514.

Related links

[Dest # 1, 2, or 3 IP address](#) on page 422

MUSIC/ANNOUNCEMENTS IP-CODEC PREFERENCES

In general, when operating across the WAN with limited bandwidth facilities, the ip-codec-set is configured only with the compressed voice codecs. Sometimes it is necessary to carry the voice calls with the normal configured compressed codecs, and the music or announcement sources are received with non-compressed G.711 codecs.

If you administer the following fields, when possible, the system overrides the ip-codec-set preference which can be configured to prefer a compressed codec, with non-compressed G.711. If the device receives the music or announcement source which has signaled support for G.711, the system attempts to use G.711.

Prefer use of G.711 by Announcement Sources

Overrides the ip-codec-set preference and establishes inter-PN or inter-gateway connections transmitting announcements with G.711. The default value is n.

Prefer use of G.711 by IP Endpoints Listening to Announcements

Overrides the ip-codec-set preference and reconfigures the IP endpoints listening to an announcement source with G.711. The default value is n.

Prefer use of G.711 by IP Endpoints Listening to Music

Overrides the ip-codec-set preference and reconfigures the IP endpoints listening to a music source with G.711. The default value is n.

Prefer use of G.711 by Music Sources

Overrides the ip-codec-set preference and establishes inter-PN or inter-gateway connections transmitting music with G.711. The default value is n.

SYSTEM PARAMETERS IP-OPTIONS: page 5

BW Management Option

Use this field to select the bandwidth management mode.

Valid entries	Usage
local-CM	With this mode, Communication Manager acts as stand-alone bandwidth management entity. Bandwidth limits for the bandwidth used by Communication Manager are set on Communication Manager. The default value is local-CM.

Table continues...

Valid entries	Usage
shared-SM	With this mode, Session Manager acts as the central authority for bandwidth management and Communication Manager obtains bandwidth for voice and multimedia IP connections from Session Manager. Bandwidth limits for the bandwidth collectively used by Communication Manager and other users are set using System Manager. For more information, see <i>Administering Avaya Aura® Session Manager</i> .

Signaling-group of Primary SM BW Mgr

Use this field to specify the assigned Session Manager SIP signaling group number. You can enter a value from 1 to 999. Do not leave the field blank.

The system displays the field only when you set the **BW Management Option** field to `shared-SM`.

Signaling-group of Secondary SM BW Mgr

Use this field to specify the assigned Session Manager SIP signaling group number. You can enter a value from 1 to 999. You can leave the field blank. This optional secondary Session Manager SIP signaling group is only used when the primary Session Manager SIP signaling group is unavailable. The secondary Session Manager SIP signaling group cannot be the same number as the primary Session Manager SIP signaling group.

The system displays the field only when you set the **BW Management Option** field to `shared-SM`.

Chapter 69: IP Routing

There is one-to-one mapping between the **Network Bits** and the **Subnet Mask** fields; entering a value in one field uniquely determines the other field. A list of Subnet Mask addresses and their corresponding Network Bits are shown in Table below.

Network Bits	Subnet Mask	Number of Hosts	Network Type
0	0.0.0.0	4,294,967,294	/ 0
1	128.0.0.0	2,147,483,646	/ 1
2	192.0.0.0	1,073,741,822	/ 2
3	224.0.0.0	536,870,910	/ 3
4	240.0.0.0	268,435,454	/ 4
5	248.0.0.0	134,217,726	/ 5
6	252.0.0.0	67,108,862	/ 6
7	254.0.0.0	33,554,430	/ 7
8	255.0.0.0	16,777,214	/ 8
9	255.128.0.0	8,388,606	/ 9
10	255.192.0.0	4,194,302	/ 10
11	255.224.0.0	2,097,150	/ 11
12	255.240.0.0	1,048,574	/ 12
13	255.248.0.0	524,286	/ 13
14	255.252.0.0	262,142	/ 14
15	255.254.0.0	131,070	/ 15
16	255.255.0.0	65,534	/ 16
17	255.255.128.0	32,766	/ 17
18	255.255.192.0	16,382	/ 18
19	255.255.224.0	8,190	/ 19
20	255.255.240.0	4,094	/ 20
21	255.255.248.0	2,046	/ 21
22	255.255.252.0	1,022	/ 22
23	255.255.254.0	510	/ 23
24	255.255.255.0	254	/ 24

Table continues...

Network Bits	Subnet Mask	Number of Hosts	Network Type
25	255.255.255.128	126	/ 25
26	255.255.255.192	62	/ 26
27	255.255.255.224	30	/ 27
28	255.255.255.240	14	/ 28
29	255.255.255.248	6	/ 29
30	255.255.255.252	2	/ 30
31	255.255.255.254	1	/31
	255.255.255.255	0	/32

Example command: `add ip-route next`

Board

Valid Entry	Usage
1 to 999	Gateway
V1 to V9	Module

Destination Node

The name of the final destination node of the IP route for this connection.

Related links

[IP Node Names](#) on page 413

Gateway

The node name of the first intermediate node consisting of a port on the procr. The first intermediate node is the Gateway. The Gateway is the local procr port.

Related links

[IP Node Names](#) on page 413

Metric

Valid Entry	Usage
0	A server that has one procr installed.

Network Bits

A 32-bit binary number that divides the network ID and the host ID in an IP address.

Valid Entry	Usage
0 to 32	The number of Network Bits associated with this IP route.

Route Number

Valid entry	Usage
digits	The IP route number. The upper limit can change depending on the system configuration. For more information, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .

Chapter 70: IP Services

Administers the connectivity for various adjuncts.

Note:

You cannot remove a service from this screen if that service has overrides defined on the Survivable Processor screen.

Example command: `change ip-services`

IP Services: page 1

[illegible]

Enabled

Enables or disables this IP service. Available only with an IP **Service Type** of AESVCS or SAT.

Related links

[Service Type](#) on page 429

Local Node

Valid Entry	Usage
procr	The Communication Manager's Processor Ethernet interface for adjunct connectivity.

Related links

[Name](#) on page 413

Local Port

The originating port number.

Valid Entry	Usage
5000 to 9999	5111 to 5117 for SAT applications. 8765 for AE Services.
0	For client applications, defaults to 0.

Remote Node

The server or switch at the far end of the link for SAT.

Valid Entry	Usage
<i>IP node name</i>	A previously-administered node name used to provide added security for SAT.
any	Any available node.

Remote Port

The port number of the destination.

Valid Entry	Usage
5000 to 64500	If this service is a client application, such as CDR or PMS. This value must match the port administered on the adjunct, Personal Computer, or terminal server that is at the remote end of this connection.
0	Default for System Management applications.

Service Type

The service provided.

Valid Entry	Usage
AESVCS	AE Services.
CDR1, CDR2	Either the primary or secondary CDR device connects over a TCP/IP link or TLS link. If you enter y for the TLS Encryption, then the primary or secondary CDR device connects over a TLS link. Else, the primary or secondary CDR device connects over a TCP/IP link.
PMS	Property Management System.
PMS_JOURNAL	The PMS journal printer connects over a TCP/IP link.
PMS_LOG	The PMS log printer connects over a TCP/IP link.
SAT	System administration terminal.
SYS_PRINT	The system printer connects over a TCP/IP link.

TLS Encryption

Allows you to configure TLS encryption for CDR1 and CDR2 service types.

Valid Entry	Usage
Y	Enables TLS encryption for CDR1 and CDR2 service types.
N	Disables TLS encryption. Entering n supports backward compatibility where TCP only is used. This is the default value.

IP Services: page 3

Enables reliable protocol for TCP/IP links, and establishes other session-layer parameters.
Available only if CDR1, CDR2, PMS_JOURNAL, or PMS_LOG is administered for the **Service Type**.

Connectivity Timer

Valid Entry	Usage
1 to 255	The amount of time in seconds that the link can be idle before Communication Manager sends a connectivity message to ensure the link is still up.

Packet Resp Timer

Valid Entry	Usage
1 to 255	The number of seconds to wait from the time a packet is sent until a response (acknowledgement) is received from the far-end, before trying to resend the packet.

Reliable Protocol

Enables or disables reliable protocol over this link. If enabled, uses reliable protocol if the adjunct on the far end of the link supports it.

Service Type

The service type for establishing parameters.

Valid Entry	Usage
CDR1, CDR2	Connects either the primary or secondary CDR device over a TCP/ IP link.
PMS_JOURNAL	Connects the PMS journal printer over a TCP/IP link.
PMS_LOG	Connects the PMS log printer over a TCP/IP link.

Session Connect Message Cntr

Valid Entry	Usage
1 to 5	The number of times Communication Manager tries to establish a connection with the far-end adjunct.

SPDU Cntr

Valid Entry	Usage
1 to 5	The number of times Communication Manager transmits a unit of protocol data before generating an error.

IP Services: page 4

Creates symbolic name and password pairs for all AE Services servers that are allowed to connect to Communication Manager. Available only if the **Service Type** is AESVCS.

Enabled

Enables or disables the AE Services server.

Password

A password for future access. Accepts 12 to 16 alphanumeric characters and must contain at least one alpha character and one numeric character.

AE Services Server

A valid AE Services Server name. The name must match the AE Services server machine name. Each name must be unique.

Server ID

Valid Entry	Usage
1 to 16	The number assigned to this server.

Status

Valid Entry	Usage
idle	The AE Services server is connected to Communication Manager.
in-use	The AE Services server is disconnected to Communication Manager.
blank	No AE Server is administered.

Chapter 71: ISDN Trunk Group

Assigns an Integrated Services Digital Network (ISDN) trunk group that supports the ISDN and Call-by-Call Service Selection service selection features. The trunk group provides end-to-end digital connectivity and supports a wide range of services including voice and non-voice services, to which users have access by a limited set of CCITT-defined, standard multipurpose interfaces.

The ISDN trunk group can contain ISDN-PRI or ISDN-BRI interfaces. However, it is not possible to use the two types of interfaces in the same trunk groups. The type of interface is chosen when the trunk members are assigned to the trunk group.

When ISDN-PRI interfaces are used on ISDN trunk groups, they can also be used to support the Wideband Switching feature. This is intended to work with the H0 (384 Kbps), H11 (1536 Kbps), H12 (1920 Kbps), and NXDS0 (128 to 1984 Kbps) data services, and to support high-speed video conferencing and data applications.

Example command: `add trunk-group n`, where *n* is the trunk group number.

ISDN Trunk Group: page 1

Auth Code

The **Auth Code** field is available only for incoming or two-way trunk groups if you enable the **Authorization Codes** feature. If you enable the **Auth Code** field, Communication Manager performs an auth code check for the incoming trunk call that is routed over another trunk.

The following table provides the situations when the caller must enter an auth code.

Calling party	Called party	Facility Restriction Level (FRL) check?	Auth Code, required if
Station	Station	No	No
	Trunk	Yes, if Route Pattern accesses the trunk	the FRL of the calling station is less than the FRL of the outgoing route pattern preference
Incoming trunk	Station	No	the Auth Code field is enabled on the incoming trunk group

Table continues...

Calling party	Called party	Facility Restriction Level (FRL) check?	Auth Code, required if
	Trunk	Yes, if Route Pattern accesses the trunk	the FRL of the incoming trunk is less than the FRL of the outgoing route pattern preference or the Auth Code field is enabled on the incoming trunk group

*** Note:**

The **Auth Code** field is unavailable if:

- the **Group Type** field is tandem
- the **Group Type** field is ISDN and the **Service Type** field is tandem.

In these situations, permissions of the caller are transmitted using Traveling Class Mark.

Related links

[Direction](#) on page 436

[Authorization Codes](#) on page 730

Busy Threshold

Valid Entry	Usage
1 to 255	The number of trunks that must be busy to alert attendants to control access to outgoing and two-way trunk groups during periods of high use. When the threshold is reached and the warning lamp for that trunk group lights, the attendant can activate trunk group control: internal callers who dial out using a trunk access code are connected to the attendant. Calls handled by AAR and ARS route patterns go out normally.

Carrier Medium

The type of transport medium interface used for the ISDN trunk group.

Valid Entry	Usage
ATM	The trunk is implemented via the ATM Interface circuit pack. Available only if ATM trunking is enabled for the system.
H.323	The trunk is implemented as an H.323 trunk group.
PRI/BRI	The trunk is implemented as a standard DS1 or BRI interface.

Related links

[Asynch. Transfer Mode \(ATM\) Trunking](#) on page 730

[SBS](#) on page 456

[Local Country Code](#) on page 275

[International Access Code](#) on page 275

[Group Type](#) on page 437

[Supplementary Service Protocol](#) on page 449

CDR Reports

Valid Entry	Usage
y	All outgoing calls on this trunk group generate call detail records. If Record Outgoing Calls Only is disabled for CDR, incoming calls on this trunk group also generate call detail records.
n	Calls over this trunk group do not generate call detail records.
r (ring-intvl)	CDR records are generated for both incoming and outgoing calls. In addition, the following ringing interval CDR records are generated: • Abandoned calls: The system creates a record with a condition code of "H" indicating the time until the call was abandoned. • Answered calls: The system creates a record with a condition code of "G" indicating the interval from start of ring to answer. • Calls to busy stations: The system creates a record with a condition code of "I" indicating a recorded interval of 0.

Related links

[Record Outgoing Calls Only](#) on page 103

Charge Advice

Determines how to accumulate and access charge information about a call. Requires that **CDR Reports** be enabled before changing this field from its default of none. Receiving Advice of Charge during the call affects system performance because of the increased ISDN message activity on the signaling channel, which might reduce the maximum call capacity.

Valid Entry	Usage
none	The system does not to collect Advice of Charge information for this trunk group.
automatic	The public network sends Advice of Charge information automatically.
end-on-request	Communication Manager requests charge information with each call, and you receive only the final call charge.
during-on-request	Communication Manager requests charge information with each call, and charges display during and at the end of a call.

COR

Valid Entry	Usage
0 to 995	The Class of Restriction (COR) for the trunk group. Classes of restriction control access to trunk groups, including trunk-to-trunk transfers. Decisions regarding the use of Class of Restriction (COR) and Facility Restriction Levels (FRLs) should be made with an understanding of their implications for allowing or denying calls when AAR/ARS/WCR route patterns are accessed.

+ Tip:

Remember that FRLs are assigned to classes of restriction. Even if two trunk groups have classes of restriction that allow a connection, different facility restriction levels might prevent operations such as off-net call forwarding or outgoing calls by remote access users.

Dial Access

Controls whether users can route outgoing calls through an outgoing or two-way trunk group by dialing its trunk access code. Allowing dial access does not interfere with the operation of AAR/ARS.

! Security alert:

Calls dialed with a trunk access code over WATS trunks bypass AAR/ARS and are not restricted by facility restriction levels. For security, leave this field disabled unless dial access is needed to test the trunk group.

Valid Entry	Usage
y	Users can gain access to the trunk group by dialing its access code.
n	Users cannot gain access to the trunk group by dialing its access code. Attendants can still select this trunk group with a Trunk Group Select button.

Direction

The direction of the traffic on this trunk group.

Available for all trunk groups except DID and CPE.

Valid Entry	Usage
incoming	Traffic on this trunk group is incoming.
outgoing	Traffic on this trunk group is outgoing
two-way	Traffic on this trunk group is incoming and outgoing. Required for Network Call Redirection .

Related links

[Disconnect Supervision-Out](#) on page 447

[Answer Supervision Timeout](#) on page 445

[Receive Answer Supervision](#) on page 558

Far End Test Line No.

The number sent to the far-end's ISDN test line extension. When the **test trunk long** command is issued, this exact number is sent to the far-end to establish a call that tests the integrity of the trunk member under test. The number does not pass through routing or undergo digit manipulation. The digits entered must be what the far-end expects. Accepts up to 15 digits.

Example

For an ISDN tandem trunk, the far-end test number should be a seven-digit ETN (Electronic Tandem Network) number.

Group Name

A unique name that provides information about the trunk group. Accepts up to 27 characters.

This field should contain names that identify the vendor and function of the trunk group rather than the group type (DID, WATS).

* Note:

Supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, 1600-Series, 9600-Series IP Deskphones, 9600-Series H.323 Deskphones, and Avaya J100 Series IP Phones. Unicode is also an option for the 2420J telephone when the **Display Character Set** is katakana. For more information about 2420J, see *2420 Digital Telephone User's Guide*.

Related links

[Display Character Set](#) on page 722

Group Number

The trunk group number.

Group Type

The type of trunk group. The fields that are displayed and available might change according to the trunk group type selected.

Valid Entry	Usage
Access	Used to connect satellite servers to the main switch in Electronic Tandem Networks (ETN). Access trunks do not carry traveling class marks (TCM) and, therefore, allow satellite callers unrestricted access to out-dial trunks on the main server. This entry allows Inband ANI.
APLT	Advanced Private Line Termination (APLT) trunks. Used in private networks. This entry allows Inband ANI.
CAMA	Used to route emergency calls to the local community's Enhanced 911 systems.
CO	Typically used to connect Communication Manager to the local telephone company central office, but can also connect adjuncts such as external paging systems and data modules.
CPE	Used to connect adjuncts, such as paging systems and announcement or music sources, to the server running Communication Manager.
DID	Used to direct callers directly to individuals within an organization without going through an attendant or some other central point. This entry allows Inband ANI.

Table continues...

Valid Entry	Usage
DIOD	Two-way trunks that are used to transmit dialed digits in both directions. In North America, tie trunks are used for applications that require two-way transmission of dialed digits. This entry allows Inband ANI.
DMI-BOS	Digital Multiplexed Interface - Bit-Oriented Signaling (DMI-BOS) trunks allow communication with systems using DMI-BOS protocol. This entry also allows Inband ANI.
FX	A local telephone company central office (CO) trunk that connects the server running Communication Manager directly to a CO outside the local exchange area. Used to reduce long-distance charges if the organization averages a high volume of long-distance calls to a specific area code.
ISDN	Used when digital trunks are needed that can integrate voice, data, and video signals and provide the bandwidth needed for applications such as high-speed data transfer and video conferencing. ISDN trunks can also efficiently combine multiple services on one trunk group. Also used for Network Call Transfer.  Note: Available only if ISDN-PRI, ISDN-BRI Trunks, or both have been enabled for the system. For descriptions of the screens and fields that are unique to ISDN trunks, see “ISDN Trunk Group”.
RLT	Used with Centralized Attendant Service in a private network.
SIP	Used to connect a server running Communication Manager to a Session Manager (SM), or to connect two Communication Manager servers.  Note: The Automatic CallBack, Priority Calling, and Whisper Page features work incorrectly if each of the call's parties is using a SIP endpoint administered on and managed by a different instance of Communication Manager.
Tandem	Used to connect tandem nodes in a private network. This entry allows Inband ANI.
Tie	Used to connect a server running Communication Manager to a local telephone company central office or to another server or switch in a private network. Tie trunks transmit dialed digits with both outgoing and incoming calls. This entry also allows Inband ANI.
WATS	Used to reduce long-distance bills when your organization regularly places many calls to a specific geographical area in North America. With outgoing WATS service, calls can be sent to certain areas (“WATS band”) for a flat monthly charge. Incoming WATS trunks allow toll-free calling to customers and employees.

Related links

[ISDN-PRI](#) on page 734

[ISDN-BRI Trunks](#) on page 734

[SBS](#) on page 456

[Local Country Code](#) on page 275

[International Access Code](#) on page 275

[Supplementary Service Protocol](#) on page 449

[Carrier Medium](#) on page 434

[Path Replacement](#) on page 462

[Call Still Held](#) on page 808

[Call Still Held](#) on page 554

Incoming Calling Number - Format

The TON/NPI encoding applied to CPN information modified by the CLI Prefix feature. This encoding does not apply to calls originating locally.

If this field is blank, Communication Manager passes on the encoding received in the incoming setup message. If the incoming setup message excludes CPN information and digits are added, the outgoing message contains these digits. If a numbering format is not administered in this case, the value defaults to pub-unk. If the numbering format is administered as unknown, the trunk group is modified to unk-unk encoding of the TON/NPI. Therefore, this field also must contain a value other than unknown.

The values for this field map to the Type of Numbering (TON) and Numbering Plan Identifier (NPI) values shown below.

Valid Entry	Type of Numbering (TON)	Numbering Plan Identifier (NPI)
blank	incoming TON unmodified	incoming NPI unmodified
natl-pub	national(2)	E.164(1)
intl-pub	international(1)	E.164(1)
locl-pub	local/subscriber(4)	E.164(1)
pub-unk	unknown(0)	E.164(1)
lev0-pvt	local(4)	Private Numbering Plan - PNP(9)
lev1-pvt	Regional Level 1(2)	Private Numbering Plan - PNP(9)
lev2-pvt	Regional Level 2(1)	Private Numbering Plan - PNP(9)
unk-unk	unknown(0)	unknown(0)

Related links

[Format](#) on page 813

[Format](#) on page 453

Incoming Calling Number Insert

Valid Entry	Usage
0 to 9	The number of digits inserted in the calling party number for all incoming calls on this trunk group. Accepts up to 15 characters.
all	
blank	

Member Assignment Method

Available only if the **Carrier Medium** is H.323 and SIP.

Valid Entry	Usage
manual	Users manually assign trunk members to a signaling group. This is the default.
auto	The system automatically generates members to a specific signaling group.

Related links

[Carrier Medium](#) on page 434

Number of Members

Available only if the **Carrier Medium** is H.323 or SIP and the **Member Assignment Method** is auto.

Valid Entry	Usage
0 to 255	The number of virtual trunk members automatically assigned to the signaling group. Default is 0.

Related links

[Carrier Medium](#) on page 434

[Member Assignment Method](#) on page 440

[Signaling Group](#) on page 442

Outgoing Display

With Outgoing Display, telephones can show the name and number of the trunk group used for an outgoing call before the call is connected.

Valid Entry	Usage
y	Displays the trunk group name and number.
n	Displays the digits the caller dials.

Queue Length

Available only for outgoing or two-way trunk groups.

Valid Entry	Usage
1 to 100	The number of outgoing calls that can wait in queue when all trunks in a trunk group are busy. Calls wait in queue in the order in which they were made. If a queue is administered, a caller hears a confirmation tone when no trunk is available for the outgoing call. The caller can then disconnect and wait; when a trunk becomes available, Communication Manager calls the extension that placed the original call. Communication Manager remembers the number the caller dialed and automatically completes the call.
0	Callers receive a busy signal when no trunks are available. Use for DCS trunks.

Related links

[Direction](#) on page 436

Service Type

The service for which this trunk group is dedicated. In addition to the predefined services or features listed as valid entries, any previously administered user-defined Network Facility **Facility Type** of 0 (feature) or 1 (service) is allowed.

Valid Entry	Usage
access	A tie trunk giving access to an Electronic Tandem Network.
accunet	ACCUNET Switched Digital Service — part of ACI (AT&T Communications ISDN) phase 2.
cbc	Call-by-Call service — provides different dial plans for different services on an ISDN trunk group. Indicates this trunk group is used by the Call-By-Call Service Selection feature.
dmi-mos	Digital multiplexed interface — message-oriented signaling.
i800	International 800 Service — A subscriber can receive international calls without a charge to the call originating party.
inwats	INWATS — provides OUTWATS-like pricing and service for incoming calls.
lds	Long-Distance Service — part of ACI (AT&T Communications ISDN) phase 2.
megacom	MEGACOM Service — an AT&T communications service that provides unbanded long-distance services using special access (switch to 4ESS switch) from an AT&T communications node.
mega800	MEGACOM 800 Service — an AT&T communications service that provides unbanded 800 service using special access (4ESS switch to switch) from an AT&T communications node.
multiquest	AT&T MULTIQUEST Telecommunications Service — dial 700 service. A terminating-user's service that supports interactive voice service between callers at switched-access locations and service provides directly connected to the AT&T Switched Network (ASN).
operator	Network Operator — provides access to the network operator.

Table continues...

Valid Entry	Usage
outwats-bnd	OUTWATS Band — WATS is a voice-grade service providing both voice and low speed data transmission capabilities from the user location to defined service areas referred to as bands; the widest band is 5.
public-ntwrk	Public network calls — It is the equivalent of CO (outgoing), DID, or DIOD trunk groups. If Service Type is public-ntwrk, Dial Access can be enabled.
sddn	Software Defined Data Network — provides a virtual private line connectivity via the AT&T switched network (4ESS switches). Services include voice, data, and video applications. These services complement the SDN service. Do not use for DCS with Rerouting.
sdn	Software Defined Network (SDN) — an AT&T communications offering that provides a virtual private network using the public switched network. SDN can carry voice and data between customer locations as well as off-net locations.
sub-operator	Presubscribed Common Carrier Operator — provides access to the presubscribed common carrier operator.
tandem	Tandem tie trunks integral to an ET.
tie	Tie trunks — general purpose.
wats-max-bnd	Maximum Banded Wats — a WATS-like offering for which a user's calls are billed at the highest WATS band subscribed to by users.

Related links

[Facility Type](#) on page 540

Signaling Group

Available only if the **Carrier Medium** is H.323 or SIP, and the **Member Assignment Method** is auto.

Valid Entry	Usage
digits or blank	Assigned H.323 signaling group number. For the maximum allowed entry, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .

Related links

[Carrier Medium](#) on page 434

[Member Assignment Method](#) on page 440

TAC

The trunk access code (TAC) that must be dialed to access the trunk group. A different TAC must be assigned to each trunk group. CDR reports use the TAC to identify each trunk group. The characters “*” and “#” can be used as the first character in a TAC. Accepts a one- to four-digit number.

TestCall BCC

The Bearer Capability Code (BCC) used for the ISDN test call.

Valid Entry	Usage
0	Voice
1	Mode 1
2	Mode 2 Asynchronous
4	Mode 0

Testcall ITC

Controls the encoding of the Information Transfer Capability (ITC) codepoint of the bearer capability Information Element (IE) in the `SETUP` message when generating an ISDN test call.

 Note:

The ISDN Testcall feature has no routing, so a testcall is never blocked due to an incompatible ITC.

Valid Entry	Usage
rest	Restricted
unre	Unrestricted

Testcall Service

The call-by-call selection for an ISDN test call. Available only if the **Service Type** is cbc. Unavailable for **Facility Type** 0 (feature), 1 (service), or 3 (outgoing) that is defined by users.

Valid Entry	Usage
access	A tie trunk giving access to an Electronic Tandem Network.
accunet	ACCUNET Switched Digital Service — part of ACI (AT&T Communications ISDN) phase 2.
cbc	Call-by-Call service — provides different dial plans for different services on an ISDN trunk group. Indicates this trunk group is used by the Call-By-Call Service Selection feature.
dmi-mos	Digital multiplexed interface — message-oriented signaling.
i800	International 800 Service — allows a subscriber to receive international calls without a charge to the call originating party.
inwats	INWATS — provides OUTWATS-like pricing and service for incoming calls.
lds	Long-Distance Service — part of ACI (AT&T Communications ISDN) phase 2.
megacom	MEGACOM Service — an AT&T communications service that provides unbanded long-distance services using special access (switch to 4ESS switch) from an AT&T communications node.
mega800	MEGACOM 800 Service — an AT&T communications service that provides unbanded 800 service using special access (4ESS switch to switch) from an AT&T communications node.

Table continues...

Valid Entry	Usage
multiquest	AT&T MULTIQUEST Telecommunications Service — dial 700 service. A terminating-user's service that supports interactive voice service between callers at switched-access locations and service provides directly connected to the AT&T Switched Network (ASN).
operator	Network Operator — provides access to the network operator.
outwats-bnd	OUTWATS Band — WATS is a voice-grade service providing both voice and low speed data transmission capabilities from the user location to defined service areas referred to as bands; the widest band is 5.
public-ntwrk	Public network calls — It is the equivalent of CO (outgoing), DID, or DIOD trunk groups. If Service Type is public-ntwrk, Dial Access can be enabled.
sdn	Software Defined Network (SDN) — an AT&T communications offering that provides a virtual private network using the public switched network. SDN can carry voice and data between customer locations as well as off-net locations.
sub-operator	Presubscribed Common Carrier Operator — provides access to the presubscribed common carrier operator.
tandem	Tandem tie trunks integral to an ET.
tie	Tie trunks — general purpose.
wats-max-bnd	Maximum Banded Wats — a WATS-like offering for which a user's calls are billed at the highest WATS band subscribed to by users.

Related links

[Service Type](#) on page 441

[Facility Type](#) on page 540

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Usage Alloc

If enabled, allocates service provided by the trunk group and enhances Network Call Redirection. Available only if the ISDN Trunk Group **Service Type** is cbc.

Related links

[Service Type](#) on page 441

ISDN Trunk Group: page 2

Administer Timers

Enables or disables administration of timers on this trunk group. The default for the ISDN trunk group type is disabled. All other trunk group types are enabled by default.

Available for all trunk group types *except* cpe, h.323, and sip.

Related links

[Group Type](#) on page 437

Answer Supervision Timeout

Valid Entry	Usage
0 to 250	The number of seconds Communication Manager waits before it acts as though answer supervision has been received from the far-end. During a cut-through operation, timing begins after each outgoing digit is sent and timing ceases after the far-end sends answer supervision. On senderized operation, the timer begins after the last digit collected is sent.

* Note:

This field's setting does not override answer supervision sent from the network.

Related links

[Receive Answer Supervision](#) on page 558

[Administer Timers](#) on page 445

Bit Rate

Valid Entry	Usage
300 1200 2400 4600 9600 19200	The baud rate used by pooled modems. The speed of the fastest modem that uses this ISDN trunk group.

Charge Advice

Determines how to accumulate and access charge information about a call. Requires that **CDR Reports** be enabled before changing this field from its default of none. Receiving Advice of Charge during the call affects system performance because of the increased ISDN message activity on the signaling channel, which might reduce the maximum call capacity.

Valid Entry	Usage
none	The system does not to collect Advice of Charge information for this trunk group.
automatic	The public network sends Advice of Charge information automatically.
end-on-request	Communication Manager requests charge information with each call, and you receive only the final call charge.
during-on-request	Communication Manager requests charge information with each call, and charges display during and at the end of a call.

Codeset to Send Display

Defines the codeset for sending the information element for display. The value depends on the type of server or switch used for the connection.

Valid Entry	Usage
0	CCITT (non-Communication Manager equipment).
6	Any other than CCITT or System 85 R2V4, 4E11.
7	System 85 R2V4, 4E11.

Codeset to Send National IEs

The codeset for sending the information element (IE) for national IEs. National IEs include all IEs previously sent only in code set 6 (such as DCS IE). Now these national IEs, including Traveling Class Marks (TCMs) and Lookahead Interflow (LAI), can be sent in code set 6 or 7. The value depends on the type of server/switch to which the user is connected.

Valid Entry	Usage
6	Other types.
7	System 85 R2V4, 4E11, or newer Avaya S8XXX Server types.

* Note:

A Traveling Class Mark (that is, the user's FRL or the user's trunk group FRL) is passed between tandem nodes in an ETN in the setup message only when the **Service Type** is tandem. It then is used by the distant tandem switch to permit access to facilities consistent with the originating user's privileges.

CONNECT Reliable When Call Leaves ISDN

Available only if **Group Type** is ISDN. The value tells the Communication Manager server whether a CONNECT received on an outgoing call that is not end-to-end ISDN is a reliable indication that the far end has answered the call.

Valid entry	Usage
y	The CONNECT is a reliable answer.

Table continues...

Valid entry	Usage
n	The CONNECT message is an unreliable answer. The message can be the result of a timer expiring. If the call was originated by a Call Center adjunct, a Call Classifier can be used instead to determine whether the call has been answered. The default value is n.

Digit Handling (in/out)

Valid Entry	Usage
enbloc/enbloc enbloc/overlap overlap/enbloc overlap/overlap	Defines whether overlap receiving and overlap sending features are enabled. enbloc disables overlap receiving and overlap sending. The first field value indicates digit receiving and the second value indicates digit sending.

Digital Loss Group

Valid Entry	Usage
1 to 19	Determines which administered two-party row in the loss plan applies to this trunk group if the call is carried over a digital signaling port in the trunk group. If values other than 18 or between 11 and 15 are administered, the system displays a warning message indicating that the loss group may not be appropriate for this trunk group.

Disconnect Supervision-Out

Indicates whether Communication Manager receives disconnect supervision for outgoing calls over this trunk group. Available for outgoing or two-way trunk groups.

Valid Entry	Usage
y	Allows trunk-to-trunk transfers involving trunks in this group. The far-end sends a release signal when the called party releases an outgoing call, and the far-end is responsible for releasing the trunk. Enhances Network Call Redirection. Available only if Answer Supervision Timeout is 0 and Receive Answer Supervision is enabled.
n	The far-end server or switch does not provide a release signal, the hardware cannot recognize a release signal, or timers are preferred for disconnect supervision on outgoing calls. Prevents trunk-to-trunk transfers involving trunks in this group.

Caution:

Verify that the far-end server or switch provides answer supervision and disconnect supervision. Most public networks do not provide disconnect supervision over analog trunks. Check with the network services provider.

Related links

[Direction](#) on page 436

[Answer Supervision Timeout](#) on page 445

[Receive Answer Supervision](#) on page 558

Duplex

* Note:

Even if the trunk group supports full-duplex transmission, other equipment in a circuit might not.

Valid Entry	Usage
full	Provides simultaneous two-way transmission, which is most efficient. Recommended in most cases.
half	Supports only one transmission direction at a time.

Group Type

Displays the type of trunk group.

Related links

[Group Type](#) on page 437

[ISDN-BRI Trunks](#) on page 734

[ISDN-PRI](#) on page 734

Max Message Size to Send

Valid Entry	Usage
128 244 256 260	The maximum number of bytes of an ISDN message for Communication Manager.

The following table indicates the expected ISDN-PRI message size from several Lucent Technologies and Avaya LLC products.

Products	Message Length (octets) Received
4ESS (4E11)	256
4ESS (4E13)	256
4ESS (4E14)	256
5ESS (5E4)	244
5ESS (5E5)	244
5ESS (5E6)	244
System 75 (all)	260

Table continues...

Products	Message Length (octets) Received
System 85 (R2V4)	128
System 85 (R2V5)	260
System 85 (R2V6)	260

Supplementary Service Protocol

The supplementary service protocol to use for services over this trunk group. Supplementary service protocols are mutually exclusive.

Valid Entry	Usage
a	National.
b	ISO/ETSI QSIG Private Network. Also used for SBS signaling trunks.
c	ETSI public network.
d	European Computer Manufacturer's Association (ECMA) QSIG private network (supports only Name Identification and Additional Network Feature Transit Counter (ANF-TC)).
e	DCS with Rerouting. Do not use with Service Type of dmi-mos or sddn.
f	ISDN Feature Plus Public network feature plus signaling.
g	ANSI.

Related links

[SBS](#) on page 456

[Local Country Code](#) on page 275

[International Access Code](#) on page 275

[Group Type](#) on page 437

[Carrier Medium](#) on page 434

Synchronization

Caution:

Do not change this field without the assistance of Avaya or the network service provider.

Valid Entry	Usage
async	Determines whether the trunk group uses synchronous or asynchronous communications.
sync	

Trunk Hunt

Defines the trunk hunt search order. Communication Manager performs a trunk hunt when searching for available channels within a facility in an ISDN trunk group. The search can be administered per ISDN-PRI trunk group, but it infers the direction of search within all ISDN-PRI facilities (or portions of those facilities) administered within the trunk group.

Valid Entry	Usage
ascend	Enables a linear trunk hunt search from the lowest to highest numbered channels. All trunks within an ISDN trunk group are selected without regard to the order in which trunks are administered within the trunk group.
cyclical	Enables a circular trunk hunt based on the sequence the trunks were administered within the trunk group. When using ISDN-BRI interfaces, only cyclical is supported. The cyclical option cannot be set if the trunk group using ISDN-PRI interfaces is to be used for Wideband operations.
descend	Enables a linear trunk hunt search from the highest to lowest numbered channels. All trunks within an ISDN trunk group are selected without regard to the order in which trunks are administered within the trunk group.

Related links

[Wideband Support](#) on page 461

Trunk Type

The type of trunk.

Related links

[Trunk Type \(in/out\)](#) on page 804

ISDN Trunk Group: page 3

ACA Assignment

Indicates whether Automatic Circuit Assurance (ACA) measurements are taken for this trunk group.

Apply Local Ringback

Enables or disables local ringback tone to the caller. If enabled, local ringback is removed when the call is connected. Available only if the **Carrier Medium** is PRI_BRI.

Related links

[Carrier Medium](#) on page 434

BSR Reply-best DISC Cause Value

Servers running Communication Manager that are polled as resources in a Best Service Routing application return data to the polling server in the ISDN DISC message. Since some cause values do not work over some networks, this field sets the cause value that the server returns in response to a BSR status poll. If this field is set incorrectly, incoming status poll calls over this trunk group are dropped before any data is returned to the polling server or switch.

Available only if **UUI IE Treatment** is shared.

Valid Entry	Usage
31	Normal-unspecified. This value is almost always used.
17	User-busy
16	Normal-call-clearing.

 Caution:

In most cases, this field is set to the appropriate value during installation. Do not change this field without the assistance of Avaya or your network service provider.

Related links

[UUI IE Treatment](#) on page 460

Charge Conversion

Available only for outgoing or two-way CO, DIOD, FX, and WATS trunk groups. Also available for isdn trunk groups, if the **Charge Advice** field is set to a value other than none.

Valid Entry	Usage
1 to 64 500	Communication Manager multiplies the number of charge units by the value of this field and displays it as a currency amount. Without a value in this field, Communication Manager displays the number of charge units without converting it to currency.

Related links

[Direction](#) on page 436

[Trunk Direction](#) on page 558

Currency Symbol

The telephone displays the symbol before the charge amount. Accepts from one to three characters. Leading and embedded spaces count as characters.

Available only for outgoing or two-way CO, DIOD, FX, and WATS trunk groups. Also available for ISDN trunk groups, if the Charge Advice field is set to a value other than none.

Related links

[Direction](#) on page 436

[Trunk Direction](#) on page 558

Data Restriction

If y, whisper page is denied on this trunk. Enter y to prevent features from generating tones on a data call that would cause erroneous data transmission.

Decimal Point

The appropriate representation for a decimal point as telephone displays it. Available only with outgoing or two-way CO, DIOD, FX, and WATS trunk groups. Also available for isdn trunk groups, if the **Charge Advice** field is set to a value other than none.

 Note:

If the received charge contains no decimals, no decimal point is displayed (that is, the administered decimal point is ignored for charge information received with no decimals). On a QSIG trunk group, unlike other trunk groups, the **Decimal Point** field is independent whether the calling display shows a decimal point. Instead, it tells what symbol should be displayed if the QSIG AOC received has a 1/10 or 1/100 or 1/1000 Multiplier.

Valid Entry	Usage
comma	If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a comma as the decimal point. Divides the charge value by 100.
period	This is the default. If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a period as the decimal point. Divides the charge value by 100.
none	No decimal point is displayed.

Related links

[Direction](#) on page 436

[Charge Advice](#) on page 435

DS1 Echo Cancellation

Enables or disables echo cancellation on a per port basis. If enabled, reduces voice call echo.

 Note:

Changes to the DS1 Echo Cancellation field is incomplete until one of the following occurs:

- Port is busied-out or released.
- Trunk group is busied-out or released.
- SAT command test trunk group is performed.
- Periodic maintenance runs.

DSN Term

Enables or disables the trunk group as a DSN termination telephone. The default is disabled.

Format

Specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number. If you have used Special Application SA9118 to generate international numbers in a Service Pack for Communication Manager Releases 5.2.1 or 6.0.1, you must set the **Format** field to intl-pub for the appropriate ISDN trunk group(s) after upgrading a Communication Manager server to Release 6.2 or later, because there is no upgrade path for intl-pub entry in Communication Manager 6.2.

The **Format** field is available only if the **Send Calling Number** or **Send Connected Number** field is enabled or restricted.

 Note:

The entries private and unk-pvt are overridden, and a public number is generated, if the called number for the outgoing call over the trunk group is encoded using E.164. The numbering plan for the called number is determined by the **Call Type** field from the ARS or AAR Analysis, or the **Numbering Format** field from the outgoing Route Pattern.

Valid Entry	Usage
intl-pub	Indicates the Numbering Plan Indicator is E.164. International (public number that includes city or area code and the country code).
natl-pub	Indicates the Numbering Plan Indicator is E.164. National (public number that includes city or area code, but excludes the country code).
private	Indicates the Numbering Plan Indicator is Private. Based on the network level of the ISDN Parameters field on the Feature-related System Parameters screen.
pub-unk	Indicates the Numbering Plan Indicator is E.164. Unknown (public number with no details of the digits included).
unk-pvt	Indicates the Numbering Plan Indicator is Private. Unknown (private number with no details about the digits included).

Related links

[Send Calling Number](#) on page 457

[Send Connected Number](#) on page 458

[Numbering-Private Format](#) on page 543

Maintenance Tests

Enables or disables hourly maintenance tests on this trunk group.

Available only for aplt, isdn, sip, or tie trunk groups.

Related links

[Group Type](#) on page 437

Maximum Size of UI IE Contents

Available only if the **UI IE Treatment** is shared.

Valid Entry	Usage
32 to 128	The maximum number of bytes of user information that the network supports.

Related links

[UUI IE Treatment](#) on page 460

Measured

Indicates if the system transmits data for this trunk group to the Call Management System.

Valid Entry	Usage
internal	Sends the data to the Basic Call Management System (BCMS), the VuStats data display, or both. Available only if BCMS (Basic) or VuStats is enabled for the system.
external	Sends the data to the CMS.
both	Collects data internally and sends it to the CMS. Available only if BCMS (Basic) or VuStats is enabled for the system.
none	Trunk group measurement reports are not required.

Related links

[VuStats](#) on page 743

[BCMS \(Basic\)](#) on page 739

Modify Tandem Calling Number

Available with outgoing or two-way trunks when the **Carrier Medium** field is set to PRI/BRI or H.323, and the **Send Calling Number** field is set to enabled or restricted.

Valid Entry	Usage
natl-intl-prefix	Adds the national or international prefixes from the Locations Parameters screen when the calling party number is appropriate.
tandem-cpn-form	Modifies the calling party number IE in the previously administered format specified for the Tandem Calling Party Number.
no	Does not modify the calling party number.

Related links

[Group Type](#) on page 437

[Direction](#) on page 436

[Carrier Medium](#) on page 434

[Send Calling Number](#) on page 457

[Modify Tandem Calling Number](#) on page 829

NCA-TSC Trunk Member

The trunk member number whose D-channel is used to route tandem NCA-TSCs or QSIG CISCs.

Network Call Redirection

Whenever the **Supplementary Service Protocol** is changed, this field resets to none to prevent an inadvertent incorrect value.

Available only if **ISDN Network Call Redirection** is enabled for the system and the **Supplementary Service Protocol** is a, c, or g.

Valid entry	Usage
deflect	Allows Network Call Deflection.
ANSI-transfer	Allows Network Call Transfer for MCI DEX 600 ISDN trunks.
Nortel-transfer	Allows Network Call Transfer for MCI DMS 250 switches.
telcordia-tbct	Allows Network Call Transfer for Lucent 5ESS or Nortel DMS100 switches.

Related links

[ISDN-PRI](#) on page 734

[ISDN/SIP Network Call Redirection](#) on page 735

[Supplementary Service Protocol](#) on page 449

Network (Japan) Needs Connect Before Disconnect

Sends an ISDN Connect message just prior to the disconnect message.

Outgoing Channel ID Encoding

Determines whether to encode the Channel ID IE as preferred or exclusive. Available only if the **Group Type** is isdn and the **Service Type** is anything except dmi-mos or sddn. Blank is an invalid entry.

Valid Entry	Usage
preferred	Default if Used for DCS is disabled or unavailable.
exclusive	Default if Used for DCS is enabled.

Related links

[Group Type](#) on page 437

[Service Type](#) on page 441

Path Replacement Method

Available only if **Basic Call Setup** and **Supplementary Services with Rerouting** are enabled for the system and when the **Supplementary Service Protocol** is either b or e and the **Group Type** is isdn.

Valid Entry	Usage
better-route	Uses the most economical route; for example, the reconfigured call does not use the same trunk group as the original call.
always	Always reconfigures the call regardless of the trunk group used.

Related links

[ISDN-PRI](#) on page 734

[ISDN-BRI Trunks](#) on page 734

[Basic Call Setup](#) on page 743

[Supplementary Services with Rerouting](#) on page 744

[Service Type](#) on page 441

[Group Type](#) on page 437

Replace Restricted Numbers

The Replace Restricted Numbers field indicates whether to replace restricted numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. If the **Replace Restricted Numbers** field is enabled, Communication Manager replaces the display irrespective of the service type of the trunks such as BRI, PRI, H.323, and SIP trunks. The **Replace Restricted Numbers** field is available if the **Group Type** is **ISDN** or **SIP**.

Related links

[Group Type](#) on page 437

Replace Unavailable Numbers

If enabled, replaces unavailable numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. The display is replaced regardless of the service type of the trunk. Applies to BRI/PRI, H.323, and SIP trunks. Also applies to analog trunks if **Analog Trunk Incoming Call ID** is enabled and **Receive Analog Incoming Call ID** is set to any value except disabled. Available only if the **Group Type** is isdn or sip.

Related links

[Group Type](#) on page 437

[Analog Trunk Incoming Call ID](#) on page 728

[Receive Analog Incoming Call ID](#) on page 832

SBS

Enables or disables Separation of Bearer and Signaling (SBS) for the trunk group.

Available only if the **Local Country Code** and **International Access Code** are administered for the system and when the **Supplementary Service Protocol** is b, the **Group Type** is isdn, the **Carrier Medium** is H.323, and **Dial Access** is disabled.

Related links

[Local Country Code](#) on page 275

[International Access Code](#) on page 275

[Group Type](#) on page 437

[Supplementary Service Protocol](#) on page 449

[Carrier Medium](#) on page 434

Send Called/Busy/Connected Number

Specifies if the dialed number, whether called (ringing), busy (busy tone), or connected (answered) is sent on incoming or tandemed ISDN calls.

Available only if **QSIG Value-Added** is enabled for the trunk group.

Valid Entry	Usage
y	The dialed number is sent on incoming or tandemed ISDN calls. This field must be enabled in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation. If enabled, the Numbering - Public/Unknown Format is accessed to construct the actual number sent, or the Numbering - Private Format is used.
n	Disables the sending of the dialed number on incoming or tandemed ISDN calls.
r	Restricted. The connected number is sent "presentation restricted".

Related links

[Numbering — Public/Unknown Format](#) on page 546

[QSIG Value-Added](#) on page 463

[Numbering-Private Format](#) on page 543

Send Calling Number

Specifies whether the calling party's number is sent on outgoing or tandemed ISDN calls.

* Note:

The Numbering - Public/Unknown Format can override the Send Calling Number administration

Valid Entry	Usage
y	The calling party's number is sent on outgoing or tandemed ISDN calls. If enabled, the Numbering - Public/Unknown Format is accessed to construct the actual number sent, or the Numbering - Private Format is used.
n	Disables the sending of the calling party's number on outgoing or tandemed ISDN calls. If disabled, an incoming number is not tandemed out again. This applies to all Supplementary Service Protocols.
r	Restricted. The calling number is sent "presentation restricted". If set to restricted, an incoming number is marked restricted when it is tandemed out again. This applies to all Supplementary Service Protocols.

Related links

[Numbering-Private Format](#) on page 543

[Incoming Number Format](#) on page 93

[Numbering — Public/Unknown Format](#) on page 546

Send Codeset 6/7 LAI IE

If enabled, the ISDN trunk transmits information in Codeset 6/7.

If the **UUI IE Treatment** is shared, then this field should be disabled. Otherwise, the same information will be sent twice and might exceed the message size.

Related links

[UUI IE Treatment](#) on page 460

Send Connected Number

Specifies if the connected party's number is sent on incoming or tandemed ISDN calls.

Available only if **QSIG Value-Added** is disabled for the trunk group.

Valid Entry	Usage
y	The connected party's number is sent on outgoing or tandemed ISDN calls. If enabled, the Numbering - Public/Unknown Format is accessed to construct the actual number sent, or the Numbering - Private Format is used. This field must be enabled for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.
n	Disables the sending of the connected party's number on outgoing or tandemed ISDN calls. If disabled, an incoming number is not tandemed out again. This applies to all Supplementary Service Protocols.
r	Restricted. The connected number is sent "presentation restricted". If this field is set to r, an incoming number is marked restricted when it is tandemed out again. This applies to all Supplementary Service Protocols.

 Note:

The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/ protocol-version combination for the media module, you should not administer the Send Connected Number as restricted, as this causes display problems. The Numbering - Public/Unknown Format overrides the Send Connected Number administration for any administrable block of extensions.

Related links

[Numbering-Private Format](#) on page 543

[QSIG Value-Added](#) on page 463

[Numbering — Public/Unknown Format](#) on page 546

Send Name

Specifies whether the calling/connected/called/busy party's administered name, or the name on a redirected call, is sent to the network on outgoing/incoming calls.

 Note:

If name information is not administered for the calling station or the connected/called/busy station, the system sends the extension number in place of the name.

Valid Entry	Usage
y	The calling/connected/called/busy party's administered name and the name on a redirected call are sent on outgoing/incoming calls. An entry of y or n is required if the Supplementary Service Protocol is e (DCS with Rerouting).
n	Disables the sending of the calling/connected/called/busy party's administered name and the name on a redirected call on outgoing/incoming calls. An entry of y or n is required if the Supplementary Service Protocol is e (DCS with Rerouting). With an entry of n, an incoming name is not tandemed out again if the Supplementary Service Protocol field is any value other than b (QSIG).
r	Restricted. The calling/connected/called/busy party's administered name is sent "presentation restricted" on outgoing/incoming calls. The name on a redirected call is not sent. This value is valid only if the Supplementary Service Protocol is a (national supplementary service), b (for called/ busy only), or d (for the QSIG Global Networking Supplementary Service Protocol). With an entry of r, an incoming name is marked restricted when it is tandemed out again. However, if the Supplementary Service Protocol is b (QSIG), then an incoming name is passed on unchanged and Send Name value is ignored.

Related links

[Supplementary Service Protocol](#) on page 449

Send UCID

If you enable the **Send UCID** field, the trunk transmits Universal Call IDs (UCIDs). This field is available only if the **UUU IE Treatment** field is set to *Shared*.

 Note:

If the incoming trunk has two UCIDs in the UUI, the outgoing trunk sends the UUI with both UCIDs to the far-end switch.

Send UUI IE

If enabled, sends UUI information on a per trunk group basis.

Show ANSWERED BY on Display

Available only for isdn pri/bri, isdn h.323, and sip trunk groups.

Valid Entry	Usage
y	The words “ANSWERED BY” display in addition to the connected telephone number on calls over this trunk. This is the default.  Note: Based on display language settings for stations, “ANSWERED BY” is translated into and displayed in the appropriate language.
n	Only the connected telephone number displays. This might be preferred when outgoing calls are over a trunk that might be redirected.

Suppress # Outpulsing

Indicates whether or not to suppress the final “#” in cases where the system would normally outpulse it. Used if end-to-end signaling begins with (and includes) “#”. This field should be enabled when the local telephone company central office or any other facility treats “#” as an error.

US NI Delayed Calling Name Update

For digital display stations, if calling name information is received after the incoming call has been delivered to the terminating telephone, enables or disables a display update.

For analog display stations, Communication Manager is enhanced to wait for the FACILITY message (that contains the name information) before delivering an incoming call to the receiver. You can administer the time duration for which Communication Manager waits for the FACILITY message.

 Note:

BRI trunks do not support display updates.

Available only if the **Carrier Medium** is either PRI/ BRI or ATM, and the **Supplementary Service Protocol** is a.

Related links

[Supplementary Service Protocol](#) on page 449

[Carrier Medium](#) on page 434

[ISDN-PRI](#) on page 734

UI IE Treatment

Specifies whether the user Information Element (IE) is shared.

Valid Entry	Usage
shared	The trunk is connected to the server.
service-provider	Service provider functionality is required.

Wideband Support

*** Note:**

This feature is not supported on the DS1 interfaces on branch gateways (G700).

Enables or disables wideband switching on this trunk group.

Available only if **Wideband Switching** is enabled for the system.

*** Note:**

Wideband trunk calls are treated as a single trunk call when Automatic Circuit Assurance (ACA) measurements are taken. This way, if an ACA referral call is generated (for short or long holding time), the wideband call only triggers a single referral call using the lowest B-channel trunk member associated with the wideband channel.

Related links

[Wideband Switching](#) on page 738

QSIG Trunk Group Options

The fields on this screen appear only when **Group Type** is isdn and **Supplementary Service Protocol** is b.

Related links

[Group Type](#) on page 437

[Supplementary Service Protocol](#) on page 449

[Trunk Group](#) on page 787

Character Set for QSIG Name

Use this field to configure the character set for transmission and display of the QSIG name data. This field is available only if:

- **Group Type** is isdn.
- **Supplementary Service Protocol** is b.
- **Display Character Set** is Roman.

Valid Entry	Usage
eurofont	The Roman Eurofont character set. Eurofont is the default setting. * Note: Avaya BRI stations support only ASCII characters. A BRI station does not correctly display non-ASCII characters, such as Eurofont or Kanafont.

Table continues...

Valid Entry	Usage
iso-8859-1	The system converts all data or characters transmitted over a QSIG trunk from Eurofont to ISO 8859-1. Eurofont is an Avaya proprietary encoding character set.  Note: ISO 8859-1, also known as ISO/IEC 8859-1 or Latin-1, is part 1 of ISO/IEC 8859, a standard character encoding defined by ISO. ISO 8859-1 encodes Latin alphabet no. 1, which consists of 191 characters from the Latin script. Each character is encoded as a single 8-bit code value.

Related links

[Group Type](#) on page 437

[Supplementary Service Protocol](#) on page 449

[Display Character Set](#) on page 722

Diversion by Reroute

Use this field to activate or deactivate the Diversion by Reroute feature. The default value of this field is y.

This field is available only if:

- **Group Type** is ISDN.
- **Supplementary Service Protocol** is b.

If the value of this field is y, Communication Manager selects the optimum path and routes the call on that path to the destination number.

If the value of this field is n, Communication Manager does not initiate a Diversion or Reroute request over the QSIG trunk. Communication Manager also rejects any Diversion or Reroute request that Communication Manager receives over the QSIG trunk.

Related links

[Supplementary Service Protocol](#) on page 449

[Group Type](#) on page 437

Path Replacement

Enables or disables the Path Replacement feature. The default is enabled. If disabled, Communication Manager does not originate a Path Replacement request over that trunk group, and rejects any Path Replacement request it receives over that trunk group. Available only if the **Group Type** is isdn and the **Supplementary Service Protocol** is b.

Related links

[Group Type](#) on page 437

[Supplementary Service Protocol](#) on page 449

Path Replacement Method

Available only if the **Group Type** is ISDN, the **Supplementary Service Protocol** is b or e, and **Supplementary Services with Rerouting** or **DCS with Rerouting** is enabled for the system.
Unavailable if **Path Replacement with Retention** is enabled.

Valid Entry	Usage
always	Use any QSIG (SSB) trunk group as the replacement trunk group. A new call is always originated, even when the original trunk group is determined to be the replacement trunk group.
BR (better route)	Route pattern preferences help determine trunk group path replacement. The original trunk group is retained if Path Replacement with Retention is enabled. Path replacement fails (and the original trunk group is retained) if Path Replacement with Retention is disabled.

Related links

[Group Type](#) on page 437
[Supplementary Service Protocol](#) on page 449
[Supplementary Services with Rerouting](#) on page 744
[DCS with Rerouting](#) on page 731
[Path Replacement with Retention](#) on page 463

Path Replacement with Retention

Available only if the **Group Type** is ISDN, the **Supplementary Service Protocol** is b or e, and **Supplementary Services with Rerouting** or **DCS with Rerouting** is enabled for the system.

Valid Entry	Usage
y	Retains the original trunk group.
n	Provides path replacement according to settings for the Path Replacement Method .

Related links

[Supplementary Service Protocol](#) on page 449
[Supplementary Services with Rerouting](#) on page 744
[DCS with Rerouting](#) on page 731
[Group Type](#) on page 437

QSIG Value-Added

Enables or disables QSIG-VALU services. Available only if **Value-Added (VALU)** is enabled for the system and **Supplementary Services Protocol** is enabled for the trunk.

Related links

[Value Added \(VALU\)](#) on page 744
[Supplementary Service Protocol](#) on page 449

QSIG-Value Coverage Encoding

The encoding method used to encode DL1, DL2, and DL3 extensions. Available only if the **Group Type** is isdn, **Supplementary Service Protocol** is b, and **QSIG Value-Added** is enabled for the system.

Valid Entry	Usage
proprietary	Communication Manager sends extension information in the normal manner. This is the default.
standard	In addition to normal extension information, Communication Manager sends the data part (as null) of the extension.

Related links

[Value Added \(VALU\)](#) on page 744

[Group Type](#) on page 437

[Supplementary Service Protocol](#) on page 449

SBS

Enables or disables Separation of Bearer and Signaling (SBS) for the trunk group.

Available only if the **Local Country Code** and **International Access Code** are administered for the system and when the **Supplementary Service Protocol** is b, the **Group Type** is isdn, the **Carrier Medium** is H.323, and **Dial Access** is disabled.

Related links

[Local Country Code](#) on page 275

[International Access Code](#) on page 275

[Group Type](#) on page 437

[Supplementary Service Protocol](#) on page 449

[Carrier Medium](#) on page 434

SIP Reference Trunk Group

The system displays SIP Reference Trunk Group only when the **Group Type** field is isdn, the **Carrier Medium** field is H.323, and the **Supplementary Service Protocol** field is b.

Valid Entry	Usage
1 to 2000	Assigns a number for the SIP trunk group. If trunk members are already assigned to this trunk group, you must not change the value of SIP Reference trunk Group field. If you change the value of this field, the system displays an error message and prompts you to remove all assigned members before enabling Q-SIP.
blank	No SIP trunk group is assigned. By default, the value is blank.

 Note:

For the maximum allowed entry per server type, see *Avaya Aura® Communication Manager System Capacities Table*.

TSC Method for Auto Callback

Controls the signaling connection method for the QSIG Temporary Signaling Connections (TSC) when Communication Manager is the terminating or the outgoing gateway PINX.

Valid Entry	Usage
drop-if-possible	QSIG Temporary Signaling Connections are released.
always-retain	QSIG Temporary Signaling Connections are always retained for successful call completion activation.

ISDN Trunk Group: Administrable Timers

This screen displays only when **Administer Timers** is enabled. This screen does not display ISDN trunk group information for trunks of **Group Type** cpe or sip.

 Note:

If the ISDN trunk group has a **Carrier Medium** value of H.323, or if the trunk group has BRI members, then this page cannot be administered. In these cases, an error message displays when you attempt to submit the screen.

 Caution:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

Related links

[Group Type](#) on page 437

[Administer Timers](#) on page 445

[Carrier Medium](#) on page 434

Programmed Dial Pause (msec)

Valid Entry	Usage
100 to 25500 in increments of 100	The exact duration of the pause used during abbreviated dialing, ARS outpulsing, and terminal dialing operations. This timer is administrable for all outgoing and two-way trunk groups. All central office (CO) media module that accept administrable timers accept this timer.

END TO END SIGNALING

Pause (msec)

Available only if the **Trunk Type** is blank.

Valid Entry	Usage
20 to 2550 in increments of 10	The interval (pause) between DTMF tones sent from a hybrid telephone. All CO, DIOD, and tie media module that accept administrable timers accept this timer.

Related links

[Trunk Type](#) on page 821

Tone (msec)

Available only if the **Trunk Type** is blank.

Valid Entry	Usage
20 to 2550 in increments of 10	The duration of the DTMF tone sent when a button on a hybrid telephone is pressed. All CO, DIOD, and tie media module that accept administrable timers accept this timer.

Related links

[Trunk Type](#) on page 821

Shared UI Feature Priorities

The fields in this page show the priorities for each type of information to be forwarded in the Shared UI. The system displays this page only on the ISDN trunk group screen when all of the following conditions are met:

- The **UI IE Treatment** is shared.
- The **Supplementary Service Protocol** is set to anything except b.

Changing the priorities in this screen might affect whether certain information will be sent.

Related links

[Supplementary Service Protocol](#) on page 449

[UI IE Treatment](#) on page 460

ASAI

User information from Adjunct/Switch Applications Interface (ASAI).

Valid Entry	Usage
1 to 7	Level of priority, with 1 being the highest. Default priority is 1.
blank	This field's information is not forwarded.

Collected Digits

Digits collected from caller (not including dial-ahead digits).

Valid Entry	Usage
1 to 7	Level of priority, with 1 being the highest. Default priority is 5.
blank	This field's information is not forwarded.

Held Call UCID

The unique tag for the last call that was put on hold by the Automatic Call Distribution (ACD) agent placing this call to another system. This Universal Call ID (UCID) can be used to identify the original or parent call that may eventually be placed into conference or transferred to the other system. This element is required for cradle-to-grave tracking with Avaya IQ release 5.0 and later.

Valid Entry	Usage
1 to 7	Level of priority, with 1 being the highest. The UCID included in the element with default priority 2 is the tag for a new call placed by the agent while the original call is on hold. Default priority is 7.
blank	This field's information is not forwarded.

In-VDN Time

Number of seconds the call has spent in vector processing.

Valid Entry	Usage
1 to 7	Level of priority, with 1 being the highest. Default priority is 3.
blank	This field's information is not forwarded.

Other LAI Information

Includes the time stamp of when the call entered the current queue, the call's priority level in its current queue, and the type of interflow.

Valid Entry	Usage
1 to 7	Level of priority, with 1 being the highest. Default priority is 6.
blank	This field's information is not forwarded.

Universal Call ID

A unique tag that identifies the call that this message is being sent for and the other information included in the User-User-Information (UUI).

Valid Entry	Usage
1 to 7	Level of priority, with 1 being the highest. Default priority is 2.
blank	This field's information is not forwarded.

VDN Name

Name of the active VDN (also called LAI DNIS).

Valid Entry	Usage
1 to 7	Level of priority, with 1 being the highest. Default priority is 4.
blank	This field's information is not forwarded.

CBC Service Trunk Group Allocation Plan Assignment Schedule

This screen administers a fixed schedule or a schedule that can change up to six times a day for each day of the week. This screen determines which CBC Service Trunk Group Allocation Plan will be in use at any given time.

Available only if the ISDN trunk group **Service Type** is cbc and the ISDN trunk group **Usage Alloc** are enabled.

Related links

[Service Type](#) on page 441

[Usage Alloc](#) on page 444

Act Time

The time the usage allocation plan will become effective.

Valid Entry	Usage
00:00 through 23:59	The time using the 24-hour clock (military time). There must be at least one entry per day.

Allocation Plan Number

Valid Entry	Usage
1 to 3 blank	The CBC Trunk Allocation Plan that is in effect if a fixed usage method has been selected. Required if the allocation plan is fixed.

Fixed

Indicates whether the allocation plan will be fixed.

Plan

Valid Entry	Usage
1 to 3 blank	The number of the usage allocation plan that will be in effect from the activation time until the activation time of the next scheduled plan change.

Scheduled

Indicates whether or not the allocation plans are implemented according to the administered schedule found on this page. If enabled, then there must be at least one entry in the schedule.

CBC Trunk Group Usage Allocation

This screen sets a minimum and maximum number of members for up to ten different Services/Features for up to three different Usage Allocation Plans (1 to 3). Available only if the **Service Type** is cbc and **Usage Alloc** is enabled.

Related links

[Service Type](#) on page 441

[Usage Alloc](#) on page 444

Max# Chan

Valid Entry	Usage
0 to 99 blank	The maximum number of members of an ISDN trunk group with a Service Type of cbc that a particular service or feature can use at any given time. This field must be administered if a service or feature has been entered in the Incoming Call Handling Treatment Table screen.

Min# Chan

Valid Entry	Usage
0 to 99 blank	The minimum number of members of an ISDN trunk group with a Service Type of cbc that a particular service or feature can use at any given time. The sum of the minimum number of members for all services or features must not exceed the total number of members of the trunk group.

Service/Feature

The ISDN services or features that can be requested at call setup time when using this trunk group. In addition to predefined services or features that can be received on a call by call basis, user-defined service types can be used.

Valid Entry	Usage
0	Feature
1	Service
2	Incoming
3	Outgoing
other	Any services or features that are not explicitly specified

Wideband Support Options

Note:

All B-channels that comprise the wideband call must reside on the same ISDN-PRI facility.

Contiguous

Available only if “N by DS-zero” (NXDS0) multi-rate service is enabled.

Valid Entry	Usage
y	Specifies the “floating” scheme. NXDS0 calls are placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without constraint on the starting channel (no fixed starting point trunk). Unavailable with H0 ISDN information transfer rate.
n	Specifies the “flexible” scheme. NXDS0 calls are placed on any set of B-channels on the same facility as long as the requested bandwidth is satisfied. There are no constraints, such as contiguity of B-channels or fixed starting points

Related links

[NxDS0](#) on page 471

[NXDS0](#) on page 573

H0

If enabled, specifies the ISDN information transfer rate for 384-kbps of data that is comprised of six B-channels. When a trunk group is administered to support H0, the trunk or hunt algorithm to satisfy a call requiring 384-kbps of bandwidth uses a fixed allocation scheme.

H11

If enabled, specifies the ISDN information transfer rate for 1536-kbps of data that is comprised of 24 B-channels. When a trunk group is administered to support H11, the trunk or hunt algorithm to satisfy a call requiring 1536-kbps bandwidth uses a fixed allocation scheme.

H12

If enabled, specifies the ISDN information transfer rate for 1920-kbps of data that is comprised of 30 B-channels. When a trunk group is administered to support H12, the trunk or /hunt algorithm to satisfy a call requiring 1920-kbps bandwidth uses a fixed allocation scheme.

NxDS0

Enables or disables the "N by DS-zero" multi-rate service.

ISDN Trunk Group: Group Member Assignments

 Note:

When supporting DCS, Member Number Assignments must be the same between nodes. For example, Member #1 must be Member #1 at the far-end trunk group.

Administered Members (min/max)

The minimum and maximum member numbers that have been administered for this trunk group.

Code

The type of media module physically installed or logically administered at the location to which this member is assigned. If no media module is installed or administered at the port address, the field is blank.

Name

The name of the trunk group member. The name should identify the trunk unambiguously. Accepts up to 10 characters.

Example

- The telephone number assigned to incoming trunks
- The Trunk Circuit Identification number assigned by the service provider

Night

The night service destination for this trunk group member is different from the night service destination administered for the trunk group. Incoming calls are routed to this destination when the system is placed in night service mode.

Valid Entry	Usage
<i>A valid extension</i>	The extension of the night destination for the trunk.
attd	Calls go to the attendant when night service is active.
blank	Not administered

Related links

[Night Service](#) on page 797

Port

When using ISDN-BRI interfaces, B-channel 1 is the port number while B channel 2 is the port number plus 16. For example, if B channel 1's port number is 01A1002, then B channel 2's port number is 01A1018.

When using ISDN-PRI interfaces, the port number will be the one allied with the B-channel. For example, if the DS1 is located in 01A10, then B channel 1 will be 01A1001, B channel 2 will be 01A1002 and so forth.

 Note:

When administering analog trunks connected to a TIM518, physical ports 17 to 24 are administered as ports 9 to 16 in Communication Manager.

Sfx

The model suffix for the type of media module physically installed at the location to which this member is assigned. If no media module is installed at the port address, the field is blank.

Sig Grp

Valid Entry	Usage
digits	The signaling group of this trunk group member. Required for IP group members. If you administer a port that resides on a DS1 board and that DS1 board belongs to one and only one signaling group, you can leave the Signaling Group blank. The appropriate default signaling group number is inserted by Communication Manager. If a DS1 board is assigned to more than one signaling group, then you must enter a signaling group number. A trunk group can contain members from different signaling groups. For the maximum allowed entry, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .

Related links

[Group Type](#) on page 437

[Signaling Group](#) on page 442

Total Administered Members

The total number of members administered in the trunk group.

Chapter 72: Language translations

The Language Translations screens display messages and their translations. Telephones display pre-translated messages from Communication Manager, which are available in the following languages:

- English
- French
- Italian
- Spanish

You can assign translations in any other language by using the Language Translations screens, but this method is deprecated. Instead, use Avaya Message Editing Tool (AMET) to assign translations for user-defined telephone messages.

If you use an editor other than AMET, the system does not update the Phone Message Files correctly and the files are not installed. For more details, see *Avaya Message Editing Tool (AMET) Job Aid*.

Note:

If you type user-defined in the Display Language field on the Station screen or the Attendant Console screen and messages are not defined for these screens, the system displays a string of asterisks on all display messages.

Warning:

If you use AMET, Communication Manager blocks access to the SAT user-defined translations-related screens. Therefore, do not use the Language Translations screens.

Warning:

If you do not use AMET, do not edit the user-defined translations-related screens through SAT.

You can download AMET from <http://www.avaya.com>.

Automatic Wakeup

Use this field to replace the English text for Automatic Wakeup messages that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related links

[Message 12](#) on page 480

[Message 15, 16, 18](#) on page 480

[Message 4](#) on page 479

[Message 6](#) on page 480

Button labels

Use this field to replace the English text for Button labels that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related links

[Message 12](#) on page 480

[Message 15, 16, 18](#) on page 480

[Message 4](#) on page 479

[Message 6](#) on page 480

Enhanced Abbreviated Dialing

Use this field to administer the Enhanced Abbreviated Dialing messages that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related links

[Message 12](#) on page 480

[Message 15, 16, 18](#) on page 480

[Message 4](#) on page 479

[Message 6](#) on page 480

Leave Word Calling

Use this field to replace the English text for Leave Word Calling messages that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related links

[Message 12](#) on page 480

[Message 15, 16, 18](#) on page 480

[Message 4](#) on page 479

[Message 6](#) on page 480

Malicious Call Trace

Use this field to replace the English text for Malicious Call Trace messages that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related links

[Message 12](#) on page 480

[Message 15, 16, 18](#) on page 480

[Message 4](#) on page 479

[Message 6](#) on page 480

Miscellaneous call identifiers

Use this field to replace the English text for miscellaneous call identifiers that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related links

[Message 12](#) on page 480

[Message 15, 16, 18](#) on page 480

[Message 4](#) on page 479

[Message 6](#) on page 480

Miscellaneous features

Use this field to replace the English text for miscellaneous features that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related links

[Message 12](#) on page 480

[Message 15, 16, 18](#) on page 480

[Message 4](#) on page 479

[Message 6](#) on page 480

Property Management Interface

Use this field to replace the English text for property management interface messages that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related links

[Message 12](#) on page 480

[Message 15, 16, 18](#) on page 480

[Message 4](#) on page 479

[Message 6](#) on page 480

Self Administration

Use this field to replace the English text for self administered messages that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related links

[Message 12](#) on page 480

[Message 15, 16, 18](#) on page 480

[Message 4](#) on page 479

[Message 6](#) on page 480

Softkey Labels

Use this field to replace the English text for softkey labels that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

To provide unique labels for abbreviated dialing button types for softkey labels, Communication Manager replaces the last two characters with digits for the 12-key 8400 and 15-key 8434D telephones.

On this screen, the digits following the **AD** are derived from the button position. If the first button is an AD button, then it is AD1 and the 15th button is AD15. All the AD buttons between 1 and 15 have the position number appended to AD.

Related links

[Message 12](#) on page 480

[Message 15, 16, 18](#) on page 480

[Message 4](#) on page 479

[Message 6](#) on page 480

Transfer Conference

Use this field to replace the English text for transfer conference messages that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related links

[Message 12](#) on page 480

[Message 15, 16, 18](#) on page 480

[Message 4](#) on page 479

[Message 6](#) on page 480

View Buttons

Use this field to replace the English text for buttons that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related links

[Message 12](#) on page 480

[Message 15, 16, 18](#) on page 480

[Message 4](#) on page 479

[Message 6](#) on page 480

Vustats

Use this field to replace the English text for Vustats that the system displays on the telephone screen.

This screen contains an English version of the display, an explanation of the English message, and a translation of the English message. A long message can be shortened on telephones that display fewer than 32 characters.

Related links

[Message 12](#) on page 480

[Message 15, 16, 18](#) on page 480

[Message 4](#) on page 479

[Message 6](#) on page 480

Message 4

The character "^" is a place holder.

English Text	Replacement Info
^-party conference in progress	"^" is replaced with the number of parties currently on the conference call.

Manually change ~ to ^ in any user-defined language. The software does not update manually.

Message 6

The character "^" is a place holder.

English Text	Replacement Info
Select line ^ to cancel or another line.	"^" is replaced with the letter of the line that is on soft hold.

Manually change ~ to ^ in any user-defined language. The software does not update manually.

Message 15, 16, 18

The character "^" is a place holder.

English Text	Replacement Info
Select line ^ to add party.	"^" is replaced with the letter of the line that is on soft hold.

Message 12

The character "^" is a place holder.

English Text	Replacement Info
Select line ^ to add party.	"^" is replaced with the letter of the line that is on soft hold.

Manually change ~ to ^ in any user-defined language. The software does not update manually.

Chapter 73: Listed Directory Numbers

With Listed Directory Numbers, Direct Inward Dialing (DID) numbers can be treated as public Listed Directory Numbers (LDNs). When one of these numbers is direct inward dialed, the calling party is routed to the attendant. The attendant display indicates a Listed Directory Number call and the name associated with the dialed extension.

Example command: `change listed-directory-numbers`

Ext

An extension number containing 16 digits.

Name

The name used to identify the Listed Directory Number. Accepts up to 27 alphanumeric characters.

Night Destination

A valid assigned extension number that receives calls to these numbers when Night Service is active. Accepts up to sixteen digits.

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Chapter 74: Locations

This screen provides daylight saving time displays. You can set the area code for each location and administer different location information for each location. If the **Multiple Locations** feature is enabled, you can administer up to 2000 location specifications, depending on the configuration of Communication Manager. Otherwise, information for Location 1 applies to all locations.

You can use the following command to modify the fields on this screen:

change locations

change locations

Page 1 of 1

LOCATIONS

ARS Prefix 1 Required For 10-Digit NANP Calls? y

Loc Name	Timezone	DST	City/	Proxy Sel
No	Offset		Area	Rte Pat
1: Main	+ 00:00	0		

ARS FAC

The Feature Access Code (FAC) for accessing Automatic Route Selection (ARS). Any valid FAC format is acceptable, up to four digits. Characters * or # are permitted, but only in the first position. Many locations are expected to share the same access code.

Related links

[Feature Access Code \(FAC\)](#) on page 212

ARS Prefix 1 Required for 10-Digit NANP Calls?

If enabled, a 1 must be dialed before all 10-digit NANP calls.

Attd FAC

The Feature Access Code (FAC) for connection to the attendant. Accepts up to two digits. Characters * or # are not permitted. Many locations are expected to share the same access code.

Available only if an Attendant Access Code has first been administered either for the Dial Plan or as a standard FAC.

 Note:

Within a dial plan, FAC/DAC codes and extensions cannot both start with the same first digits. Either the FAC/DAC entries or the block of extensions must be changed to have a different first digit.

Related links

[Call Type](#) on page 189

[Attendant Access Code](#) on page 213

Disp Parm

The administered display parameters for the location.

Related links

[Display Parameters](#) on page 200

Loc Parm

Required when administering multiple locations.

Valid Entry	Usage
1 to 50 blank	The number of the corresponding Location Parameter set for this location. Default is blank.

Loc Number

Valid Entry	Usage
1 to 2000	The location number.

Name

A name you use for the location. Names are easier to remember than location numbers. Accepts up to 15 alphanumeric characters.

City/Area

The three-digit numbering plan area code for each location.

Prefix

Valid Entry	Usage
0 to 99999	Location prefix used with the Uniform Dialing Plan to identify the location of an originating call. All or part of the prefix is added to the front of the dialed string.

Related links

[Insert Digits](#) on page 865

Proxy Sel Rte Pat

Valid Entry	Usage
1 to 999 blank	The routing pattern used to get to the proxy server.

Related links

[Route Pattern](#) on page 591

DST

Required for each administered location.

Valid Entry	Usage
0	No Daylight Saving.
1 to 15 blank	The number for the Daylight-Saving Rule applied to this location.

Timezone Offset

Specifies how much time each location differs from the system time. Required for each administered location. Use +00:00 for the time zone offset for a single location media server.

Valid Entry	Usage
+	The time set on this location is a certain amount of time ahead of the system time.
-	The time set on this location is a certain amount of time behind the system time.

Valid Entry	Usage
0 to 23	The number of hours difference between this location and system time.

Valid Entry	Usage
0 to 59	The number of minutes difference between this location and system time.

Chapter 75: Location Parameters

Sets or changes certain administrable characteristics that determine part of a location's behavior. These include recall timing intervals and loss plans for two-party and conference calls.

Multiple instances of the Location Parameters screen are accessible if **Multiple Locations** is enabled for the system. If **Multinational Locations** is enabled, Location Parameters 2-50 contain the same fields as for Location Parameters 1. If **Multinational Locations** is disabled, the system does not display the following fields for Location Parameters 1:

- **Tone Generation Plan**
- **DCP Terminal-parameters Plan**
- **Country Code for CDR**

Location Parameters: page 1

Analog Ringing Cadence

The country code identifies the ringing cadence to be used by analog telephones in the system.

Valid entry	Usage
1 to 25	The location country code.  Note: This field must be set to 1 (US) for the Message Waiting Indicator on the Station to be set to neon.

Analog Line Transmission

The country code identifies the transmission and signaling parameters.

Valid entry	Usage
1 to 25	The location country code.

Companding Mode

The companding algorithm used by system hardware.

Valid Entry	Usage
A-Law	Generally used outside the U.S.
Mu-law	Generally used in the U.S.

Convert ISDN Public Numbers to International for Display

If you set the field to values other than **no**, Communication Manager expands the number to the international numbering format. The default value of this field is **no**.

Valid entries	Usage
All-stations	Communication Manager expands calling and connected ISDN numbers to international format when the number is displayed on any station.
All-digital	Communication Manager expands calling and connected ISDN numbers to international format when the number is displayed on any digital station.
DCP stations	Communication Manager expands calling and connected ISDN numbers to international format when the number is displayed on any DCP station.
H.323 stations	Communication Manager expands calling and connected ISDN numbers to international format when the number is displayed on any H.323 station.
No	Calling and connected ISDN numbers appear unchanged on stations with displays.

Country code for CDR

Available only if the Multinational Locations feature is enabled for the system.

Valid Entry	Usage
1 to 999	The country code to be used for Call Detail Recording information for a location. Default is 1.

Date Format on Terminals

The **Date Format on Terminals** field enables you to set the date format on the terminals, based on its location. The system displays this field only if you set the **Multinational location** field on the System Parameter Customer Options screen to y.

Valid Entry	Usage
mm/dd/yy	(month/day/year)
dd/mm/yy	(day/month/year)
yy/mm/dd	(Year/month/day)

DCP Terminal-parameters Plan

This field corresponds to the DCP terminal transmission parameters administered for the location. The system displays this field only if the Multinational Locations feature is enabled.

Valid entry	Usage
1 to 25	The terminal-parameters plan number. The default value is 1.

End OCM After Answer (msec)

If the **End OCM After Answer** field contains a non-blank value, Communication Manager starts the End OCM timer when Communication Manager receives an answer signal. The End OCM timer ensures that an outgoing call using OCM call classification is answered by an agent or an announcement within a specified time.

Valid Entry	Usage
100 to 25,000	The timeout value is in milliseconds. If the timer expires, Communication Manager disconnects the call classifier and connects the call to the administered intercept extension. If the call classifier classifies the call before the timer expires, the timer is cancelled and the call routed or treated appropriately.
blank	Indicates that the timer has no limit.

End of OCM Intercept Extension

Valid Entry	Usage
extension	The extension number Communication Manager connects the call to when the End OCM After Answer field expires. The extension number can be a recorded announcement, a vector directory number, or a hunt group extension.
blank	Indicates that the extension number is empty. If the End OCM After Answer field is set to a non-blank value, the End of OCM Intercept Extension field cannot be left blank.

International Access Code

An up to five-digit International Access Code. Default is blank.

Line Load Control

Line Load Control (LLC) indicates the level of restriction per location. If the restriction level is set, then users with the same or higher restriction level set on their COR will be restricted from originating calls. The default value of this field is 0.

Note:

The location parameter index can be associated with multiple locations. If the restriction level is modified on the location parameter, it will apply to all the associated locations.

Local E.164 Country Code

An up to three-digit E.164 Country Code. Default is blank.

For a list of country codes, see the *International Telecommunications Union List of ITU-T Recommendation E.164 Assigned Country Codes*.

Long Distance Access Code

Valid Entry	Usage
0 to 9	The long distance access code you want the system to prefix to the telephone number. Accepts up to five digits.
blank	Not administered. This is the default.

Off-PBX Feature Name Extension Set

Valid Entry	Usage
0 to 10 for a medium configuration 1 to 99 for a large configuration	Feature Name Extension (FNE) set that should be used for location based FNEs.
blank	Default FNE set is used. This is the default.

Time Format on Terminals

With the **Time Format on Terminals** field, you can control the format for displaying the time on non-SIP endpoints. The system displays the **Time Format on Terminals** field even when the Multinational Locations feature is not enabled. Therefore, if all users are in a single country, you can change the time format for all users. To use different time formats for users in different locations, you must enable the Multinational Locations feature.

You can change the time format for SIP endpoints from the endpoint itself.

Valid entry	Usage
12-hour	To display the time in the 12-hour format on non-SIP display sets. With this option, the system overrides the display language and SA8885 settings.
24-hour	To display the time in the 24-hour format on non-SIP display sets. With this option, the system overrides the display language and SA8885 settings.
based-on-language	To use the display language and SA8885 to determine the time format for non-SIP display sets. This option is the default value.

Tone Generation Plan

The value in this field corresponds to the tone generation characteristics administered for the location. The system displays this field only if the Multinational Locations feature is enabled for the system.

Valid Entry	Usage
1 to 25	The tone-generation plan number. The default value is 1.

RECALL TIMING

Disconnect Timing (msec)

Available only if a Flashhook Interval is not required.

Valid Entry	Usage
80 to 1250 (in increments of 10)	An on-hook that lasts for a period of time less than this value will be ignored; greater than or equal to this value is regarded as a disconnect.

Flashhook Interval

Enables or disables requiring a flashhook interval (recall window).

Forward Disconnect Timer (msec)

Valid Entry	Usage
25 to 1500 (in increments of 25)	Specifies the duration of a momentary disconnect sent by the server/switch to an analog station user when that user is the last party still off-hook on a call.

Lower Bound (msec)

Available only if a flashhook interval is required.

Valid Entry	Usage
80 to 1250 (in increments of 10)	The lower bound of the station-to-switch recall signal timing interval in milliseconds. Specifies the lower bound of the flashhook interval.

MF Interdigit Timer (sec)

Valid Entry	Usage
1 to 255	The maximum number of seconds Communication Manager waits for the first forward signal (digit) to arrive, and for subsequent digits to arrive. Intercept returns to the calling party if this timer expires. This number must be less than the number of seconds administered for the short interdigit timer. Applies only to multifrequency signaling trunks.

Outgoing Shuttle Exchange Cycle Timer (sec)

Available only if the **Incoming Call Type** is group-ii-mfc or non-group-ii-mfc and the **Outgoing Call Type** is group-ii-mfc or none for the Multifrequency Signaling-Related System Parameters. This field applies only to multifrequency signaling calls made from Avaya Communication Manager.

Valid Entry	Usage
1 to 25	The number of seconds to time an exchange cycle (starts when the far end requests a digit until Avaya Communication Manager sends the requested digit).

Upper Bound (msec)

Available only if the flashhook interval is required.

Valid Entry	Usage
80 to 1250 (in increments of 10)	The upper bound of the flashhook interval. Specifies the upper bound of the station-to-switch recall signal timing interval in milliseconds. A flash of 50 msec to 130 msec is always acceptable from a 2500-type set regardless of the setting of the Upper and Lower bounds and is treated as the digit one.

Replace International Access Code with '+'

If you set the field to values other than **no**, Communication Manager the plus sign (+) instead of the configured international access code in front of international calling and connected numbers received from an ISDN trunk group. The default value of this field is **no**.

Valid entries	Usage
All-stations	Communication Manager adds the plus sign (+) to international calling and connected ISDN numbers that appear on all stations that includes analog with a display.
All-digital	Communication Manager adds the plus sign (+) to international calling and connected ISDN numbers that appear on digital stations with a display.
DCP stations	Communication Manager adds the plus sign (+) to international calling and connected ISDN numbers that appear on DCP stations.
H.323 stations	Communication Manager adds a plus sign (+) to international calling and connected ISDN numbers that appear on the H.323 stations.
No	International calling and connected ISDN numbers appear with the international access code, and no plus sign (+) on stations with a display.

Location Parameters: page 2

LOSS PLANS

2-Party Loss Plan/Tone Loss Plan

This field provides the default values for digital loss plan and for *n*-party conference loss.

Valid entry	Usage
1 to 25	A country code for the location.

Customize

Enables customization on the corresponding loss plan table. When **Customize** is enabled for the End-to-End total loss (dB) in a *n*-party conference, values can be changed by the administrator. When **Customize** is disabled, the End-to-End total loss (dB) in a *n*-party conference values are reset to the values that they would have had under the 2 Party Loss Plan. Available only if Digital Loss Plan Modification is enabled for the system.

End-to-End total loss (dB) in a *n*-party conference

Valid Entry	Usage
0 to 99	Provides total loss for a conference call with the designated number of parties. The higher the number listed for a call with a fixed number of parties, the more loss Communication Manager adds into a conference call with that number of parties; therefore, the conference call is quieter.

 Note:

The End-to-End total loss for multi-party conference calls that is administered for this field is not always applied to a specific call.

Inter-location Loss Group

When inserting loss for a call, the server treats parties on the call who are in separate locations as if the location with the most parties were connected by an equal number of IP tie trunks as there are parties at other locations. The field specifies the digital loss group number that is used by these virtual IP tie trunks. The system displays this field only if the Multinational Locations feature is enabled.

Valid entry	Usage
1 to 19	The digital loss group number to use on inter-location calls involving this location. The default value is 18.

Line Load Control Restriction

Restriction Level

Use **Restriction Level** field to impose restrictions on certain calls during critical scenarios or when experiencing a high volume of incoming calls.

Valid Entry	Usage
0	Feature not active (no restrictions). This is the default.
2	Restrict stations with a COR assigned to LLC levels 2, 3, and 4.
3	Restrict stations with a COR assigned to LLC levels 3, and 4.
4	Restrict stations with a COR assigned to LLC level 4.

Feature Access code Authentication Required

Use **Feature Access Code Authentication Required** field to control, if the user needs to be authenticated before the system accepts or denies change in locations LLC level.

Valid Entry	Description
N (no)	Select this option if you don't require authentication for accessing or modifying the LLC Restriction level. This is default.
y (yes)	Select this option if you want authentication to be required when accessing or changing the LLC Restriction level. This means that users will need to provide valid authentication credentials before making changes to the LLC restrictions.

Location Parameters: page 3

FROM / TO

Valid Entry	Usage
-3 to 15	Identifies the variable digital loss values. An unsigned number is a decibel loss, while a number preceded with a minus sign is a decibel gain.

Chapter 76: Login Administration

Beginning with Communication Manager 4.0, there is no longer a Login Administration screen. For details on screens used for login administration, see *Maintenance Commands for Avaya Aura® Communication Manager, branch Gateways and Servers* and “AAA Services” in *Avaya Aura® Communication Manager Feature Description and Implementation*.

Chapter 77: Logging levels

Administers logging of SAT activities. Specifies that commands associated with specific actions shown on this screen are logged by the system. The amount of detail to be logged is the same for all enabled actions and is specified by the **Log Data Values** field.

 Note:

The defaults on this screen provide the same amount and type of logging as in Communication Manager releases prior to 4.0.

Example command: `change logging-levels`

Related links

[Log Data Values](#) on page 495

Logging Levels: page 1

Enable Command Logging

If enabled, SAT activity is logged for selected commands.

Log Data Values

Valid Entry	Usage
none	Only the object, the qualifier, and the command action are logged.
new	The new value of any field is logged. The old value is not logged.
both	Both the prior field value and the field value after the change are logged.

Logging Levels: page 2

Log All Submission Failures

Form submission failures due to a security violation are always logged and are unaffected by this field.

Valid Entry	Usage
y	Record submission failures on the history log. An event is logged when Avaya Communication Manager rejects a form submission for any reason, such as an invalid entry in a field or a missing value.
n	Submission failures are not recorded on the history log.

Log CTA/PSA/TTI Transactions

Enables or disables logging transactions when extensions and physical telephones move between ports without additional administration from the administrator of Communication Manager.

If enabled, the system records Customer Telephone Activation (CTA), Personal Station Activation (PSA), and TTI transactions in the system history log.

Available only if Terminal Translation Initialization (TTI) is enabled for the system.

Related links

[Terminal Trans. Init. \(TTI\)](#) on page 738

Log IP Registrations and Events

If enabled, the IP registrations can be logged in the history log.

Log PMS/AD Transactions

If enabled, the system records Property Management System (PMS) and Abbreviated Dialing (AD) events to the log.

Chapter 78: Loudspeaker Paging

The Loudspeaker Paging screen administers voice paging, deluxe voice paging, and chime paging.

 Note:

To set up paging on a branch gateway, connect the paging system to a port on an MM711 and administer the port as an analog station on the Station screen. No entries on the Loudspeaker Paging screen are required.

Example command: `change paging loudspeaker`

CDR

If enabled, Communication Manager collects CDR data on the paging ports.

Code Calling — COR

Valid Entry	Usage
0 to 995	Assigns a Class of Restriction to a paging zone for chime paging.
blank	An unused paging zone.

Code Calling Playing Cycles

Valid Entry	Usage
1 to 3	The number of times a chime code plays when a user makes a chime page.
blank	Cannot be blank when administering chime paging (code calling).

Code Calling — TAC

Assigns a Trunk Access Code (TAC) to a paging zone for chime paging. Users dial this code to make a page to a specific zone. One TAC must be assigned to each zone used. Two zones cannot have the same TAC. A TAC in the zone designated ALL means that users can activate speakers in all the zones by dialing that code.

Valid Entry	Usage
1 to 4 digits	A Trunk Access Code (TAC) allowed by the dial plan.
*	Can be used as first digit.
#	Can be used as first digit.
blank	An unused paging zone.

Code Calling — TN

Valid Entry	Usage
1 to 100	Assigns a paging zone to a tenant partition for chime paging when Tenant Partitioning is in use.

Related links

[Tenant Partitioning](#) on page 738

Location

Assigns a descriptive name for the physical location corresponding to each zone when Tenant Partitioning is in use. Accepts up to 27 characters.

Example

Typical entries might be “conference room A”, “warehouse”, or “storeroom”.

Port

Assigns a port on an auxiliary trunk port to a paging zone.

Valid Entry	Usage
1 to 250	Gateway

Table continues...

Valid Entry	Usage
VI to V9	Module
01 to 31	Circuit
blank	An unused paging zone

Voice Paging — COR

Valid Entry	Usage
0 to 995	Assigns a Class of Restriction to a paging zone for voice paging.
blank	An unused paging zone.

Voice Paging — TAC

Use this field to assign a Trunk Access Code (TAC) to a paging zone for voice paging. Users dial this code to make a page to a specific zone. One TAC must be assigned to each zone. Two zones cannot have the same TAC. If you assign a TAC to the zone designated ALL, users can activate speakers in all the zones by dialing the code.

Valid entry	Usage
digits	Up to four digit TAC according to the dial plan
*	Can only be used as first digit
#	Can only be used as first digit
blank	An unused paging zone

Voice Paging Timeout (sec)

Limits the duration of voice pages. When this interval ends, calls are disconnected. To determine the best setting, time the typical pages that are expected to broadcast and then add another 4 to 5 seconds.

Valid Entry	Usage
10 to 60 seconds	The maximum number of seconds any page lasts.
blank	The field cannot be blank when voice paging is administered.

Voice Paging — TN

Valid Entry	Usage
1 to 100	Assigns a paging zone to a tenant partition for voice paging if Tenant Partitioning is in use.

Related links

[Tenant Partitioning](#) on page 738

Chapter 79: Maintenance-related system parameters

This screen is described in *Maintenance Commands for Avaya Aura® Communication Manager, Branch Gateways and Servers*.

Chapter 80: Media-Gateway

This screen is described in *Maintenance Commands for Avaya Aura® Communication Manager, Branch Gateways and Servers*.

Chapter 81: Media Server

Use the Media Server screen to add media servers to Communication Manager.

To modify the fields on the Media Server screen, run the following command:

`change media-server n`, where *n* the media server number. Where, *n* has a value from 1 to 250.

To view the status of the Media Server screen, run the following command:

`status media-server .`

Signaling Group

The SIP signaling group specifies the SIP signaling group that Communication Manager uses to communicate with the media server. The signaling-group must be populated with a **Peer Server** of type *AMS*. The signaling-group number entered on the media-server form may not be assigned to any other media-server.

Valid entry	Usage
Node Name	Far-End Node Name of the signaling group form.
Network Region	Far-End Network Region from the signaling group form.
Location	Location obtained from the Network Region form.
Announcement Storage Area	A display-only field that specifies the location on the media-server where the announcements for Communication Manager are stored. This value is same for all media-servers used by a specific Communication Manager. If the media-server is shared between different Communication Managers, each Communication Manager has a separate announcement storage area.

Voip Channel License Limit

The **Voip Channel License Limit** field represents the maximum number of simultaneous Voip channel licenses, which can be used on the specific media server. The field represents an upper limit. You can type a value from 1 to 9999 or leave the field blank.

If the field is left blank, the upper limit of simultaneous Voip channel licenses that can be used on the media-server are determined as follows:

- By the capacity of the media-server
- By the availability of Voip channel licenses

Dedicated Voip Channel Licenses

The **Dedicated Voip Channel Licenses** field represents the number of Voip channel licenses dedicated to the specified media server. **Dedicated Voip Channel Licenses** field is left blank or has a value between 1 and 9999. Blank implies no channel licenses are dedicated to the media-server.

Capacity Profile Max Sessions

Capacity profile max session is a value provided by AMS as part of its capacity profile and is given to the communication manager when the AMS comes into service. This value represents the maximum number of sessions of any type (audio, video, control), which this AMS server could establish. There are no limitations on the number of sessions, but rather a hard limit assuming an optimal selection of codecs and enough processing power and memory.

Capacity Profile MPU Estimate

Capacity profile Media Processing Unit (MPU) estimate is a value provided by AMS as part of its capacity profile. It is given to Communication Manager when the AMS comes into service. This is an estimate of the MPU capacity of the physical server. Capacity profile MPU value is estimated by AMS based on server type, clock speed, number of physical processor cores, logical processors, RAM, disc space, and NIC speed. Different AMS actions require different amount of MPU. For example, a simple G711, non-encrypted channel requires 1 MPU. More complex audio or video codecs require higher MPUs.

Cluster Type

AMS is configured as 1 of 3 available cluster types: Stand-alone, High-available, or Load-sharing. When AMS is a client of the Communication Manager, only the Stand-alone or High-available cluster types are supported. If AMS is incorrectly configured as part of a Load-sharing cluster, Communication Manager will not place the AMS into service. In this case, the **state** field of the status form indicates this with the new string: "OOS-ILLEGAL loadshare cluster". Additionally,

the "list media-server" report will also indicate the AMS is out-of-service with this new state: "OOS-LS".

Chapter 82: Message Waiting Indication Subscriber Number Prefixes

If the voice mail server is an AUDIX server connected using BX.25, you must fill in the Message Waiting Indication (MWI) Subscriber Number Prefixes to send or cancel a Message Waiting Indication to a served user. The served user is a user on the Private Telecommunication Network (PTN) to whom the MWI is sent on activation by the Message Center or Message Center PBX.

Using the QSIG MWI Supplementary Service, Message Center/Message Center PBX sends MWI Activate or MWI Deactivate message to the served user as a result of storage or retrieval of messages. Also, the served user is allowed to query the Message Center for MWI status.

Example command: `change isdn mwi-prefix`

AUDIX Mach ID

Valid entry	Usage
1 to 99	AUDIX number serving the switch of the AUDIX number field from the Switch Interface Administration screen.
blank	Available only if all the other records on the screen are blank.

Inserted Digits

Accepts up to 12 digits.

Valid entry	Usage
0 to 9	Digits prefixed to the AUDIX subscriber number to provide complete subscriber number to the Message Center PBX.
blank	No digits prefixed to the AUDIX subscriber number.

Machine ID

Available only if Machine ID is assigned to the qsig-mwi application in the Processor Channels Communication Interface.

Valid entry	Usage
1 to 20	Unique ID associated with each processor channel qsig-mwi application

Message Center ID

Accepts up to 20 digits. These are the Message Center ID digits to be sent a QSIG MWIActivate or MWIDeactivate message. It is necessary to enter the Message Center ID only if the system at the far-end of the QSIG connection needs it.

Routing Digits

Accepts up to four digits.

Valid entry	Usage
0 to 9, *, #	Digits comprising the AAR or ARS access code.
blank	No Routing Digits prefixed to the AUDIX subscriber number. For example, if UDP is used to route the call.

Send QSIG Message Center ID

Valid Entry	
y	Insert a message center ID in the MWI Activate and MWI Deactivate messages. For SIP-connected messaging servers, Communication Manager can be administered to insert a Message Center ID over QSIG connections to other servers by entering the Message Center ID, along with the Local Node Number for this Communication Manager server, in any record of this form.
n	Does not insert the message center ID in the MWI Activate and MWI Deactivate messages.

Chapter 83: Mode Code Related System Parameters

Establishes parameters associated with the Mode Code Voice Mail System Interface. Available only if the **Mode Code Interface** feature is enabled.

Example command: `change system-parameters mode-code`

Related links

[Mode Code Interface](#) on page 261

MODE CODES (FROM SWITCH TO VMS)

Direct Dial Access-Trunk

The mode code that the media server or switch sends when an external caller dials the Voice Mail System (VMS) access number. Accepts up to six digits. Also accepts #, *, and #00.

Direct Inside Access

The mode code that the media server or switch sends when a caller at an internal extension dials the Voice Mail System (VMS) access number. Accepts up to six digits. Also accepts #, *, and #00.

External Coverage

The mode code that the media server or switch sends when an external caller tries to reach a user at another extension and the call goes to the user's voice mail coverage. Accepts up to six digits. Also accepts #, *, and #00.

Internal Coverage

The mode code that Communication Manager sends when an internal caller tries to reach a user at another extension and the call goes to the user's voice mail coverage. Accepts up to six digits. Also accepts #, *, and #00.

Refresh MW Lamp

The mode code that Communication Manager sends during a system level 3 or higher reset that requests the VMS to refresh the Message Waiting (MW) lamps. Accepts up to six digits. Also accepts #, *, and #00.

System In Day Service

This value is used by the Voice Mail System (VMS) to indicate that Communication Manager has changed from Night to Day Service. Accepts up to six digits. Also accepts #, *, and #11.

System In Night Service

This value is used by the Voice Mail System (VMS) to indicate that Communication Manager has changed from Day to Night Service. Accepts up to six digits. Also accepts #, *, and #12.

OTHER RELATED PARAMETERS

DTMF Duration On

Valid Entry	Usage
75 to 500 — in multiples of 25	The duration in milliseconds of mode code digits sent to the VMS. This field cannot be blank.

Off

Valid Entry	Usage
Between 75 and 200 — in multiples of 25	The pause in milliseconds between mode code digits as they are sent to the VMS. This field cannot be blank.

Remote VMS Extensions- First

The first remote UDP VMS hunt group extension. Available only if **Mode Code for Centralized Voice Mail** is enabled for the system.

Related links

[Mode Code for Centralized Voice Mail](#) on page 735

Remote VMS Extensions - Second

The second remote UDP VMS hunt group extension. This extension cannot be the same as the first Remote VMS Extension. Available only if **Mode Code for Centralized Voice Mail** is enabled for the system.

Related links

[Mode Code for Centralized Voice Mail](#) on page 735

Sending Delay

Valid Entry	Usage
75 to 1000 — in multiples of 25	The delay in milliseconds between the time answer supervision is received from the VMS and the time the first mode code digit is sent. This field cannot be blank.

VMS Hunt Group Extension

The extension of a hunt group containing VMI extensions. A check is made to verify that a valid hunt group extension is entered, but a check is not made to verify that the hunt group members are VMI extensions.

Chapter 84: MOH Group

Use the MOH Group screen to define a collection of analog station and/or aux trunk ports that are connected to external audio sources for use with the Music on Hold feature.

Example command: `change moh-analog-group`

add moh-analog-group 1

Page 1 of 4

MOH GROUP 1

Group Name: _____

MOH SOURCE LOCATION

1: _____	16: _____	31: _____	46: _____	61: _____
2: _____	17: _____	32: _____	47: _____	62: _____
3: _____	18: _____	33: _____	48: _____	63: _____
4: _____	19: _____	34: _____	49: _____	64: _____
5: _____	20: _____	35: _____	50: _____	65: _____
6: _____	21: _____	36: _____	51: _____	66: _____
7: _____	22: _____	37: _____	52: _____	67: _____
8: _____	23: _____	38: _____	53: _____	68: _____
9: _____	24: _____	39: _____	54: _____	69: _____
10: _____	25: _____	40: _____	55: _____	70: _____
11: _____	26: _____	41: _____	56: _____	71: _____
12: _____	27: _____	42: _____	57: _____	72: _____
13: _____	28: _____	43: _____	58: _____	73: _____
14: _____	29: _____	44: _____	59: _____	74: _____
15: _____	30: _____	45: _____	60: _____	75: _____

MOH Source Location

The Music-on-hold analog or aux-trunk port location

Valid Entry	Usage
1 to 999	Gateway
V1 to V9	Module
01 to 31	Circuit

Group Name

The name that identifies the Music-on-hold (MOH) group.

Chapter 85: Multifrequency-Signaling-Related Parameters

Sets the system or location parameters associated with multifrequency signaling.

 Note:

With the Multinational Locations feature enabled, you can assign MFC signal sets per trunk group, rather than system-wide.

Example command: `change multifrequency-signaling`

Related links

[Multinational Locations](#) on page 736

Multifrequency-Signaling-Related Parameters: page 1

ANI Prefix

The prefix to apply to an extension when ANI is sent to the local telephone company central office. Accepts one to six digits.

Available only if the **Outgoing Call Type** is group-ii-mfc or mfe.

Related links

[Outgoing Call Type](#) on page 517

Backward Cycle Timer (sec)

Available only if the **Incoming Call Type** is mfe.

Valid Entry	Usage
1 to 255	The number of seconds the system waits to send the check frequency after receiving an MFE signal.

Related links

[Incoming Call Type](#) on page 515

Collect All Digits Before Seizure

Available only if the **Outgoing Call Type** is group-ii-mfc or mfe.

Valid Entry	Usage
y	The system collects all the digits before seizing the trunk. Values administered for ANI Req for AAR and ARS Digit Conversion do not apply.
n	ANI collection is controlled by ARS administration.

Related links

[ANI Req](#)

[Outgoing Call Type](#) on page 517

Convert First Digit End-of-Dial To

The digit used when the incoming initial end-of-ani or end-of-dial MF signal is converted on a per-switch basis.

Available only if **Private Group II Permissions and Public Interworking** is enabled.

Related links

[Private Group II Permissions and Public Interworking](#) on page 518

Default ANI

Available only if the **Outgoing Call Type** is group-ii-mfc or mfe.

Valid Entry	Usage
2 to 15 digits	The switch or server identification number that is sent to the local telephone company central office (CO) when ANI is requested (by the CO) on a particular call but is unavailable, such as on tandem tie trunk calls.
blank	Use for tandem switching. If blank, administer a value to send for ANI on outgoing calls when it is unavailable.

Related links

[ANI Not Available](#) on page 523

[Outgoing Call Type](#) on page 517

Forward Cycle Timer (sec)

Available only if the **Incoming Call Type** is mfe.

Valid Entry	Usage
1 to 255	The number of seconds the system waits to receive the check frequency after sending an MFE signal. Communication Manager drops the call if the time runs out before it receives check frequency.

Related links

[Incoming Call Type](#) on page 515

Group II Called Party Category

The type of Group II signals that should be used on the outgoing R2-MFC call. Available only if the **Outgoing Call Type** is group-ii-mfc and **Use COR for All Group II Responses** is disabled.

Valid Entry	Usage
user-type	The type of telephone making the call determines the type of Group II signal that the switch or server sends (normal = ordinary telephone set, attendant = attendant console, data-call = data modules and similar data endpoints).
call-type	The dialed digits determine the type of Group II signal that the server sends.

Related links

[Outgoing Call Type](#) on page 517

[Use COR for All Group II Responses](#) on page 519

Incoming Call Type

Valid Entry	Usage
group-ii-mfc non-group-ii-mfc	The signal type that a local telephone company central office uses to place an incoming call to the server.
mfe	Multi-frequency Espanol used for Spain

Incomplete Dial Timer (sec)

Available only if the **Incoming Call Type** is mfe.

Valid Entry	Usage
45 to 255	The number of seconds Communication Manager waits from the start of a call until the end of the check frequency of the last signal. Communication Manager drops the call if the time runs out before it receives the check frequency.

Related links

[Incoming Call Type](#) on page 515

Maintenance Call Type

Available only if the **Incoming Call Type** is group-ii-mfc or non-group-ii-mfc.

Valid Entry	Usage
1	The Belgium maintenance sequence is indicated when the local telephone company central office (CO) sends an MFC maintenance tone.

Table continues...

Valid Entry	Usage
2	The Saudi Arabian sequence is indicated when the CO sends an MFC maintenance tone.
none	Not administered.

Related links

[Incoming Call Type](#) on page 515

Maximum Resend Requests

The threshold number of resend type MFC signals the server running Communication Manager accepts during an outgoing call.

Valid Entry	Usage
1 to 99	The number of resend requests.
1	The call is dropped if one resend signal is received.
blank	An unlimited number of resend requests is allowed.

MF Signaling Intercept Treatment - Incoming

Valid Entry	Usage
y	Sends the group B signal for the intercept to the local telephone company central office and play intercept tone on the trunk.
n	Uses normal DID/TIE/ISDN intercept treatment.

MF Signaling Intercept Treatment - Outgoing

Defines the treatment for outgoing calls that cannot be completed as dialed. Available only if **Outgoing Call Type** is group-ii-mfc.

Valid Entry	Usage
announcement	Plays a recorded announcement for outgoing calls that cannot be completed as dialed. You select and record the message. Requires entry of the extension number for the announcement.
tone	Plays intercept tone for outgoing calls that cannot be completed as dialed.

Related links

[Outgoing Call Type](#) on page 517

MFE Type

Available only if the **Incoming Call Type** is mfe.

Valid Entry	Usage
2/5	Determines which public signaling Communication Manager uses.
2/6	

Related links

[Incoming Call Type](#) on page 515

Outgoing Call Type

Valid Entry	Usage
group-ii-mfc	The signal type that the switch or server uses to place an outgoing call to a local telephone company central office.
mfe	Multi-frequency Espanol used in Spain.
none	Not administered.

Outgoing Forward Signal Absent Timer (sec)

Available only if the **Outgoing Call Type** is group-ii-mfc.

Valid Entry	Usage
11 to 255	The maximum time to elapse between forward signals on outgoing calls. The timer starts and restarts when a forward tone is taken off the link, and it stops when the next forward tone is applied to the link.

Related links

[Outgoing Call Type](#) on page 517

Outgoing Forward Signal Present Timer (sec)

Available only if the **Outgoing Call Type** is group-ii-mfc.

Valid Entry	Usage
1 to 255	The maximum time to elapse between signals on a call. This timer runs when MFC tones are being sent or received on an outgoing call. The timer starts (and restarts) when Communication Manager begins sending a forward signal and stops when Communication Manager receives the backward signal.

Related links

[Outgoing Call Type](#) on page 517

Outgoing Start Timer (sec)

Available only if the **Incoming Call Type** is mfe.

Valid Entry	Usage
1 to 255	The number of seconds from seizure until the beginning of the first Group A signal from the receiving end, and from the end of the check frequency until the beginning receipt of the first digit following the Group II signal.

Related links

[Incoming Call Type](#) on page 515

Overlap Sending on Link-to-Link Tandem Calls

If enabled, and calls are tandemed between servers, then ANI for the switch or server is sent to the terminating switch if that switch requests ANI before Communication Manager receives it from the originating server or switch. The terminating server or switch can request ANI before the receipt of the last address digit if it is not running Communication Manager, or if Communication Manager is administered to request the call category at the start of the call.

If enabled, Communication Manager sends and receives digits one digit at a time instead of enbloc. With enbloc, digits are not sent until the entire group of digits is received.

Available only if **Collect All Digits Before Seizure** is disabled.

Related links

[Collect All Digits Before Seizure](#) on page 514

[Request Call Category at Start of Call](#) on page 522

Private Group II Permissions and Public Interworking

If enabled, then Communication Manager:

- Sends the category for MFC ANI for the COR of the originating party for non-private-MFC-trunk to MFC-private-trunk calls.
- Sends the Group II category received over the incoming private trunk as the outgoing Group II category on tandem private MFC calls.
- Applies MFC group II-CPC termination restrictions on incoming MFC private trunk calls.
- Checks station permissions for call forward off-net calls.

Available only if the **Incoming Call Type** is group-ii-mfc or non-group-ii-mfc and the **Outgoing Call Type** is group-ii-mfc or none.

Related links

[Incoming Call Type](#) on page 515

[Outgoing Call Type](#) on page 517

Received Signal Gain (dB)

Valid Entry	Usage
-15 to 3	The number for the loss or gain when the MFC port listens to the trunk port. Communication Manager listens with a range of -5 to -35. This value moves the range (for example, a value of -5 provides a range of -10 to -40). Also applies to Russian MF Shuttle trunks.

Request Incoming ANI (non-AAR/ARS)

If enabled, ANI is requested on incoming R2-MFC calls.

Available only if **Incoming Call Type** is group-ii-mfc or mfe and the **Outgoing Call Type** is group-ii-mfc or mfe. Applies only if the incoming call via the R2-MFC trunk is terminating to a local station on this PBX.

Related links

[Incoming Call Type](#) on page 515

[Outgoing Call Type](#) on page 517

Transmitted Signal Gain (dB)

Valid Entry	Usage
-15 to 3	The number for the loss or gain when the trunk port listens to the MFC port. The MFC port generates at -5 for MFC and -8 for MFE. This field adds gain or loss to the starting value of -5. Also applies to Russian Shuttle trunks and Russian multi-frequency ANI.

Use COR for All Group II Responses

If enabled, the COR administered category is used for both the calling party and called party categories.

Available only if the **Outgoing Call Type** is group-ii-mfc.

Related links

[Outgoing Call Type](#) on page 517

Use COR for Calling Party Category

Indicates the category to send with ANI if requested on an outgoing R2-MFC call. Available only if the **Outgoing Call Type** is group-ii-mfc and **Use COR for All Group II Responses** is disabled.

Valid Entry	Usage
y	The calling facility's COR is used to determine category.
n	The calling party's user-type COR is used to determine category.

Related links

[Outgoing Call Type](#) on page 517

[Use COR for All Group II Responses](#) on page 519

NEXT ANI DIGIT**Related links**

[Incoming Call Type](#) on page 515

[Outgoing Call Type](#) on page 517

Incoming

Available only if the **Incoming Call Type** is group-ii-mfc and the **Outgoing Call Type** is group-ii-mfc or mfe.

Valid Entry	Usage
next-digit next_ani_digit send-ani	Determines whether the Next ANI Digit signal is the same as the send-ani signal, the next-digit signal, or another signal defined as next_ani_digit.

Outgoing

Available only if the **Outgoing Call Type** is group-ii-mfc.

Valid Entry	Usage
next-digit next_ani_digit send-ani	Determines whether the Next ANI Digit signal is the same as the send-ani signal or the next-digit signal or another signal defined as next_ani_digit.

Related links

[Outgoing Call Type](#) on page 517

Multifrequency-Signaling-Related Parameters: page 2

These fields define call category and ANI information. For India, the ANI can be requested without the call category information.

Address Digits Include End-of-Digits Signal

If enabled, indicates that an outgoing forward Group I end-of-digit signal is always sent after completion of address digits upon request from the local telephone company central office for outgoing calls.

ANI Source for Forwarded & Covered Calls

Valid Entry	Usage
caller	Send the calling party's ANI when calls are redirected.
forwarder	Send the forwarding party's ANI when calls are redirected.

Call Category for Vector ii-digits

If enabled, allows the use of the call category digit, which is part of ANI, as the ii-digits on call vector steps.

Do Not Send Group B Signals to CO

Available only if the **Incoming Call Type** is group-ii-mfc.

Valid Entry	Usage
y	Does <i>not</i> send Group-B signals to complete an incoming call.
n	Sends Group-B signals to complete an incoming call.

Related links

[Incoming Call Type](#) on page 515

Number of Incoming ANI Digits

Valid Entry	Usage
0 to 15	The number of ANI digits for incoming MFC calls.

Number of Outgoing ANI Digits

Valid Entry	Usage
0 to 15	The number of ANI digits for outgoing MFC calls. In India or any country where end-of-ani and end-of-digits are not defined for Tones to CO on Outgoing Forward Calls - Group I, Avaya Communication Manager appends ANI-Not-Available digits to ANI digits if the actual ANI length is less than the number entered in this field. If end-of-ani or end-of-digits are defined, this field is used in conjunction with Truncate Station Number in ANI as a maximum ANI length. For India, even if the length of ANI is defined, if the timeout occurs during the ANI collection, the call is routed with the ANI digits already collected. Applies to Russian shuttle trunks, and MFC and MFE trunks.

Outgoing II by COR

Available only if either **Use COR for Calling Party Category** or **Use COR for All Group II Responses** are enabled.

Valid Entry	Usage
1 to 10	The Group II signal sent to the local telephone company central office on outgoing calls can be administered per COR (Class of Restriction) and per trunk group. The Group II signal is administered per COR. That per-COR value in turn can be mapped into a different outgoing signaling parameter set. The values for this field administer that outgoing mapping.

Related links

[Use COR for All Group II Responses](#) on page 519

[Use COR for Calling Party Category](#) on page 519

Request Call Category at Start of Call

Indicates that the Send-ANI backward signal requesting for the caller-category information is sequenced differently in the MFC signaling flow. The Caller-category Request backward signal is disjointed from the ANI request.

If enabled, the Send-ANI backward signal corresponds exclusively to the caller-category request. In response to this signal, Communication Manager sends a forward signal containing the caller-category information on outgoing calls. On incoming calls, Communication Manager sends the Send-ANI backward signal upon receipt of the first address signal.

Request CPN at Start of Call

With Request CPN at Start of Call, Communication Manager can collect ANI and call category immediately after receipt of the first address digit.

If enabled, provides ANI Calling Party Number (CPN) and call category immediately after receiving the first address digit.

Available only if the **Incoming Call Type** is group-ii-mfc.

Related links

[Incoming Call Type](#) on page 515

Restart ANI from Caller Category

If enabled, Avaya Communication Manager sends the caller-category signal later again when the signals for Caller-Category and ANI requests are the same and this signal is received after the Next-Digit forward signals have been received.

Truncate Station Number in ANI

Applies to Russian shuttle trunks, and MFC and MFE trunks.

Valid Entry	Usage
beginning	Defines the side of the extension number from which to truncate when station ANI is sent to the local telephone company central office and the combined length of the ANI prefix and extension number is greater than the administered Number of Outgoing ANI Digits. The ANI prefix (either MFC or COR) is not truncated. There is no effect if ANI for switch or server is sent.
ending	
no	

Related links

[Number of Outgoing ANI Digits](#) on page 521

INCOMING / OUTGOING

ANI Available

Valid Entry	Usage
1 to 15	The signal number used for incoming ANI-Available.
blank	

ANI Not Available

Required if the **Default ANI** is not administered.

Valid Entry	Usage
1 to 15	The signal number used for outgoing ANI-Available. Communication Manager outputs the End-of-Dial backward signal when the ANI-Not-Available forward signal is received on incoming calls. Communication Manager outputs the ANI-Not-Available forward signal to the local telephone company central office on outgoing calls where ANI is not possible.
blank	

Multifrequency-Signaling-Related Parameters: page 3

These fields define the meaning of MFC tones for calls originated at the local telephone company central office.

When the system initially displays the screen, either of two sets of default values is possible. One set is for the Group II call type; the other set is for non-Group II call type. In each set, the default value for each field is set to the most common value.

Available only if the **Incoming Call Type** is group-ii-mfc or non-group-ii-mfc.

Related links

[Incoming Call Type](#) on page 515

INCOMING FORWARD SIGNAL TYPES (Tones from CO)

Group I

Displays message codes 11 to 15. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.) A Group I signal type can be administered for each code.

Group I signals are a set of forward signals generated by the originating server.

Valid Entry	Usage
ani-avail	Used in Hungary. If this signal is defined and Automatic Number Identification (ANI) is requested on outgoing R2-MFC calls, ANI is sent to the local telephone central office before ANI caller digits are sent. This signal is sent after the ANI caller category signal. Available if Incoming Call Type is group-ii-mfc.
ani-not-avail	Used on Direct Outward Dialing calls in Brazil and Columbia. Communication Manager sends this signal to the local telephone company central office when it receives an Automatic Number Identification (ANI) request and the caller's number is unavailable. Available if Incoming Call Type is group-ii-mfc.
drop	When this signal is received from the local telephone company central office, Avaya Communication Manager starts the disconnect sequence and drops the call. Available if Incoming Call Type is group-ii-mfc or non-group-ii-mfc.
end-of-ani	This signal is used on Direct Outward Dialing and Direct Inward Dialing calls. Communication Manager sends this signal to indicate the end-of-ANI digits when Automatic Number Identification (ANI) digits are sent to the local telephone company central office. Available if Incoming Call Type is group-ii-mfc.
end-of-dial	This signal is used when open numbering is used on Direct Inward Dialing calls. The local telephone company central office sends this signal to indicate the end-of-dial digits and Communication Manager responds with a request for a Group II signal. Available if Incoming Call Type is group-ii-mfc.
ignored	If this signal is received from the local telephone company central office, Communication Manager sends a corresponding signal (A.1, and so on) but no action is taken in the response and it is not counted as a digit. In Belgium, this signal is not acknowledged. Available if Incoming Call Type is group-ii-mfc or non-group-ii-mfc.
maint-call	The local telephone company central office (CO) sends a signal to indicate that a call is a maintenance call and Communication Manager prepares the special maintenance call sequences for the CO. This signal can be used on Direct Inward Dialing calls in Saudi Arabia. Available if Incoming Call Type is group-ii-mfc.

Table continues...

Valid Entry	Usage
send-congest	When Communication Manager receives this signal from the local telephone company central office (CO) on a Direct Inward Dialing call, it returns a congestion signal (Group A), in compel (not pulse) mode, to the CO. Available if Incoming Call Type is group-ii-mfc.

Group II

Displays message codes 1 to 15. A Group II signal type can be administered for each code.

Group II signals are a more elaborate set of forward signals generated by the originating server.

Valid Entry	Usage
attendant	If Communication Manager receives this signal on Direct Inward Dialing (DID) calls, the call terminates at an attendant regardless of the extension dialed. On Direct Outward Dialing (DOD) calls, this signal is sent to the local telephone company central office (CO) if the CO requests calling-category information and the originating extension is an attendant. This signal is used on both DID and DOD calls.
busy-rt-attd	If Communication Manager receives this signal on Direct Inward Dialing (DID) calls, the call terminates at an attendant if the called extension is busy or at the called extension if it is not busy. This signal is used on DID calls.
data-call	If Communication Manager receives this signal on Direct Inward Dialing (DID) calls, it sends intercept treatment. On Direct Outward Dialing (DOD) calls, this signal is sent to the local telephone company central office (CO) if the CO requests calling-category information and the originating extension is a data extension. This signal is used on both DID and DOD calls.
data-verify	If Communication Manager receives this signal on Direct Inward Dialing (DID) calls and the terminating extension is not a data extension, it sends intercept treatment. On Direct Outward Dialing (DOD) calls, this signal is sent to the local telephone company central office (CO) if the CO requests calling-category information and the originating extension is a data extension. This signal is used on both DID and DOD calls.
drop	When this signal is received from the local telephone company central office (CO), Communication Manager starts the disconnect sequence and drops the call.
maint-call	The local telephone company central office (CO) sends a signal to indicate that a call is a maintenance call and Avaya Communication Manager prepares the special maintenance call sequences for the CO.
normal	This signal indicates that the caller is a normal subscriber. If it is received on a Direct Inward Dialing (DID) call, the call is terminated at the called extension. For an outgoing MF signaling call that uses Group II signaling, this signal is sent to the local telephone company central office (CO) when the CO requests calling-category information and the originating extension is a station. This signal is used in both DID and Direct Outward Dialing calls.
send-intercept	If Communication Manager receives this signal from the local telephone company central office (CO) on a Direct Inward Dialing call, it returns Group B intercept signal to the CO.

Table continues...

Valid Entry	Usage
toll-auto	This signal is used in China. This signal indicates that a call is an automatic toll call. When the call terminates at a busy station and a special busy signal is defined, the busy signal is sent to the local telephone company central office. A special busy signal can be defined by choosing the option toll-busy on the incoming Group B signals.
toll-operator	This signal, used in China, is treated as a normal subscriber signal. See the normal definition.

INCOMING BACKWARD SIGNAL TYPES (Tones to CO)

Group A

Displays message codes 11 to 15. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.) A Group A signal type can be administered for each code.

Group A signals are backward signals generated by the destination server or switch.

Valid Entry	Usage
congestion	The local telephone company central office sends this signal to indicate that it is experiencing network congestion. When Communication Manager receives this signal on Direct Outward Dialing (DOD) calls, it drops the trunk and plays reorder tone to the calling party. This signal is used on DOD calls.
end-of-dial	This signal is sent to indicate the end of the address digit string. For MF Group II calls, this signal requests a Group II signal and switches the sender over to the Group B signaling mode. This signal is used on both Direct Inward Dialing and Direct Outward Dialing calls.
intercept	The local telephone company central office sends this signal to indicate the call has been terminated to an invalid destination. When Communication Manager receives this signal on Direct Outward Dialing (DOD) calls, it drops the trunk and plays intercept tone to the calling party. This signal is used on DOD calls.
next-ani-digit	Communication Manager sends this signal to request the next ANI digit. This signal is used on Direct Inward Dialing and Direct Outward Dialing calls.
next-digit	Communication Manager sends this signal to request the next digit. This signal is used on both Direct Inward Dialing and Direct Outward Dialing calls.
send-ani	The local telephone company central office (CO) sends this signal to request calling-party category and sends additional signals to request ANI digits. This signal is sent to the CO when Avaya Communication Manager requests ANI digits on Direct Inward Dialing (DID) calls. This signal is used on both Direct Outward Dialing and DID calls.
setup-sppath	The local telephone company central office sends this signal to Communication Manager to set up a speech path. This signal is used on Direct Outward Dialing calls and on Direct Inward Dialing calls in Belgium.

Group B

Displays message codes between 1 and 15. A Group B signal type can be administered for each code

Group B signals enhance Group A signals for backward signaling from the destination end by providing the status of the called party. In addition, if the originating server uses Group II signals, the destination end answers with Group B signals.

Not available if **Do Not Send Group B Signals to CO** is enabled.

Valid Entry	Usage
busy	This signal is sent to indicate that the called party is busy. On Direct Inward Dialing calls, the signal is sent to the local telephone company central office if there is no coverage point to terminate the call. If Communication Manager receives this signal on Direct Outward Dialing calls, it plays busy tone to the calling party and drops the trunk.
congestion	This signal is sent to indicate that the system is congested and the call cannot be terminated successfully. On Direct Inward Dialing calls, the signal is sent to the local telephone company central office to indicate that a resource is not available. On Direct Outward Dialing calls, if Communication Manager receives this signal, reorder tone is played to the calling party and the trunk is dropped.
free	This signal indicates that the called party is idle. On Direct Inward Dialing calls, the signal is sent to the local telephone company central office to indicate that the called party is idle and the call is terminated successfully. If Communication Manager receives this signal on Direct Outward Dialing calls, it connects the trunk to the calling party.
intercept	This signal indicates that the called party number is not in service or is incorrect. On Direct Inward Dialing calls, if intercept treatment is set to provide a tone, tone is sent to the local telephone company central office to indicate that the called number is invalid. If Communication Manager receives the signal on Direct Outward Dialing calls, it plays intercept tone to the calling party and drops the trunk.
mct	This signal identifies the call as one that needs to be traced by the local telephone company central office (CO). Avaya Communication Manager then generates an MFC Call Trace Backward Signal administered for multifrequency signaling during call setup instead of the "free" signal. If the terminating station's Class of Restriction (COR) is enabled, the CO collects trace information before releasing the calling party. If the station's COR has MF Incoming Call Trace enabled and the "mct" signal is not defined, then the "free" signal is sent.
tariff-free	This signal is sent when the trunk group provides an 800 service. Avaya Communication Manager generates an MFC tariff-free backward signal during call setup instead of the "free" signal, facilitating local telephone company central office billing. If the trunk is administered as a tariff-free trunk and the "tariff-free" signal is not defined, then the "free" signal is sent.
tie-free	This signal is used only when an incoming call is received and defined and the incoming facility is a tie trunk. Otherwise, the free signal is used.
toll-busy	This signal, used in China, is sent to indicate that the called party is busy if the call is an automatic toll call.

Related links

[Do Not Send Group B Signals to CO](#) on page 521

Multifrequency - Signaling- Related Parameters: page 4

The fields shown on this page define the meaning of MFC tones for calls originated at the switch or server.

The system displays this screen only if **Outgoing Call Type** is group-ii-mfc or mfe.

Related links

[Outgoing Call Type](#) on page 517

OUTGOING FORWARD SIGNAL TYPES (Tones to CO)

Group I

Displays message codes 11 to 15. (Numbers 1 through 10 are assigned to the digits of the destination telephone number.) A Group I signal type can be administered for each code.

Group I signals are a set of forward signals generated by the originating server.

Valid Entry	Usage
ani-avail	Used in Hungary. If this signal is defined and Automatic Number Identification (ANI) is requested on outgoing R2-MFC calls, ANI is sent to the local telephone central office before ANI caller digits are sent. This signal is sent after the ANI caller category signal.
ani-not-avail	Used on Direct Outward Dialing calls in Brazil and Columbia. Communication Manager sends this signal to the local telephone company central office when it receives an Automatic Number Identification (ANI) request and the caller's number is unavailable.
end-of-ani	This signal is used on Direct Outward Dialing and Direct Inward Dialing calls. Communication Manager sends this signal to indicate the end-of-ANI digits when Automatic Number Identification (ANI) digits are sent to the local telephone company central office.
end-of-digits	This signal is sent by the originating server that makes outgoing calls, sends digits, and receives a next-digit Group A signal from the destination server or switch when there are no more digits to be sent. This signal is also sent when Communication Manager does not have end-of-ani assigned, makes an outgoing call, sends ANI, and receives a call-info-ani Group A signal from the destination end when there are no more Automatic Number Identification (ANI) digits to be sent. If both end-of-digits and end-of-ani are assigned, Communication Manager uses end-of-ani after it sends the last ANI digit and end-of-digits after sending the last called-number digit.

Group II

Displays message codes between 1 and 15. A Group II signal type can be administered for each code. Each entry can only appear once in the Group II column.

Group II signals are a more elaborate set of forward signals generated by the originating server.

Valid Entry	Usage
attendant	If Communication Manager receives this signal on Direct Inward Dialing (DID) calls, the call terminates at an attendant regardless of the extension dialed. On Direct Outward Dialing (DOD) calls, this signal is sent to the local telephone company central office (CO) if the CO requests calling-category information and the originating extension is an attendant. This signal is used on both DID and DOD calls.
data-call	If Communication Manager receives this signal on Direct Inward Dialing (DID) calls, it sends intercept treatment. On Direct Outward Dialing (DOD) calls, this signal is sent to the local telephone company central office (CO) if the CO requests calling-category information and the originating extension is a data extension. This signal is used on both DID and DOD calls.
normal	This signal indicates that the caller is a normal subscriber. If it is received on a Direct Inward Dialing (DID) call, the call is terminated at the called extension. For an outgoing MF signaling call that uses Group II signaling, this signal is sent to the local telephone company central office (CO) when the CO requests calling-category information and the originating extension is a station. This signal is used in both DID and Direct Outward Dialing calls.
toll-auto	This signal is used in China. This signal indicates that a call is an automatic toll call. When the call terminates at a busy station and a special busy signal is defined, the busy signal is sent to the local telephone company central office. A special busy signal can be defined by choosing the option toll-busy on the incoming Group B signals.

OUTGOING BACKWARD SIGNAL TYPES (Tones from CO)

Group A

Displays message codes between 1 and 15. A Group A signal type can be administered for each code.

Group A signals are backward signals generated by the destination server or switch.

Valid Entry	Usage
congestion	The local telephone company central office sends this signal to indicate that it is experiencing network congestion. When Communication Manager receives this signal on Direct Outward Dialing (DOD) calls, it drops the trunk and plays reorder tone to the calling party. This signal is used on DOD calls.
drop	When this signal is sent, the receiving end starts the disconnect sequence.

Table continues...

Valid Entry	Usage
end-of-dial	This signal is sent to indicate the end of the address digit string. For MF Group II calls, this signal requests a Group II signal and switches the sender over to the Group B signaling mode. This signal is used on both Direct Inward Dialing and Direct Outward Dialing calls.
intercept	The local telephone company central office sends this signal to indicate the call has been terminated to an invalid destination. When Communication Manager receives this signal on Direct Outward Dialing (DOD) calls, it drops the trunk and plays intercept tone to the calling party. This signal is used on DOD calls.
last-2-digits	Communication Manager sends this signal to adjust the outpulsing pointer so that the last three digits can be resent. This signal is used on Direct Outward Dialing calls.
last-3-digits	Communication Manager sends this signal to adjust the outpulsing pointer so that the last four digits can be resent. This signal is used on Direct Outward Dialing calls.
last-digit	Communication Manager sends this signal to adjust the outpulsing pointer so that the last two digits can be resent. This signal is used on Direct Outward Dialing calls.
next-ani-digit	Communication Manager sends this signal to request the next ANI digit. This signal is used on Direct Inward Dialing and Direct Outward Dialing calls.
next-digit	Communication Manager sends this signal to request the next digit. This signal is used on both Direct Inward Dialing and Direct Outward Dialing calls.
restart	Communication Manager sends this signal to request the whole digit string again. This signal is used on Direct Outward Dialing calls.
resend-digit	Communication Manager sends this signal to adjust the outpulsing pointer so that the last digit can be resent again. This signal is used on Direct Outward Dialing calls.
send-ani	The local telephone company central office (CO) sends this signal to request calling-party category and sends additional signals to request ANI digits. This signal is sent to the CO when Avaya Communication Manager requests ANI digits on Direct Inward Dialing (DID) calls. This signal is used on both Direct Outward Dialing and DID calls.
setup-sppath	The local telephone company central office sends this signal to Communication Manager to set up a speech path. This signal is used on Direct Outward Dialing calls and on Direct Inward Dialing calls in Belgium.

Group B

Displays message codes between 1 and 15. A Group B signal type can be administered for each code.

Group B signals enhance Group A signals for backward signaling from the destination end by providing the status of the called party. In addition, if the originating server uses Group II signals, the destination end answers with Group B signals.

Valid Entry	Usage
busy	This signal is sent to indicate that the called party is busy. On Direct Inward Dialing calls, the signal is sent to the local telephone company central office if there is no coverage point to terminate the call. If Communication Manager receives this signal on Direct Outward Dialing calls, it plays busy tone to the calling party and drops the trunk.
congestion	This signal is sent to indicate that the system is congested and the call cannot be terminated successfully. On Direct Inward Dialing calls, the signal is sent to the local telephone company central office to indicate that a resource is not available. On Direct Outward Dialing calls, if Communication Manager receives this signal, reorder tone is played to the calling party and the trunk is dropped.
free	This signal indicates that the called party is idle. On Direct Inward Dialing calls, the signal is sent to the local telephone company central office to indicate that the called party is idle and the call is terminated successfully. If Communication Manager receives this signal on Direct Outward Dialing calls, it connects the trunk to the calling party.
intercept	This signal indicates that the called party number is not in service or is incorrect. On Direct Inward Dialing calls, if intercept treatment is set to provide a tone, tone is sent to the local telephone company central office to indicate that the called number is invalid. If Communication Manager receives the signal on Direct Outward Dialing calls, it plays intercept tone to the calling party and drops the trunk.
mct	This signal identifies the call as one that needs to be traced by the local telephone company central office (CO). Avaya Communication Manager then generates an MFC Call Trace Backward Signal administered for multifrequency signaling during call setup instead of the "free" signal. If the terminating station's Class of Restriction (COR) is enabled, the CO collects trace information before releasing the calling party. If the station's COR has MF Incoming Call Trace enabled and the "mct" signal is not defined, then the "free" signal is sent.
tariff-free	This signal is sent when the trunk group provides an 800 service. Avaya Communication Manager generates an MFC tariff-free backward signal during call setup instead of the "free" signal, facilitating local telephone company central office billing. If the trunk is administered as a tariff-free trunk and the "tariff-free" signal is not defined, then the "free" signal is sent.
toll-busy	This signal, used in China, is sent to indicate that the called party is busy if the call is an automatic toll call.

Chapter 86: Multiple Level Precedence & Preemption (MLPP) Parameters

Use this screen to set up system parameters for the Multiple Level Precedence & Preemption feature.

Example command: `change system-parameters mlpp`

ANNOUNCEMENTS

ASAC Enabled

 Note:

The **ASAC Enabled** field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

ASAC is an extension to the existing **Multiple Level Precedence and Preemption (MLPP)** calling function which provides directionalization (inbound and outbound awareness) attribute to a calling budget. Administrators can restrict the available call capacity in the inbound and outbound direction with ASAC.

This field is available only when you set the **Multiple Level Precedence and Preemption** field on the System Parameters Customer Options screen to **y**.

Valid entry	Usage
y	Makes all ASAC-related fields available on the SIP Trunk Group screen. ASAC also allows Destination Code Control (DCC) functionality.
n	Makes ASAC-related fields unavailable on the SIP Trunk Group screen. Therefore, the ASAC functionality is unavailable for calls. The default value is n.

Attendant Queue Announcement

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

The extension of the attendant queue announcement for incoming precedence calls that are above the Routine precedence level and are placed in the attendant waiting queue.

To enable **Attendant Queue Announcement**, you must set the **Annc Type** field on the Announcements/Audio Sources screen to *Integrated*.

Blocked Precedence Level

The extension of the Blocked Precedence Level announcement.

Busy, Not Equipped

The extension of the Busy, Not Equipped for Preemption announcement.

Service Interruption

The extension of the Service Interruption announcement.

Unauthorized Precedence Level

The extension of the Unauthorized Precedence Level announcement.

Vacant Code

The extension of the Vacant Code announcement.

PRECEDENCE CALLING-DIALED DIGIT ASSIGNMENT

Caution:

Do not change the default Precedence Calling dialed digits unless you are coordinating this change with other companion networks in your system. If the Precedence Calling digits are different across networks, the system does not properly process the calls. Each of the Precedence Calling digits must be different. You cannot use the same digit for two different precedence levels.

Attendant Diversion Timing (sec)

Valid Entry	Usage
10 to 99 blank	Controls how many seconds this type of call rings before the call is routed to the Remote Attendant Route String that consists of any valid telephone number on the network. Attendant Diversion Timing is usually a backup answering position for the remote attendant console. The Remote Attendant Route String does not raise the precedence level of the call.

Default Route Digit

Available only if Worldwide Numbering Dial Plan is enabled. A valid digit is required in this field.

Valid Entry	Usage
0	Voice call (the default value)
1	Circuit switched data call
2	Satellite avoidance call
3	(reserved)
4	(reserved)
5	Hotline voice grade call
6	Hotline data grade call
7	(reserved)
8	(reserved)
9	(reserved)

Related links

[Worldwide Numbering Dial Plan Active](#) on page 536

Default Service Domain

Valid Entry	Usage
0 to 16777215	The system service domain number. This number must be unique within a switching network. The system uses the system service domain to determine eligibility for precedence calling when interswitch precedence calls over non-ISDN trunks occur.

Flash

The digit assignment for Flash precedence level calls. The default is 1.

Flash Override

The digit assignment for Flash Override precedence level calls. The default is 0.

Immediate

The digit assignment for Immediate precedence level calls. The default is 2.

ISDN Precedence Call Timeout (sec)

Valid Entry	Usage
4 to 30	The timeout seconds used instead of the Precedence Call Timeout when the call is from a Multiple Level Precedence and Preemption (MLPP) ISDN-PRI trunk. Default is 30 seconds.

Line Load Control Restriction Level

Determines what stations, based on their COR, are restricted from originating calls.

Valid Entry	Usage
0	Feature not active (no restrictions). This is the default.
2	Restrict stations with a COR assigned to LLC levels 2, 3, and 4.
3	Restrict stations with a COR assigned to LLC levels 3 and 4.
4	Restrict stations with a COR assigned to LLC level 4.

Precedence Call Timeout (sec)

Valid Entry	Usage
4 to 30	The number of seconds a precedence call remains in call waiting status before it is diverted. A busy user receives a precedence call waiting tone only if the incoming call cannot be connected and cannot preempt the user. The called party hears the tone every 10 seconds until answered or the administered time-out occurs. If ignored, the caller is diverted to an attendant or a call-forwarded station. The default is 30 seconds.

Preempt Emergency Call

If enabled, allows preemption of an Emergency 911 call made from a preemptable station by a higher precedence call.

 Note:

Emergency call preemption occurs only in case of trunk and endpoint preemption. In case of network preemption the emergency calls are not preempted, irrespective of the settings for `Preempt Emergency Call?` flag.

Priority

The digit assignment for Priority precedence level calls. The default is 3.

Remote Attendant Route String

A user-defined telephone to which a precedence call can be routed when no console or night telephone is administered. Accepts from 1 to 24 digits.

Routine

The digit assignment for Routine precedence level calls. The default is 4.

WNDP Emergency 911 Route String

This route string is outpulsed when a user dials either 911 and waits for the interdigit timeout, or dials 911 followed by #. This dialing option only works when the WNDP Flash FAC is 91. Use a trunk access code (TAC), the AAR or the ARS access code, a WNDP access code, or an extension. For a WNDP access code, use the access code for the lowest precedence calling level in the system. Accepts from 1 to 24 digits.

Note:

An Emergency/911 call is a call that routes using the ARS table with the call type defined as either “alrt” or “emer”.

Worldwide Numbering Dial Plan Active

Enables or disables the Worldwide Numbering Dial Plan. Disabled by default.

Chapter 87: Music Sources

Use this screen to define music sources for Tenant Partitions. Each of these music sources can be used by one or more Tenant Partitions. However, a partition can have only one music source.

*** Note:**

If you use equipment that rebroadcasts music or other copyrighted materials, you might be required to obtain a copyright license from or pay fees to a third party. You can purchase a Magic Hold system which does not require a license from Avaya LLC or Avaya Business Partners.

Available only if **Tenant Partitioning** is enabled for the system.

Example command: `change music-sources`

change music-sources				Page	1 of	7
MUSIC SOURCES						
Source No.	Type	Source	Description			
1:	<u>music</u>	Type: <u>ext</u> <u>360-5080</u>	<u>test</u>			
2:	<u>none</u>					
3:	<u>none</u>					
4:	<u>none</u>					
5:	<u>none</u>					
6:	<u>none</u>					
7:	<u>none</u>					
8:	<u>none</u>					
9:	<u>none</u>					
10:	<u>none</u>					
11:	<u>none</u>					
12:	<u>none</u>					
13:	<u>none</u>					
14:	<u>none</u>					
15:	<u>none</u>					

Related links

[Tenant Partitioning](#) on page 738

Description

A description of the administered music source. Accepts up to 20 alphanumeric characters. Available only if music or tone is administered for treatment for hold.

Related links

[Type \(Column\)](#) on page 538

Source

Available only if music is administered for treatment for hold.

Valid Entry	Usage
ext	Audio source extension for a single or group audio source.
group	A Music-on-Hold analog group number.
port	An analog or auxiliary trunk source location.

Related links

[Type \(Column\)](#) on page 538

Source No

The number assigned to this music source.

Type (Column)

Valid Entry	Usage
music	The type of treatment to be provided by the music source.
tone	Only one music source can use the tone value.
none	

Type (field)

Available only if music is administered for treatment for hold.

Valid Entry	Usage
ext group port	The source for the music on hold. The source can be an announcement extension, an audio group, or a port on a VAL board.  Note: A source identifier (extension number, audio group number, or port number) must be administered with the source type.

Related links

[Type \(Column\)](#) on page 538

Chapter 88: Network Facilities

Used to administer new network-provided service or feature names and corresponding ISDN PRI (network specific facilities information element) encodings, for call-by-call trunk groups. Values for pre-defined facilities are displayed at the top of the screen and are display-only. User-defined facilities and services can be entered in the fields below.

When **Usage Allocation Enhancements** is enabled for the system, the system displays the Network Facilities screen. Additional user-defined entries can be administered on this screen.

Example command: `change isdn network-facilities`

Facility Coding

The ISDN-specified value for this service or feature.

Facility Type

The facility type. For types 2 and 3, **Usage Allocation Enhancements** must be enabled for the system.

Valid Entry	Usage
0 - feature	Predefined features.
1 - service	Predefined services.
2 - incoming	An incoming-type user-defined entry.
3 - outgoing	An outgoing-type user-defined entry.

Related links

[Usage Allocation Enhancements](#) on page 738

Name

The name for the feature or service.

Chapter 89: Network Region Group

A network region group is a collection of network regions that share one network connection to the core and have infinite bandwidth between them. A network region group is used for bandwidth management.

To administer the fields on the Network Region Group screen, use the following command:

`change network-region-group n`, where *n* is the network region group number.

Name

Use this field to assign a unique name to the network region group. You can enter up to 20 alphanumeric characters. The name must not start or end with a blank space, but can contain internal blank spaces. The name is case-sensitive and must not contain the following characters: `<>^%$@#*!'?.`

This name must exactly match a location name administered through System Manager on the **Elements > Routing > Locations** page.

Do not set this field unless the **BW Management Option** field is set to `shared-SM`.

Chapter 90: Node Number Routing

Specifies the routing pattern associated with each node in a public or private network. Node Number Routing is a required capability for Extension Number Portability (ENP) and is associated with the Uniform Dial Plan (UDP).

Example command: `change node-routing n`, where *n* is the node number.

Node Number

The node number.

Partitioned Group Number

The partitioned group number associated with the node numbers being administered.

Route Pat

Valid Entry	Usage
1 to 2000 or blank	The routing pattern associated with the corresponding node number.

Chapter 91: Numbering-Private Format

Use this screen to specify the digits to be inserted in the calling number Information Element (IE), the connected number IE, and the QSIG party number for extensions in the Private Numbering Plan.

Communication Manager can generate private network numbers up to 15 digits long.

Communication Manager can receive and display private network numbers up to 21 digits long.

Use the Numbering-Private Format screen to:

- Administer the extensions of all calls traversing the SIP trunks.
- Configure the private numbering table.
- Populate the calls to extensions terminating on a SIP-integrated Modular Messaging.

To change the value of the fields on the Numbering-Private Format screen, use the following command:

change private-numbering *n*, where *n* is the length of the extension.

The command **change private-numbering 0** starts at the first entry.

Ext Code

In the case of a four-digit **Ext Len**, an **Ext Code** of 12 is the equivalent of all extensions of the screen 12xx, excluding any explicitly listed longer codes. If a code of 123 is also listed, the 12 code is equivalent of all extensions of the screen 12xx except extensions of the screen 123x. The coding precludes having to list all the applicable 12xx extensions.

Valid Entry	Usage
0 to 16 blank	Accepts up to 16 digits depending on the administered extension length. When 0 alone is entered, the administered extension length must be 1 and the DDD number must be 10 digits.
attd	Generates a private calling number for a call from the attendant group.

Related links

[Ext Len](#) on page 544

Ext Len

Valid Entry	Usage
0 to 16	The number of digits the extension can have.
blank	Corresponds to the extension lengths allowed by the dial plan.

Maximum Entries

The maximum number of private numbering entries that can be administered on the system.

Private Prefix

The number that is added to the beginning of the extension to form a Private Identification Number. The length of the prefix and the extension must at least equal the total length.

Total Administered

The number of private numbering entries currently administered on the system.

Total Len

Valid Entry	Usage
0 to 16	The total number of digits to send.

Trk Grp(s)

Communication Manager generates the station's identification number if **Ext Code** is administered, and this field is administered with the trunk group number carrying the call. Accepts one to seven digits. If blank, the identification numbers are independent on the trunk group carrying the call.

Related links

[Ext Code](#) on page 543

Chapter 92: Numbering — Public/Unknown Format

Use this screen to specify the required digits for the Calling Number Information Element (IE) and the Connected Number Information Element (IE), in addition to the QSIG Party number, for any extension in Public and/or Unknown Number Plans. Use this screen for Automatic Route Selection (ARS) public trunks as well as SIP trunks. This screen supports the ISDN Call Identification Display feature. This feature provides a name and number display for display-equipped stations within an ISDN network. The system displays the name and number of the caller on the telephone display of the called party and displays the name and the number of the called party on the telephone display of the caller.

You must administer this screen if either **Send Calling Number** or **Send Connected Number** is specified, or **Supplementary Service Protocol** is b on the Trunk Group screen.

You must administer the following in the public numbering table on the Numbering-Public/Unknown Format screen:

- Extensions of all calls traversing SIP trunks
- Extensions terminating on SIP integrated Modular Messaging. However, do not configure a prefix for the SIP trunk group by which the Modular Messaging is integrated.

Note:

If an entry applies to a SIP connection to Avaya Aura® Session Manager, the resulting number must be a complete E.164 number. Communication Manager automatically inserts the plus (+) digit.

You can use the following command to modify the fields on this screen:

`change public-unknown-numbering n`, where *n* is the extension length from 0 to 16.

Related links

[Format](#) on page 453

[Send Calling Number](#) on page 457

[Send Connected Number](#) on page 458

[Supplementary Service Protocol](#) on page 819

CPN Prefix

Use this field to enter the number that is added to the beginning of the extension to form a Calling Party Number (CPN) number.

You must enter only digits. You cannot enter leading spaces, or spaces in between the digits. You can enter up to 15 digits. Communication Manager formulates the CPN number in the following manner:

- If the length of **CPN Prefix** matches **Total CPN Length**, the extension number is not used to formulate the CPN number.
- If the number of digits in **CPN Prefix** plus the extension length exceeds the administered **Total CPN Length**, Communication Manager deletes excess leading digits of the extension when formulating the CPN number.
- If the number of **CPN Prefix** digits plus the extension length is less than the **Total CPN Length**, Communication Manager does not allow the entry.
- If the **Total CPN Length** is 0, no calling party number information is provided to the called party and no connected party number information is provided to the calling party.

 Note:

If you leave this field blank, the extension is sent unchanged.

Ext Code

In the case of a four-digit **Ext Len**, an **Ext Code** of 12 is the equivalent of all extensions of the screen 12xx, excluding any explicitly listed longer codes. If a code of 123 is also listed, the 12 code is equivalent of all extensions of the screen 12xx except extensions of the screen 123x. The coding precludes having to list all the applicable 12xx extensions.

Valid Entry	Usage
0 to 16 blank	Accepts up to 16 digits depending on the administered extension length. When 0 alone is entered, the administered extension length must be 1 and the DDD number must be 10 digits.
attd	Generates a private calling number for a call from the attendant group.

Related links

[Ext Len](#) on page 544

Ext Len

Valid Entry	Usage
0 to 16	The number of digits the extension can have.
blank	Corresponds to the extension lengths allowed by the dial plan.

Total CPN Len

Valid Entry	Usage
0 to 15	The number of digits the extension can have.
blank	Used when deleting an entry. This is the default.

Trk Grp(s)

Communication Manager generates the station's identification number if **Ext Code** is administered, and this field is administered with the trunk group number carrying the call. Accepts one to seven digits. If blank, the identification numbers are independent on the trunk group carrying the call.

Related links

[Ext Code](#) on page 543

Chapter 93: Off-PBX Telephone Mobile Feature Extensions

Example command: `change off-pbx-telephone mobile-feature-ext`

Mobile Call (CTI) Extension

A CTI call to this Mobile Feature Extension (MCE) creates an OPTIM call under CTI influence. A call to the MCE triggers an OPTIM extend-call from a desk phone to its mapped cell phone number and to the destination. All calls made using the MCE appear to the destination as if they were dialed from the desk phone.

Chapter 94: Partition Routing Table

Identifies routing for partition groups associated with an ARS analysis entry.

Example command: `change partition-route-table n`, where *n* is the routing index.

PGN 1 (through PGN 8)

The routing for each partition group associated with each route index number.

Valid Entry	Usage
1 to 2000 for all templates except 254 for Avaya Aura® Main Embedded template.	The route pattern used to route the call.
r1 to r32	The remote home numbering plan area table used to route the call.
node	Node number routing.
deny	Blocks the call.

Chapter 95: Personal CO Line Group

Sets up a personal central office line group.

Example command: `add personal-co-line n`, where *n* is the personal central office line group number.

Personal CO Line Group: page 1

CDR Reports

Valid Entry	Usage
y	All outgoing calls on this personal central office line group generate call detail records. If Record Outgoing Calls Only is disabled for CDR, incoming calls on this personal central office line group also generate call detail records.
n	Calls over this personal central office line group do not generate call detail records.
r (ring-intvl)	CDR records are generated for both incoming and outgoing calls. In addition, the following ringing interval CDR records are generated: • Abandoned calls: The system creates a record with a condition code of “H” indicating the time until the call was abandoned. • Answered calls: The system creates a record with a condition code of “G” indicating the interval from start of ring to answer. • Calls to busy stations: The system creates a record with a condition code of “I” indicating a recorded interval of 0.

Related links

[Record Outgoing Calls Only](#) on page 103

Coverage Path

Valid Entry	Usage
digits	The number of the call coverage path to use for incoming calls. For the maximum allowed entry, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .
t1 to t999	The number of a time-of-day coverage table.
blank	No coverage path is assigned. Assigning a coverage path is optional.

Data Restriction

Enables or disables data restriction that is used to prevent tones, such as call-waiting tones, from interrupting data calls. Data restriction provides permanent protection and cannot be changed by the telephone user. Cannot be assigned if **Auto Answer** is administered as all or acd. If enabled, whisper page to this station is denied.

Related links

[Auto Answer](#) on page 668

Group Name

A unique name that provides information about the personal CO line group. Accepts up to 27 characters.

This field should contain names that identify the vendor and function of the personal CO line group rather than the group type (WATS).

 Note:

Supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, 1600-Series, 9600-Series IP Deskphones, 9600-Series H.323 Deskphones, and Avaya J100 Series IP Phones. Unicode is also an option for the 2420J telephone when the **Display Character Set** is katakana. For more information about 2420J, see *2420 Digital Telephone User's Guide*.

Related links

[Display Character Set](#) on page 722

Group Number

The personal CO line group number.

Group Type

The type of personal central office line group. The fields that are displayed and available might change according to the personal central office line group type selected.

Valid Entry	Usage
CO	Typically used to connect Communication Manager to the local telephone company central office, but can also connect adjuncts such as external paging systems and data modules.
FX	A local telephone company central office (CO) trunk that connects the server running Communication Manager directly to a CO outside the local exchange area. Used to reduce long-distance charges if the organization averages a high volume of long-distance calls to a specific area code.

Table continues...

Valid Entry	Usage
WATS	Used to reduce long-distance bills when your organization regularly places many calls to a specific geographical area in North America. With outgoing WATS service, calls can route to certain areas ("WATS band") for a flat monthly charge. With incoming WATS trunks, toll-free calling can be used to call customers and employees.

Outgoing Display

With Outgoing Display, telephones can show the name and number of the personal central office line group used for an outgoing call before the call is connected.

Valid Entry	Usage
y	Displays the personal central office line group name and number.
n	Displays the digits the caller dials.

Security Code

The station security code that users must dial to retrieve voice messages and to use the Demand Print Message feature. Accepts from three to eight digits.

TAC

The trunk access code (TAC) that must be dialed to access the personal central office line group. A different TAC must be assigned to each personal central office line group. CDR reports use the TAC to identify each personal central office line group. The characters "*" and "#" can be used as the first character in a TAC. Accepts a one- to four-digit number.

TRUNK PARAMETERS

Analog Loss Group

Determines which administered two-party row in the loss plan applies to this personal central office line group if the call is carried over an analog signaling port in the personal central office line group.

Valid Entry	Usage
1 to 17	The index into the loss plan and tone plan. If values are administered other than in between 6 and 10 or 15 and 17, a warning message displays stating that the loss group might not be appropriate for this personal central office line group.

Answer Supervision Timeout

Valid Entry	Usage
0 to 250	The number of seconds Communication Manager waits before it acts as though answer supervision has been received from the far-end. During a cut-through operation, timing begins after each outgoing digit is sent and timing ceases after the far-end sends answer supervision. On senderized operation, the timer begins after the last digit collected is sent.

 Note:

This field's setting does not override answer supervision sent from the network.

Related links

[Receive Answer Supervision](#) on page 558

[Administer Timers](#) on page 445

Call Still Held

If enabled, the system prevents glare by extending the Incoming Glare Guard timer and delaying an outgoing seizure of a trunk for at least 140 seconds after it is released from an incoming call. This field is used when the receiving end media server or switch initiates the disconnection of incoming calls. This field affects only TN438B, TN465B, and TN2147 ports and is used primarily when the **Country** code is 2.

Available only for co or fx personal central office line groups.

Related links

[Group Type](#) on page 552

Charge Conversion

Available only for outgoing or two-way CO, FX, and WATS trunk groups.

Valid Entry	Usage
1 to 64 500	Communication Manager multiplies the number of charge units by the value of this field and displays it as a currency amount. Without a value in this field, Communication Manager displays the number of charge units without converting it to currency.

Related links

[Direction](#) on page 436

[Trunk Direction](#) on page 558

Charge Type

Text string used to describe charges related to a telephone call. The system displays these words or characters on telephone screen after the charge amount. Typically uses either the currency

symbol or the charge type, but not both. Accepts up to seven characters. Embedded spaces count as characters.

Available only for outgoing or two-way CO, FX, and WATS personal central office line groups.

Related links

[Direction](#) on page 436

Country

The country code that corresponds to the protocol used by the local telephone company central office (CO) where the personal central office line group terminates.

Available only for personal central office line groups that connect Communication Manager to a CO in the public network — CO, FX, and WATS personal central office line groups.

Valid Entry	Usage
1 to 25, except for 19	For a list of country codes, see the <i>Country code table</i> .

Related links

[Country options table](#) on page 718

[Trunk Gain](#) on page 558

[Trunk Termination](#) on page 560

Currency Symbol

The telephone displays the currency symbol before the charge amount. Accepts from one to three characters. Leading and embedded spaces count as characters.

Available only for outgoing or two-way CO, FX, and WATS personal central office line groups.

Related links

[Trunk Direction](#) on page 558

[Direction](#) on page 436

Decimal Point

The appropriate representation for a decimal point as seen on the telephone displays. Available only with outgoing or two-way CO, FX, and WATS personal central office line groups. Also available for isdn personal central office line groups, if the **Charge Advice** field is set to a value other than none.

Note:

If the received charge contains no decimals, no decimal point is displayed (that is, the administered decimal point is ignored for charge information received with no decimals). On a QSIG personal central office line group, unlike other personal central office line groups, the **Decimal Point** field is independent of whether the calling screen displays a decimal point.

Instead, it tells what symbol should be displayed if the QSIG AOC received has a 1/10 or 1/100 or 1/1000 Multiplier.

Valid Entry	Usage
comma	If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a comma as the decimal point. Divides the charge value by 100.
period	This is the default. If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a period as the decimal point. Divides the charge value by 100.
none	No decimal point is displayed.

Related links

[Charge Advice](#) on page 435

[Direction](#) on page 436

[Trunk Direction](#) on page 558

Digital Loss Group

Valid Entry	Usage
1 to 19	Determines which administered two-party row in the loss plan applies to this personal central office line group if the call is carried over a digital signaling port in the personal central office line group. If values other than 18 or between 11 and 15 are administered, a warning message displays stating that the loss group is inappropriate for this personal central office line group.

Disconnect Supervision-In

Indicates whether Communication Manager receives disconnect supervision for incoming calls over this personal central office line group.

Available only for incoming or two-way personal central office line groups.

Valid Entry	Usage
y	Allows trunk-to-trunk transfers involving trunks in this group. The far-end server or switch sends a release signal when the calling party releases an incoming call, and the far-end server or switch is responsible for releasing the trunk. Enhances Network Call Redirection.
n	The far-end server or switch does not provide a release signal, the hardware cannot recognize a release signal, or timers are preferred for disconnect supervision on incoming calls. Prevents trunk-to-trunk transfers involving trunks in this group.

Caution:

In general, U.S. local telephone company central offices provide disconnect supervision for incoming calls but not for outgoing calls. Public networks in most other countries do

not provide disconnect supervision for incoming or outgoing calls. Check with the network services provider.

Related links

[Direction](#) on page 436

[Trunk Direction](#) on page 558

Outgoing Dial Type

Sets the method used to transmit digits for an outgoing call. Usually, this method should match what the local telephone company central office provides.

Available for CO, FX, and WATS personal central office line groups. This field is also available for the trunks with the **group type** as Tie and when the **Trunk Signaling Type** is blank, cont, or dis.

Valid Entry	Usage
tone	Uses Dual Tone Multifrequency (DTMF) addressing, also known as “touchtone” in the U.S. With this, the personal central office line group can support both DTMF and rotary signals. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use tone or mf.
rotary	With this, only the dial pulse addressing method can be used by non-touch tone telephones. For example, this value is appropriate for an internal full touch tone system and for a connection to a local telephone company central office that only supports rotary dialing.
automatic	For tie trunks if the Trunk Signaling Type is not administered. This provides “cut-through” operation to outgoing callers who dial a trunk access code, connecting them directly to local telephone company central office dial tone and bypassing any toll restrictions administered on Communication Manager.
mf	Use this option if the Trunk Signaling Type field is blank. The system displays this field only if you set the Multifrequency Signaling field on the System Parameters Customer-Options screen to y. You cannot use this option if the Used for DCS field is y. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use tone or mf.
r1mf	Enter r1mf for CAMA trunk groups. It is the only outgoing dial type allowed on CAMA trunk groups. Enter r1mf to allow Russian MF Packet Signaling on outgoing trunks. Russian MF Packet Signaling carries calling party number and dialed number information. Group type field must be set to co.
mf2/6	Use this entry for signaling in Spain. This signaling type provides normal public network service and a virtual private network digital trunk service called IBERCOM. Use this entry only if the Trunk Signaling Type field is blank. The system displays this entry only if you enable Multifrequency Signaling for the system. The system does not display this entry if the trunk is used for DCS.

Related links

[Group Type](#) on page 437

[Multifrequency Signaling](#) on page 735

[Trunk Signaling Type](#) on page 803

Prefix-1

If enabled, the prefix “1” is added to the beginning of the digit string for outgoing calls. Use this field for outgoing and two-way personal central office line groups handling long distance service. Disable for personal central office line groups in AAR or ARS route patterns.

Available only for CO and FX personal central office line groups.

Receive Answer Supervision

If enabled, the Communication Manager provides answer supervision for a trunk group. For Outbound Call Management applications, use for trunks supporting network answer supervision. For trunks that do not receive a real answer, this field determines when the CallVisor Adjunct-Switch Application Interface (ASAI) connect event is sent.

Related links

[Answer Supervision Timeout](#) on page 445

[Administer Timers](#) on page 445

Trunk Direction

Important:

This setting must match the provider's settings.

The direction of the traffic on this personal central office line group.

Valid Entry	Usage
incoming	Traffic on this personal central office line group is incoming.
outgoing	Traffic on this personal central office line group is outgoing.
two-way	Traffic on this personal central office line group is incoming and outgoing. Required for Network Call Redirection .

Related links

[Charge Conversion](#) on page 451

Trunk Gain

Specifies the amplification applied to the trunks in this group. With the values administered for **Trunk Termination** and **Country** code, the value in this field also determines the input and trans-hybrid balance impedance for TN465B, TN2146, TN2147, and TN2184 ports. All other CO circuit packs are set automatically to high.

Valid Entry	Usage
high	Used if users complain of low volume.
low	Used if users complain of squeal or feedback.

Related links

[Trunk Termination](#) on page 560

[Country](#) on page 555

Trunk Port

The port to which the trunk is connected.

Trunk Type

Controls the seizure and start-dial signaling used on this personal central office line group. Entries in this field vary according to the function of the personal central office line group and must match the corresponding setting on the far-end server or switch.

Available only for CO, FX, and WATS personal central office line groups.

Valid Entry	Usage
ground-start	Use ground-start signaling for two-way trunks whenever possible. Ground-start signaling avoids glare and provides answer supervision from the far end.
loop-start	In general, loop-start signaling is used only for one-way trunks. Loop-start signaling is susceptible to glare and does not provide answer supervision.
auto/auto auto/delay auto/immed auto/wink	The term before the slash tells Communication Manager how and when it receives incoming digits. The term after the slash tells Communication Manager how and when it should send outgoing digits. • auto — Used for immediate connection to a single preset destination (incoming central office trunks, for example). No digits are sent, because all calls terminate at the same place. • delay — The sending server running Communication Manager does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end server or switch, indicating that it is ready to receive the digits. • immed — The sending server running Communication Manager sends digits without waiting for a signal from the far-end server or switch. • wink — The sending server running Communication Manager does not send digits until it receives a wink start (momentary off-hook) signal from the far-end server or switch, indicating that it is ready to receive the digits.
2-wire-ac 2-wire-dc 3-wire	These entries are used with local telephone company central office (CO) trunks in Russia. The specific CO should match one of these values. Available only if the Country code is 15 and the CO trunks use ports on a TN2199 circuit board.

Related links

[Country](#) on page 555

Trunk Termination

Adjusts the impedance of the personal central office line group for optimal transmission quality.

Valid Entry	Usage
600ohm	The distance to the local telephone company central office (CO) or to the server at the other end of the trunk is less than 3,000 feet.
rc	The distance to the CO or to the server at the other end of the trunk is more than 3,000 feet.

Personal CO Line Group: page 2

Ext

The extension of telephones that have a **CO Line** button.

Name

The name assigned to telephones that have a **CO Line** button.

Chapter 96: Pickup Group

Implements call pickup groups. A pickup group is a group of users authorized to answer calls to a telephone extension within that group of users. A telephone extension can belong to only one pickup group, and a pickup group can have up to 50 extensions.

Example command: `add pickup-group n`, where *n* is the pickup group number.

GROUP MEMBER ASSIGNMENTS

Ext

The extension of the pickup group number. A VDN cannot be assigned to a Call Pickup group.

Name

The name assigned to the pickup group number extension.

Extended Group Number

Available only if the **Extended Group Call Pickup** is administered as flexible.

Valid Entry	Usage
1 to 100 blank	The Extended Group number. The extended group is a collection of pickup groups that can answer calls to a telephone extension within that other pickup group.

Related links

[Extended Group Call Pickup](#) on page 251

Group Number

The pickup group number.

Chapter 97: Policy Routing Table

With Policy Routing Table, you can distribute calls among a set of call centers based on specified percent allocation. Various types of incoming calls that arrive at a particular VDN can be directed to a Policy Routing Table (PRT) instead of to a vector. The PRT then distributes the calls to the administered Route-to VDNs based on the specified percent allocation targets. Use this screen to implement and monitor percentage allocation routing by assigning destination routes and target percentages.

Example command: `change policy-routing-table n`, where *n* is the policy routing table number.

Actual %

The actual percent of total calls routed to a VDN. Calculated to six decimal places, but only the first decimal place is displayed.

Call Counts

The current number of calls routed to a VDN.

Index

The sequential number of the row in a Policy Routing Table.

Name

The name of the Policy Routing Table (PRT). Accepts a string of up to 15 alphanumeric characters.

Number

The number of the Policy Routing Table (PRT).

Period

The period for resetting the call counts and actual percentages.

Valid Entry	Usage
100_count	Resets the call counts (and displayed %) when total calls for the PRT reach 100. At this point, the total calls match the target routing pattern percentages. This ensures that the routing points have equal distribution of calls all the time. This is the default.
max_count	Call counts are maintained until calls delivered to at least one of the VDNs exceed 65,400. At this point, calls continue to be distributed over the VDNs but the call counts are reset when the actual percentages equal the targets for all of the VDNs at the same time.
Half-hour	Resets the call counts at the top of the hour and at the 30-minute point.
hour	Resets the call counts at the top of the hour.
daily	Resets the call counts at midnight, every night.
weekly	Resets the call counts at midnight on Saturday.

Route-to VDN

VDN extension to which calls are to be routed. Accepts up to 15 extensions containing 1 to 16 digits.

Target %

Valid Entry	Usage
0 to 100	The target percent of total calls to be routed to a VDN. Use whole numbers only, no decimal fractions.

Totals

Displays totals for **Target %** and **Call Counts** for all the assigned VDNs in the PRT. The total for **Target %** is always 100 for form submittal.

Type

The type of algorithm the Policy Routing Table (PRT) supports.

VDN Name

The name of a VDN, if a name has been assigned previously.

Chapter 98: Precedence Routing Digit Analysis Table

Communication Manager compares dialed numbers with the dialed strings in this table and determines the route pattern of an outgoing Multiple Level Precedence and Preemption (MLPP) call.

Example command: `change precedence-routing analysis n`, where *n* is the precedence-routing digit analysis table number.

Destination Code Control

 Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Use Destination Code Control (DCC) to restrict outbound calls to a specific called number or a range of numbers to keep the trunk free for important calls.

DCC limits outbound calls from the outgoing budget (IPBo).

To enable the DCC feature you must first activate DCC by using FAC in COS.

Valid entry	Usage
y	DCC is activated.
n	DCC is not activated. The default value is n.

Dialed String

Communication Manager matches the dialed numbers with the entry in the **Dialed String** field that most closely matches the dialed number. You can enter up to 18 digits that the call-processing server analyzes. You can also enter the wildcard characters, `x` and `X`.

Max

Use this field to specify the maximum number of user-dialed digits that the system must collect to match the dialed string.

Min

Use this field to specify the minimum number of user-dialed digits that the system must collect to match the dialed string.

Percent Full

This field displays the percentage of system memory resources used for the AAR and the ARS digit analysis. For more information about the system memory resources, see *Avaya Aura® Communication Manager System Capacities Table*.

Percentage IPBO Blocked

 Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

With **Percentage IPBO Blocked**, Communication Manager supports blocking a percentage of calls to a destination for the DCC feature. You can enter a value from 1% to 100%. The default value is 100%.

Preempt Method

The preemption method used by the server running Communication Manager for this dialed string.

Valid Entry	Usage
group	The system checks the first trunk group in the route pattern to determine if any trunks are idle. If the system finds an idle trunk, the system connects the call. This is the default.
route	The system checks each trunk group in the route pattern to determine if any trunks are idle. If the system finds an idle trunk, the call is connected.

Chapter 99: Precedence Routing Digit Conversion Table

Assigns the Precedence Routing digit conversion. Digit conversion takes digits dialed on incoming calls and converts the digits to local telephone numbers, usually extension numbers.

Example command: `change precedence-routing digit-conversion n`, where *n* is the precedence routing digit conversion table number.

Conv

Use this field to enable additional digit conversion.

Del

Use this field to enter the number of digits that the system must delete from the starting of the dialed string.

Matching Pattern

Use this field to specify the pattern that the system must use to match the dialed number. The pattern can be up to 18 characters long and consist of digits and the wildcard characters, `x` and `X`.

Max

Use this field to specify the maximum number of user-dialed digits that the system must collect to match the dialed string.

Min

Use this field to specify the minimum number of user-dialed digits that the system must collect to match the dialed string.

Net

Valid Entry	Usage
ext	Communication Manager routes the call as an extension using the Dial Plan Analysis and Uniform Dial Plan forms, as appropriate.
pre	Precedence routing. Uses the Precedence Analysis Tables to route the call.

Replacement String

Use this field to specify the digit string that must replace the deleted portion of the dialed number.

Valid entry	Usage
blank	To delete the digits without replacement.
0 to 9, *	To type the digit string. The digit string can have up to 18 characters.
#	To indicate the end of dialing.

Chapter 100: PRI Endpoint

Administers PRI Endpoints for the Wideband Switching feature.

 Note:

A PRI Endpoint with a width greater than 1 can be administered only if Wideband Switching has been enabled for the system.

A PRI Endpoint is an endpoint application connected to line-side ISDN-PRI facilities and has standard ISDN-PRI signaling interfaces to the system.

A PRI Endpoint is defined as 1 to 31 adjacent DS0s/B-channels, addressable via a single extension, and signaled via a D-channel (Signaling Group) over a standard T1 or E1 ISDN-PRI interface.

Example command: `add pri-endpoint n`, where *n* is the extension number.

Related links

[Wideband Switching](#) on page 738

COR

Valid Entry	Usage
0 to 995	The class of restriction (COR) used to determine calling and called party privileges.

COS

Valid Entry	Usage
0 to 15	The Class of Service (COS) used to determine the features that can be activated by, or on behalf of, the endpoint.

Extension

The extension number used to access the PRI endpoint.

Maintenance Tests

If enabled, runs hourly maintenance tests on this PRI Endpoint.

Name

The name of the endpoint. Accepts up to 27 alphanumeric characters.

Originating Auto Restoration

If enabled, automatically restores calls originating from this PRI Endpoint (while maintaining endpoint call status) in the case of network failure if the call is over SDDN network facilities.

Signaling Group

Valid Entry	Usage
digits or blank	The D-channel or D-channel pair that provides the signaling information for the set of B-channels that make up the PRI Endpoint. For the maximum allowed entry, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .

Simultaneous Calls

If enabled, specifies that multiple simultaneous calls can be placed to/from the PRI Endpoint.

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Width

Valid Entry	Usage
1 to 31	The number of adjacent DS0 ports beginning with the administered Starting Port, that make up the PRI Endpoint. This field cannot be blank. A width of 6 defines a PRI Endpoint that can support data rates up to 384 Kbps.

WIDEBAND SUPPORT OPTIONS

Contiguous

Available only if “N by DS-zero” (NXDS0) multi-rate service is enabled.

Valid Entry	Usage
y	Specifies the “floating” scheme. NXDS0 calls are placed on a contiguous group of B-channels large enough to satisfy the requested bandwidth without constraint on the starting channel (no fixed starting point trunk). Unavailable with H0 ISDN information transfer rate.
n	Specifies the “flexible” scheme. NXDS0 calls are placed on any set of B-channels on the same facility as long as the requested bandwidth is satisfied. There are no constraints, such as contiguity of B-channels or fixed starting points

Related links

[NxDS0](#) on page 471

[NXDS0](#) on page 573

H0

If enabled, specifies the ISDN information transfer rate for 384 Kbps of data, which is comprised of six B-channels. When a PRI Endpoint is administered to support H0, the hunt algorithm to satisfy a call requiring 384 Kbps of bandwidth uses a fixed allocation scheme.

H11

If enabled, specifies the ISDN information transfer rate for 1536 Kbps of data, which is comprised of 24 B-channels. When a PRI Endpoint is administered to support H11, the hunt algorithm to satisfy a call requiring 1536 Kbps of bandwidth uses a fixed allocation scheme.

H12

If enabled, specifies the ISDN information transfer rate for 1920 Kbps data, which includes 30 B-channels. When a PE is administered to support H12, the hunt algorithm to satisfy a call requiring 1920 Kbps of bandwidth uses a fixed allocation scheme.

NXDS0

If enabled, specifies the NXDS0 multi-rate service.

Chapter 101: Processor Channel Assignment

Assigns each local processor channel to an interface link channel, and defines the information associated with each processor channel on an Ethernet link.

*** Note:**

You cannot remove a service from this screen if that service has overrides defined on the Survivable Processor screen.

Example command: `change communication-interface processor-channels`

Appl

Specifies the server application type or adjunct connection used on this channel.

Valid Entry	Usage
audix	Voice Messaging.
ccr	Contact Center Reporting, now known as Avaya IQ.
dcs	Distributed Communication System.
fp-mwi	ISDN Feature Plus Message Waiting Indication. This channel passes message waiting light information for subscribers on the messaging system, from a messaging adjunct on a main switch for a telephone on a satellite switch. The terminating location (far end) of this channel must be a Communication Manager system compatible with ISDN Feature Plus proprietary protocol.
gateway	Supports an X.25 connected AUDIX connected to an ISDN DCS network.
gateway-tcp	Supports a TCP-connected voice messaging system connected to an ISDN DCS network.
mis	Management Information System, otherwise known as the Call Management System.
qsig-mwi	QSIG Message Waiting Indication. Used with a QSIG-based interface to a messaging system, this channel passes message waiting light information for subscribers on the messaging system.

Destination Node

Identifies the server or adjunct at the far end of this link. The destination node can be an adjunct name, server name, far end IP address, or node name for services local to the Avaya server. For ppp connections, match the Destination Node Name administered for the ppp data module.

Destination Port

Valid Entry	Usage
5000 to 64500	The number of the destination port.
0	Any port can be used.

Enable

Enables or disables this processor channel on the main server.

Gtwy to

A number that identifies the processor channel to which the specified processor channel is serving as a gateway.

Interface Channel

The channel number or the TCP/IP listen port channel to carry this processor (virtual) channel.

Valid Entry	Usage
5000 to 64500	For ethernet or ppp. For TCP/IP, interface channel numbers are in the range 5000 to 64500. The value 5001 is recommended for CMS, and 5003 is recommended for DCS.
0	Any port can be used.

Interface Link

Valid Entry	Usage
1 to 254	The physical link carrying this processor (virtual) channel.
pv4 or pv6	Communication Manager's Processor Ethernet interface used for adjunct connectivity.
blank	Not administered.

Mach ID

Valid Entry	Usage
1 to 63 for MWI 1 to 63 for DCS 1 to 99 for voice messaging blank	The destination server ID defined on the dial plan of the destination server.

Related links

[Mach ID](#) on page 632

Mode

Valid Entry	Usage
c(lient) s(erver) blank	Indicates whether the IP session is passive (client) or active (server). This field must be blank if the interface link is procr-intf. This field cannot be blank if the type of interface link is ethernet or ppp.

Proc Chan

The number assigned to each processor channel.

Session - Local/Remote

Valid Entry	Usage
1 to 384 blank	The Local and Remote Session numbers. For each connection, the Local Session number on the Avaya server must equal the Remote Session number on the remote server and vice versa. It is allowed, and sometimes convenient, to use the same number for the Local and Remote Session numbers for two or more connections. Local and Remote Session numbers must be consistent between endpoints.

Chapter 102: QSIG to DCS TSC Gateway

This screen determines when and how to convert messages from a QSIG NCA-TSC to an administered AUDIX NCA-TSC. This screen maps the QSIG subscriber number to the appropriate AUDIX signaling group and TSC index.

Available only if **Interworking with DCS** is enabled for the system.

Example command: `change isdn qsig-dcs-tsc-gateway`

Related links

[Interworking with DCS](#) on page 744

Sig Grp

Valid Entry	Usage
digits	The assigned signaling group number. For the maximum allowed entry, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .

Subscriber Number

A subscriber number up to 20 characters in length. Accepts *, x, and X as wildcard characters.

Note:

If the Subscriber Number is local, the form does not send any DCS messages over the listed Sig Grp/TSC Index. Therefore, you can put a block entry like '2xxx' with the understanding that if extension 2001 is local, no DCS message will be sent.

TSC Index

The TSC Index for each machine ID.

Valid Entry	Usage
1 to 64	The assigned signaling group number for the qsig-mwi application type.

Chapter 103: Reason Code Names

Assigns names to reason codes. Each reason code can have a different name for Aux Work and for Logout. The system displays the screen only when you set the **Two-Digit Aux Work Reason Codes** to y.

 Note:

Logout reason codes can only be in the range of 0 to 9, even if **Two-Digit Aux Work Reason Codes** is active.

Example command: `change reason-code-names`

Aux Work

The name associated with a reason code when the agent uses the reason code to enter Aux Work mode. Accepts up to 16 alphanumeric characters.

Default Reason Code

The default reason code names. A separate name can be administered for the Aux Work Reason Code of 0 and for the Logout Reason Code of 0. If an agent changes to Aux Work mode and the Aux Work Reason Code Type is set to none, the agent is put into Aux Work mode with the default Aux Work reason code, even if a different reason code is administered for the **Aux** button. If an agent logs out when the Logout Reason Code Type is set to none, the agent is logged out with the default Logout reason code.

Accepts up to 16 alphanumeric characters.

Interruptible

Specifies whether or not each reason code is interruptible.

 Note:

The Default Reason Code, Auto-answer IP Failure Aux Work Reason Code, and Maximum Agent Occupancy Aux Work Reason Code cannot be made interruptible.

Logout

The name associated with a reason code when the agent uses the reason code to log out.
Accepts up to 16 alphanumeric characters.

Chapter 104: Remote Access

Implements the Remote Access feature. Remote Access permits a caller located outside the system to access the system through the public or private network and then use the features and services of the system.

Security alert:

The Remote Access feature, when properly administered, enables the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes, and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed concerning the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of remote access features. In such an event, applicable tariffs require the customer pay all network charges for traffic. Avaya will not be responsible for such charges, and will not make any allowance or give any credit for charges that result from unauthorized access.

Example command: `change remote-access`

Authorization Code Required

If enabled, requires an authorization code be dialed by Remote Access users to access the system's Remote Access facilities. The use of an authorization code in conjunction with a barrier code increases the security of the Remote Access feature.

Barrier Code

The number users must dial to use Remote Access. Can be used with an authorization code. Accepts a four- to seven-digit number in any combination of digits. Must conform to the administered **Barrier Code Length**. The none value is required if a **Barrier Code Length** is not administered.

Related links

[Barrier Code Length](#) on page 583

Barrier Code Length

Valid Entry	Usage
4 to 7 blank	The length of the barrier code. A barrier code length of 7 provides maximum security.

Calls Used

The number of calls placed using the corresponding barrier code. A usage that exceeds the expected rate indicates improper use.

COR

Valid Entry	Usage
0 to 995	The number of the class of restriction (COR) associated with the barrier code. To provide maximum security, assign the most restrictive COR that provides only the level of service required.

COS

Valid Entry	Usage
0 to 15	The number of the class of service (COS) associated with the barrier code. To provide maximum security, assign the most restrictive COS that provides only the level of service required.

Disable Following a Security Violation

Available only if SVN Authorization Code Violation Notification is enabled for the system.

Valid Entry	Usage
y	Disables the Remote Access feature following detection of a Remote Access security violation. The system administrator can re-enable Remote Access using the <code>enable remote-access</code> command.

Table continues...

Valid Entry	Usage
n	Enables the Remote Access feature following detection of a Remote Access security violation.

Related links

[SVN Authorization Code Violation Notification Enabled](#) on page 602

Expiration Date

The date the barrier code expires. Assign an expiration date based on the expected length of time the barrier code is needed. For example, if the barrier code is expected to be used for a two-week period, assign a date two weeks from the current date. If both an **Expiration Date** and **No. of Calls** are assigned, the corresponding barrier code expires when the first of these criteria is satisfied.

No. of Calls

Valid Entry	Usage
1 to 9999 blank	The number of Remote Access calls that can be placed using the associated barrier code. If both an Expiration Date and No. of Calls are assigned, the corresponding barrier code expires when the first of these criteria is satisfied.

Related links

[Expiration Date](#) on page 584

Permanently Disable

If enabled, permanently blocks remote access to the administration interface. Reactivation of remote access to the interface requires the intervention of Avaya Services.

Remote Access Dial Tone

Enables or disables the user hearing dial tone as a prompt for entering the authorization code. Disabling this feature provides maximum security. Available only if Authorization Codes are required for Remote Access.

Related links

[Authorization Code Required](#) on page 582

Remote Access Extension

The extension assigned to handle Remote Access calls. The remote access extension is used as if it was a DID extension. Only one DID extension can be assigned as the remote access extension. Calls to that number are treated the same as calls on the remote access trunk.

When a trunk group is dedicated to Remote Access, the remote access extension number is administered as the trunk group's incoming destination.

A Vector Directory Number (VDN) extension cannot be used as the remote access extension.

Can be blank if no barrier codes are administered.

Related links

[Incoming Destination](#) on page 796

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Chapter 105: Remote Call Coverage Table

Provides automatic redirection of certain calls to alternate non-local answering positions in a coverage path.

Non-local numbers can be any ARS or AAR number, any number on the public network, any international number, or a UDP/DCS extension up to 16 digits or blank, which includes any ARS/AAR facility access code, any trunk dial access code (TAC), long distance dialing code, or international dial code.

Example command: `change coverage remote n`, where *n* is the remote coverage number.

change coverage remote 1

Page 1 of 23

REMOTE CALL COVERAGE TABLE

ENTRIES FROM 1 TO 1000

01: _____	16: _____	31: _____
02: _____	17: _____	32: _____
03: _____	18: _____	33: _____
04: _____	19: _____	34: _____
05: _____	20: _____	35: _____
06: _____	21: _____	36: _____
07: _____	22: _____	37: _____
08: _____	23: _____	38: _____
09: _____	24: _____	39: _____
10: _____	25: _____	40: _____
11: _____	26: _____	41: _____
12: _____	27: _____	42: _____
13: _____	28: _____	43: _____
14: _____	29: _____	44: _____
15: _____	30: _____	45: _____

01–1000

The identifier for the destination coverage point. Accepts up to 16 digits. Two places are used for L, D, %, and ,.

Valid Entry	Usage
*	DTMF digit asterisk
#	DTMF digit pound
L	Uses a coverage point only when in Survivable Remote or Survivable Core Server mode
D	Represents the called extension digits
%	Remaining digits are for end-to end signaling
,	Pauses for 1.5 seconds
blank	Not administered

Chapter 106: Remote Office

Supports the Remote Office feature, an arrangement whereby a user can set up a remote office without having an on-premises physical desk-set. An R300 is issued to connect remote DCP and analog telephones, IP telephones, and H.323 trunks to the Communication Manager server using IP.

Example command: `change remote-office n`, where *n* is the assigned remote office number.

Location

Valid Entry	Usage
1 to 2000	(Depending on your server configuration, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .) Assigns the location number to the remote office comprised of the associated time zone and the appropriate numbering plan. See the Location sections in <i>Avaya Aura® Communication Manager Feature Description and Implementation</i> , for the other ways to assign a location, and for a list of features that use location.
blank	Obtains the location from the PROCR or the gateway that the endpoint registered with. By default, the value is blank.

Network Region

Valid Entry	Usage
1 to 2000	The network region assigned to all stations supported at this remote office.
blank	

Node Name

The node name of the remote office.

Site Data

Any required site information. Accepts up to 30 alphanumeric characters.

Chapter 107: RHNPA Table

Defines route patterns for specific three-digit codes, usually direct distance dialing (DDD) prefix numbers.

Example command: `change rhnpa n`, where *n* is the prefix number.

CODES

The 100-block of codes being administered.

Code-Pattern Choice Assignments

Valid Entry	Usage
1 to 24	A pattern choice number associated with each office code.

Pattern Choices

Valid Entry	Usage
1 to 999 1 to 2000 blank	The maximum number of route patterns that you can configure depends on your system configuration. The route pattern number associated with each code. If you use one pattern for most of the codes, assign that pattern to choice 1.

RHNPA TABLE

The RHNPA table number.

Chapter 108: Route Pattern

Use this screen to define route patterns. Each route pattern contains a list of trunk groups that Communication Manager uses to route a call. The maximum number of route patterns and trunk groups allowed depends on the system configuration and the memory available in the system.

Use this screen to convert the dialed number into a format that the network uses for routing the call. For example, you can:

- Insert or delete digits when Automatic Alternate Routing (AAR) or Automatic Route Selection (ARS) calls route over different trunk groups.
- Convert an AAR number into an international number.
- Insert an area code into an AAR number to convert an on-network number to a public network number.
- Insert the dial access code for an alternative carrier. The system uses this alternative carrier when a call directly accesses a local telephone company central office (CO) and if the long distance carrier that the CO provides is unavailable.

Network regions are added to the Route Pattern form. In the Route Pattern form, you can specify the network region that can be used against each route preference. Network region is applicable for SIP trunks and Off-PBX Telephone Integration and Mobility (OPTIM) end points like EC500 and Telecommuter.

A network region can be populated for each of the route patterns. This network region will be used only when routing SIP trunk calls (SRE Calls) over the route preference (Trunk Group). Network region field on the route pattern applies only for trunk and OPTIM such as EC500 and telecommuter calls but not for SIP station calls. This overrides the network region field on the SIP signaling group for the above mentioned applicable calls.

To modify the value of the fields on the Route Pattern screen, use the following command:

`change route-pattern n`, where *n* is the route pattern number.

Band

A number that represents the OUTWATS band number (US only).

Available only if **Services/Features** is administered as outwats-bnd and **ISDN-PRI** or **ISDN-BRI Trunks** are enabled for the system. **Band** is required by Call-by-Call Service Selection.

Related links

[ISDN-BRI Trunks](#) on page 734

[ISDN-PRI](#) on page 734

[Service/Feature](#) on page 599

BCC Value

Identifies the type of call appropriate for a trunk group. Available only if **ISDN-PRI** or **ISDN-BRI Trunks** are enabled for the system.

Valid Entry	Usage
y/n	If enabled, (0, 1, 2, 3, 4, or W) indicates the BCC value is valid for the routing preference. A trunk group preference can have more than one BCC.

BCC values:

BCC Value	Description
0	Voice-Grade Data and Voice
1	56-kbps Data (Mode 1)
2	64-kbps Data (Mode 2)
M	Multimedia call
4	64-kbps Data (Mode 0)
W	128 to 1984-kbps Data (Wideband)

Related links

[ISDN-BRI Trunks](#) on page 734

[ISDN-PRI](#) on page 734

BCIE (Bearer Capability Information Element)

Determines how to create the ITC codepoint in the setup message. Applies to ISDN trunks. Available only if the Information Transfer Capability (ITC) is administered as both.

Valid Entry	Usage
ept	endpoint
unr	unrestricted

Related links

[ITC \(Information Transfer Capability\)](#) on page 595

CA-TSC Request

For the DCS+ features, Communication Manager sends a Call-Associated Temporary Signaling Connection (CA-TSC) request to send messages other than normal call control messages.

Valid entry	Usage
as-needed	Communication Manager sends the CA-TSC request only when needed.
at-setup	Communication Manager sends the CA-TSC request for every B-channel call, whether or not it is needed.
none	Communication Manager does not send the CA-TSC request Use this entry to allow Communication Manager to send Non-Call-Associated Temporary Signaling Connection (NCA-TSC) requests.

DCS/QSIG Intw

Enables or disables DCS/QSIG Voice Mail Interworking. Available only if **Interworking with DCS** is enabled for the system.

Related links

[Interworking with DCS](#) on page 744

FRL

Valid Entry	Usage
0 to 7	The Facility Restriction Level (FRL) associated with the group routing preference. 0 is the least restrictive, and 7 is the most restrictive. The calling party's FRL must be greater than or equal to this FRL to access the associated trunk group.

Security alert:

For system security reasons, use the most restrictive FRL possible.

Grp No

Valid Entry	Usage
1 to 2000	The trunk group number associated with the routing preference.

Hop Lmt

The number of hops for each routing preference. Communication Manager blocks a hop equal to or greater than the number entered.

Valid Entry	Usage
blank	There is no limit to the number of hops for this preference.
1 to 9	Valid for limiting the number of hops if using the tandem hop feature.
1 to 32	Valid for limiting the number of hops if using the transit feature.

Inserted Digits

The digits to insert for routing. Communication Manager can send up to 52 digits. This includes up to 36 digits entered here plus up to 18 digits originally dialed. Special symbols count as two digits each.

Valid Entry	Usage
*	When * is in the route pattern and the outgoing trunk is signaling type “mf”, the MFC tone for the “end-of-digits” is sent out to the local telephone company central office (CO) in place of the *.
#	When # is in the route pattern and the outgoing trunk is signaling type “mf”, the MFC tone for the “end-of-digits” is sent out to the CO in place of the #.
, (comma)	Creates a 1.5 second pause between digits being sent. Do not use as the first character in the string unless absolutely necessary. Misuse can result in some calls, such as Abbreviated Dialing or Last Number Dialed, not completing.
+	Wait for dial tone up to the Off Premises Tone Detection Timer and then send digits or intercept tone depending on system-wide feature administration. The length of waiting time is controlled by the Off-premises Tone Detect Timeout Interval field on the Feature-related system parameters screen.
%	Start End-to-End Signaling.
!	Wait for dial tone without timeout and then send DTMF digits.
&	Wait for ANI that is used for Russian pulse trunks.
p	The associated trunk group must be of type sip. The single digit p is used for fully qualified E.164 numbers. Communication Manager inserts a plus (+) digit to the digit string.

Related links

[Off-Premises Tone Detect Timeout Interval \(seconds\)](#) on page 242

[Outpulse Without Tone](#) on page 279

ITC (Information Transfer Capability)

Identifies the type of data transmission or traffic that this routing preference can carry. The ITC applies only to data calls (BCC 1 through 4).

This field must be both or unre for a **BCC Value** of W.

Valid Entry	Usage
both	Calls from restricted and unrestricted endpoints can access the route pattern.
rest(ri)cted	Calls from restricted endpoints can access the route pattern.
unre(stri)cted	Calls from unrestricted endpoints can access the route pattern.

Related links

[BCC Value](#) on page 592

IXC

Identifies the carrier, such as AT&T, used for calls that route using an Inter-Exchange Carrier (IXC), and for Call Detail Recording (CDR).

Available only if **ISDN-PRI** or **ISDN-BRI Trunks** are enabled for the system.

Valid Entry	Usage
Valid carrier code	Identifies the carrier for IXC calls.
user	For presubscribed carrier. Used when an IXC is not specified.
none	Must be none for non-ISDN trunk groups and for Telcordia Technologies NI-2 Operator Service Access. If it is necessary to send an IXC code for a non-ISDN trunk group, the IXC code can be entered as Inserted Digits .

Related links

[Inserted Digits](#) on page 594

[ISDN-BRI Trunks](#) on page 734

[ISDN-PRI](#) on page 734

LAR

The routing-preference for Look Ahead Routing.

Valid Entry	Usage
next	Go to the next routing preference and attempt the call again.

Table continues...

Valid Entry	Usage
rehu	Re-hunt within the current routing preference for another trunk to attempt the call again.
none	Look Ahead Routing is disabled for the preference.

No. Del. Digits

Modifies the dialed number so an AAR or ARS call routes over different trunk groups that terminate in servers or switches with different dial plans.

Valid Entry	Usage
0 to 28 blank	Inter-Exchange Carrier (IXC) digits are deleted if the field is administered to any non-blank value. If the field is blank, no IXC digits are deleted. Communication Manager administers the IXC digits on the Inter-Exchange Carrier (IXC) Codes screen. Use for calls that route: • To or through a remote server. • Over tie trunks to a private network server. • Over local telephone company central office (CO) trunks to the serving CO.

No. Dgts Subaddress

Available only if **ISDN Feature Plus** is enabled for the system.

Valid Entry	Usage
1 to 5 blank	The number of dialed digits to send in the calling party subaddress IE. Allows a caller to reach a number where the media server's digit processing deletes the dialed number and inserts the listed directory number (LDN). The LDN then is sent to the destination address and the dialed extension is sent in the calling party subaddress information element (IE). At the receiving end, the call terminates to the user indicated by the subaddress number instead of the attendant.

Related links

[ISDN Feature Plus](#) on page 734

NPA

The three-digit Numbering Plan Area (NPA) or area code for the terminating endpoint of the trunk group. Not required for AAR.

The local telephone company can verify this number. For WATS trunks, the terminating NPA is the same as the home NPA unless the Local Exchange Carrier requires 10 digits for local NPA calls.

Leave blank for AAR calls and for tie trunks.

Numbering Format

Use this field to specify the numbering format of calls that Communication Manager routes over trunk group of the preferred routing pattern. This field applies to ISDN and SIP trunk groups.

Valid entry	Numbering Plan Identifier	Type of Numbering
blank	Derived from the value of the Call Type field on the ARS Analysis screen or the AAR Analysis screen.	Derived from the value of the Call Type field on the ARS Analysis screen or the AAR Analysis screen.
natl-pub	E.164(1)	national(2)
intl-pub	E.164(1)	international(1)
locl-pub	E.164(1)	local/subscriber(4)
pub-unk	E.164(1)	unknown(0)
lev0-pvt	Private Numbering Plan - PNP(9)	local(4)
lev1-pvt	Private Numbering Plan - PNP(9)	Regional Level 1(2)
lev2-pvt	Private Numbering Plan - PNP(9)	Regional Level 2(1)
unk-unk	unknown(0)	unknown(0)

 Note:

To gain access to Telcordia Technologies NI-2 Operator Service Access, type `unk-unk` in the **Inserted Digits** field.

For Network Call Redirection or Transfer, type `lev0-pvt` in the **Numbering Format** field.

Pattern Name

An alphanumeric name for the route pattern.

Pattern Number

The route pattern number.

Prefix Mark

Sets the requirements for sending a prefix digit 1, indicating a long-distance call. Prefix Marks apply to 7- or 10-digit Direct Distance Dialing (DDD) public network calls. A prefix digit 1 is sent only when call type is foreign number plan area (FNPA) or home numbering plan area (HNPA) in the ARS Digit Analysis table.

Not required for AAR. ARS requires a number from 0 to 4 or blank.

For a WATS trunk, the **Prefix Mark** is the same as the local telephone company central office (CO) trunk.

Valid Entry	Usage
0	• Suppresses a user-dialed prefix digit 1 for 10-digit FNPA calls. • Leaves a user-dialed prefix digit 1 for 7-digit HNPA calls. • Leaves a prefix digit 1 on 10-digit calls that are not FNPA or HNPA calls. Should not be used in those areas where all long-distance calls must be dialed as 1+10 digits. Check with the local network provider.
1	Sends a 1 on 10-digit calls, but not on 7-digit calls. Used for HNPA calls that require a 1 to indicate long-distance calls.
2	Sends a 1 on all 10-digit and 7-digit long-distance calls. Refers to a Toll Table to define long-distance codes.
3	Sends a 1 on all long-distance calls and keep or insert the NPA (area code) so that all long-distance calls are 10-digit calls. The NPA is inserted when a user dials a prefix digit 1 plus 7 digits. Refers to a Toll Table to define long-distance codes.
4	Always suppress a user-dialed prefix digit 1. Used, for example, when ISDN calls route to a server that rejects calls with a prefix digit 1.
blank	For tie trunks.

SCCAN

If enabled, indicates that the route pattern supports incoming SCCAN calls. Available only if Enhanced EC500 is enabled for the system.

Related links

[Enhanced EC500](#) on page 732

Secure SIP

If enabled, specifies using the SIP or SIPS prefix when the call is routed to a SIP trunk preference. If SIP trunks are not specified, the call is routed over whatever trunk is specified.

Service/Feature

An identifier of the Service/Feature carried by the information element (IE) in a call in this route preference. Required by Call-by-Call Service Selection, and Network Call Redirection Transfer. Accepts up to 15 characters.

* Note:

User-defined service types for network facilities can also be used. Any user-defined **Facility Type** of 0 (feature), 1 (service), or 3 (outgoing) is allowed.

Available only if **ISDN-PRI** or **ISDN-BRI Trunks** are enabled for the system.

Valid Entry		
accunet	multiquest	sdn (allows Network Call Redirection/Transfer)
i800	operator	sub-operator
lds	oper-meg (operator and megacom)	sub-op-meg (sub-operator and megacom)
mega800	oper-sdn (operator and sdn)	sub-op-sdn (sub-operator and sdn)
megacom	outwats-bnd	wats-max-bnd

Related links

[Facility Type](#) on page 540

[ISDN-BRI Trunks](#) on page 734

[ISDN-PRI](#) on page 734

Toll List

Valid Entry	Usage
1 to 32 blank	The number of the ARS Toll Table associated with the terminating NPA of the trunk group. Required for Prefix Mark 2 or 3. Not required for AAR.

Related links

[Prefix Mark](#) on page 598

TSC

Use the **Temporary Signaling Connection (TSC)** field to support feature transparency on DCS+ calls and to support QSIG call completion. You must enable **TSC** to allow Communication Manager to send Call-Associated TSC (CA-TSC) and Non-Call-Associated TSC (NCA-TSC) requests for a routing preference.

Used for SIP Stations

Use the **Used for SIP Stations** field to set the preferred route pattern.

Valid entry	Usage
Yes	Communication Manager automatically populates the values of Primary, Secondary, Third, and Fourth Session Manager node names in the Primary SM , Secondary SM , Third SM , and Fourth SM fields.
No	The default option is No.

Note:

When the policy-based assignment is enabled in System Manager, you can administer up to four Session Managers for a SIP station.

Chapter 109: Security related system parameters

Determines when Avaya Communication Manager reports a security violation. Many of the fields on this screen repeat for each type of security violation. They are explained once here, but the usage is the same for all.

Example command: `change system-parameters security`

Security related system parameters: page 1

Hide Post-Dialing DTMF On List Trace

Enable this field to display asterisk (*) characters in place of digits in the output of the list trace command. The default value of this field is n.

SECURITY VIOLATION NOTIFICATION PARAMETERS

Announcement Extension

The announcement extension where the Security Violation Notification (SVN) announcement resides. The server running Communication Manager calls the referral destination, then plays this announcement upon answer.

Originating Extension

The extension that initiates the referral call in the event of a security violation. It also sends the appropriate alerting message or display to the referral destination. If notification for more than one type of security violation is established, a different extension must be assigned to each one. When Communication Manager generates a referral call, this extension and the type of violation appear on the display at the referral destination.

Referral Destination

The extension that receives the referral call when a security violation occurs. The referral destination telephone must have a display, unless it is an Announcement Extension. The extension can be the telephone, attendant console, or vector directory number (VDN) that

receives the referral call for each type of violation. This can be the same extension for all type of violations.

The **Announcement Extension** field is used for a VDN. Call Vectoring Time-of-Day routing is used to route the referral call to different destinations based on the time of day or the day of the week.

Related links

[Announcement Extension](#) on page 601

SVN Authorization Code Violation Notification Enabled

Enables or disables Authorization Code Violation Security Notification. By using **SVN Remote Access Violation Notification Enabled**, you can do the following:

- Establish parameters for remote access security violations.
- Enable or disable the feature.

A remote access violation occurs if a user enters incorrect barrier codes. The system cannot disable remote access following a security violation unless **SVN Remote Access Violation Notification Enabled** field is enabled.

Related links

[SVN Remote Access Violation Notification Enabled](#) on page 602

SVN Login (Violation Notification, Remote Access, Authorization Code) Enabled

Enables or disables login violation notification. If enabled, Communication Manager sends a notification when a login violation occurs.

SVN Remote Access Violation Notification Enabled

Use this field to enable or disable Remote Access Violation Notification. Use with **SVN Authorization Code Violation Notification Enabled** to enable or disable remote access security violations. A remote access violation occurs if a user enters incorrect barrier codes. The system cannot disable remote access following a security violation unless this field is enabled.

Related links

[SVN Authorization Code Violation Notification Enabled](#) on page 602

Time Interval

Valid Entry	Usage
0:01 to 7:59	This time range, in conjunction with Login Threshold, determines if a security violation has occurred. The range for the time interval is one minute to eight hours, entered in the screen x:xx. For example, one minute is entered as 0:01 and seven and one-half hours is entered as 7:30.

Security related system parameters: page 2

ACCESS SECURITY GATEWAY PARAMETERS

These fields are available only if **Access Security Gateway (ASG)** is enabled for the system.

Related links

[Access Security Gateway \(ASG\)](#) on page 728

INADS

Indicates whether or not any entry attempt through a port that is a direct connection to the Initialization and Administration System (INADS) receives a challenge response. INADS is used to remotely initialize and administer Communication Manager

MGR1

Indicates whether or not any entry attempt through a port that is a direct connection to the system administration and maintenance access interface located on the processor receives a challenge response.

NET

Indicates whether or not any entry attempt through a port that is a dialed-in or dialed-out connection to the Network Controller receives a challenge response.

Translation-ID Number Mismatch Interval (days)

Valid Entry	Usage
1 to 90	The number of days the system allows access to system administration commands. When this interval expires, only init logins have the ability to execute system administration commands to modify translation data.

REMOTE MANAGED SERVICES

Port Board Security Notification

Enables or disables port board denial of service notification. Available only if **RMS Feature Enabled** is enabled.

Related links

[RMS Feature Enabled](#) on page 604

Port Board Security Notification Interval

Valid Entry	Usage
60 to 3600 in increments of 10	The interval in seconds between port board Denial of Service notifications (traps). Default is 60.  Note: There is no delay before the first trap is sent. The interval administered in this field applies only to the period between the sending of the traps. Available only if Remote Managed Services and Port Board Security Notification are enabled.

Related links

[Port Board Security Notification](#) on page 603

[REMOTE MANAGED SERVICES](#) on page 603

RMS Feature Enabled

Enables or disables Remote Managed Services.

SECURITY VIOLATION NOTIFICATION PARAMETERS

SVN Station Security Code Violation Notification Enabled

Use this field to enable or disable the security violation notification for Station Security Codes. Station Security Codes are used to validate logins to a particular extension.

The default value is n.

STATION SECURITY CODE VERIFICATION PARAMETERS

Minimum Station Security Code Length

Valid Entry	Usage
3 to 8	The minimum required length of the station security codes. Longer codes are more secure. If station security codes are used for external access to telecommuting features, the minimum length should be seven or eight.

Receive Unencrypted from IP Endpoints

Allows or blocks unencrypted data from IP endpoints.

Security Code for Terminal Self Administration Required

Specifies whether or not a Personal Station Access code is required to enter the Self-Administration mode.

Related links

[Personal Station Access \(PSA\)](#) on page 736

Chapter 110: Service Hours Table

Specifies office service hours using up to 99 different tables. Available only if basic vectoring is enabled.

Example command: `change service-hours-table n`, where *n* is the service hours table number.

change service-hours-table 1Page 1 of 1

SERVICE HOURS TABLE:

Number: 1

Description: _____

Use time adjustments from location: _____

MON		TUE		WED		THU		FRI	
Start	End	Start	End	Start	End	Start	End	Start	End
—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—
—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—
—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—
—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—
—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—	—:—

SAT		SUN	
Start	End	Start	End
—:—	—:—	—:—	—:—
—:—	—:—	—:—	—:—
—:—	—:—	—:—	—:—
—:—	—:—	—:—	—:—
—:—	—:—	—:—	—:—

Description

A name that provides a description of the table. Accepts up to 27 characters. The default is blank.

Example

Call-ahead Reservations

Number

The table number.

Start/End

The range of office hours for each day of the week.

A time is in the table from the first second of the start time (for example, 08:00:00) until the last second of the end time (for example, 17:00:59).

Valid Entry	Usage
0 to 23	Hour range. The hour range must be within the specified day, from 00:00 (midnight) until 23:59.
0 to 59	Minute range.

Use Time Adjustments from Location

The location number that specifies how time zone offset and daylight saving rule adjustments are performed.

Related links

[Loc Number](#) on page 484

Chapter 111: Signaling group

Establishes signaling group parameters for ISDN-PRI, H.323, atm and SIP trunks. Because these trunk types vary in the types of parameters needed, the system displays these fields on this screen change depending on the **Group Type**.

Related links

[Signaling group: page 1](#) on page 608

Signaling group: page 1

Alert Incoming SIP Crisis Calls

This field is used to enable the Emergency call routing for H.323 visiting users feature.

The Emergency call routing for H.323 visiting users feature support crisis alert for H.323 visiting users. Emergency calls made by H.323 visiting users from an enterprise network are routed to the nearest possible emergency response agencies (PSAPs) at the location visited by the user.

This field is available only for SIP signaling groups.

Valid entry	Usage
y	The attendant endpoints are notified for the incoming emergency call.
n	The attendant endpoints are not notified for the incoming emergency call. The default value is n.

Alternate Route Timer

Available only if **Group Type** is sip.

Note:

You must the **LAR** field to a value other than none. If you set the **LAR** field to none, the call drops.

Valid Entry	Usage
2 to 30	The time in seconds Communication Manager waits for a Ringing message from the far end of the trunk before trying for an alternate route to establish a call. The default value is 6 seconds.

Related links

[Group Type](#) on page 614

Associated Signaling

Available only if **Group Type** field is isdn-pri.

Valid Entry	Usage
y	Enables associated signaling.
n	Enables non-facility associated signaling.

Related links

[Group Type](#) on page 614

Bypass If IP Threshold Exceeded

Available only if **Group Type** is h.323 or sip.

Valid Entry	Usage
y	Automatically remove from service the trunks assigned to this signaling group when IP transport performance falls below administered system-wide limits.
n	Signaling group trunks stay in service when IP transport performance falls below limits.

Related links

[Group Type](#) on page 614

Calls Share IP Signaling Connection

Available only if **Group Type** is h.323 or sip.

Valid Entry	Usage
y	Enables inter-connection between servers running Avaya Communication Manager. Unavailable if a RAS-Location Request (LRQ) is enabled.  Note: When the near and far-end servers are running Avaya Communication Manager, this value must match the value administered for enabling the Layer 3 test.
n	Disables inter-connection between servers running Avaya Communication Manager. Used if the local or remote server is <i>not</i> running Avaya Communication Manager.

Related links

[Group Type](#) on page 614

[LRQ Required](#) on page 618

[Enable Layer 3 Test](#) on page 612

Circuit Type

Available only if **Group Type** is atm.

Valid Entry	Usage
T1	U.S., Canadian, and Japanese digital transmission format.
E1	European digital transmission format.

Related links

[Group Type](#) on page 614

Clustered

Valid Entry	Usage
y	Allows the signaling group to administer Cluster ID. When it is set to y, the Far-end Node Name field is replaced with Cluster ID field.
n	Does not allow the signaling group to administer Cluster ID..

Connect

To control communications at layers 2 and 3 of the ISDN-PRI protocol, this field specifies what is on the far end of the link.

Available only if **Group Type** is atm.

Valid Entry	Usage
host	The link connects Communication Manager to a computer.
network	The link connects Communication Manager to a local telephone company central office or any other public network switch.
pbx	The link is connected to another switch in a private network.

Related links

[Group Type](#) on page 614

Country Protocol

The country protocol used by the local telephone company central office where the link terminates. This value must match the country protocol used by the far-end server. For connections to a public network, the network service provider can tell you which country protocol they are using.

Available only if **Group Type** is atm.

Valid Entry	Usage
1 to 25	The country code.
etsi	The network service provider uses the protocol of the European Telecommunications Standards Institute (ETSI). Used only if the Signaling Mode is isdn-pri.

Related links

[Signaling Mode](#) on page 625

[Country options table](#) on page 718

[Group Type](#) on page 614

DCP/Analog Bearer Capability

Sets the information transfer capability in a bearer capability IE of a setup message to speech or 3.1kHz. Available only if **Group Type** is atm or h.323.

Valid Entry	Usage
3.1kHz	Provides 3.1kHz audio encoding in the information transfer capability. This is the default.
speech	Provides speech encoding in the information transfer capability.

Related links

[Group Type](#) on page 614

Direct IP-IP Audio Connections

Allows or disallows direct audio connections between H.323 endpoints. Direct audio connections save bandwidth resources and improve sound quality of voice over IP (VoIP) transmissions. For SIP trunk groups, allows direct audio connections between SIP endpoints. Available only if **Group Type** is h.323 or sip.

For the SIP Resiliency feature, ensure that the **Direct IP-IP Audio Connections** field is set to Y.

Related links

[Group Type](#) on page 614

DTMF over IP

The **DTMF over IP** field specifies the touchtone signals that are used for dual-tone multifrequency (DTMF) telephone signaling. The **DTMF over IP** field is available if the **Group Type** field is set to h.323 or sip.

Valid Entry	Usage
in-band	All G711 and G729 calls pass DTMF in-band. DTMF digits encoded within existing RTP media stream for G.711/G.729 calls. G.723 is sent out-of-band. It also supports the SIP trunks.
in-band-g711	Only G711 calls pass DTMF in-band. The system displays the in-band-g711 option if the Group Type field is set to h.323.

Table continues...

Valid Entry	Usage
out-of-band	All IP calls pass DTMF out-of-band. For IP trunks, the digits are done with either Keypad IEs or H245 indications. This is the default for newly added H.323 signaling groups. The out-of-band option enables out of band signaling for the SIP Signaling group. When you select the out-of-band option, Communication Manager sends all outgoing DTMF messages as SIP INFO messages over a SIP signaling group. This option is interoperable with H.323 networks. You can connect an Avaya non-SIP endpoint or trunk to a voice mail system on the H.323 network that is integrated with Session Manager and Communication Manager through SIP. Examples of Avaya non-SIP endpoints or trunks are H.323, Analog, DCP, and Digital trunks.
rtp-payload	This is the method specified by RFC 2833. This is the default value for newly added SIP signaling groups. Support for SIP trunks requires the default entry of rtp-payload.

Related links

[Group Type](#) on page 614

Enable Layer 3 Test

Enables or disables a Layer 3 test. Communication Manager runs the Layer 3 test to verify that all connections known at the near-end are recognized at the far-end. Available only if **Group Type** is H.323 and SIP. This field is enabled by default and is in read-only mode.

* Note:

The Layer 3 test should be enabled for an H.323 signaling group type when the **Calls Share IP Signaling Connection** is enabled and the far-end is Communication Manager. Requires administration of a **Far-end Node Name**.

The Layer 3 Test must be enabled for the call reconstruction.

Related links

[Calls Share IP Signaling Connection](#) on page 609

[Far-end Node Name](#) on page 614

[Group Type](#) on page 614

Enforce SIPS URI for SRTP

The system displays this when the **Group Type** field is sip. Use this field to enable or disable Communication Manager to enforce SIPS URI for any incoming SIP message with security descriptions (used for SRTP negotiation) in SDP.

Valid Entry	Usage
y	Enable Communication Manager to enforce SIPS URI for any incoming SIP message with security descriptions (used for SRTP negotiation) in SDP. Note: The system displays y as the default value for the Enforce SIPS URI for SRTP field.
n	Disable Communication Manager from enforcing SIPS URI for any incoming SIP message with security descriptions (used for SRTP negotiation) in SDP. For the SIP Resiliency feature, you must set this field to n if the SIP signaling is not supported.

ETSI CCBS Support

Available only if **Group Type** is isdn-pri and **TSC Supplementary Service Protocol** is set to c for ETSI.

Valid Entry	Usage
none	Interface supports neither incoming nor outgoing ETSI CCBS. This is the default.
incoming	Interface supports only incoming ETSI CCBS.
outgoing	Interface supports only outgoing ETSI CCBS.
both directions	Interface supports incoming and outgoing ETSI CCBS. * Note: When upgrading from a version of Communication Manager that is earlier than 5.1, this is the default.

Related links

[TSC Supplementary Service Protocol](#) on page 626

[Group Type](#) on page 614

Far-end Domain

The name of the IP domain that is assigned to the far-end of the signaling group. For example, to route Session Manager calls within an enterprise, the domain assigned to the proxy server is used. For external SIP calling, the domain name could be that of the SIP service provider. Available only if **Group Type** is sip.

Valid Entry	Usage
<i>Character string</i>	A maximum of 40 characters used to define the name of the IP domain for which the far-end proxy is responsible, if different than the near-end domain.
blank	Far-end domain is unspecified and the far-end IP address is used. Use if the domains are the same.

Related links

[Group Type](#) on page 614

Far-end Listen Port

Available only if **Group Type** is h.323 or sip.

Valid Entry	Usage
1 to 65535	Must match the number administered for Near-end Listen Port . Typically, this is the default of 5061 for SIP over TLS.
blank	Far-end listen port is unspecified.

Related links

[Near-end Listen Port](#) on page 620

[Group Type](#) on page 614

Far-end Network Region

Available only if **Group Type** is h.323 or sip.

Valid Entry	Usage
1 to 2000	The number of the network region that is assigned to the far-end of the signaling group. The region is used to obtain the codec set used for negotiation of trunk bearer capability.
blank	Selects the region of the near-end node and the far-end network region is unspecified.

Related links

[Group Type](#) on page 614

Far-end Node Name

Use this field to specify the node name for the far-end IP interface that is used for trunks assigned to the Signaling group. You must administer the node name on the Node Names screen.

The system displays the field only if the **Group Type** field is set to h.323 or sip.

Related links

[Node Name](#) on page 394

[Group Type](#) on page 614

Group Number

The signaling group number.

Group Type

The type of protocol used with the signaling group.

Valid Entry	Usage
h.323	h.323 protocols or SBS signaling trunks
isdn-pri	Integrated Service Digital Network Primary Rate Interface
sip	For SIP

H.235 Annex H Required

Enables or disables the requirement that the Communication Manager server uses the H.235 Annex H (now called H.235.5) protocol for authentication during registration. Available only if **Group Type** is h.323. Requires that RAS-Location Request (LRQ) is enabled.

Related links

[LRQ Required](#) on page 618

[Group Type](#) on page 614

H.245 DTMF Signal Tone Duration (msec)

Available only if **DTMF over IP** is set to out-of-band.

Valid Entry	Usage
80 to 350 blank	Specifies the duration of DTMF tones sent in H.245-signal messages. The default is blank.

Related links

[DTMF over IP](#) on page 611

[Group Type](#) on page 614

H.323 Station Outgoing Direct Media

Available if **Group Type** is h.323 and **Direct IP-IP Audio Connections** is enabled. Also available if **Group Type** is SIP and **Direct IP-IP Audio Connections** is enabled.

Valid Entry	Usage
y	A call from an H.323 station over a trunk that uses this signaling group starts as a direct media call. The IP address and port of the H.323 station are sent as the media and media control channel addresses in the SETUP/INVITE message.  Note: On an outgoing Direct Media call from an IP (H.323) telephone over this trunk, if an attempt is made to transfer the call, conference another party, or put the call on hold while the call is still in ringing state, the operation fails.
n	The IP address of the MEDPRO board is sent in the SETUP/INVITE message. This is the default.

Related links

[Direct IP-IP Audio Connections](#) on page 611

[Group Type](#) on page 614

Idle Code

An eight-digit string that is compatible with the protocol used by the far-end switch or server. Sets the signal sent out over idle DS0 channels.

Available only if **Group Type** is atm.

Related links

[Group Type](#) on page 614

IMS Enabled

Enables or disables accepting SIP requests that match the domain in the **Far-End Domain**. This field enables the signaling group to use the IMS signaling. Available only if **Group Type** is sip. If you are configuring Communication Manager as a feature server, set the **IMS Enabled** field to y. Setting this field to y only applies when Communication Manager is being configured as a feature server. If you are configuring Communication Manager as an evolution server, leave the **IMS Enabled** field set to n.

Related links

[Far-end Domain](#) on page 613

[Group Type](#) on page 614

Incoming Dialog Loopbacks

The system displays this on the Signaling Group screen when the **Group Type** field is sip.

Valid entries	Usage
allow	Communication Manager software connects the call and allows the SIP trunks to remain in the looparound connection. If the trunk group controlled by this SIP signaling group is used for IGAR calls, set the value allow.
eliminate	Communication Manager software connects the call and eliminates the SIP trunks from the looparound connection. The default value is eliminated.

Initial IP-IP Direct Media

The system displays the **Initial IP-IP Direct Media** field on the Signaling Group screen only when you set the **Group Type** field to sip and the **Direct IP-IP Audio Connections** field to y. The **Initial IP-IP Direct Media** field enables or disables the SIP direct media feature between SIP endpoints. The **Initial IP-IP Direct Media** field also enables or disables the direct media feature from SIP endpoints to Avaya H.323 stations. The default value for this field is n.

Valid Entry	Usage
n	Disables the SIP direct media feature. The default value is n.
y	Enables the SIP direct media feature and SIP-H.323 direct media feature.

 Note:

If **Initial IP- IP Direct Media?** is set to *y* in the SIP Signaling Group then **Initial INVITE with SDP for secure calls?** must also be set to *y* in the system parameter features forms.

For the SIP Resiliency feature, ensure that the **Initial IP- IP Direct Media?** and the **Initial INVITE with SDP for secure calls?** field must be set to *y*.

Related links

[Direct IP-IP Audio Connections](#) on page 611

[Group Type](#) on page 614

Interface

Controls how your server negotiates glare with the far-end switch. Available only for a far-end PBX connection.

Valid entry	Usage
U.S. private network applications	
network	The server overrides the other end when glare occurs. For servers connected to a host computer.
user	The server releases the contested circuit and looks for another when glare occurs. For servers connected to a public network.
Private networks, including QSIG networks, applications outside the U.S.	
peer-master	The switch overrides the other end when glare occurs.
peer-slave	The switch releases the contested circuit and looks for another when glare occurs.

Related links

[Connect](#) on page 610

Interface Companding

The companding algorithm expected by the system at the far end.

Valid Entry	Usage
a-law	Algorithm expected at the far-end for E1 service.
mu-law	Algorithm expected at the far-end for T1 service.

Interworking Message

Determines what message Communication Manager sends when an incoming ISDN trunk call is routed over a non-ISDN trunk group.

Valid Entry	Usage
PROGress	Requests the public network to cut through the B-channel and let the caller hear tones such as ringback or busy tone provided over the non-ISDN trunk. Normally-selected value.
ALERTing	Causes the public network in many countries to play ringback tone to the caller. This value is used only if the DS1 is connected to the public network, and it is determined that callers hear silence (rather than ringback or busy tone) when a call incoming over the DS1 interworks to a non-ISDN trunk.

IP Video

Enables or disables IP video capability for this signaling group. Available only if the signaling group type h.323 and sip.

Link Loss Delay Timer (sec)

Specifies how long to hold the call state information in the event of an IP network failure or disruption. Communication Manager preserves calls and starts this timer at the onset of network disruption (signaling socket failure). If the signaling channel recovers before the timer expires, all call state information is preserved and the signaling channel is recovered. If the signaling channel does not recover before the timer expires, the system:

- Raises an alarm against the signaling channel
- Maintains all connections with the signaling channel
- Discards all call state information about the signaling channel

Valid Entry	Usage
1 to 180	The number of seconds to delay the reaction of the call controller to a link bounce. Default is 90.

Location for Routing Incoming Calls

The system displays this only when the **Group Type** field on the Signaling Group screen is h.323 and **Multiple Locations** field on the Optional Feature screen is y.

Valid Entry	Usage
1 to 2000	(Depending on your server configuration, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .) Determines the location-based digit analysis which is applied to the destination digits dialed by the incoming trunk call.
blank	The location of the Processor Ethernet (PE) is taken for routing the call. By default, the value is blank.

LRQ Required

Enables or disables RAS-Location Request (LRQ) messages. Available only if **Group Type** is h.323.

Valid Entry	Usage
y	Allows IP trunk availability to be determined on a per call basis. An LRQ message is sent to the far-end gatekeeper prior to each call over the IP trunk. The far-end gatekeeper responds with a RAS-Location Confirm (LCF) message, and the call proceeds. Required if H.235 Annex H Required is enabled. Calls Share IP Signaling Connection must be disabled.
n	For far-end servers running Communication Manager, unless H.235 Annex H Required is enabled.

Related links

[H.235 Annex H Required](#) on page 615

[Calls Share IP Signaling Connection](#) on page 609

[Group Type](#) on page 614

Max number of CA TSC

Available only if **Group Type** is atm, h.323, or isdn-pri.

Valid Entry	Usage
0 to 619	The maximum number of simultaneous call-associated Temporary Signaling Connections that can exist in the signaling group. Typically this is the number of ISDN-PRI trunk group members controlled by this signaling group.

Related links

[Group Type](#) on page 614

Max number of NCA TSC

Available only if **Group Type** is atm, h.323, or isdn-pri.

Valid Entry	Usage
0 to 256	The maximum number of simultaneous non-call-associated Temporary Signaling Connections. The TSCs carry signaling for features not associated with a specific call, for example, signals to turn on Leave Word Calling.

Related links

[Group Type](#) on page 614

Media Encryption

Enables or disables encryption for trunk calls assigned to this signaling group. If encryption for the signaling group is disabled, then trunk calls using this signaling group do not get encrypted regardless of IP Codec Set administration. Available only if Media Encryption is enabled in Communication Manager and the **Group Type** is h.323.

Related links

[Group Type](#) on page 614

Name

A name that identifies the signaling group. Accepts up to 15 alphanumeric characters.

 Note:

Avaya BRI deskphones support only ASCII characters because non-ASCII characters, such as Eurofont and Kanafont, show up incorrectly.

Related links

[Group Type](#) on page 614

Near-end Listen Port

Available only if **Group Type** is h.323 or sip.

Valid Entry	Usage
5000 to 9999	An unused port number for the near-end listen port.
1719	For LRQ-enabled ports
1720	For h.323-enabled ports

Related links

[H.323 Station Outgoing Direct Media](#) on page 615

[LRQ Required](#) on page 618

[Group Type](#) on page 614

Near-end Node Name

Available only if **Group Type** is h.323 or sip. This field can be set to `procr` or `procr6`, for IPv4 and IPv6 signaling.

Related links

[Name](#) on page 413

[Node Name](#) on page 394

[Group Type](#) on page 614

Network Call Transfer

Enables or disables Network Call Transfer so that D-channels support Explicit Network Call Transfer (ENCT). Available only if **Group Type** is atm.

Related links

[Group Type](#) on page 614

Passphrase

The passphrase used to generate a shared “secret” for symmetric encryption of the media session key. The same passphrase must be assigned to the corresponding signaling groups at both ends of an IP trunk. The passphrase:

- Consists of 8 to 30 alphanumeric characters
- Is case sensitive
- Must contain at least one alphabetic and at least one numeric
- Valid characters also include letters, numbers, and these symbols: !&*?;'^(),.-:-

The passphrase is used for both Media Encryption and authentication. This field cannot be left blank.

Available only if Media Encryption is enabled or the **H.235 Annex H Required** is enabled.

Related links

[H.235 Annex H Required](#) on page 615

Peer Detection Enabled

The values for this field are `n(o)` and `y(es)`. The default value is `y(es)`. This field is only available if **Group Type** is `sip`. When set to `y(es)` then Communication Manager will automatically set the value in the **Peer Server** field when the sip signaling group comes into service.

Peer Server

The values for this field are `Others`, `SM`, and `AMS`. The default value is `Others`. This field is only available if **Group Type** is `sip`. If this sip signaling group terminates to an Avaya Session Manager, then Communication Manager will automatically set value in this field to `SM` if the **Peer Detection Enabled** field is set to `y(es)`.

If **Clustered** field is set to `y`, then the value in this field is automatically changed to `SM`, and this field will be in read-only mode.

Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers

If you set the **Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers** field on the SIP Signaling Group screen to `y` and if Communication Manager uses the public-unknown-numbering table, Communication Manager inserts a leading plus (+) digit in calling, alerting, diverting, or connected numbers. The default value is `n`.

If the **Peer Server** field on the SIP Signaling Group screen is `SM`, the system sets the **Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers** field to display-only, with the default value of `y`. If the **Peer Server** field on the SIP Signaling Group screen is `Others`,

the system sets the **Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers** field to n and as editable.

Primary D Channel

Available only if **Group Type** is isdn-pri.

Valid Entry	Usage
1 to 999	First to third are the media gateway number.
V	Fourth character is the board.
1 to 8	Fifth character is the board number.
01 to 04 (Analog TIE trunks) 09 to 32	Six and seventh characters are the circuit number.
xxxVmpp	Specifies the Branch Gateway. • xxx is the Branch Gateway number, which is in the range 001 to 999. • m is the module number, which is in the range 1 to 9. • pp is the port number, which is in the range 01 to 32.

Related links

[Group Type](#) on page 614

Priority Video

Enables or disables the priority video function that specifies that incoming video calls have an increased likelihood of receiving bandwidth, and are also allocated a larger maximum bandwidth per call.

Available only if:

- **Group Type** is h.323 or sip
- Multimedia SIP trunking is enabled for the system

Related links

[Multimedia IP SIP Trunking](#) on page 736

[Group Type](#) on page 614

Protocol Version

Available only if **Group Type** is atm.

Valid Entry	Usage
a	In countries whose public networks allow multiple layer-3 signaling protocols for ISDN-PRI service. Selects the protocol that matches the network service provider's protocol.
b	
c	
d	

Related links

[Group Type](#) on page 614

Q-SIP

The system displays this only when the **Group Type** field is h.323 or sip.

Valid Entry	Usage
y	Enables the QSIG over SIP (Q-SIP) feature for the signaling group.
n	The QSIG over SIP feature is disabled. By default, the value is n.

QSIG Signaling Group

The system displays this only when the **Group Type** field is sip and the **Q-SIP** field is set to y.

Valid Entry	Usage
1 to 999	Assigns a number for the QSIG signaling group.
blank	No QSIG signaling group is assigned. By default, the value is blank.

Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers

If you set the **Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers** field on the SIP Signaling Group screen to y, Communication Manager removes the leading plus (+) digit, if present, from both calling and called numbers. The default value is y.

This field interacts with the **Peer Server** field in the following manner:

- If the **Peer Server** field on the SIP Signaling Group screen is SM, the system sets the **Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers** field to n. The **Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers** field is editable.
- If the **Peer Server** field on the SIP Signaling Group screen is Others, the system sets the **Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers** field to y. The **Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers** field is editable.

Remote Office

Enables or disables administering this signaling group for a remote office. Available only if **Group Type** is h.323.

Related links

[Group Type](#) on page 614

RFC 3389 Comfort Noise

The system displays this on the Signaling Group screen when the **Group Type** field is sip.

Valid entries	Usage
y	This enables SIP signaling for comfort noise. If RFC 3389 Comfort Noise field is set to y, this field overrides the Silence Suppression field on the IP Codec Set screen.
n	This disables SIP signaling for comfort noise. Default is n.

RRQ Required

Enables or disables the requirement that a vendor registration be sent. Available only if **Group Type** is h.323.

Related links

[Group Type](#) on page 614

SBS

Enables or disables the Separation of Bearer and Signaling (SBS) trunk groups. If this field is enabled, both the **Trunk Group for NCA TSC** and the **Trunk Group for Channel Selection** must be set to the signaling group number administered for the SBS trunk group. Available only if **Group Type** is set to h.323.

Related links

[Trunk Group for NCA TSC](#) on page 626

[Trunk Group for Channel Selection](#) on page 625

[Group Type](#) on page 614

Session Establishment Timer (min)

Use this field to specify the time in minutes that Communication Manager waits before tearing down a ring no answer call. You can enter a value from 3 to 240. The default value is 3.

The system displays this field only when the **Group Type** field is set to sip.

Related links

[Group Type](#) on page 614

Signaling Mode

Displays the isdn-pri signaling mode. Available only if **Group Type** is atm.

Related links

[Group Type](#) on page 614

SIP Signaling Group

The system displays this only when the **Group Type** field is h.323 and the **Q-SIP** field is set to y.

Valid Entry	Usage
1 to 999	Assigns a number for the SIP signaling group.
blank	No SIP signaling group is assigned. By default, the value is blank.

T303 Timer (sec)

Available only if **Group Type** is h.323.

Valid Entry	Usage
2 to 10	The number of seconds the system waits for a response from the far end before invoking Look Ahead Routing. Default is 10.

Related links

[Group Type](#) on page 614

Transport Method

Available only if **Group Type** is sip.

Valid Entry	Usage
tcp	Transport is accomplished using Transmission Control Protocol (TCP).
tls	Transport is accomplished using Transport Layer Security (TLS). This is the default. For the SIP Resiliency feature, ensure that the TLS is selected as a transport method.

Related links

[Group Type](#) on page 614

Trunk Group for Channel Selection

Available only if **Group Type** is atm, h.323, or isdn-pri.

Valid Entry	Usage
1 to 2000	Trunk group number used for channel selection.

Related links

[Group Type](#) on page 614

Trunk Group for NCA TSC

Available only if **Group Type** is atm, h.323, or isdn-pri.

Valid Entry	Usage
1 to 2000 blank	Trunk group number used for non-call-associated Temporary Signaling Connections.

Related links

[Group Type](#) on page 614

TSC Supplementary Service Protocol

The supplementary service protocol used for temporary signaling connections.

Available only if **Group Type** is atm, h.323, or isdn-pri.

Valid Entry	Usage
a	AT&T, Telcordia Technologies, Nortel.
b	ISO QSIG. Also, for SBS signaling groups.
c	ETSI. Available only if Group Type is isdn-pri.
d	ECMA QSIG
e	Allows DCS with rerouting. DCS with Rerouting must be enabled, and Used for DCS must be enabled for the trunk group.
f	Feature Plus
g	ANSI. Available only if ISDN-PRI , or ISDN-BRI , or Used for DCS are enabled for the system.

Related links

[DCS with Rerouting](#) on page 731

[Used for DCS](#) on page 838

[ISDN-BRI Trunks](#) on page 734

[ISDN-PRI](#) on page 734

[Group Type](#) on page 614

Virtual Channel Identifier

Available only if **Group Type** is atm.

Valid Entry	Usage
32 to 1023 blank	The number used as a Virtual Channel Identifier.

Related links

[Group Type](#) on page 614

Virtual Path Identifier

Available only if **Group Type** is atm.

Valid Entry	Usage
32 to 1023 blank	The number used as a Virtual Path Identifier.

Related links

[Group Type](#) on page 614

X-Mobility/Wireless Type

The type of X-Mobile endpoints allowed. Available only if **Group Type** is isdn-pri.

Valid Entry	Usage
DECT	The remote end of the trunk group controlled by the signaling group is a DECT mobility controller. This allows X-Mobility to work over ISDN-PRI trunks between the server/switch and adjunct.
none	Not administered.

Related links

[Group Type](#) on page 614

Signaling group: page 2

This screen must be filled in for ATM signaling groups. It provides two functions:

- Defines fractional T1 and fractional E1 facilities, specifying how many and which channels to use.
- Specifies the port numbers to use. Port numbers must be unique for all signaling boards on the same ATM board.

Available only if **Group Type** is atm.

LIMIT SIGNALING GROUP USAGE

The LIMIT SIGNALING GROUP USAGE screen is available only if the **Group Type** field is set to h.323 or sip, and the **Near-end Node Name** field is set to procr. Use this screen to control where H.323 and SIP trunks are used.

Related links

[Group Type](#) on page 614

[Near-end Node Name](#) on page 620

Enable on Survivable Processors (ESS and LSP)

Valid Entry	Usage
all	To enable the signaling group and the associated trunks on both - Survivable core server (Enterprise Survivable Server (ESS)) and Survivable remote server (Local Survivable Processor (LSP)).
ess-all	To enable the signaling group and the associated trunks on Survivable core server (Enterprise Survivable Server) but not on Survivable remote server (Local Survivable Processor).
none	To block the signaling group and the associated trunks on survivable processors and free system resources.
selected	To specify which survivable processors can use the signaling group and the associated trunks.

Enable on the Main Processor(s)

Use the **Enable on the Main Processor(s)** field to enable or disable the signaling group and the associated trunks on the main server.

Valid Entry	Usage
y	To enable the signaling group and the associated trunks on the main server.
n	To disable the signaling group and the associated trunks on the main server.

Selected Survivable Processor Node Names

The **Selected Survivable Processor Node Names** field is available only if you set the **Enable on Survivable Processors (ESS and LSP)** field to selected. Use this field to specify the node names of the survivable processors that can use the signaling group and the associated trunks.

Related links

[Enable on Survivable Processors \(ESS and LSP\)](#) on page 628

SIGNALING GROUP

Chan Port

Valid Entry	Usage
009 to 256	The port number for non-signaling channels.
blank	The signaling channel (port 16 for an E1 and port 24 for a T1) must be a port between 9 and 32. A port number used here cannot be used on any other ATM signaling group on the same board. The channels used must match exactly the channels used on the other end of the signaling group. For example, if your T1 is set up to use channels 1 through 5, 7, and 24 (the signaling channel), the far end must also use channels 1 through 5, 7, and 24.

Enable Failover Event Package Subscription

* Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

This field is available only when you set the **Multiple Level Precedence & Preemption** field on the System parameters customer options screen to *y*.

For Communication Manager to send and receive failover notify messages, you must configure **CCA-ID** on the Multiple Level Precedence & Preemption screen.

* Note:

MLPP is a customer option. Therefore, you can change the parameters only through a license change.

Valid Entry	Usage over
y	Displays the option request parameters on the screen.
n	Does not display the option request parameters on the screen. The default value is n.

Failover/failback Signaling-group Pair

* Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Type the Signaling group that connects to the primary Multi-Function Soft Switch (MFSS) or secondary MFSS.

A lower signaling group number is always the number assigned to the primary MFSS from the pair of signaling groups.

Valid Entry	Usage
SIP signaling-group number	The number that connects to the far-end MFSS.
Blank	The field is blank by default.

OPTIONS Request Parameters

Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

The system displays the OPTIONS Request Parameters fields on the screen only when you set the **Enable Failover Package Subscription** field to *y*.

Communication Manager creates and sends periodic OPTIONS messages to the MFSS. OPTIONS Request Parameters on the Signaling group screen monitor the interval between OPTIONS messages.

Interval between OPTIONS messages (seconds)

The interval time for Communication Manager to send OPTIONS requests to MFSS.

You can enter a number from 30 to 900 in increments of 10. The default value is 60.

Number of OPTIONS message failure before failover

The number of OPTIONS messages that Communication Manager sends to the far-end MFSS without getting a response. A failover is triggered when the far-end MFSS does not respond to the OPTIONS messages.

You can enter a number from 1 to 10. The default value is 2.

Number of OPTIONS message successes before failback

The number of successful OPTIONS messages that Communication Manager sends to the far-end MFSS. A failback is triggered when the far-end MFSS starts responding to the OPTIONS messages.

You can enter a number from 1 to 10. The default value is 2.

Wait time before failback (seconds)

The time taken for Communication Manager to failback to the primary MFSS after the far-end MFSS starts responding to the OPTIONS messages.

You can enter 1 or any number from 10 to 250 in increments of 10.

Signaling group: NCA-TSC Assignment page

Appl.

Specifies the application for this administered NCA-TSC.

Valid Entry	Usage
audix	The ISDN-PRI D-channel DCS Audix feature
dcs	The DCS Over ISDN-PRI D-channel feature.
gateway	The administered NCA-TSC is used as one end in the gateway channel.
masi	The NCA-TSC is one end of a multimedia application server interface.
qsig-mwi	Message conversion from an administered AUDIX NCA-TSC to a QSIG CISC. A Machine ID between 1 and 20 is required.

As-needed Inactivity Time-out (min)

Valid Entry	Usage
10 to 90	Applies only to as-needed NCA-TSCs.
blank	

Dest. Digits

An ISDN interface extension number. Accepts up to 15 characters including * and #.

Enabled

Enables or disables the administered NCA-TSC.

Established

The strategy for establishing this administered NCA-TSC.

Valid Entry	Usage
permanent	The administered NCA-TSC can be established by either the near end or the far end.
as-needed	The administered NCA-TSC is established the first time the administered NCA-TSC is needed. The NCA-TSC can be set up either by the near end or far end switch.

Local Ext

The local extension of the ISDN interface.

Mach ID

Valid Entry	Usage
1 to 63 for DCS, 1 to 99 for AUDIX, 1 to 15 for MASI, and 1 to 63 for QSIG-MWI	A unique machine ID. You cannot specify an ID that you already administered for the Processor Channel.

Related links

[Mach ID](#) on page 576

Service/Feature

Valid Entry	Usage
accunet i800 inwats lds mega800 megacom multiquest operator sdn sub-operator wats-max-band	The assigned service or feature. In addition to pre-defined services or features, any user-defined Facility Type of 0 (feature) or 1 (service) is allowed.

Chapter 112: SIT Treatment for Call Classification

The treatment of Special Information Tones (SITs) used for Outbound Call Management type calls with USA tone characteristics. Branch Gateway internal tone detector resources in classified mode are used to detect SITs. The classifiers are capable of detecting the following SITs:

- SIT Ineffective Other
- SIT Intercept
- SIT No Circuit
- SIT Reorder
- SIT Vacant Code
- SIT Unknown
- AMD (Answering Machine Detected) Treatment

Available only if **ASAI Link Core Capabilities** and **ASAI Link Plus Capabilities** are enabled for the system.

Example command: `change sit-treatment`

```
change sit-treatment                                     Page 1 of 1
SIT TREATMENT FOR CALL CLASSIFICATION

SIT Ineffective Other: dropped
SIT Intercept: answered
SIT No Circuit: dropped
SIT Reorder: dropped
SIT Vacant Code: dropped
SIT Unknown: dropped

AMD Treatment: dropped
Pause Duration (seconds): 1.0
Talk Duration (seconds): 1.5
```

Related links

[ASAI Link Core Capabilities](#) on page 729

[ASAI Link Plus Capabilities](#) on page 729

AMD Treatment

Specifies the treatment for Answering Machine Detected. An ASAI adjunct can request AMD for a call. If an Answering Machine is detected, one of two treatments is specified.

Valid Entry	Usage
answered	Calls are classified as answered, and are therefore sent to an agent.
dropped	Calls are classified as unanswered, and are therefore not sent to an agent. This is the default.

Pause Duration

Fractions of a second pause duration, as opposed to **Talk Duration** that is for full seconds.

Valid Entry	Usage
0.1 to 2.0 seconds in increments of 0.1 seconds	The amount of time Communication Manager looks for a pause before it classifies the call as a live person. The Pause Duration timer should be set longer than the typical silence between words in an answering machine greeting, but shorter than the typical space between words in a live greeting. Defaults to 0.5 seconds.

SIT Ineffective Other

The treatment for “Ineffective Other” messages. An example of an Ineffective Other message is “You are not required to dial a 1 when calling this number”.

Valid Entry	Usage
answered	Calls are classified as answered, and are therefore sent to an agent.
dropped	Calls are classified as unanswered, and are therefore not sent to an agent. This is the default.

SIT Intercept

The treatment for intercept messages that direct callers to another number or extension.

Valid Entry	Usage
answered	Calls are classified as answered, and are therefore sent to an agent. This is the default.
dropped	Calls are classified as unanswered, and are therefore not sent to an agent.

Example

An example of an intercept message is “XXX-XXXX has been changed to YYY-YYYY, make a note of it”.

SIT No Circuit

The treatment for calls with circuit problems, such as busy.

Valid Entry	Usage
answered	Calls are classified as answered, and are therefore sent to an agent.
dropped	Calls are classified as unanswered, and are therefore not sent to an agent. This is the default.

SIT Reorder

The treatment for reorder calls.

Valid Entry	Usage
answered	Calls are classified as answered, and are therefore sent to an agent.
dropped	Calls are classified as unanswered, and are therefore not sent to an agent. This is the default.

SIT Unknown

The treatment for calls when a situation or condition that is unknown to the network is encountered.

Valid Entry	Usage
answered	Calls are classified as answered, and are therefore sent to an agent.
dropped	Calls are classified as unanswered, and are therefore not sent to an agent. This is the default.

SIT Vacant Code

The treatment for calls when a vacant code is encountered.

Valid Entry	Usage
answered	Calls are classified as answered, and are therefore sent to an agent.
dropped	Calls are classified as unanswered, and are therefore not sent to an agent. This is the default.

Talk Duration

Full second talk duration, as opposed to **Pause Duration** that is for fractions of a second.

Valid Entry	Usage
0.1 to 5.0 seconds in increments of 0.1 seconds	The amount of time Communication Manager looks for voice energy. If it finds that much continuous speech, Communication Manager classifies the call as an answering machine. The Talk Duration timer should be set to a time longer than it takes to say a typical live greeting. Defaults to 2.0 seconds.

Chapter 113: Site Data

Provides descriptive information about the buildings, floors, and telephone set colors. Values must be supplied here before the site data can be entered for the station.

From Release 8.1.3.1 onwards, the Site Data screen displays 14 pages of building strings that can hold 1000 entries, instead of two pages containing 150 entries.

To modify the fields on the Site Data screen, run the following command: **change site-data**.

Chapter 114: Station

Administers individual telephone sets or virtual telephones. Some of the fields are used for specific telephone types, while others are used for all telephone types.

To change the value of the fields on the Station screen, run the following command:

change station n , where n is the station number.

To add a station, run the following command:

```
add station next
```

```

add station next
                                     STATION
Do not Share Port with Unnamed Registration? n
      Conf/Trans on Primary Appearance? n
      Bridged Appearance Origination Restriction? n
      Bridging Tone for This Extension? n

Enhanced Callr-Info Display for 1-Line Phones? n

                                     ENHANCED CALL FORWARDING
                                     Forwarded Destination
Unconditional For Internal Calls To: _____ n
                                     External Calls To: _____ n
      Busy For Internal Calls To: _____ n
                                     External Calls To: _____ n
      No Reply For Internal Calls To: _____ n
                                     External Calls To: _____ n

SAC/CF Override: n

```

The following table lists various set types.

Note:

The table also contains older set types. However, Avaya does not provide support for End of Manufacturers Support (EoMS) set types. For more information on EoMS set types, see <https://downloads.avaya.com/css/P8/documents/100172510>.

Telephone type	Model	Administered as
Single-line analog	500	500
	2500 and 2500 with Message Waiting Adjunct	2500
	6210	6210
	6211	6210
	6218	6218
	6219	6218
	6220	6220
	6221	6220
CallerID	Analog telephone with caller ID	CallrID
	7101A and 7102A	7101A
	7103A Programmable and Original	7103A
	7104A	7104A
	8110	8110
	DS1FD	DS1FD
	7302H, 7303H	7303S
	Voice Response Unit (VRU) with C and D tones	VRU
	VRU without C and D tones	2500
Single-line DS1/DSO (Lineside T1/DS1)	DS1 device without forward disconnect	OPS
	VRU with forward disconnect without C and D tones	DS1FD or DS1SA
	VRU with forward disconnect without C and D tones	VRUFD or VRUSA
Terminals	510D	510
	515BCT	515
Multi-appearance hybrid	7303S	7303S, 7313H
	7305H	7305S
	7305S	7305S, 7316H, 7317H
	7309H	7309H, 7313H
	7313H	7313H
	7314H	7314H
	7315H	7315H
	7316H	7316H
	7317H	7317H

Table continues...

Telephone type	Model	Administered as
Multi-appearance digital	2402	2402
	2410	2410
	2420	2420
	6402	6402
	6402D	6402D
	6408	6408
	6408+	6408+
	6408D	6408D
	6408D+	6408D+
	6416D+	6416D+
	6424D+	6424D+
	7401D	7401D
	7401+	7401+
	7403D	7403D
	7404D	7404D
	7405D	7405D
	7406D	7406D
	7406+	7406+
	7407D	7407D
	7407+	7407+
	7410D	7410D
	7410+	7410+
	7434D	7434D
	7444D	7444D
	8403B	8403B
	8405B	8405B
	8405B+	8405B+
	8405D	8405D
	8405D+	8405D+
	8410B	8410B
	8410D	8410D
	8411B	8411B
	8411D	8411D
	9408	9408
	9404	9404

Table continues...

Telephone type	Model	Administered as
	CALLMASTER I	602A1
	CALLMASTER II, CALLMASTER III, CALLMASTER IV	603A1, 603D1, 603E1, 603F1
	CALLMASTER VI	606A1
	IDT1	7403D
	IDT2	7406D
IP Telephone	4601+  Note: When adding a new 4601 IP telephone, you must use the 4601+ station type. This station type has the Automatic Callback feature enabled.	4601+
	4602+  Note: When adding a new 4602 IP telephone, you must use the 4602+ station type. This station type has the Automatic Callback feature enabled.	4602+
	4606	4606
	4610	4610
	4612	4612
	4620SW IP with G3.5 hardware	4620
	4621	4621
	4622	4622
	4624	4624
	4625	4625
	4690	4690
	9611	9611
	9621	9621
	9630	9630
	9641	9641
	9650	9650

Table continues...

Telephone type	Model	Administered as
SIP IP Telephone	• 4602SIP with SIP firmware • 4610SIP with SIP firmware • 9608SIPCC • 9611SIPCC • 9621SIPCC • 9641SIPCC • 4620SIP with SIP firmware • SIP Softphone or Avaya one-X Desktop • Toshiba SP-1020A * Note: Any model telephone with SIP firmware used for SIP networking must be administered as a 4620SIP, 96xxSIP, or 16CC telephone. * Note: Communication Manager Release 6.2 and later do not support 1616SIP CC and 4620SIP CC telephones.	4620SIP
	9601SIP	9608SIP
	9620, 9630, 9630G, 9640, 9640G with SIP firmware	96xx or 96xxSIP telephone
	* Note: • Communication Manager Release 8.0.1 and later internally maps the Avaya B199 Conference Phone conference phones with 9630SIP set types. • You can administer Avaya B199 Conference Phone conference phones from the System Manager user interface only.	
	9608 with SIP firmware	—
	9611 with SIP firmware	9611SIP
	9621 with SIP firmware	9621SIP
	9641 with SIP firmware	9641SIP
	Avaya J129 IP Phone	J129
	Avaya J169 IP Phone	J169

Table continues...

Telephone type	Model	Administered as
	Avaya J179 IP Phone	J179
	J169CC IP Phone	J169CC
	J179CC IP Phone	J179CC
	AvyaSIP	J139, J159, J189
	AvyaSIPCC	J189CC
H.323 SoftPhone	Road Warrior application	H.323 or DCP type
	Native H.323	H.323
	Single connect	H.323 or DCP type
	Any NI-BRI N1 and N2 telephone	NI-BRI
	7505D	7505D
	7506D	7506D
	7507D	7507D
	8503D	8503D
	8510T	8510T
	8520T	8520T
Personal computer	6300 or 7300	Personal computer
Voice or data	6538 or 6539	Constellation
Test line	ATMS	105TL
Administered without hardware	—	XDID or XDIDVIP
Key telephone system interface	—	K2500
Administered without hardware	any digital set	—
	Computer telephony integration (CTI) station	CTI
CTI	CTI station	CTI
XMOBILE	EC500, DECT, or PHS	XMOBILE
ISDN-BRI data module	7500	7500
Separation of Bearer and Signaling (SBS) extension	SBS test extension without hardware	SBS

Table continues...

Telephone type	Model	Administered as
CS1K Endpoints	• CS1k-39xx • CS1k-1col • CS1k-2col • CS1k-ana • CS1k-IP • CS1k-IPCC	CS1k

Station: Page 1

Actual

Displays the actual endpoint configured in System Manager for the AvyaSIP and AvyaSIPCC set types. Actual is a read-only field.

In System Manager, you can add the following J-Series endpoints for the AvyaSIP and AvyaSIPCC set types:

J139, J159, J189, and J189CC.

Attendant

Use this field to configure SIP attendant. This field appears for set type 9641SIP only.

Valid Entry	Usage
n	Disables SIP attendant. This is the default value.
y	Enables SIP attendant. When you set the SIP value to y, ensure that: • Other stations should not have Bridge Appearance of this station. • This station should not be part of any group. For example pickup group, hunt group, or term-extension group. • This station should not have coverage path configured. • Call diversion feature should not be enabled on stations. For example, SAC and call forward • Call restriction feature should not be enabled on stations. For example LNCC and Restrict Last Appearance

BCC

Indicates voice or voice-grade data when the value is set to 0. The Bearer Capability Classes (BCC) value is used to determine compatibility when non-ISDN facilities are connected to ISDN facilities (ISDN Interworking). Available only if **ISDN-PRI** or **ISDN-BRI Trunks** are enabled for the system.

Bearer

Used when Secure Terminal Equipment (STE) telephones are administered as 8510 telephones. The system the **Bearer** field for 8503, 8510, and 8520 stations in Communication Manager 2.1 and 2.2 only. **Secure Terminal Equip** is for Bearer functionality in Communication Manager 3.0 and later.

Valid Entry	Usage
speech	Forces the Bearer Cap IE to "speech" before a call is delivered to the 85xx BRI station.
3.1khz	Leaves the Bearer Cap IE unchanged. Use 3.1khz to let secure calls from Secure Terminal Equipment (STE) telephones to work properly.

Button Module Pages

This field can be configured for CS1000 station types only.

Buttons per Page

This field can be configured for CS1000 station types only. This field appears if **Button Module Pages** field is set to a non-zero value. This field displays a default value of 24 and allows the values 8, 12, 18, 22, and 24. If the value in this field is lower than 24, Communication Manager will do one of the following:

- Allows the Button Module Page to display 24 buttons. However, it blocks form submission if there is a non-blank button in a position higher than the configured number.
- Reduce the number of button fields on the Button Module Pages to the configured number.

Coverage Path 1 or Coverage Path 2

The coverage-path number or time-of-day table number assigned to the station.

Note:

If Modified Misoperation is active, a Coverage Path must be assigned to all stations on Communication Manager.

Cell Phone Number

This field can be configured when the set type is XMOBILE. The unformatted cell telephones published external numbers consisting of 1 to 15 digits. This field can contain a three-digit area

code plus the seven-digit main number. If the same Cell Phone Number is administered for multiple XMOBILE stations, then the Dial Prefix associated with each instance of the Cell Phone Number must be the same. This number consist of a full 10-digit Cell Phone Number regardless of whether the cell telephone is local or not.

Caller ID Message Waiting Indication

Allows or prevents aliasing of various non-Avaya telephones and adjuncts. Available only if **Type** is CallrID. For CallrID type telephones or analog telephones with Caller ID adjuncts only.

Note:

The Caller ID Message Waiting Indication administration is independent of the administration of LED or NEON-lamp Communication Manager Message Waiting Indication (MWI). For example, it is possible to administer a Caller ID telephone with **Caller ID Message Waiting Indication** is disabled and **Message Waiting Indicator** set to neon.

Customizable Labels

Enables or disables the Increase Text for Feature Buttons feature for this station. This feature expands the text labels associated with Abbreviated Dial buttons from the current five uppercase alphanumeric characters to a maximum of 13 upper and lower case alphanumeric characters. Ensures that there will always be sufficient customized button resources to support VIP users. Available only when the station type is one of the following:

- 2410 (Release 2 or later)
- 2420 (Release 4 or later)
- 4610 (IP Telephone Release 2.2 or later)
- 4620 (IP Telephone Release 2.2 or later)
- 4621 (IP Telephone Release 2.2 or later)
- 4622 (IP Telephone Release 2.2 or later)
- 4625 (IP Telephone Release 3.1 or later)

COR

Class of Restriction (COR) number with the required restriction.

COS

The Class of Service (COS) number used to select allowed features.

Configuration Set

Configuration Set field can be configured when the set type is XMOBILE.

Valid Entry	Usage
1 to 10	The configuration set number that contains the call treatment options required for the XMOBILE station. This field must be administered if: • The XMOBILE Type is EC500. • The Mobility Trunk Group is a trunk group number and the administered trunk group is non-DECT or non-PHS. • The Mobility Trunk Group is aar or ars.
blank	If the Mobility Trunk Group is a trunk group number and the administered trunk group is DECT or PHS, this field can be left blank.

Calls Allowed

Identifies the Extension to Cellular call filter type for an XMOBILE station. This field allows an XMOBILE station to function as a bridge and still be restricted. Available only with an EC500-type XMOBILE station and with a **Mapping Mode** of termination or both.

Valid Entry	Usage
internal	External calls are blocked. Internal calls terminate to the XMOBILE station. Attendant-originated and attendant-delivered calls are not delivered
external	Internal calls are blocked. External calls terminate to the XMOBILE station.
all	All calls terminate to the XMOBILE station.
none	Prevents calls from terminating to the XMOBILE station. Can be used to prevent business-related calls from accruing telephone charges on cellular telephones that are lost, being transferred to a new user, or being disabled for other business reasons.

* Note:

Interswitch calls on DCS trunks are treated as internal calls. When this field is set to:

- internal or all, DCS calls are delivered to the cell telephone
- external or none, DCS calls are not delivered

Incoming calls from other Extension to Cellular users are internal if office caller ID is enabled for the XMOBILE station associated with the cell telephone. When this field is set to:

- internal or all, calls from other Extension to Cellular users are delivered
- external or none, calls from other Extension to Cellular users are not delivered

Dial Prefix

This field can be configured when the set type is XMOBILE. The unformatted sequence of up to four digits or characters that are prepended to the cell telephone's published cell telephone number before dialing. Accepts the * and # characters. If the same Cell Phone Number is administered on multiple XMOBILE stations, then the Dial Prefix associated with each instance of the Cell Phone Number must be the same.

Data Module

Indicates whether or not this telephone has an associated data module.

Display Caller ID

Enables or disables transmission of calling party information to the Caller ID telephone or adjunct. For CallrID type telephones or analog telephones with Caller ID adjuncts only. Available only if the station type is CallrID.

Display Cartridge

Enables or disables displaying the cartridge associated with the station. For 7404 D telephones only.

Display Language

Valid Entry	Usage
english french italian spanish user-defined	The language that displays on stations. Time of day is displayed in 24-hour format (00:00 - 23:59) for all languages except English, which is displayed in 12-hour format (12:00 a.m. to 11:59 p.m.).
unicode	Displays English messages in a 24-hour format. If no Unicode file is installed, displays messages in English by default.  Note: Unicode display is only available for Unicode-supported telephones. Currently, 4610SW, 4620SW, 4621SW, 4622SW, 16xx, 96xx, 96x1, 9600-series telephones (Avaya one-X Deskphone Edition SIP R2 or later), and Avaya J100 Series IP Phones support Unicode display. Unicode is also an option for DP1020 (aka 2420J) and SP1020 (Toshiba SIP Phone) telephones when enabled for the system.

Extension

The extension for this station.

For a virtual extension, a valid physical extension or a blank can be entered. With blank, an incoming call to the virtual extension can be redirected to the virtual extension “busy” or “all” coverage path.

The extension length must be within 16 digits.

Endpt Init

Indicates the terminal's endpoint initialization capability. Endpoint initialization is a procedure, required for multipoint operation, by which User Service Order Profile (USOP) is associated with an endpoint on the ISDN-BRI. This association is made through the SPID, administered into the system, and entered into the ISDN-BRI terminal.

Available only if **MIM Support** is enabled.

Valid Entry	Usage
y	The terminal supports Telcordia Technologies ISDN-1 terminal initialization procedures.
n	For all other country protocols.

Expansion Module

Indicates whether or not this telephone has an expansion module. Enables the administration of the buttons for the expansion module.

EMU Login Allowed

Enables or disables using the station as a visited station by an Enterprise Mobility User (EMU).

Endpt ID

Available only if Endpt Init is enabled.

Valid Entry	Usage
00 to 62	A unique two-digit number for this endpoint. Each Endpt ID field must have a unique value for each endpoint on the same port.

Fixed TEI

Indicates whether or not the endpoint has a fixed Terminal Endpoint Identifier (TEI). The TEI identifies a unique access point within a service. TEIs must be administered for fixed TEI terminals.

Available only for ISDN-BRI data modules, NI-BRI telephones and WCBRI data modules.

Hunt-to Station

The extension the system must hunt to for this telephone when the telephone is busy. You can create a station hunting chain by assigning a hunt-to station to a series of telephones.

IP Softphone

Indicates whether or not this extension is either a PC-based multifunction station or part of a telecommuter complex with a call-back audio connection.

Available only for DCP station types and IP Telephones.

IP Video Softphone

Indicates whether or not this extension is a video softphone. Available only if **IP Softphone** is enabled.

IP Video

Use this field to specify whether the extension has IP video capability. The system displays this field for H.323 and SIP station types.

ITC (Information Transfer Capability)

The type of transmission facilities used for ISDN calls originating from this endpoint. Unavailable for voice-only or BRI stations.

Valid Entry	Usage
restricted	Either restricted or unrestricted transmission facilities are used to complete the call. A restricted facility is a transmission facility that enforces 1's density digital transmission. In other words, a sequence of eight digital zeros are converted to a sequence of seven zeros and a digital one.
unrestricted	Only unrestricted transmission facilities are used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1's density digital transmission. In other words, digital information is sent exactly as is.

Lock Messages

Controls access to voice messages by other users.

Valid Entry	Usage
y	Restricts other users from reading or canceling the voice messages, or retrieving messages using Voice Message Retrieval.
n	Allows other users to read, cancel, or retrieve messages.

Location

The system displays this field only when you set the **Multiple Locations** field on the system parameters customer options screen to y, and set the **Type** field to H.323 or SIP station types.

Valid entry	Usage
1 to 2000	(Depending on your server configuration, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .) Assigns the location number to a particular station. Allows IP telephones and softphones connected through a VPN to be associated with the branch an employee is assigned to. This field is one way to associate a location with a station. For the other ways and for a list of features that use location, see the Location sections in <i>Avaya Aura® Communication Manager Feature Description and Implementation</i> .
blank	Indicates that the existing location algorithm applies. By default, the value is blank.

Loss Group

Valid Entry	Usage
1 to 17	Determines which administered two-party row in the loss plan applies to each station. Is not displayed for stations that do not use loss, such as x-mobile stations.

Mapping Mode

This field can be configured when the set type is XMOBILE. Controls the mode of operation in which the cell telephone operates when mapped to this XMOBILE extension. An XMOBILE station can be bridged to a deskset. These restrictions or modes exist because the COR of a bridge is ignored; instead the principal's COR is used. With Mapping Mode, an XMOBILE station can function as a bridge and still be restricted.

When a cell telephone is mapped to more than one XMOBILE station, then only one of the mapped XMOBILE stations can have origination or both as its Mapping Mode. Therefore, only one of the XMOBILE stations mapped to the cell telephone number is permitted to originate calls.

Valid Entry	Usage
both	The cell telephone can be used to originate and terminate calls from its associated XMOBILE extension. This is the default when the XMOBILE type is PHS or DECT.
none	The XMOBILE station is disabled administratively and cannot originate and terminate calls from its associated internal extension.
origination	The cell telephone can be used only to originate calls from its associated internal XMOBILE extension by dialing into the office server running Communication Manager.
termination	The cell telephone can be used only to terminate calls from its associated internal XMOBILE extension. This is the default when the XMOBILE type is EC500.

Message Lamp Ext

The extension of the station tracked with the message waiting lamp.

Media Complex Ext

When used with Multi-media Call Handling, indicates which extension is assigned to the data module of the multimedia complex. Users can dial this extension to place either a voice or a data call, and voice conversion, coverage, and forwarding apply as if the call were made to the 1-number.

Valid Entry	Usage
A valid BRI data extension	For MMCH, enter the extension of the data module that is part of this multimedia complex.
H.323 station extension	For 4600 series IP Telephones, enter the corresponding H.323 station. For IP Softphone, enter the corresponding H.323 station. If you enter a value in this field, you can register this station for either a road-warrior or telecommuter/Avaya IP Agent application.
blank	Leave this field blank for single-connect IP applications.

Message Waiting Indicator

Specifies the type of message waiting indicator. This field is independent of the administration of the Caller ID Message Waiting Indication for CallrID telephones. Must be set to a value other than none when the station type is set to H.323.

Available only for ISDN-BRI data modules and for 500, 2500, K2500, 7104A, 6210, 6218, 6220, 8110, H.323 and VRU telephones.

Valid Entries	Usage
led	The message waiting indicator is a light-emitting diode (LED).
neon	The message waiting indicator is a neon indicator.  Note: The neon message waiting indicator is supported only on a small subset of boards, including older US-only boards, such as the TN746 and the TN793. Check the documentation for your board to see if neon is supported. Available only if the Analog Ringing Cadence is set to 1 (US) for the location parameters.
none	No message waiting indicator is selected. This is the default.

MIM Mtce/Mgt

Indicates if the telephone supports MIM Maintenance and Management capabilities other than endpoint initialization. Available only if **MIM Support** is enabled.

MIM Support (Management Information Message Support)

Enables or disables MIM endpoint initialization (SPID support) and other Maintenance or Management capabilities.

Message Waiting Type

This field can be configured when the set type is XMOBILE.

Valid Entry	Usage
DISPL	The MW text is added to the end of the display line during an incoming or outgoing call.
ICON	The MWI message is used to update to the Wireless Terminal.
NONE	No MWI message is sent to the Wireless Terminal

Mobility Trunk Group

Associates the XMOBILE station to a trunk.

Valid Entries	Usage
2000	A valid trunk group number for mobility routing. This trunk group is used for routing.
aar	The Automatic Alternate Routing (AAR) capabilities of Communication Manager are used to direct the call to an ISDN trunk. If no ISDN trunk is available, the call is not extended out of the server. It provides ringback to the calling company and might eventually go to coverage.
ars	The Automatic Route Selection (ARS) capabilities of Communication Manager are used to direct the call to an ISDN trunk. If no ISDN trunk is available, the call is not extended out of the server. It provides ringback to the calling company and might eventually go to coverage.
blank	Not administered.

Model

The model of the NI-BRI telephone.

Valid Entry	Usage
L-3 Communication STE	The NI-BRI telephone is a model L-3 Communication STE.
Tone Commander	The NI-BRI telephone is a model 6210 and 6220 Tone Commander.
Other	The NI-BRI telephone is another model (for example, a Nortel 5317T).

Name

The name of the person associated with this telephone or data module. The system uses this value to create the system directory.

 Note:

This field is supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW telephones.

For more information on Unicode language display, see *Administering Avaya Aura® Communication Manager*.

 Note:

In the display for emergency notification when completing the **Name** field, fill the most important identifying information at the beginning of the field. When an emergency call is made and a crisis alert station with a 27-character display is notified, only 17 characters of the **Name** field appear on the first display line, followed by the extension. The second line contains the last three characters of the **Name** field, followed by the word `EMERGENCY`. Characters 18 through 24 of the **Name** field are not displayed at all.

Off Premises Station

Available only for analog telephones.

Valid Entries	Usage
y	This telephone is located in a different building with the system. Requires administration of R Balance Network.
n	The telephone is located in the same building with the system.

Personalized Ringing Pattern

Defines the personalized ringing pattern for the station. Personalized Ringing allows users of some telephones to have one of 8 ringing patterns for incoming calls. For virtual stations, this field dictates the ringing pattern on its mapped-to physical telephone.

L = 530 Hz, M = 750 Hz, and H = 1060 Hz

Valid Entries	Usage
1	MMM (standard ringing)
2	HHH
3	LLL
4	LHH
5	HHL
6	HLL
7	HLH
8	LHL

Port

The Auxiliary and Analog ports assigned to the station are as follows.

Valid Entry	Usage
x or X	Indicates that there is no hardware associated with the port assignment since the switch was set up, and the administrator expects that the extension has a non-IP set. Or, the extension had a non-IP set, and it dissociated. Use x for Administered WithOut Hardware (AWOH) and Computer Telephony (CTI) stations, as well as for SBS Extensions.
IP	Indicates that there is no hardware associated with the port assignment since the switch was set up, and the administrator expects that the extension would have an IP set. This is automatically entered for certain IP station set types, but you can enter for a DCP set with softphone permissions. This changes to the s00000 type when the set registers.
xxxVmpp	Specifies the Branch Gateway. • xxx is the Branch Gateway number, which is in the range 001 to 999. • m is the module number, which is in the range 1 to 9. • pp is the port number, which is in the range 01 to 32.
Analog Trunk port	Analog trunk port is available with: • MM711 and MM714 media modules

Passive Signalling Station

The entries y and n indicate the call processing status of the analog trunk port user. This field is available only when you select an analog trunk port as the station type.

Valid Entry	Usage
y	The analog station port does not process calls when off hook.
n	The analog station port processes and receives calls as an analog station. This is the default value.

R Balance Network

You must enter a value in this field if you set the **Off-Premise Station** field to y.

Valid Entry	Usage
y	Selects the Resistor Only Balance network. The Resistor Only Balance network is used when Communication Manager is connected to terminal equipment, such as an SLC carrier or an impedance compensator, that provides an input impedance of 600 ohm and Communication Manager is less than 3,000 feet from the equipment.
n	Selects the Resistor and Capacitor (RC) balance network. RC balance network, when connected to off-premise stations, provide better signal transmission than Resistor only balance network.

Remote Office Phone

Indicates whether or not this station is used as an endpoint in a remote office configuration.

Set

Appears only when the selected station type is CS1000.

Security Code

The security code required by users for specific system features and functions are as follows:

- Extended User Administration of Redirected Calls
- Personal Station Access
- Redirection of Calls Coverage Off-Net
- Leave Word Calling
- Extended Call Forwarding
- Station Lock
- Voice Message Retrieval
- Terminal Self-Administration
- Enterprise Mobility User
- Extension to Cellular
- Call Forwarding
- Posted Messages
- Security Violation Notification
- Demand Printing

The required security code length is administered system wide.

Short/Prefixed Registration Allowed

If the **IP Stations** or the **IP Softphone** field is set to y, the system displays the **Short/Prefixed Registration Allowed** field for this H.323 IP set types: 1603, 1608, 1616, 4601, 4601P, 4602, 4602P, 4606, 4610, 4612, 4620, 4622, 4624, 4630, 9610, 9620, 9630, and 9650.

Valid Entry	Usage
y	Call processing allows the short registration to proceed per station basis. This entry overrides the System Parameters IP Options screen field value for that station.
n	Call processing rejects the short registration per station basis. This entry overrides the System Parameters IP Options screen field value for that station.
default	Call processing uses the Short/Prefixed Registration Allowed field option on the System Parameters IP Options screen for short registration. The default value is default.

Related links

[Short/Prefixed Registration Allowed](#) on page 415

Speakerphone

Controls the behavior of speakerphones.

Valid Entry	Usage
1-way	Indicates that the speakerphone listen-only.
2-way	Indicates that the speakerphone is both talk and listen.
grp-listen	With Group Listen, a telephone user can talk and listen to another party with the handset or headset while the telephone's two-way speakerphone is in the listen-only mode. Others in the room can listen, but cannot speak to the other party through the speakerphone. The person talking on the handset acts as the spokesperson for the group. Group Listen provides reduced background noise and improves clarity during a conference call when a group needs to discuss what is being communicated to another party. Available only with 6400-series and 2420/2410 telephones.
none	Not administered for a speakerphone.

Signaling Type

The **Signaling Type** field is used to specify how the A and B bits are used on an Off Premise Station (OPS) line.

Valid Entry	Usage
Fx	The signaling bits are manipulated as if the port is terminating on a Foreign Exchange (FX) unit in a D4 channel bank. This value is default.
Inverted	The signaling bits are manipulated as if the port is terminating on a Special Access Unit (SAU) in a D4 channel bank.

Survivable COR

Sets a level of restriction for stations to be used with the survivable dial plan to limit certain users to only to certain types of calls. You can list the restriction levels in order from the most restrictive to least restrictive. Each level has the calling ability of the ones above it.

Available for all analog and IP station types.

Valid Entries	Usage
emergency	This station can only be used to place emergency calls.
internal	This station can only make intra-switch calls. This is the default.
local	This station can only make calls that are defined as locl, op, svc, or hnpa in the Survivable Gateway Call Controller's routing tables.
toll	This station can place any national toll calls that are defined as fnpa or natl on the Survivable Gateway Call Controller's routing tables.
unrestricted	This station can place a call to any number defined in the Survivable Gateway Call Controller's routing tables. Those strings marked as deny are also denied to these users.

Survivable Trunk Dest

Designates certain telephones as not being allowed to receive incoming trunk calls when the Branch Gateway is in survivable mode.

Available for all analog and IP station types.

Valid Entry	Usage
y	Allows this station to be an incoming trunk destination while the Branch Gateway is running in survivability mode. This is the default.
n	Prevents this station from receiving incoming trunk calls when in survivable mode.

Survivable GK Node Name

Any valid previously-administered IP node name. Identifies the existence of other H.323 gatekeepers located within gateway products that offer survivable call features. For example, the MultiTech MVPxxx-AV H.323 gateway family and the SLS function within the Branch Gateways. When a valid IP node name is entered into this field, Communication Manager adds the IP address of this gateway to the bottom of the Alternate Gatekeeper List for this IP network region. As H.323 IP stations register with Communication Manager, this list is sent down in the registration confirm message. With this, the IP station can use the IP address of this Survivable Gatekeeper as the call controller of last resort.

If blank, there are no external gatekeeper nodes within a customer's network. This is the default value.

Available only if the station type is an H.323 station for the 46xx or 96xx models.

Secure Terminal Equip

Used when Secure Terminal Equipment (STE) telephones are administered as 8510 telephones. Available only for 8503, 8510, and 8520 stations in Communication Manager 3.0 and later.

Valid Entries	Usage
y	Leave the Bearer Cap IE unchanged. 3.1khz is used to let secure calls from Secure Terminal Equipment (STE) telephones work properly.
n	Force the Bearer Cap IE to "speech" before a call is delivered to the 85xx BRI station.

SPID — (Service Profile Identifier)

The Service Profile Identifier (SPID) for this station. The SPID is a variable length parameter required for ISDN-BRI stations.

The SPID is a numeric string, which means that the value of 00 is different from 000. The SPID must be different for all terminals on the BRI and from the Service SPID. The SPID should always be assigned. If the SPID is not assigned for the first BRI on a port, any other BRI assignment to that port is blocked.

 Note:

If you have not administered a port for an ISDN-BRI extension and intend to use Terminal Translation Initialization (TTI) to assign the port, then the SPID number must equal the station number.

Available only if the terminal supports ISDN terminal initialization procedures.

Special Dialing Option

Identifies the type of dialing for calls when this data module originates calls.

Valid Entry	Usage
hot-line	The Hot Line Service destination number is stored in an Abbreviated Dialing List. When the user goes off-hook on a Data Hot Line call, the system dials the AD number.
default	The system dials the default dialing AD number.
blank	Normal keyboard dialing.

System ID

This field can be configured for CS 1000 station types only. This field allows you to leave the field blank or enter a string of up to 9 characters. With Release 8.0 and later, more than one station can use the combination of **System ID** and **Terminal Number**.

With Release 8.0.1 and later, each station must have a unique combination of **System ID** and **Terminal Number**.

TEI

Available only if the station has a fixed Terminal Endpoint Identifier (TEI).

Valid Entry	Usage
0 to 63	A one- or two-digit number that identifies the TEI of the station.

Tests

Valid Entry	Usage
y	Enables port maintenance tests.
n	Disables port maintenance tests. Required if the equipment (dictaphone) connected to the port does not support these tests.

Time of Day Lock Table

Valid Entry	Usage
1 to 5	Assigns the station to a Time of Day (TOD) Lock/Unlock table. The assigned table must be administered and active.
blank	Indicates no TOD Lock/Unlock feature is active. This is the default.

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Type

Use this field to specify the telephone type. You must administer the station type for each station that you add to the system.

The following table lists the telephones, virtual telephones, and personal computers that you can administer on Communication Manager. Telephones that are not in the table require an alias to a supported set type.

 Note:

If Terminal Translation Initialization is enabled, you cannot change the analog telephones administered with hardware to a virtual extension.

Telephone type	Model	Administered as
Single-line analog	500	500
	2500, 2500 with Message Waiting Adjunct	2500
	6210	6210
	6211	6210
	6218	6218
	6219	6218
	6220	6220
	6221	6220
CallerID	Analog telephone with Caller ID	CallrID
	7101A, 7102A	7101A
	7103A Programmable and Original	7103A
	7104A	7104A
	8110	8110
	DS1FD	DS1FD
	7302H, 7303H	7303S

Table continues...

Telephone type	Model	Administered as
	Voice Response Unit (VRU) with C&D tones	VRU
	VRU without C&D tones	2500
Single-line DS1/DSO	DS1 device without forward disconnect	ops
	VRU with forward disconnect without C&D tones	ds1fd or ds1sa
	VRU with forward disconnect without C&D tones	VRUFD or VRUSA
Terminals	510D	510
	515BCT	515
Multi-appearance hybrid	7303S	7303S, 7313H
	7305H	7305S
	7305S	7305S, 7316H, 7317H
	7309H	7309H, 7313H
	7313H	7313H
	7314H	7314H
	7315H	7315H
	7316H	7316H
	7317H	7317H
Multi-appearance digital	2402	2402
	2410	2410
	2420	2420
	6402	6402
	6402D	6402D
	6408	6408
	6408+	6408+
	6408D	6408D
	6408D+	6408D+
	6416D+	6416D+
	6424D+	6424D+
	7401D	7401D
	7401+	7401+
	7403D	7403D
	7404D	7404D
	7405D	7405D
	7406D	7406D

Table continues...

Telephone type	Model	Administered as
	7406+	7406+
	7407D	7407D
	7407+	7407+
	7410D	7410D
	7410+	7410+
	7434D	7434D
	7444D	7444D
	8403B	8403B
	8405B	8405B
	8405B+	8405B+
	8405D	8405D
	8405D+	8405D+
	8410B	8410B
	8410D	8410D
	8411B	8411B
	8411D	8411D
	8434D	8434D
	9404	9404
	9408	9408
	CALLMASTER I	602A1
	CALLMASTER II, III, IV	603A1, 603D1, 603E1, 603F1
	CALLMASTER VI	606A1
	IDT1	7403D
	IDT2	7406D
IP Telephone	4601+  Note: When you add a new 4601 IP telephone, you must use the 4601+ station type. This station type has the Automatic Callback feature enabled.	4601+

Table continues...

Telephone type	Model	Administered as
	4602+	4602+
	 Note: When you add a new 4602 IP telephone, you must use the 4602+ station type. This station type has the Automatic Callback feature enabled.	
	4606	4606
	4610	4610
	4612	4612
	4620SW IP (G3.5 hardware)	4620
	4621	4621
	4622	4622
	4624	4624
	4625	4625
	4690	4690
	9608	9608
	9610	9610
	9611	9611
	9620	9620
	9621	9621
	9630	9630
	9640	9640
	9641	9641
	9650	9650

Table continues...

Telephone type	Model	Administered as
SIP IP Telephone	• 4602SIP with SIP firmware • 4610SIP with SIP firmware • 4620SIP with SIP firmware • 4620SIP CC (Call Center) • SIP Softphone/Avaya one-X Desktop • Toshiba SP-1020A  Note: You must administer any telephone with SIP firmware used for SIP networking as a 4620SIP, 96xxSIP, or 16CC SIP telephone.  Note: Communication Manager Release 6.2 and later do not support 1616SIP CC and 4620SIP CC telephones.	4620SIP
	9601SIP	9608SIP
	9620, 9630, 9630G 9640, 9640G with SIP firmware	96xx or 96xxSIP telephone
	9608 with SIP firmware	9608SIP
	9611 with SIP firmware	9611SIP
	9621 with SIP firmware	9621SIP
	9641 with SIP firmware	9641SIP
	9608 with SIP firmware (for call center)	9608SIPCC
	9611 with SIP firmware (for call center)	9611SIPCC
	9621 with SIP firmware (for call center)	9621SIPCC
	9641 with SIP firmware (for call center)	9641SIPCC
	AvyaSIP	J139, J159, J189
	AvyaSIPCC	J189CC
H.323 SoftPhone	application Road Warrior	H.323 or DCP type
	Native H.323	H.323
	Singleconnect	H.323 or DCP type
	Any NI-BRI (N1 and N2) telephone	NI-BRI
	7505D	7505D
	7506D	7506D
	7507D	7507D

Table continues...

Telephone type	Model	Administered as
	8503D	8503D
	8510T	8510T
	8520T	8520T
Personal computer	6300/7300	PC
(voice/data)	6538/9	Constellation
Test Line	ATMS	105TL
No hardware assigned at the time of administration.		• Use XDID when Communication Manager assigns a DID number to the station. • Use XDIDVIP when the administrator later assigns a DID number to the station. • Use virtual to map the station and other extensions to one physical telephone.
Key telephone system interface	—	K2500
AWOH	any digital set	—
	CTI station	CTI
CTI	CTI station	CTI
XMOBILE	EC500, DECT, PHS	XMOBILE
ISDN-BRI data module	7500	7500
SBS Extension	SBS test extension (no hardware)	sbs

Terminal Number

This field can be configured for CS 1000 station types only. You can enter numbers in the following range: 0.0.0.0 to 252.1.15.31.

The first digit must be divisible by 4. For example: 0, 4, 8, ..., 252.

Unicode Name

This read-only field shows the assignment of Unicode Name to the station.

Valid entry	Usage
n	Unicode Name is not assigned to the station.
y	Unicode Name is assigned to the station.

For more information on Unicode Language Display, see *Administering Avaya Aura® Communication Manager*.

XOIP Endpoint type

Available only for 500, 2500, K2500 and CallrID station types.

Valid Entries	Usage
auto modem fax tty	Identifies the endpoint type of the analog station. Use this field for older or non-standard external equipment such as modems, fax, and TTY devices that are not easily recognized by VoIP resources within Communication Manager. By identifying this external equipment through administration, VoIP firmware can determine whether to immediately attempt to put a call in pass-through mode, or allow the system to handle it normally. Default is auto.  Note: This field is intended for exception cases only. For the majority of stations, use the default setting of auto.

XID

Use this field to enable the Layer 2 XID testing capability. The system displays this field only for an ISDN-BRI data module.

XMOBILE Type

The type of XMOBILE station. Available only if the **Type** field is XMOBILE and the **Mobility Trunk Group** field is administered.

Valid Entry	Usage
DECT	The DECT Access System or the AGCS (ROAMEO) IS-136 (TDMA cellular). This represent the ISDN-based DECT system.
EC500	Any public cellular networks.
IPDECT	The H.323 IP-based DECT system. This can either be the ASCOM or the DeTeWe product.
PHS	The DENSO 300M.

Station: Page 2

Att. Call Waiting Indication

With attendant call waiting, attendant-originated or attendant-extended calls to a busy single-line telephone can wait and sends distinctive call-waiting tone to the single-line user.

Valid Entry	Usage
y	Activates Call Waiting for the telephone without Caller ID information. This feature must be enabled when the Type is set to H.323. This is the default.
n	Call Waiting is disabled for the station. Disable this feature if: • Data Restriction is enabled • Switchhook Flash field is disabled • Data Privacy is enabled for the telephone's class of service (COS)
c	Enables the Caller ID Delivery with Call Waiting feature, which displays CID information on for the waiting call. Available only if the station Type is CallrID.

Active Station Ringing

Defines how calls ring to the telephone when it is off-hook without affecting how calls ring at this telephone when the telephone is on-hook.

Valid Entry	Usage
continuous	All calls to this telephone ring continuously.
single	Calls to this telephone receive one ring cycle and then ring silently.
if-busy-single	Calls to this telephone ring continuously when the telephone is off-hook and idle. Calls to this telephone receive one ring cycle and then ring silently when the telephone is off-hook and active.
silent	All calls to this station ring silently.

AUDIX Name

The voice messaging system associated with the station. Must contain a user-defined adjunct name that was previously administered.

Always Use

Use this field to enable the following emergency call handling settings:

- A softphone can register irrespective of the emergency call handling settings the user has entered into the softphone. If a softphone dials 911, the value administered in the **Emergency Location Extension** field is used as the calling party number. The user-entered emergency call handling settings of the softphone are ignored.
- If an IP telephone dials 911, the value administered in the **Emergency Location Extension** field is used as the calling party number.
- If an agent dials 911, the physical station extension is used as the calling party number, overriding the value administered in the **LoginID for ISDN Display** field.

Does not apply to SCCAN wireless telephones, or to extensions administered as type H.323.

Auto Answer

In an Expert Agent Environment (EAS) environment, the auto answer setting for an Agent LoginID overrides the endpoint settings when the agent logs in. In EAS environments, the auto answer setting for the Agent LoginID can override a station's setting when an agent logs in.

Valid entry	Usage
all	All ACD and non-ACD calls to an idle station cut through immediately. The agent cannot use automatic hands-free answer for intercom calls. With non-ACD calls, the station rings while the call is cut through. To prevent the station from ringing, activate the ringer-off feature button, provided the Allow Ringer-off with Auto-Answer feature is enabled for the system.
acd	Only ACD split, ACD skill, and direct agent calls cut through. Non-ACD calls to the station ring audibly. For analog stations: • Only the ACD split or skill calls and direct agent calls cut through. • Non-ACD calls receive busy treatment. If the station is active on an ACD call and a non-ACD call arrives, the agent receives call-waiting tone.
none	All calls to the station receive an audible ringing.
icom	The user can answer an intercom call from the same intercom group without pressing the intercom button.

Audible Message Waiting

Audible Message Waiting field enables or disables an audible message waiting tone indicating the user has a waiting message consisting of a stutter dial tone when the user goes off-hook.

This field does not control the Message Waiting lamp.

Available only if **Audible Message Waiting** is enabled for the system.

Automatic Moves

With Automatic Moves, you can unplug a DCP telephone from one location and moved to a new location without additional Communication Manager administration. Communication Manager automatically associates the extension to the new port.

Caution:

When a DCP telephone is unplugged and moved to another physical location, the **Emergency Location Extension** must be changed for that extension or the USA Automatic Location Identification data base must be manually updated. If the **Emergency Location Extension** is not changed or if the USA Automatic Location Identification data base is not updated, the DID number sent to the Public Safety Network could send emergency response personnel to the wrong location.

Valid Entry	Usage
always	The DCP telephone can be moved anytime without additional administration by unplugging from one location and plugging into a new location.
once	The DCP telephone can be unplugged and plugged into a new location once. After a move, the field is set to done the next time that routine maintenance runs on the DCP telephone. Use when moving a large number of DCP telephones so each extension is removed from the move list. Also use to prevent automatic maintenance replacement.
no	Requires administration to move the DCP telephone.
done	Communication Manager sets the field to done after the telephone is moved and routine maintenance runs on the DCP telephone.
error	Communication Manager sets the field to error, after routine maintenance runs on the DCP telephone, when a non-serialized telephone is set as a movable telephone.

Auto Select Any Idle Appearance

Auto Select Any Idle Appearance field enables or disables automatic selection of any idle appearance for transferred or conferenced calls. Communication Manager first attempts to find an idle appearance that has the same extension number as the call being transferred or conferenced has. If that attempt fails, Communication Manager selects the first idle appearance.

Adjunct Supervision

Available only if the station type is 500, 2500, k2500, 8110, ops, ds1fd, ds1sa, VRU, VRUFD, or VRUSA.

Valid Entry	Usage
y	An analog disconnect signal is sent automatically to the port after a call terminates. Analog devices, such as answering machines and speakerphones, use this signal to turn the devices off after a call terminates.
n	Required hunt group agents are alerted to incoming calls. In a hunt group environment, the disconnect signal blocks the reception of zip tone and incoming call notification by an auto-answer station when a call is queued for the station.

Busy Auto Callback without Flash

Enables or disables automatic callback for a calling analog station without flashing the hook. Available for analog telephones only if **Without Flash** is enabled for the system.

Bridged Call Alerting

Controls how the user is alerted to incoming calls on a bridged appearance.

Valid Entry	Usage
y	The bridged appearance rings when a call arrives at the primary telephone.
n	The bridged appearance flashes but does not ring when a call arrives at the primary telephone. This is the default. If disabled and Per Button Ring Control is also disabled, audible ringing is suppressed for incoming calls on bridged appearances of another telephone's primary extension.

Bridged Idle Line Preference

Use this field to specify that the line that the system selects when you go off hook is always an idle call appearance for incoming bridged calls.

Valid entry	Usage
y	The user connects to an idle call appearance instead of the ringing call.
n	The user connects to the ringing bridged appearance.

CDR Privacy

Enables or disables Call Privacy for each station. With CDR Privacy, digits in the called number field of an outgoing call record can be blanked on a per-station basis. The number of blocked digits is administered system-wide as CDR parameters.

Coverage Msg Retrieval

Allows or denies users in the telephone's Coverage Path to retrieve Leave Word Calling (LWC) messages for this telephone. Applies only if the telephone is enabled for LWC Reception.

Coverage After Forwarding

Governs whether an unanswered forwarded call is provided coverage treatment.

Valid Entry	Usage
y	Coverage treatment is provided after forwarding regardless of the administered system-wide coverage parameters.
n	No coverage treatment is provided after forwarding regardless of the administered system-wide coverage parameters.
s(ystem)	Administered system-wide coverage parameters determine treatment.

Call Waiting Indication

User, attendant-originated, and outside calls to a busy single-line telephone can wait and sends a distinctive call-waiting tone to the single-line user. Not available if **Data Restriction** is enabled, **Switchhook Flash** is disabled, or if Data Privacy is active by way of the telephone COS assignment.

Valid Entry	Usage
y	Activates Call Waiting without Caller ID information for the telephone. This is the default.
n	Call Waiting is disabled for the station.
c	Enables the Caller ID Delivery with Call Waiting feature, which displays CID information on for the waiting call. This value can only be entered when the station type is CallrID.

Call Appearance Display Format

Specifies the display format for the station. Bridged call appearances are not affected by this field. This field is available only on telephones that support downloadable call appearance buttons, such as the 2420 and 4620 telephones.

Note:

This field sets the administered display value only for an individual station.

Valid Entry	Usage
loc-param-default	The system uses the administered system-wide default value. This is the default.
inter-location	The system displays the complete extension on downloadable call appearance buttons.
intra-location	The system displays a shortened or abbreviated version of the extension on downloadable call appearance buttons.

Conf/Trans On Primary Appearance

Enables or disables the forced use of a primary appearance when the held call to be conferenced or transferred is a bridge. This is regardless of the administered value for **Auto Select Any Idle Appearance**.

Data Restriction

Enables or disables data restriction that is used to prevent tones, such as call-waiting tones, from interrupting data calls. Data restriction provides permanent protection and cannot be changed by the telephone user. Cannot be assigned if **Auto Answer** is administered as all or acd. If enabled, whisper page to this station is denied.

Display Client Redirection

Enables or disables the display of redirection information for a call originating from a station with Client Room Class of Service and terminating to this station. When disabled, only the client name and extension or room display. Available only if Hospitality is enabled for the system.

*** Note:**

This field must be enabled for stations administered for any type of voice messaging that needs display information.

Direct IP-IP Audio Connections

Use this field to enable direct audio connections between IP endpoints. Direct audio connections save bandwidth resources and improve the sound quality of voice over IP transmissions.

*** Note:**

This field has no impact on a SIP station (OPS) on shuffling. For SIP stations, this field is controlled by the "Direct IP-IP Audio Connections" field in the Signaling group form.

Distinctive Audible Alert

Use this field to enable the distinctive audible alerts to allow telephones to receive three different types of ringing patterns that identify the type of incoming calls. Distinctive ringing might not work properly for off-premises telephones.

Emergency Location Ext

The Emergency Location Extension for this station. This extension identifies the street address or nearby location when an emergency call is made. Defaults to the telephone's extension. Accepts up to sixteen digits.

*** Note:**

On the ARS Digit Analysis Table in Communication Manager, 911 must be administered to be call type emer or alrt for the E911 Emergency feature to work properly.

Related links

[Remote Soft Phone Emergency Calls](#) on page 675

EC500 State

Use this field to observe or change the EC500 state of the station .This status is also observed and changed by administering an EC500 button on the station.

Valid Entry	Usage
enabled	When EC500 state is enabled, a call to that station generates a new outbound call from the Communication Manager to the configured EC500 destination, typically a mobile phone.
disabled	When EC500 state is disabled, the outbound call to the EC500 destination is not generated.

H.320 Conversion

Use this field to enable the conversion of H.320-compliant calls to voice-only calls for the attendant console.

 Note:

The system can handle only a limited number of conversion calls. Therefore, the number of attendant consoles with H.320 conversion must be limited.

Ignore Rotary Digits

Indicates whether or not rotary digits from the set are ignored. If enabled, the short switch-hook flash (50 to 150) from a 2500-type set is ignored.

LWC Reception

Use this field to specify the location where the system must store the LWC messages.

Valid entry	Usage
spe	Use this option to store the LWC messages on Switch Processor Element (SPE).
none	Use this option if you do not want to store the LWC messages.
audix	Use this option to store the LWC messages on the voice messaging system.

Related links

[AUDIX Name](#) on page 358

LWC Activation

Activates or deactivates the Leave Word Calling (LWC) feature. With LWC, internal telephone users on this extension can leave short pre-programmed messages for other internal users.

You must use LWC if:

- The system has hospitality and the guest-room telephones require LWC messages indicating that wakeup calls failed
- The LWC messages are stored in a voice-messaging system

LWC Log External Calls

This field can be configured when the set type is XMOBILE. Determines whether or not unanswered external call logs are available to end users. When external calls are not answered, Communication Manager keeps a record of up to 15 calls provided information on the caller identification is available. Each record consists of the latest call attempt date and time.

Multimedia Early Answer

Enables or disables multimedia early answer on a station-by-station basis.

You must enable the station for the Multimedia Early Answer feature if the station receives coverage calls for multimedia complexes, but is not multimedia-capable. This ensures that calls are converted and the talk path is established before ringing at this station.

MWI Served User Type

Controls the auditing or interrogation of a served user's message waiting indicator (MWI).

Valid Entries	Usage
fp-mwi	The station is a served user of an fp-mwi message center.
qsig-mwi	The station is a served user of a qsig-mwi message center.
sip-adjunct	Used to audit message waiting lamps.
blank	The served user's MWI is not audited or if the user is not a served user of either an fp-mwi or qsig-mwi message center.

Per Button Ring Control

Using this option you can enable or disable ring control for every button, provided you have the station user credentials.

Valid Entries	Usage
y	To enable Automatic Abbreviated and Delayed ring transition for each call-appr on the station, select ring behavior individually for each call-appr or brdg-appr option. To prevent the system from automatically moving the line selection to a silently alerting call, unless the call was audibly ringing earlier.  Note: The abrdg-appr option is unavailable for SIP station.
n	To enable the calls on call-appr buttons always to ring the station To enable the calls on brdg-appr buttons always ring or not ring based on the Bridged Call Alerting value To move line selection to a silently alerting call, if the call is not audibly ringing the station  Note: The abrdg-appr option is unavailable for SIP station.

Per Station CPN - Send Calling Number

Determines Calling Party Number (CPN) information sent on outgoing calls from this station.

Valid Entries	Usage
y	All outgoing calls from the station deliver the CPN information as "Presentation Allowed."

Table continues...

Valid Entries	Usage
n	No CPN information is sent for the call.
r	Outgoing non-DCS network calls from the station delivers the Calling Party Number information as "Presentation Restricted."
blank	The sending of CPN information for calls is controlled by administration on the outgoing trunk group the calls are carried on.

Precedence Call Waiting

Activates or deactivates Precedence Call Waiting for this station.

Redirect Notification

Enables or disables redirection notification that gives a half ring at this telephone when calls to this extension are redirected through Call Forwarding or Call Coverage. Must be enabled if LWC messages are stored on a voice-messaging system.

Restrict Last Appearance

Valid Entries	Usage
y	Restricts the last idle call appearance used for incoming priority calls and outgoing call originations only.
n	Last idle call appearance is used for incoming priority calls and outgoing call originations.

Recall Rotary Digit

Enables or disables the Recall Rotary Digit dialing that enables this user to perform conference and transfer operations.

Available only if the station type is 500 or 2500.

Remote Soft Phone Emergency Calls

Tells Communication Manager how to handle emergency calls from the IP telephone.

Caution:

An Avaya IP endpoint can dial emergency calls (for example, 911 calls in the U.S.). It only reaches the local emergency service in the Public Safety Answering Point area where the telephone system has local trunks. You cannot use an Avaya IP endpoint to dial to and connect with local emergency service when dialing from remote locations that do not have local trunks. Avoid using an Avaya IP endpoint to dial emergency numbers for emergency services when dialing from remote locations. Avaya LLC is not responsible or liable for any damages resulting from misplaced emergency calls made from an Avaya endpoint. Your use of this product indicates that you have read this advisory and agree to use an alternative telephone to dial all emergency calls from remote locations. If you have

questions about emergency calls from IP telephones, go to the Avaya Support website at <http://support.avaya.com>.

Available only if the station is an IP Softphone or a remote office station.

Valid Entry	Usage
as-on-local	If the emergency location extension that corresponds to this station's IP address is not administered (left blank), the value as-on-local sends the station emergency location extension to the Public Safety Answering Point (PSAP). If the administrator populates the IP address mapping with emergency numbers, the value as-on-local functions as follows: • If the station emergency location extension is the same as the IP address mapping emergency location extension, the value as-on-local sends the station's own extension to the Public Safety Answering Point (PSAP). • If the station emergency location extension is different from the IP address mapping emergency location extension, the value as-on-local sends the IP address mapping extension to the Public Safety Answering Point (PSAP).
block	Prevents the completion of emergency calls. Use this entry for users who move around but always have a circuit-switched telephone nearby, and for users who are farther away from the server than an adjacent area code served by the same 911 Tandem office. When users attempt to dial an emergency call from an IP Telephone and the call is blocked, they can dial 911 from a nearby circuit-switched telephone instead.
cesid	Allows Communication Manager to send the CESID information supplied by the IP Softphone to the PSAP. The end user enters the emergency information into the IP Softphone. Use this entry for IP Softphones with road warrior service that are near enough to the server that an emergency call reaches the PSAP that covers the softphone's physical location. If the server uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the IP Softphone. If the server uses CAMA trunks for emergency calls, the end user enters a specific digit string for each IP Softphone location, based on advice from the local emergency response personnel.
option	Allows the user to select the option (extension, block, or cesid) that the user selected during registration and the IP Softphone reported. This entry is used for extensions that can be swapped back and forth between IP Softphones and a telephone with a fixed location. The user chooses between block and cesid on the softphone. A DCP or IP telephone in the office automatically selects the extension.

Service Link Mode

Use this field to specify the duration of a service link connection. The service link is the combined hardware and software multimedia connection between an H.320 Desktop Video Conferencing (DVC) system and Communication Manager.

The service link is established when a user receives or makes a call during a multimedia, IP softphone, or IP telephone session.

Valid entry	Usage
as-needed	For multimedia, IP softphone, and IP telephone users. The service link remains connected for 10 seconds after the user disconnects a call so that the user can immediately make or receive another call. After 10 seconds, the link is disconnected, and a new link must be established to make or receive a call.
permanent	For call center agents who are constantly making or receiving calls during the multimedia, IP softphone, or IP telephone session. The service link remains connected for the entire duration of the session.

Switchhook Flash

Valid Entry	Usage
y	Allows users to use the switchhook flash function to activate Conference/Transfer/ Hold and Call Waiting. Required for H.323 station types.
n	Disables the flash function so that when the switchhook is pressed while active on a call, the call drops. Requires that Call Waiting Indication is disabled.

Select Last Used Appearance

Valid Entry	Usage
y	Indicates a station's line selection is not to be moved from the currently selected line button to a different, non-alerting line button. The line selection on an on-hook station only moves from the last used line button to a line button with an audibly alerting call. If there are no alerting calls, the line selection remains on the button last used for a call.
n	The line selection on an on-hook station with no alerting calls can be moved to a different line button that might be serving a different extension.

Station: Page 3

Access Code

A five-digit access code used to place a wireless terminal into service. The access code is a temporary, shorter version of the complete User Authentication Key (UAK) required by the system when the terminal is first put into service. It is used to automatically generate a unique UAK for that wireless terminal over-the-air.

Available only if a wireless terminal model number is selected as the station type.

Bridged Appearance Origination Restriction

Restricts or allows call origination on the bridged appearance.

Valid Entry	Usage
y	Call origination on the bridged appearance is restricted.
n	Call origination on the bridged appearance is allowed. This is normal behavior, and is the default.

Bridging Tone for This Extension

Allows you to enable or disable a single burst of tone when a station bridges on to the principal's call.

Valid Entry	Usage
y	Enables a single burst of tone when a station bridges on to the principal's call.
n	Disables a single burst of tone when a station bridges on to the principal's call. This is the default value for a new SIP station, or when a SIP station upgrades to Communication Manager 8.0 or later from a previous release.

Do not Share Port with Unnamed Registration

Controls whether x-port station can be shared or not when a request comes from an un-named IP address. This field is applicable for DCP phones with x-port only. This field appears only when **Unnamed Registrations and PSA for IP Telephones** field is set to y on the system-parameters features screen and DCP station.

Valid Entry	Usage
y	Communication Manager does not allow x-port station to be shared when a request comes from un-named IP address.
n	Communication Manager allows x-port station to be shared when a request comes from un-named IP address. This is the default value.

Enhanced Callr-Info display for 1-line display phones

Use this field to enable or disable the **Callr-Info Persists** feature. Use this feature with telephones that have a one-line display and a **callr-info** button that displays the collected digits. For more information on **Callr-info Persists**, see *Avaya Aura® Call Center Elite Feature Reference*.

Valid entry	Usage
n	This entry is the default. If an agent presses the callr-info button, the telephone displays the callr-info digits. The callr-info digits are cleared after: - the timeout period, defined by the Callr-info Display Timer field, expires or - the agent disconnects the call
y	On receiving an incoming call with callr-info digits associated, the indicator of the callr-info button flashes. - If the agent presses the callr-info button, the caller information is displayed on the telephone and the status lamp of the button becomes steady. If the agent presses the callr-info button after the telephone displays the caller information, the telephone displays the caller information again and the indicator of the callr-info button winks. - If the agent does not press the button, the status lamp continues to flash during the call.

Forwarded Destination

A destination extension for both internal and external calls for each of the three types of enhanced call forwarding (Unconditional, Busy, and No Reply). Accepts up to 18 digits. The first digit can be an asterisk *.

Requires administration to indicate whether the specific destination is active (enabled) or inactive (disabled).

Home

Indicates the roaming status of the wireless user. Available only when a wireless terminal model number is selected as the station type.

Valid Entry	Usage
y	The user's home. This is the default.
n	The roaming system.

IP Phone Group ID

Available for H.323 and SIP station types.

Valid entry	Usage
0 to 999 blank	The Group ID number for the station.

IPEI

Available when a wireless terminal model number is selected as the station type.

Valid Entry	Usage
0 to 9 a to f A to F blank	The International Portable Equipment Identifier of the wireless terminal. Accepts a unique nine-character hexadecimal ID number.

Offline Call Logging

Use the **Offline Call Logging** field to back up and restore call logs for an offline user.

Valid entry	Usage
y	Communication Manager tracks the calls that the user missed when the telephone was in the logged-out state, and updates the call logs on the telephone when the user logs in. The default option is y.
n	Communication Manager does not track the calls that the user missed when the telephone was in the logged-out state.

Require Mutual Authentication if TLS

The TLS server, Communication Manager, and the TLS client, 96x1 H.323 phone, must both exchange Identity certificates and mutually authenticate the other side certificate.

If an H.323 phone has a certificate and the **Require Mutual Authentication if TLS** field is set to:

Valid entry	Usage
y	Communication Manager validates the certificate. If the certificate is valid, Communication Manager allows TLS setup with H.323 phones. If the certificate is invalid, Communication Manager rejects the TLS request.
n	Communication Manager does not check for the phone certificate. Instead Communication Manager checks for the root certificate. If the root certificate is valid, Communication Manager allows TLS setup with H.323 phones to proceed without an endpoint certificate. If the root certificate is invalid or expired, Communication Manager rejects the TLS request.

If an H.323 phone does not have a certificate and the **Require Mutual Authentication if TLS** field is set to:

Valid entry	Usage
y	Communication Manager does not accept the security profile, H323TLS, in the Gatekeeper Confirmation (GCF). If the endpoint attempts to set up TLS, Communication Manager rejects the TLS request.
n	Communication Manager does not check for the phone certificate. Instead, Communication Manager checks for the root certificate. If the root certificate is valid, Communication Manager allows TLS setup with H.323 phones to proceed without an endpoint certificate. If the root certificate is invalid or expired, Communication Manager rejects the TLS request.

SAC/CF Override

With **SAC/CF Override**, the user of the calling station can override the redirection set by the called station.

Valid entry	Usage
ask	The system prompts the user of the calling station whether the call must follow the redirection path or override the redirection path. The user can type y or n.
no	The user of the calling station cannot override the redirection path of the call. The call follows the redirection path.
yes	The user of the calling station can override the redirection path of the call, provided the called station has at least one idle call appearance.

Station: Page 4

Auto-A/D

Enables or disables automatic abbreviated/delayed ringing for a call appearance. Available only if **Per Button Ring Control** is enabled.

Abbreviated Dialing List 1, List 2, List 3

Assigns up to three abbreviated dialing lists to each telephone.

Valid Entry	Usage
enhanced	Telephone user can access the enhanced system abbreviated dialing list.
group	Telephone user can access the specified group abbreviated dialing list. Requires administration of a group number.

Table continues...

Valid Entry	Usage
personal	Telephone user can access and program their personal abbreviated dialing list. Requires administration of a personal list number.
system	Telephone user can access the system abbreviated dialing list.

Building

A valid building location.

Cable

Identifies the cable that connects the telephone jack to the system.

Cord Length

The length of the cord attached to the receiver. This is a free-form entry, and can be in any measurement units.

Floor

A valid floor location.

Headset

Indicates whether or not the telephone has a headset.

HOT LINE DESTINATION — Dial Code

Hot Line Destination - Dial code is the dial code in the specified abbreviated dialing list where the hotline destination number is stored. This field can be configured when the set type is XMOBILE or 8510T (and other BRI types).

HOT LINE DESTINATION — Abbreviated Dialing Dial Code

This field is available only if **Special Dialing Option** is hot-line.

Hot Line Service is used when very fast service is required and when a telephone is used only for accessing a certain facility. This field can be configured when the set type is XMOBILE or 8510T (and other BRI types).

HOT LINE DESTINATION — Abbreviated Dialing List Number

The abbreviated dialing list where the hotline destination number is stored. This field can be configured when the set type is XMOBILE or 8510T (and other BRI types).

Jack

Alpha-numeric identification of the jack used for this station.

Mounting

Indicates whether the station mounting is d(esk) or w(all).

Rg

Valid Entry	Usage
a(bbreivated-ring) d(elayed-ring) n(o-ring) r(ing)	The type of automatic abbreviated/delayed ringing for each call appearance when per button ring control is enabled. Default is r.

Room

Valid Entry	Usage
<i>Telephone location</i>	Identifies the telephone location. Accepts up to 10 characters.
<i>Guest room number</i>	Identifies the guest room number if this station is one of several to be assigned a guest room and the Display Room Information in Call Display is enabled for the system. Accepts up to five digits.

Speaker

Indicates whether or not the station is equipped with a speaker.

Set Color

Indicates the set color. Valid entries include the following colors: beige, black, blue, brown, burg (burgundy), gray, green, ivory, orng (orange), red, teak, wal (walnut), white, and yel (yellow).

You can change the list of allowed set colors by using the Valid Set Color fields on the site-data screen.

Voice mail Number

Valid Entries	Usage
digits (1 to 9)	Provides a voice mail retrieval AUX dial button on 4xx, 46xx, and 96xx telephones. When a number is entered in this field, the telephone's fixed voice mail retrieval button acts as an autodial button, dialing the number entered in this field to access voice mail. Accepts up to 24 digits.  Note: If this field is left blank, the telephone's fixed voice mail retrieval button acts as a "transfer to voice mail" button that only works for Intuity Audix or QSIG-integrated voice mail systems. For Intuity Audix and QSIG-integrated voice mail systems, leave the field blank. For non-QSIG integrated voice mail systems, set the field to an appropriate number.
*	
#	
~p (pause)	
~w/~ (wait)	
~m (mark)	
~s (suppress)	

Station :Page 5

BUTTON ASSIGNMENTS

The feature assigned to each button on the station. Feature buttons are assigned by entering the abbreviated feature. For a list of feature buttons, see Telephone Feature Buttons Table in *Administering Avaya Aura® Communication Manager*.

 Note:

To use Terminal Translation Initialization (TTI), a call appearance (call-appr) must be assigned to the first button position. TTI needs the button on the first call appearance to get dial tone.

Data Extension

Use this field to assign an extension to the data module. You can enter up to five digits. This value must conform to the dial plan.

Default Dialing Abbreviated Dialing Dial Code

The list number associated with the abbreviated dialing list.

When the user goes off-hook for a data call and presses the **Enter** button following the DIAL prompt, the system dials the AD number.

Available only if the **Special Dialing Option** is set to default.

1-Step Clearing

Enables or disables call termination at the WCBRI terminal when the user drops from the call.

Station: Page 6

Enable Reachability for Station Domain Control

Using the **Enable Reachability for Station Domain Control** field, you can enable or disable options polling for specific stations.

*** Note:**

When the system-parameters features form is set to `disable-all`, the station form field is overridden or made inaccessible.

Enable Reachability for Station Domain Control field has the following options:

Valid Entry	Usage
y	To enable the polling feature for an individual SIP station.
n	To disable the polling feature for an individual SIP station.
system	To take action based on the option selected on the system-parameters features form, type <code>system</code>. * Note: The default option is set to <code>system</code>. The administration option has effect only if the station is domain-controlled.

Primary Session Manager, Secondary Session Manager, Third Session Manager, and Fourth Session Manager

These four fields are populated by System Manager and are read-only on Communication Manager. You cannot modify these fields from the Communication Manager SAT interface.

If the call to this SIP station is routed to a SIP trunk with a clustered signaling group, then the SIP Trunk Optimization feature sends the Invite to the station on the primary Session Manager. If primary Session Manager is not reachable, the Invite is sent to the secondary Session Manager.

The Session Manager selection for routing to a SIP station does not apply to a call between two SIP stations. In this case, Session Manager decides the destination Session Manager for the called SIP station.

*** Note:**

IP addresses are allocated to these fields, when the policy-based assignment is enabled in System Manager.

Route Selection

This field displays the value of the Trunk Selection of the corresponding off-pbx-telephone station mapping entry if there is one. The valid entries are:

- ars

- aar
- Trunk group number: tg1 to tg2000
- Route pattern number: rp1 to rp2000

SIP URI

Use this field to place a call using the URI. URI is an alphanumeric string such as letters, numbers, and special characters. You can enter up to 70 characters, including handle and domain name. For example:

help.desk@example.com

helpdesk123@example.com

helpdesk@example.com

Following are not supported:

- Unicode characters
- Eurofont characters
- Using same URIs across hunt groups, VDNs, and stations within the same Communication Manager.

Type of 3PCC Enabled

Indicates if the telephone can be controlled by an Avaya third Party Call Control (3PCC) adjunct or a Computer Telephony Integration (CTI) adjunct. The valid entries are none and Avaya.

Assigned Member — Ext

The extension of the user who has an associated **Data Extension** button and shares the module. This field is available only if the station type is 6408D and data module is set to y.

Assigned Member — Name

The name associated with the extension of the user who has an associated **Data Extension** button and shares the module. This field is available only if the station type is 6408D and data module is set to y.

Automatic Selection of DID Numbers

Enables or disables the Automatic Selection of DID Numbers for Guest Rooms feature. This feature assigns a two- to five-digit extension from a predetermined list of numbers to a hotel room telephone number that is disassociated with the room number.

Mute Button Enabled

Enables or disables the mute button on the station.

Country Protocol

The protocol that corresponds to the supported initialization and codesets. The Country Protocol must match any previously-administered endpoint on the same port.

Coverage Module

Indicates whether or not a coverage module is connected to the station.

Custom Selection of VIP DID Numbers

The selection or removal of a DID number assigned to a room when a guest checks in. Available only if **Automatic Selection of DID Numbers** is enabled.

Related links

[Automatic Selection of DID Numbers](#) on page 331

Data Option

Valid Entry	Usage
analog	A second line on the telephone is administered on the I-2 channel.

Table continues...

Valid Entry	Usage
data module	A second line on the telephone is <i>not</i> administered on the I-2 channel.
none	The data option is not administered.

DSN Number

You can enter a 10-digit extension, or leave the field blank.

Feature Module

Indicates whether or not the station is connected to a feature module.

Idle/Active Ringing (Callmaster)

Defines how a call rings to the telephone when it is on-hook. Applies to CALLMASTER telephones. This field is available only if station type is 602A1.

Valid Entry	Usage
continuous	All calls to this telephone ring continuously.
if-busy-single	Calls to this telephone ring continuously when the telephone is off-hook and idle, and calls to this telephone receive one ring cycle and then ring silently when the telephone is off-hook and active.
silent-if-busy	Calls ring silently when this station is busy.
single	Calls to this telephone receive one ring cycle and then ring silently.

Idle Appearance Preference

Indicates which call appearance is selected when the user lifts the handset and there is an incoming call.

Valid Entry	Usage
y	The user connects to an idle call appearance instead of the ringing call.
n	The Alerting Appearance Preference is set and the user connects to the ringing call appearance.

Map-to Station

The extension of a physical telephone used for calls to a virtual extension. Cannot be used with an xmobile, xdid or any other virtual extension.

This field is applicable only for the virtual endpoints.

Message Server Name

Specifies which Message Server is associated with the station. Must contain a user-defined adjunct name that was previously administered. Names must be administered in alphabetical order.

PCOL/TEG Call Alerting

Enables or disables alerting at the station for Personal CO Line/Terminating Extension Group calls.

Available only for 510 telephones.

Public (PSTN) Number

You can enter an 11-digit extension, or leave the field blank.

Chapter 115: Stations With Off-PBX Telephone Integration

Maps an office telephone to a cell phone through the Extension to Cellular feature. The office telephone can be a standard office number or an administration without hardware (AWOH) station.

Example command: `add off-pbx-telephone station-mapping`

Stations With Off-PBX Telephone Integration: page 1

Application

The type of off-PBX application that is associated with the office telephone. More than one application can be assigned to an office telephone.

Valid Entry	Usage
blank	Default is blank.
CSP	cell phone with Extension to Cellular provided by the cellular service provider
EC500	cell phone with Extension to Cellular
HEMU	Home Enterprise Mobility User
OPS	SIP enabled phone
PBFMC	Public Fixed Mobile Convergence
PVFMC	Private Fixed Mobile Convergence
SCCAN	wireless SIP phone and cell phone
SPFMC	Service Provider Fixed-Mobile Convergence
VEMU	Visited Enterprise Mobility User
VIEMU	Visited Initial Enterprise Mobility User
ONE-X	It is used by Avaya one-X® Client Enablement Services. You cannot enter any data for the ONE-X application types, but you can see this on the display command if it is active.

Table continues...

Valid Entry	Usage
EMX	Ring extension A station extension can have a maximum of four simultaneous ring extensions. With Communication Manager R10.1.3, when you register simultaneous ring extensions on the Avaya Workplace Client, you can specify which extensions must simultaneously ring when a call is made to the primary station. You cannot enter data for the EMX application type using Communication Manager. You must use the Avaya Workplace Client to enter data for the EMX application type. However, on Communication Manager, you can view the data for the EMX application type using the following commands: display off-pbx-telephone station-mapping, list off-pbx-telephone station-mapping.

CC

The country code associated with the extension. Accepts up to three digits. Multiple entries that use the same telephone number must also have the same country code.

Country code changes made to existing stations or XMOBILE entries are applied to all instances of the telephone number.

SAFE (Self-Administered Feature Access Code for EC500) must not be used with an extension that has an administered country code. Origination mapping can occur with or without a country code. Default is blank.

Configuration Set

Valid Entry	Usage
1 to 99	The Configuration Set number that contains the required call treatment options for the Extension to Cellular station. The SCCAN application requires two different configuration sets selected for each station. The first set is the value for the WLAN followed by a slash. The second is the value for the cellular network.
blank	Enterprise Mobility User (EMU) and ONE-X application type display a blank entry.. The default setting is blank.

Dial Prefix

The dial prefix the system prepends to the off-switch telephone number before dialing the off-switch telephone. The system deletes the dial prefix when a user enters their cell phone number using the Self Administration Feature (SAFE) access code. The routing tables must be set properly so that the dial prefix 1 is not necessary for correct routing. Communication Manager inserts a plus (+) digit before initiating a call to a SIP endpoint or a cell phone.

- asterisk (*) or hash (#) must be in the first digit position
- 1 must be used if the telephone number is long-distance

- 011 must be used if the telephone number is international

Dual Mode

The **Dual Mode** field provides functionality for dual mode wireless phones that operate both in the public cellular network using GSM or CDMA and in a wireless network using SIP. The **Dual Mode** field indicates which application is part of a dual-mode pair. The dual Mode pair determines the pair of channels that the wireless phone is using, one channel over GSM/CDMA and the other channel over SIP. The dual Mode pairing is required to transfer a call to the correct channel when the call is moving between GSM/CDMA and SIP networks.

Valid Entry	Usage
blank	The application is not part of a dual-mode pair.
DM1	The application belongs to a dual-mode pair. Applications of the same pair have the same value.
DM2	The application belongs to a dual-mode pair. Applications of the same pair have the same value.
DMX	Supported only for Private Fixed Mobile Convergence application for pairing with one-X application.

* Note:

dm1 or dm2 can be assigned to any two of the Public Fixed Mobile Convergence (PBFMC), Private Fixed Mobile Convergence (PVFMC), and Service Provider Fixed-Mobile Convergence (SPFMC) applications. However, one of the applications must be PVFMC. The other application can be PBFMC or SPFMC.

Phone Number

The **Phone Number** field accepts:

- The telephone number of the off-switch telephone.
- The telephone number of an on-PBX extension for the one-X[®] application type.

The **Phone Number** field can be blank for:

- the first EC500, CSP, or PBFMC telephones you administer
- EC500, CSP, and PBFMC telephones that support self-administered Feature Access Code for EC500 (SAFE)

Station Extension

The number of the office telephone that is mapped to the cell telephone. Accepts up to sixteen digits. This number must be an administered extension. Default is blank.

Trunk Selection

The Trunk Selection field indicates the trunk group selection method used for the outgoing call to Session Manager only if the application field is set to ops. If the application field is set to any other value, for example EC500, then the selected trunk will not be Session Manager. The selected trunk will be some other destination, for example, to a cell phone in the PSTN. In the half-call model, indicates a different trunk group than the trunk group that eventually connects the call to the telephone network.

The valid options are:

- AAR
- ARS
- TG1 to TG2000
- RP1 to RP2000

Stations With Off-PBX Telephone Integration: page 2

Bridged Calls

Determines if bridged call appearances extend to the Extension to Cellular cell telephone.

 Note:

For an Avaya SIP AST phone, this field must be set to **none**.

Valid Entry	Usage
termination	Users can use their Extension to Cellular cell phone to only receive calls from the associated office telephone. Users cannot use the cell phone to originate calls from the associated office telephone. Calls originating from the cell phone independent of the office phone are independent of Extension to Cellular and behave exactly as before enabling Extension to Cellular.  Note: The on-PBX extensions that the ONE-X application type enables are in termination mapping mode only.
origination	Users can originate Extension to Cellular cell phone calls only from the associated office telephone. Users cannot use the cell phone to receive calls from the associated office telephone.
both	Users can originate Extension to Cellular cell phone calls from the associated office telephone, as well as receive calls from the associated office telephone.
none	Users cannot originate or receive calls from the office telephone with the cell phone. This is the default value.

Call Limit

Use this field to set the maximum number of calls that can be made simultaneously by using an application, such as Extension to Cellular (EC500) or Off-Premise Station (OPS), on a station. You can enter a value from 1 to 10. For example, if you set the **Call Limit** field for an EC500 application to 2 and the **Call Limit** field for an OPS application to 4, the station can have up to two EC500 calls and up to four OPS calls.

Different scenarios are as follows:

- For SIP station type, the call limit is not administered using the OPS form. Instead, the call limit is automatically counted by the call appearance and bridge button administered on the Station screen. In this scenario, the Station screen is used for SIP station only.
- For H.323 station type that has an associate SIP configuration, the call limit for OPS station is administered using the OPS form, which is independent of the button admin on the Station screen. The Station screen is used for H.323 configuration only. For SIP configuration, administration is done using the OPS form.
- For EC500, the call limit is administered using the OPS form for both SIP and H.323 stations.

Calls Allowed

Identifies the call filter type for an Extension to Cellular station. Determines the type of calls to the office telephone a user can receive on an Extension to Cellular cell telephone.

With Communication Manager R10.1.3, you can view the EMX simultaneous ring extensions mapped to the primary station.

Valid Entry	Usage
all	The cell telephone receives internal and external calls. The default value is all. If the value is set to “all” for an EMX simultaneous ring extension, and if a call is made to the primary station, the ring extension rings with the primary station.
internal	The cell telephone receives only internal calls.
external	The cell telephone receives only external calls.
none	The cell telephone does not receive any calls to the associated office telephone. If the value is set to “none” for EMX simultaneous rings for an extension, then that extension does not ring when a call is made to the primary station.

Location

Valid entry	Usage
1 to 2000	(Depending on your server configuration, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .) The location value for each administered OPS, PBFMC, SPFMC or PVFMC application. Typically, set this to the same location number being used by the desk set. For more information on location, see <i>Avaya Aura® Communication Manager Feature Description</i> .
blank	Trunk location is used for the outgoing calls from Off-PBX endpoints and location of station is used for incoming calls to Off-PBX endpoints. Blank is the default.

Related links

[Application](#) on page 690

[Loc Number](#) on page 484

Mapping Mode

The mode of operation for the Extension to Cellular cell phone. These modes control the degree of integration between the cell phone and the office telephone. The modes are valid for Extension to Cellular calls only. For each office telephone, only one cell phone can be assigned as the origination mode. A cell phone cannot be assigned as either the origination or both mode more than once.

Valid Entry	Usage
termination	Users can use their Extension to Cellular cell phone to only receive calls from the associated office telephone. Users cannot use the cell phone to originate calls from the associated office telephone. Calls originating from the cell phone independent of the office phone are independent of Extension to Cellular and behave exactly as before enabling Extension to Cellular.  Note: The on-PBX extensions that the ONE-X application type enables are in termination mapping mode only.
origination	Users can originate Extension to Cellular cell phone calls only from the associated office telephone. Users cannot use the cell phone to receive calls from the associated office telephone.
both	Users can originate Extension to Cellular cell phone calls from the associated office telephone, as well as receive calls from the associated office telephone.
none	Users cannot originate or receive calls from the office telephone with the cell phone. This is the default value.

Stations With Off-PBX Telephone Integration: page 3

Share Level

Indicates the sharing level. Enter the value from 1-9.

Labels

Displays the label assigned by Avaya Workplace Client for a given EMX simultaneous ring extension. For example, you can label extension 88888888 as Home-num and 99999999 as Office-num. Similarly, you can label each sim-ring extension of a primary station. You can also leave the label blank. The maximum number of characters that the Communication Manager displays is ten. By default, the field is blank.

Chapter 116: Survivable ARS Analysis Table

Communication Manager compares dialed numbers with the dialed strings in this table and determines the route pattern for the number.

Example command: `change survivable-ars-analysis`

Call Type

Valid Entry	Usage
emer npa hnpa intl iop locl natl op svc	The call type used for the dialed string.

Deny

Indicates whether or not the dialed string should be blocked. The system denies a dialed string that does not match an entered pattern.

Dialed String

Communication Manager matches the dialed numbers with the entry in the **Dialed String** field that most closely matches the dialed number. You can enter up to 18 digits that the call-processing server analyzes. You can also enter the wildcard characters, `x` and `X`.

Total Length

Valid Entry	Usage
0 to 28	The minimum number of digits required to validate the route. The minimum value when the dial string is populated is the length of the dialed string entry with a maximum value up to 28. Default is blank.

Trunk Grp No

The trunk group number that specifies the destination route for the dial plan analysis of this dialed string.

Chapter 117: Survivable Processor

Adds information specific to a Survivable Remote Server (Local Survivable Processor) or to connect certain adjuncts to a Survivable Remote or Survivable Core server (Enterprise Survivable Server). Before administering this screen, you must first assign node names for each Survivable Remote and Survivable Core media server on the IP Node Names screen.

While this screen is administered on the active main server, the information entered applies only to Survivable Remote and Survivable Core servers and does not apply to main servers. When translations are copied to a Survivable Remote or Survivable Core server, the Survivable Remote Server/Survivable Core Server replaces like translations for the main server with the overrides administered on the Survivable Processor screens. That is, use the Survivable Processor screen to administer overrides against adjunct links that have already been administered for the main servers. For more information about survivable core server, see *Avaya Aura® Communication Manager Survivable Options*.

Example command: `add survivable-processor n`, where *n* is the assigned node name.

Survivable processor: page 1

Cluster ID

Valid Entry	Usage
1 to 999	The Cluster ID for the survivable core server. The Cluster ID corresponds to the Module ID as found on the Server Role web page of the Survivable Core Server or Survivable Remote Server.  Note: The <code>statuslicense -v</code> shell command on the survivable core server or the survivable remote server displays the Module ID.

Community Size

Valid Entry	Usage
Sngl_PN	For a survivable core server, the value must be Sngl_PN.
all	Default is all.

Enable PE for H.248 Gateways

Enables or disables using the PE interface of the survivable core server for gateways.

Enable PE for H.323 Endpoints

Enables or disables using the PE interface of the survivable core server for H.323 devices such as telephones.

IP Address

The IP address that corresponds to the node name.

There are three IP addresses, one for each node name if the survivable processor is a duplicated survivable core.

Related links

[Name](#) on page 413

Node Name

The previously-administered name used to identify the server. If the survivable processor is duplicated, there are three node names, one each for the duplicated server pair and one for the server that is active at a given point.

Priority Score

Valid Entry	Usage
1 to 100	The priority score for this survivable core server. Default is 1.

Processor Ethernet Network

Valid Entry	Usage
1 to 2000	The network region in which the PE interface of the Survivable Remote or Survivable Core server resides.

Serial Number

If you configure the **Topology** field as **Edge**, provide the 12-character serial number of Communication Manager LSP server. Do the following to know the serial number:

- In the Communication Manager LSP server, go to **Communication Manager System Management Interface > Administration > Server (Maintenance) > Server Role**.
- In the VMware ESXi, go to S8300 **VMware ESXi Host user interface > Host**.

Server A — Server ID

Valid Entry	Usage
1 to 256	Server A ID corresponds to the Server ID configured on the survivable core server. The administration on the main server and the configuration on the survivable core server must match for the survivable core server to register to the main server.

Server B — Server ID

Valid Entry	Usage
1 to 256	For duplicated survivable core servers, displays the node name of Server B.

Survivable License Grace Period (in days)

The **Survivable License Grace Period (in days)** field defines how many days the Enterprise Survivable Server (ESS) can control the communication devices after the main Communication Manager server encounters an error. The ESS functions as a backup for the main Communication Manager server. This setting is configurable only on main Communication Manager server and not on ESS.

If the grace period expires and the main Communication Manager server error is not resolved, the ESS changes to a restricted mode and does not manage communication devices.

For example, if the grace period is set to **60 days**, the ESS manages the communication devices for up to 60 days after the main Communication Manager server encounters an error.

 Note:

This field is applicable only to ESS and not to LSP.

Valid entry	Usage
30, 60, 90, and 180	On the main Communication Manager server, go to the Survivable Processor screen and enter the preferred grace period (in days) in the Survivable License Grace Period (in days) field. When the ESS is active, translation sync does not occur between main Communication Manager and ESS. The default value is 30 days.

System Preferred

Valid Entry	Usage
y	Allows one survivable core server to replace the main server to keep as much of the system network intact as possible. Disables Local Preferred and Local Only and cannot be changed. This is the default except for an S8400 survivable core server.
n	The default for an S8400 survivable core server and cannot be changed.

Topology

Use the **Topology** field to configure the network topology on the LSP, either in the Enterprise mode with a single shared network connecting the main Communication Manager and LSP or in an Edge topology within a separate sub-network.

Valid Entry	Description
Edge	Use the Communication Manager LSP server deployed for redundancy in a cloud-connected branch office or a separate sub-network behind a Network Address Translation (NAT) device.
Enterprise	Use the Communication Manager LSP server in a traditional role in an enterprise topology. This is the default setting.

Type

Valid Entry	Usage
lsp simplex_ess duplex_ess	The survivable processor type.

Survivable Processor: page 3 (IP Services page)

Used when an AESVCS or a CDR connects to the Survivable Remote or Survivable Core server.

Enabled

Specifies how data for each specified service type is transferred to the survivable processor.

Valid Entry	Usage
i(nherit)	This link is to be inherited by the Survivable Remote or Survivable Core server. The survivable processor inherits this service type just as it is administered on the main server. Used in the following situations: The main server connects to the adjuncts using the main server's PE interface and you want the Survivable Remote or Survivable Core server to connect to the adjunct using its PE interface
n(o)	This IP services link is disabled on the Survivable Remote or Survivable Core server. This is the default.
o(overwrite)	Overwrites the processor channel information sent in the file sync from the main server. With the entry o(overwrite), the administered CDR or AE Services attributes can be modified uniquely for each individual Survivable Remote or Survivable Core server.

Local Node

Displays the previously-administered node name.

Related links

[Name](#) on page 413

Local Port

The originating port number. For client applications such as Call Detail Recording (CDR), this field defaults to 0.

Priority with respect to Media Servers

Use this field to assign the priorities to the media servers.

The priority of main server is always 1, so you must enter a priority value from 2 to 9999.

Remote Node

The name at the far end of the link for the CDR. Does not apply for AESVCS.

Remote Port

Valid Entry	Usage
5000 to 65500	The port number of the destination for CDR or AESVCS. This number must match the port administered on the CDR or AESVCS server.

Service Type

Valid Entry	Usage
CDR1 CDR2 AESVCS	The service provided.

Related links

[Service Type](#) on page 429

Store to disk

Enables or disables the storage of the CDR data on the local hard drive of the Survivable Remote or Survivable Core server. Pertains only to service types set to CDR1 or CDR2.

Survivable Processor: page 4 (IP Services — Session Layer Timers page)

Available only if CDR1 or CDR2 is administered, and if processor channel information is administered to be overwritten.

Related links

[Service Type](#) on page 703

[Enabled](#) on page 702

Connectivity Time

Valid Entry	Usage
1 to 255	The amount of time that the link can be idle before Communication Manager sends a connectivity message to ensure the link is still up. Default is 60.

Packet Resp Timer

Valid Entry	Usage
1 to 255	The number of seconds to wait from the time a packet is sent until a response (acknowledgement) is received from the far-end, before trying to resend the packet. Default is 30.

Reliable Protocol

Enables or disables using a reliable protocol over this link. A reliable protocol should be used if the adjunct on the far end of the link supports it.

Service Type

Displays the previously-administered service type.

Related links

[Service Type](#) on page 703

Session Connect Message Cntr

Valid Entry	Usage
1 to 5	The number of times Communication Manager tries to establish a connection with the far-end adjunct.

SPDU Cntr

Valid Entry	Usage
1 to 5	The number of times Communication Manager transmits a unit of protocol data before generating an error.

Chapter 118: System Capacity

Provides a status of administered capacity information and a snapshot status of system resources. For more information, see *Maintenance Commands for Avaya Aura® Communication Manager, Branch Gateways and Servers*. Detailed system capacity information can be found in *Avaya Aura® Communication Manager System Capacities Table*.

Chapter 119: System Configuration

The System Configuration screen shows all the boards on the system that are available for connecting telephones. Used to view the board number, board type, and status of each board's ports.

Chapter 120: System Parameters Call Coverage/Call Forwarding

Sets the system-wide parameters for call coverage and call forwarding.

Example command: `change system-parameters coverage-forwarding`

System parameters call coverage/ call forwarding: page 1

```
change system-parameters coverage-forwarding Page 1 of 2
SYSTEM PARAMETERS CALL COVERAGE / CALL FORWARDING
CALL COVERAGE/FORWARDING PARAMETERS
  Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings): 2
  Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings): 2
  Coverage - Caller Response Interval (seconds): 4
  Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls: 1
Location for Covered/Forwarded Calls: called VDN/Hunt Group Location: all
  PGN/TN/COR for Covered and Forwarded Calls: caller
  COR/FRL check for Covered and Forwarded Calls? n
  QSIG/SIP Diverted Calls Follow Diverted to Party's Coverage Path? n

COVERAGE
  Criteria for Logged Off/PSA/TTI Stations? n
  Keep Held SBA at Coverage Point? y
  External Coverage Treatment for Transferred Incoming Trunk Calls? n
  Immediate Redirection on Receipt of PROGRESS Inband Information? n
  Maintain SBA At Principal? y
  QSIG VALU Coverage Overrides QSIG Diversion with Rerouting? n
  Station Hunt Before Coverage? n

FORWARDING
  Call Forward Override? n Coverage After Forwarding? y
```

CALL COVERAGE / FORWARDING PARAMETERS

Coverage - Caller Response Interval (seconds)

Valid Entry	Usage
0 to 10	The time in seconds an internal caller has before a call redirects to the called party's first coverage point.

COR/FRL check for Covered and Forwarded Calls

Valid Entry	Usage
y	Communication Manager checks the following: • COR of the calling and forwarded-to parties for local forwarding. • COR and FRL of the calling party and outgoing trunk for remote forwarding and remote coverage calls.
n	Communication Manager does not check the following: • COR of the calling and forwarded-to parties for local forwarding. • COR and FRL of the calling party and outgoing trunk for remote forwarding and remote coverage calls. This is the default value.

Diverted Party Identification

The **Diverted Party Identification** field is available only if the **QSIG/SIP Diverted Calls Follow Diverted to Party's Coverage Path** field is enabled. The default setting is last-fwd.

The field settings do not work if the incoming call is over a non-QSIG or non-SIP trunk or the call is routed through a route-to vector step.

Valid entry	Usage
principal	To play voice mail greeting of the principal and save the voice message at the voicemail box of the principal.
last-fwd	To play voice mail greeting of the last-forwarded party and save the voice message at the voicemail box of the last-forwarded party. The default configuration.

Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings)

Valid Entry	Usage
1 to 99	Specifies: • The number of rings applied at a local coverage point before a call redirects to the next coverage point • The number of rings applied at the principal before a call forwards when Call Forwarding Busy/Don't Answer is activated  Note: When ringing local destinations, such as an office environment, a short interval often is appropriate because the intended party either is near the telephone or not present. However, if the call is left at an off-net destination for only a short interval, the call can be redirected to the next destination before the intended party has any real chance of answering the call.

Location for Covered and Forwarded Calls

Determines the location number used for coverage and forwarding.

Valid Entry	Usage
called	• If the called party is registered or in-service, coverage and forwarding use the called party's physical phone's location number. • If the called party is AWOH (x-port) or unregistered, coverage and forwarding use a location based on the type of set. - IP set: location 1 - DCP set: location 0 (all) • When the forwarding or coverage destination is to UDP instead of to an external destination starting with the ARS FAC, routing is always based on the caller's physical phone's location regardless of how this field is administered. This is the default.
caller	Coverage and forwarding use the caller's physical phone's location number.

Related links

[ARS FAC](#) on page 482

VDN/Hunt Group Location

Determines the location to be used when a VDN or hunt group launches a call.

Valid Entry	Usage
all	The Communication Manager uses the default Dial Plan, Automatic Route Selection (ARS), and Automatic Alternate Routing (AAR) forms to route the call. These forms have the Location set to "all".
caller	The Communication Manager uses the Dial Plan, Automatic Route Selection (ARS), and Automatic Alternate Routing (AAR) forms associated with the location of the calling station or trunk to route the call. These forms have a specific location in the range 1-2000. As per normal routing rules, Communication Manager prefers the specific-location entries over the all-location entries if both tables match the called number.

Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings)

Valid Entry	Usage
1 to 99	Specifies: - The number of rings applied at an off-net coverage point before a call is redirected to the next coverage point - The number of rings applied at an off-net forwarded-to destination before the call is redirected to coverage.  Note: When ringing local destinations, such as an office environment, a short interval often is appropriate because the intended party either is near the telephone or not present. However, if the call is left at an off-net destination for only a short interval, the call can be redirected to the next destination before the intended party has any real chance of answering the call.

PGN/TN/COR for Covered and Forwarded Calls

Valid Entry	Usage
called	Communication Manager checks for permissions (tenant number, partition group number, and COR) between the actual called party (forwarding party) and forwarded-to party. When the check is successful, the call is routed to the forwarded-to party. Permission check for COR is only done if the COR/FRL check for Covered and Forwarded Calls field is set to y.
caller	Communication Manager checks for permissions (tenant number, partition group number, and COR) between the calling party and forwarded-to party. When the check is successful, the call is routed to the forwarded-to party. Permission check for COR is only done if the COR/FRL check for Covered and Forwarded Calls field is set to y. This is the default value.

QSIG/SIP Diverted Calls Follow Diverted to Party's Coverage Path

When a party with a defined coverage path does not answer the call, Communication Manager determines whether to continuously ring the call or redirect the call to a station or a messaging system. The calls follow the coverage path of the diverted-to party only if the calls route over QSIG and SIP trunks.

Valid entry	Usage
y	Communication Manager forwards the call to the coverage path of the last redirected party. The Diverted Party Identification field displays the default value as last-fwd.
n	The default configuration. The call continuously rings at the extension of the redirected party even if that extension has a coverage path. The call does not reach the messaging system of the last party in the coverage path.

Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls

Valid Entry	Usage
1 to 7	If the number of incoming trunk calls routes within the Call Forward timer, the block commences. Applies for those occasions when an incoming call to a station redirects off-net. At that time, the Call Forward timer activates to block any further incoming calls to that station from being redirected off-net until the timer expires.
n (all)	Call processing never activates the Call Forward timer. Therefore, any number of calls to a principal can be redirected off-net.

COVERAGE

Criteria for Logged Off/PSA/TTI Stations

Enables or disables the call coverage criteria for logged-off IP/PSA/TTI stations. By default, the value is n.

Default treatment for logged off phones is a busy tone. If you set the **Criteria for Logged Off/PSA/TTI Stations** field to y, calls follow the coverage treatment as configured on the station screen. The call is redirected to coverage after the number of rings exceed the number specified in the **Number of Rings** field for **Criteria for Logged Off/PSA/TTI Stations**.

Related links

[Active](#) on page 165

External Coverage Treatment for Transferred Incoming Trunk Calls

Enables or disables external coverage treatment for incoming trunk calls that redirect to coverage.

Immediate Redirection on Receipt of PROGRESS Inband Information

This field determines if a call is immediately redirected to coverage when an ISDN PROGRESS message is received. This field pertains only to CCRON and QSIG VALU coverage calls redirected over end-to-end ISDN facilities.

However, the message might indicate that the cell telephone is not available to receive calls and should be redirected.

The system displays this field only if:

- The **Coverage of Calls Redirected Off-Net** field shows y.
- The Value-Added Avaya (VALU) feature is enabled for the system.

Valid entry	Usage
y	Immediately redirect an off-net coverage/forwarded call to the next coverage point. Users in European countries following the ETSI standard and redirecting to GSM cellular telephones should enable the field.

Table continues...

Valid entry	Usage
n	Does not immediately redirect an off-net coverage/forwarded call to the next coverage point. Users in the United States should disable this field because PROGRESS messages with the Progress Indicator field set to inband information are sent for a variety of reasons that are not associated with unavailable cellular telephones.

Related links

[Cvg Of Calls Redirected Off-net](#) on page 731

[Value Added \(VALU\)](#) on page 744

Keep Held SBA at Coverage Point

Determines how a covering user who has placed an answered coverage call on hold is treated if the original principal bridges onto the call.

Valid Entry	Usage
y	Keeps the coverage party on the call. The coverage party remains on hold, but might enter the call along with the principal and the calling party.
n	Drops the coverage party from the call.

Maintain SBA At Principal

Using this, a user can maintain a simulated bridged appearance (SBA) when a call redirects to coverage.

Valid Entry	Usage
y	Maintains a simulated bridged appearance on the principal's telephone when a call redirects to coverage. DCS with rerouting will not be attempted after coverage.
n	No SBA is maintained on the principal's telephone. DCS with rerouting is attempted, and if successful, the principal loses the bridged appearance and the ability to bridge onto the coverage call.

QSIG VALU Coverage Overrides QSIG Diversion with Rerouting

Specifies whether or not, with both QSIG Diversion with Rerouting and QSIG VALU turned on, Coverage After Forwarding for the station works for calls that go to remote coverage.

Available only if **Basic Supplementary Services** and **Supplementary Services with Rerouting** are both enabled for the system.

Valid Entry	Usage
y	QSIG VALU call coverage takes precedence. If Coverage After Forwarding is enabled for a station, the call can receive coverage after rerouting.
n	With QSIG Diversion with Rerouting turned on, the local system passes control of a forwarded call to the remote QSIG server on which the forwarding destination resides. The forwarded call cannot return to coverage for the user who originally received the call.

Related links

[Basic Supplementary Services](#) on page 743

[Supplementary Services with Rerouting](#) on page 744

Station Hunt Before Coverage

Determines whether or not a call to a busy station performs station hunting before going to coverage.

FORWARDING

Call Forward Override

Specifies how to treat a call from a forwarded-to party to the forwarded-from party.

Valid Entry	Usage
y	Overrides the Call Forwarding feature by allowing a forwarded-to station to complete a call to the forwarded-from station.
n	Directs the system to forward calls to the forward-to station even when the calls are from the forwarded-to station.

Coverage After Forwarding

Determines whether or not an unanswered forwarded call is provided coverage treatment.

Valid Entry	Usage
y	Coverage treatment is provided to unanswered forwarded calls.
n	No coverage treatment is provided to unanswered forwarded calls. The call remains at the forwarded-to destination.

System parameters call coverage/ call forwarding: page 2

```

change system-parameters coverage-forwarding                               Page  2 of  2
SYSTEM PARAMETERS CALL COVERAGE / CALL FORWARDING

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)

      Coverage Of Calls Redirected Off-Net Enabled? y
Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point? y
      Ignore Network Answer Supervision? n
      Disable call classifier for CCRON over ISDN trunks? n
      Disable call classifier for CCRON over SIP trunks? n

CHAINED CALL FORWARDING
      Maximum Number Of Call Forwarding Hops: 3
      Station Coverage Path For Coverage After Forwarding: principal

```

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)

Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point

Determines whether or not a simulated bridge appearance (SBA) is maintained on the principal when a call is directed to a final off-net coverage point. This field has no consequence when the off-net call is carried end-to-end by ISDN facilities; the SBA is maintained and there is no cut-through delay.

Available only if **Coverage of Calls Redirected Off-Net Enabled** is enabled.

Valid Entry	Usage
y	Maintains a simulated bridged appearance on the principal when redirecting to a final off-net coverage point.
n	Drops the SBA on the principal's telephone when the call redirects off-net at the last coverage point, eliminating the cut-through delay inherent in CCRON calls, but sacrificing the principal's ability to answer the call.

Related links

[Coverage Of Calls Redirected Off-Net Enabled](#) on page 716

Coverage Of Calls Redirected Off-Net Enabled

Controls the Coverage of Calls Redirected Off-Net (CCRON) feature. Disables this feature if the demand on the call classifier port resources degrades other services provided by Communication Manager.

Available only if **Coverage of Calls Redirected Off-Net** is enabled for the system.

Valid Entry	Usage
y	Communication Manager monitors off-net coverage/forwarded calls and provides further coverage treatment for unanswered calls.
n	Communication Manager does not monitor off-net coverage/ forwarded calls. No further coverage treatment is provided if the calls are unanswered.

Related links

[Cvg Of Calls Redirected Off-net](#) on page 731

Disable call classifier for CCRON over ISDN trunks

Enables or disables the use of a call classifier on a CCRON call over ISDN facilities. When a CCRON call routes offnet over ISDN end-to-end facilities, no call classifier is attached to the call. If, subsequently during the call, an ISDN PROGRESS or ALERT message is received that indicates that interworking has occurred, a call classifier is normally attached to the call and has precedence over ISDN trunk signalling. This field can direct Communication Manager to dispense with the call classifier on interworked calls and rely on the ISDN trunk signalling messages.

Valid Entry	Usage
y	Disables the call classifier for CCRON calls over interworked trunk facilities.
n	Enables the call classifier for CCRON calls over interworked trunk facilities.

Disable call classifier for CCRON over SIP trunks

Valid Entry	Usage
y	Disables the call classifier for CCRON calls over interworked trunk facilities. Directs Communication Manager to dispense with the call classifier on interworked calls and rely on the SIP trunk signalling messages.
n	Enables the call classifier for CCRON calls over interworked trunk facilities.

Ignore Network Answer Supervision

Administers whether or not a call classifier for network answer supervision is used to determine when a call is answered. CCRON might use a call classifier port to determine whether an off-net coverage or forwarded call has been answered, discarding other information that might indicate an answered state.

Available only if **Coverage of Calls Redirected Off-Net Enabled** is enabled.

Valid Entry	Usage
y	Ignore network answer supervision and rely on the call classifier to determine when a call is answered. Provides accurate answer supervision for tandem calls redirected to the public network.
n	Treat network answer supervision as a true answer. Preserves network answer supervision information.

Related links

[Coverage Of Calls Redirected Off-Net Enabled](#) on page 716

CHAINED CALL FORWARDING

Maximum Number of Call Forwarding Hops

Available only if Chained Call Forwarding is enabled.

Valid Entry	Usage
3 to 10	The number of hops allowed in the forwarding chain.

Related links

[Chained Call Forwarding](#) on page 302

Station Coverage Path For Coverage After Forwarding

This field specifies what coverage path the call must follow. The system displays the **Station Coverage Path For Coverage After Forwarding** field only when you enable the **Chained Call Forwarding** and the **Coverage After Forwarding** fields.

Valid entry	Usage
principal	The call follows the coverage path of the principal.
last-fwd	The call follows the coverage path of the last-forward-to party.

Related links

[Chained Call Forwarding](#) on page 302

[Coverage After Forwarding](#) on page 714

Chapter 121: System Parameters Country Options

This screen implements parameters associated with certain international, including North American, call characteristics. This screen cannot be changed. Go to the Avaya Support website at <http://support.avaya.com> for current documentation, product notices, knowledge articles on how to modify any of the values here. This table shows the country codes that are used in Communication Manager. The Country Code is used by various fields and screens throughout the system.

Example command: `change system-parameters country-options`

Country options table

Code	Country	Ringling Signal Voltage, Frequency, and Cadence
1	United States, Canada, Korea, India	300v peak to peak, < 200v peak to ground; < 70 Hz; < 5s on > 1s off Korea: 20 Hz, 75 to 85 Volts (AC), Cadence: 1 sec on, 2 sec off
2	Australia, New Zealand	75 +/- 20 VRMS superimposed on 48 V dc at 14.5 to 55 Hz with cadence 400ms on, 200ms off, 400ms on, 2000ms off New Zealand: Ringing voltage at the customer's premises not less than 38 V rms (25Hz) on top of 50V d.c; 20 Hz; 400ms on, 200ms off, 400ms on, 2000ms off
3	Japan	75 VRMS(75-10VRMS <= x <= 75+8VRMS), 15-20 Hz and cadence of 1second on and 2 seconds off is required
4	Italy	20 to 50 Hz, 26 to 80 Volts rms superimposed on 48 V dc, 1 sec on, 4 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off

Table continues...

Code	Country	Ringling Signal Voltage, Frequency, and Cadence
5	Netherlands	25 Hz, 35 to 90 Volts rms superimposed on 66 V dc, 1 sec on, 4 sec off. Note that 50 Hz should be used, and another cadence can be 0.4 sec on, 0.2 sec off, 0.4 sec on, 4 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
6	Singapore	75V at 24Hz with a cadence of 0.4 seconds on, 0.2 seconds off, 0.4 seconds on and 2.0 seconds off.
7	Mexico	25 Hz, 70 +/- 20 Vrms superimposed on 48Vdc Cadence 1 sec on, 4 sec off, flashhook is 100 ms
8	Belgium, Luxembourg, Korea	25 Hz, 25 to 75 Volts rms superimposed on 48 V dc, 1 sec on, 3 sec off Korea: 20 Hz, 75 to 85 Volts (AC), Cadence: 1 sec on, 2 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
9	Saudi Arabia	
10	United Kingdom	U.K.: 15 to 26.25 Hz, 25 to 100 Volts rms superimposed on 48 V dc, 0.35 on, 0.22 off then start in at any point in: 0.4 sec on, 0.2 sec off, 0.4 sec on, 2 sec off. Note 1: 48v DC can be present during the whole cadence or can be confined to silent periods. Note 2: Some exchanges provide a facility known as immediate ring; in this case an initial burst of ringing 20 msec to 1 sec in length immediately precedes switching to any point in the normal ringing cycle. Ireland: 25 Hz, 30 to 90 Volts rms superimposed on 50 V dc, 0.4 sec on, 0.2 sec off, 0.4 sec on, 2 sec off another possible cadence is 0.375 sec on, 0.250 sec off, 0.375 sec on, 2 sec off. ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
11	Spain	20 to 30 Hz, 35 to 75 Volts rms superimposed on 48 V dc, 1 to 1.5 sec on, 3 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off

Table continues...

Code	Country	Ringin Signal Voltage, Frequency, and Cadence
12	France	50 Hz, 28 to 90 Volts rms superimposed on 0.45 to 54 V dc, 1.5 sec on, 3.5 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
13	Germany	Germany: 25 Hz, 32 to 75 Volts rms superimposed on 0 to 85 V dc, 1 sec on, 4 sec off Austria: 40 to 55 Hz, 25 to 60 Volts rms superimposed on 20 to 60 V dc, 1 sec on, 5 sec off +/- 20% ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
14	Czech Republic, Slovakia	
15	Russia (CIS)	25+/-2 Hz, 95+/-5 Volts eff, local call cadence: first ring 0.3-4.5 sec then 1 second on 4 seconds Off toll automatic cadence 1 sec On 2 sec Off toll operator: manual sending
16	Argentina	25Hz; 75 Vrms superimposed on 48 Vdc; 1s on 4s off
17	Greece	
18	China	25Hz +/- 3Hz; 75 +/- 15 Vrms; Harmonic Distortion <= 10%; 1 sec ON, 4 secs OFF
19	Hong Kong	75 +/- 20 VRMS superimposed on -40 to -48 V dc at 25 Hz +/- 10% with cadence 0.4 s on, 0.2 s off, 0.4 s on, 3.0 s off
20	Thailand	
21	Macedonia	
22	Poland	
23	Brazil	25Hz +/-2.5Hz; minimum of 40 Vrms; 1s on, 4s off for equipment supporting up to six trunks only otherwise 25Hz +/-2.5Hz; minimum of 70+/-15 Vrms at a continuous emitting condition under no load, overlapping a DC level.

Table continues...

Code	Country	Ringling Signal Voltage, Frequency, and Cadence
24	Nordic	Finland: 25 Hz, 35 to 75 Volts rms superimposed on 44 to 58 V dc, 1 sec on, 4 sec off 25 Hz, 40 to 120 Volts rms superimposed on 44 to 56 V dc, 0,75 on, 7,5 off +/- 20 % 25 Hz, 28 to 90 Volts rms superimposed on 24 to 60 V dc, 1 sec on, 4 sec off 25 and 50 Hz, 30 to 90 Volts rms superimposed on 33 to 60 V dc, 1 sec off, 5 sec off ETSI countries: 30 Volts rms, superimposed on a DC voltage of 50 Volts, 25 or 50 hz, cadence of 1 sec on, 5 sec off
25	South Africa	

Dial Tone Validation Timer (sec)

Available only when Tone Detection Mode is 4 or 5.

Valid Entry	Usage
0 to 6375	Displays number of milliseconds in increments of 25 that the dial tone validation routine uses to sample transmissions.

Related links

[Tone Detection Mode](#) on page 723

Disconnect on No Answer by Call Type

Enables or disables the system from disconnecting calls that are not answered. Drops outgoing trunk calls, except DCS and AAR, that users leave unanswered too long.

Directory Search Sort Order

Available only for the Cyrillic or Ukrainian display character set.

Valid Entry	Usage
Cyrillic	Cyrillic Collation is used for integrated directory name search and result sorting. This is the default value.
Roman	Eurofont Latin Collation is used for directory name search and result sorting. The letters to be searched in the specified order for dial pad button presses are defined in the row for each key.

Related links

[Display Character Set](#) on page 722

Display Character Set

Valid Entry	Usage
Cyrillic Katakana Roman Ukranian	The character set used for all non-native name values without an ASCII-only restriction.

 Note:

Cyrillic, Roman, and Ukrainian map to the Eurofont character set. For Katakana, the Optrex font is used. If a Communication Manager server uses non-English in any name field, characters on a BRI station are displayed incorrectly.

 Warning:

Changing the value in this field might cause some telephones to perform improperly, and can cause non-ASCII data in non-native names to display incorrectly on telephones. To correct this, non-native names of previously administered stations must be removed and re-administered using non-ASCII characters. This includes any display messages that have been administered.

Enable Busy Tone Disconnect for Analog Loop-start Trunks

Enables or disables Busy Tone Disconnect. When enabled, Communication Manager recognizes a busy tone from the local telephone company central office as a disconnect signal.

Howler After Busy

Enables or disables howler tone when users leave their analog telephone off-hook too long.

Set Layer 1 timer T1 to 30 seconds

Specifies whether or not the Layer 1 timer is set to 30 seconds.

tone detection parameters

Interdigit Pause

Specifies the maximum length of the inter-digit pause. Breaks lasting less than this range will be bridged or ignored.

Valid Entry	Usage
short	5–30ms
long	20–40ms

Tone Detection Mode

Use this field to specify the type of tone-detection algorithm that Communication Manager must use.

Valid entry	Usage
1	The precise Italian tone-detection algorithm.
2	The precise Australian tone-detection algorithm.
3	The precise UK tone-detection algorithm.
4	The imprecise normal broadband-filter algorithm.
5	The imprecise wide broadband-filter algorithm.
6	The precise USA tone-detection algorithm.

Chapter 122: System Parameters Customer Options

Shows which optional features are enabled for the system, as determined by the installed license file. All fields on this screen are display only. For more information about disabling or enabling one of the features, go to Avaya Support website at <http://support.avaya.com/>.

Example command: `display system-parameters customer-options`

System parameters customer options: page 1

G3 Version

Identifies the version of Avaya Communication Manager being used.

Location

Indicates the location of this Avaya server or switch.

Valid Entry	Usage
1	Canada or the United States
2	Any other location. Allows the use of International Consolidation telephones.

Platform

Displays the platform being used.

Valid Entry	Usage
28	VCM
29	VCM_ESS
30	VCM_LSP

Software Package

Indicates whether the software package license is Standard or Enterprise.

Used

The actual current usage as compared to the system maximum.

Platform Maximum Ports

The number of active ports.

Maximum Stations

Displays the maximum number of stations allowed in the system.

Maximum XMOBILE Stations

Specifies the maximum number of allowable XMOBILE stations. In general, each XMOBILE station is assigned to a wireless handset. Each XMOBILE station counts as a station and a port in terms of system configuration.

Maximum Off-PBX Telephones - EC500

Stations that are administered for any Extension to Cellular (EC500/ CSP) application count against this limit.

The “license max” value is defined as follows:

- On legacy systems, the upper limit is 1/2 of the maximum number of administrable stations. Legacy platforms do not support SIP trunks.
- On Linux systems, the upper limit is the maximum number of administrable stations.

Maximum Off-PBX Telephones - OPS

Stations that are administered for any SIP Extension to Cellular/OPS application count against this limit. The “license max” value is defined as follows:

- On legacy systems, the upper limit is 1/2 of the maximum number of administrable stations. Note that legacy platforms do not support SIP trunks.
- On Linux systems, the maximum number of administrable stations.

Maximum Off-PBX Telephones — PBFMC

Number of stations administered for Public Fixed-Mobile Convergence. Each station is allowed only one PBFMC application. The “license max” value is defined as follows:

- On legacy systems, the upper limit is 1/2 of the maximum number of administrable stations. Legacy platforms do not support SIP trunks.
- On Linux systems, the upper limit is the maximum number of administrable stations.

Maximum Off-PBX Telephones - PVFMC

Number of stations administered for Private Fixed-Mobile Convergence. Each station is allowed only one PVFMC application. The “license max” upper limit is:

- On legacy systems, 1/2 of the maximum number of administrable stations. Legacy platforms do not support SIP trunks.
- On Linux systems, the maximum number of administrable stations.

Maximum Off-PBX Telephones - SCCAN

The “license max” value is defined as follows:

- SCCAN is only available on Linux systems. The upper limit is the maximum number of administrable stations.

Maximum Off-PBX Telephones - EMX

Displays the maximum number of ring extensions available for stations to configure and the number of ring extensions the stations have already configured. You can administer one ring extension with each license. For example, if station X configured four ring extensions, Communication Manager consumes four licenses. A station extension can have a maximum of four simultaneous ring extensions.

System parameters customer options: page 2

Maximum Administered Ad-hoc Video Conferencing Ports

Defines the number of ad-hoc ports allowed for the system; one for each simultaneous active conference port. The maximum number of ad-hoc video conferencing ports allowed is the sum of the maximum allowed IP trunks and the maximum allowed SIP trunks on your system.

Maximum Administered IP Trunks

Defines the maximum number of IP trunks administered.

Maximum Administered Remote Office Trunks

Defines the maximum number of IP endpoints based on the endpoint.

Maximum Administered SIP Trunks

Defines the maximum number of SIP trunks administered.

Maximum Concurrently Registered IP eCons

Specifies the maximum number of IP SoftConsoles that can be registered at one time. The maximum number depends on the type of system.

Maximum Concurrently Registered IP Stations

Specifies the maximum number of IP stations that can be registered at one time.

Maximum G700 VAL Sources

Specifies the maximum number of VAL announcement sources.

Maximum Number of DS1 Boards with Echo Cancellation

Displays the number of DS1 board that can have echo cancellation.

Maximum Video Capable IP Softphones

The maximum number of IP Softphones that are video-capable. The maximum number depends on the type of system.

Maximum Video Capable Stations

The maximum number of stations that are video-capable. The maximum number depends on the type of system.

Used

For each item with a capacity listed, the USED value is the actual number of units currently in use.

System parameters customer options: page 3

Maximum Number of Avaya Media Server VoIP Channels

This field displays the maximum number of active VoIP channels, which is the number of licenses currently used on active media connections.

Maximum Number of Expanded Meet-me Conference Ports

The maximum number of Expanded Meet-me Conference ports on the system.

Maximum TN2501 VAL Boards

The maximum number of TN2501AP (Voice Announcement over LAN) boards allowed in this system.

Maximum TN2602 Boards with 80 VoIP Channels

The total number of TN2602AP boards that can be administered with 80 VoIP channels.

Maximum TN2602 Boards with 320 VoIP Channels

The total number of TN2602AP boards that can be administered with 320 VoIP channels.

System parameters customer options: page 4

Abbreviated Dialing Enhanced List

Provides the capability to store and retrieve dialing lists that simplify or eliminate dialing. The stored entries are organized in number lists. There are three types of number lists: personal, group, and enhanced.

Access Security Gateway (ASG)

Provides an additional level of security for remote administration.

A/D Grp/Sys List Dialing Start at 01

Used to number Abbreviated Dialing group or system lists starting with 01, rather than simply 1.

Analog Trunk Incoming Call ID

Used to collect and display the name and number of an incoming call on analog trunks.

Answer Supervision by Call Classifier

Indicates if the system can use a call classifier to do answer supervision. If the system has a call classification resource, the field is set to y. The field can be set to y even if the system has no call classifier circuit packs. This circuit pack detects tones and voice-frequency signals on the line and determines whether a call has been answered.

ARS

Provides access to public and private communications networks. Long-distance calls can be routed over the best available and most economical routes. Provides partitioning of ARS routing patterns.

ARS/AAR Dialing without FAC

Provides for Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) calls without dialing a feature access code (FAC).

ARS/AAR Partitioning

Provides the ability to partition AAR and ARS into eight user groups within a single server running Avaya Communication Manager. Can establish individual routing treatment for each group.

ASAI Link Core Capabilities

Provides linkage between Avaya Communication Manager and adjuncts. CallVisor ASAI improves the call handling efficiency of ACD agents and other system users by providing an adjunct to monitor, initiate, control, and terminate calls on the server running Communication Manager.

Note:

This field applies only to links administered as type `asai-ip`.

ASAI Link Plus Capabilities

Provides linkage between Avaya Communication Manager and adjuncts. If enabled, then the following ASAI capability groups are also enabled:

- Adjunct Routing
- Answering Machine Detection
- Selective Listening
- Switch Classified Outbound Calls
- ISDN Redirecting Number Information - the original dialed number information is provided within the ASAI messages if it arrives in ISDN SETUP messages from the public networks as either Original Dialed Number or Redirecting Party Number.

Note:

This field applies only to links administered as type `asai-ip`.

Asynch. Transfer Mode (ATM) PNC

PNC ATM PNC can be enabled only if:

- All prior fiber-link administration has been removed

- All “switch-node” and “dup-switch-node” carrier types have been removed.

Asynch. Transfer Mode (ATM) Trunking

If ATM trunking is enabled, multiple ISDN-PRI T1 or E1 trunks can be emulated on one ATM pipe. Enables circuit emulation service (CES).

Related links

[Carrier Medium](#) on page 434

ATMS

Provides for voice and data trunk facilities to be measured for satisfactory transmission performance.

ATM WAN Spare Processor

Indicates whether or not an ATM WAN spare processor is part of the system. An ATM WAN spare processor acts as a PPN in the event of network failure, and can function as an SPE if the main PPN is not functional.

Attendant Vectoring

Enables or disables Attendant Vectoring.

Audible Message Waiting

Enables or disables audible message waiting.

Authorization Codes

Enables or disables the use of Authorization Codes. Authorization Codes provide levels of calling privileges that override in-place restrictions. In addition to facilities access, authorization codes are used for unique identification for billing security purposes.

CAS Branch

Enables or disables Centralized Attendant Service - Branch.

CAS Main

Enables or disables multi-location customers served by separate switching vehicles to concentrate attendant positions at a single, main Avaya Communication Manager location. The main Avaya Communication Manager is served by an attendant queue that collects calls from all locations (main and branch). Each branch location switches all of its incoming calls to the centralized attendant positions over release link trunks (RLTs). The calls are then extended back to the requested extension at the branch server/switch over the same RLT. When the call is answered, the trunks to the main server are dropped and can be used for another call.

Change COR by FAC

Provides certain users the ability to change the class of restriction of local extensions and local attendants via a telephone by using a feature access code (FAC).

Computer Telephony Adjunct Links

Use this field to administer a Computer Telephony Adjunct (CTI) link between Communication Manager and the adjunct.

Note:

This field only applies to links administered as type `adj-ip`.

Cvg Of Calls Redirected Off-net

Provides continued monitoring for calls redirected to off-network (remote) coverage points. Uses call classification via ISDN trunk signaling.

DCS (Basic)

Provides transparent operation of selected features across a Distributed Communications System (DCS). Users on one server running Communication Manager can use features located on another server. Includes 4- and 5-digit uniform dialing and 1 to 4 digit steering. Does not support a 6/7-digit dial plan.

DCS Call Coverage

Provides DCS-based transparency of the call coverage feature across a DCS network of media servers or switches.

DCS with Rerouting

Provides for rerouting calls transferred among DCS nodes, enabling rerouting of the call for more effective use of facilities.

Related links

[TSC Supplementary Service Protocol](#) on page 626

[Used for DCS](#) on page 838

[ISDN-BRI Trunks](#) on page 734

[ISDN-PR](#) on page 734

[Group Type](#) on page 614

Digital Loss Plan Modification

Allows or disallows permission to customize the digital loss and digital tone plans.

DS1 MSP

Allows or disallows permission to administer values for the DS1 board without removing the related translations of all trunks from the trunk group.

DS1 Echo Cancellation

Removes perceivable echo from the system.

System parameters customer options: page 5

Emergency Access to Attendant

Provides for emergency calls to be placed to an attendant. These calls can be placed automatically by Avaya Communication Manager or dialed by users.

Enable 'dadmin' Login

Provides Avaya Business Partners the ability to install, administer, and maintain Avaya servers and switches. The dadmin login has access to all the same commands as other logins with the exception of **Go** and **WP**. **Go** is used for **go tcm** and **go debug** as well as **go server**. **WP** is for writing memory.

Enhanced Conferencing

Enables or disables the use of Meet-me Conference, Expanded Meet-me Conference, the No Hold Conference features, Selective Conference Party: Display, Drop, or Mute.

Enhanced EC500

Indicates if Extension to Cellular is enabled. EC500 refers to the Extension to Cellular feature.

Enterprise Survivable Server

Identifies the server is a Survivable Core Server (Enterprise Survivable Server).

ESS Administration

Indicates if administration of survivable core server (Enterprise Survivable Servers) is enabled.

Extended Cvg/Fwd Admin

Enables or disables Extended Coverage and Forwarding Administration.

External Device Alarm Admin

Provides for analog line ports to be used for external alarm interfaces. Allows identification of port location, adjunct associated with port location, and the alarm level to report.

Enterprise Wide Licensing

Enterprise Wide Licensing. Go to the Avaya Support website at <http://support.avaya.com/> for current documentation, product notices, knowledge articles on Enterprise Wide Licensing.

Flexible Billing

Provides an internationally accepted standard interface for end-to-end digital connectivity. Used with a T1 interface and supports twenty-three 64-KBPS voice or data B-Channels and one 64-Kbps signaling D Channel for total bandwidth of 1.544 Mbps.

Forced Entry of Account Codes

Allows system administration to force account users to enter account codes based on user or trunk class of restriction, or by an option on the Toll Analysis table. FEAC provides an easy method of allocating the costs of specific calls to the correct project, department, and so on.

Global Call Classification

Provides call classification outside of North America. Listens for tones and classifies tones detected. Required for Call Coverage Off Net and Outgoing Call Management.

Hospitality (Basic)

Provides access to basic features including: Attendant Crisis Alert, Attendant Room Status, Automatic Wakeup, Custom Selection of VIP DID Numbers, Do Not Disturb, Names Registration, Single-Digit Dialing, and Mixed Station Numbering.

Hospitality (G3V3 Enhancements)

Software required for Property Management System and Automatic Wakeup. Property Management System Interface activates Forward PMS Messages to INTUITY Lodging and PMS Protocol Mode (transmit in ASCII mode).

 Note:

Standard hospitality features are included in basic system software.

IP Attendant Consoles

Controls permission to administer the IP Attendant Console.

IP Stations

Controls permission to administer H.323 and/or SoftPhone stations. Must be enabled for IP telephones.

IP Trunks

Controls permission to administer H.323 trunks. Must be enabled for IP trunks.

ISDN-BRI Trunks

Provides the capability to add ISDN-BRI trunks to Communication Manager.

Related links

[Service/Feature](#) on page 599

[Facility Type](#) on page 540

[ISDN-PRI](#) on page 734

[TSC Supplementary Service Protocol](#) on page 626

[DCS with Rerouting](#) on page 731

[Used for DCS](#) on page 838

[ISDN-PRI](#) on page 734

[Group Type](#) on page 614

ISDN Feature Plus

Provides ISDN Feature Plus signaling.

ISDN-PRI

Provides Integrated Services Digital Network (ISDN-PRI) software for either a switching-hardware platform migration only or a switching-hardware platform migration in combination with a software release upgrade. Also provides signaling support for H.323 signaling. Must be enabled for IP trunks.

Related links

[Service/Feature](#) on page 599

[Facility Type](#) on page 540

[ISDN-BRI Trunks](#) on page 734

[TSC Supplementary Service Protocol](#) on page 626

[DCS with Rerouting](#) on page 731

[Used for DCS](#) on page 838

[ISDN-BRI Trunks](#) on page 734

[Group Type](#) on page 614

ISDN/SIP Network Call Redirection

Redirects an incoming ISDN/SIP call from a server running Avaya Communication Manager to another PSTN endpoint. It is used in call centers with Best Service Routing and Lookahead Interflow.

Local Survivable Processor

Indicates that the server is a Survivable Remote Server (Local Survivable Processor). The Survivable Remote Server is configured to provide standby call processing in case the primary media server is unavailable.

Malicious Call Trace

Provides the ability to retrieve certain information related to a malicious call.

Media Encryption Over IP

Enables or disables media encryption over IP network.

 Note:

For SIP Resiliency feature, ensure that the **Media Encryption Over IP** field is set to **Y**.

Mode Code for Centralized Voice Mail

Provides the ability to share a Voice Mail System (VMS) among several servers/ switches using the Mode Code - Voice Mail System Interface.

Multifrequency Signaling

Provides for multi-frequency signaling between Communication Manager and the local telephone company central office.

Multimedia Appl. Server Interface (MASI)

Allows users of the Multimedia Communications Exchange (MMCX) to take advantage of certain Avaya Communication Manager telephony features.

Multimedia Call Handling (Basic)

Allows administration of desktop video-conferencing systems as data modules associated with Avaya Communication Manager voice stations in a multimedia complex. Users can dial one number to reach either endpoint (voice or data) in the complex. Also provides support for IP SoftPhones.

Multimedia Call Handling (Enhanced)

Allows a multifunction telephone to control a multimedia call like a standard voice call.

Multimedia IP SIP Trunking

Extends applicability of the H.323 video station licensing/control to all non-IP Softphones.

System parameters customer options: page 6

Multinational Locations

Provides the ability to use a single Enterprise Communication Server (ECS) with stations, remote offices, or gateways in multiple countries. Allows administration of location parameters such as companding, loss plans, and tone generation per location, instead of system-wide.

Multiple Level Precedence and Preemption

Multiple Level Precedence and Preemption (MLPP) provides users the ability to assign levels of importance to callers, and when activated, to give higher-priority routing to individual calls based on the level assigned to the caller.

Multiple Locations

Valid Entry	Usage
y	(Depending on your server configuration, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .) Allows numbering plans, time zone and daylight saving plans, digit analysis, companding, call progress tone generation, loss plan, and analog line board parameters that are specific for each Gateway, Remote Office, network regions, signaling groups, and stations. See the Location sections in <i>Avaya Aura® Communication Manager Feature Description</i> , for the other ways, and for a list of features that use location.
n	The location is fixed. The location is 1 for endpoints and all for digit analysis. This is the default value.

Personal Station Access (PSA)

Provides basic telecommuting package capability for Personal Station Access.

Posted Messages

Supports the ability for users to post messages, selected from among a set of as many as 30 (15 fixed, 15 administrable), and shown on display telephones.

Private Networking

Indicates upgrading of PNA or ETN software RTU purchased with earlier systems.

Processor and System MSP

Allows for maintenance of the processor.

Processor Ethernet

Indicates if the Ethernet card resident in the processor cabinet is used by the Communication Manager Call Processing software. The Processor Ethernet interface is always enabled for duplex media servers.

For the Avaya supported servers, see *Avaya Aura® Communication Manager Hardware Description and Reference*.

Remote Office

Allows administration of a remote office.

Restrict Call Forward Off Net

Allows the system to monitor the disposition of an off-call and, if it detects busy, bring the call back for further processing, including call coverage.

Secondary Data Module

Provides the ability to use any data module as a secondary data module.

Station and Trunk MSP

Allows for maintenance of the station and trunk.

Station as Virtual Extension

Allows multiple virtual extensions to be mapped to a single physical analog telephone. A specific ringing pattern can be administered for each virtual extension. Useful in environments such as college dormitories, where three occupants can have three different extensions for one physical telephone.

System Management Data Transfer

Indicates Communication Manager is accessible by network administration.

Tenant Partitioning

Provides for partitioning of attendant groups and/or stations and trunk groups. Typically this is used for multiple tenants in a building or multiple departments within a company or organization.

 Note:

If SIP attendant is enabled on console parameters screen, you cannot change the **Tenant Partitioning?** field value to y.

Terminal Trans. Init. (TTI)

Allows administrators of Terminal Translation Initialization (TTI) to merge an station administered with X in the **Port** field, to a valid port by dialing a system-wide TTI security code and the extension from a terminal connected to that port.

Time of Day Routing

Provides AAR and ARS routing of calls based on the time of day and day of the week to take advantage of lower calling rates during specific times.

Related links

[Time of Day Chart](#) on page 118

Uniform Dialing Plan

Enables or disables three- to seven-digit Uniform Dial Plan (UDP) and one- to seven-digit steering. Also allows use of Extended Trunk Access and Extension Number Portability features.

Usage Allocation Enhancements

Enables the user to use user-defined entries for the incoming and outgoing ISDN calls independent of the NSF value. The Usage Allocation Enhancements feature lets the user administer the non-unique Facility Type and Facility Coding combinations. With Usage Allocation Enhancements, the customer can reserve a minimum number of trunk members for the incoming and outgoing calls at all times.

 Note:

The Usage Allocation Enhancements feature is helpful outside the US, where the default Network Facilities available in the system are not very useful.

Wideband Switching

Provides wideband data software for switching video or high-speed data. DSO channels can be aggregated up to the capacity of the span. Wideband supports H0, H11, and H12 standards, where applicable, as well as customer-defined data rates.

Wireless

Provides right to use for certain wireless applications.

System parameters customer options: page 7

ACD

Provides the software required for the Call Center Basic, Plus, Deluxe, and Elite features for the number of agents specified. Automatic Call Distribution (ACD) automatically distributes incoming calls to specified splits or skills.

BCMS (Basic)

Provides real-time and historical reports about agent, ACD split, Vector Directory Number (VDN) and trunk group activity.

BCMS/VuStats Service Level

Provides for hunt groups or Vector Directory Numbers (VDNs) with an acceptable service level. An acceptable service level defines the number of seconds within which a call must be answered to be acceptable.

Business Advocate

Enables or disables Avaya Business Advocate. Business Advocate establishes different levels of service for different types of calls. For example, a company might decide that a premium customer gets faster service than other types of customers.

Call Center Release

The call center release installed on the system.

Call Work Codes

Allows agents to enter digits for an ACD call to record customer-defined events such as account codes or social security numbers.

DTMF Feedback Signals For VRU

Provides support for the use of C and D Tones to voice response units (VRUs).

Dynamic Advocate

Enables or disables the Dynamic Advocate feature. While Business Advocate assigns reserve agents and sets overload thresholds to determine when those reserve agents get engaged, the Dynamic Advocate feature, also known as Dynamic Threshold Adjustment, takes this a step further. Dynamic Advocate automatically adjusts the thresholds as needed to help maintain defined service levels.

EAS-PHD

Increases the number of skills an agent can log in to from 4 to 20. Increases the number of agent skill preference levels from 2 to 16.

Expert Agent Selection (EAS)

Enables or disables skills-based routing of calls to the best-qualified agent.

Forced ACD Calls

See Multiple Call Handling.

Least Occupied Agent

Allows call center calls to be routed to the agent who has been the least busy, regardless of when the agent last answered a call.

Lookahead Interflow (LAI)

Provides Look-Ahead Interflow to balance the load of ACD calls across multiple locations.

Multiple Call Handling (Forced)

Forces an agent to be interrupted with an additional ACD call while active on an ACD call. Splits or skills can be one forced, one per skill, or many forced.

Multiple Call Handling (On Request)

Allows agents to request additional calls when active on a call.

PASTE (Display PBX Data on Phone)

Provides an interface between the display of a DCP telephone set and PC-based applications.

Reason Codes

Allows agents to enter a numeric code that describes their reason for entering the AUX work state or for logging out of the system.

Service Level Maximizer

Allows an administrator to define a service level whereby X% of calls are answered in Y seconds. When Service Level Maximizer (SLM) is active, the software verifies that inbound calls are matched with agents in a way that ensures that the administered service level is met.

Service Observing (Basic)

Allows a specified user to observe an in-progress call on a listen-only or listen-and-talk basis.

Service Observing (Remote/By FAC)

Allows users to service observe calls from a remote location or a local station using this feature's access codes.

Service Observing (VDNs)

Provides the option of observing and/or monitoring another user's Vector Directory Number (VDN).

Timed ACW

Places an auto-in agent in ACW for an administered length of time after completion of the currently active ACD call.

Vectoring (ANI/II-Digits Routing)

Provides for ANI and II-Digits vector routing used to make vector routing decisions based on caller identity and the originating line.

Vectoring (Basic)

Provides basic call vectoring capability.

Related links

[Basic](#) on page 86

Vectoring (Best Service Routing)

Enables or disables the Best Service Routing feature. Through special vector commands, Best Service Routing allows the system to compare splits or skills at local and remote locations and queue a call to the resource that will give the caller the best service.

Vectoring (CINFO)

Enables or disables the Caller Information Forwarding (CINFO) feature that allows the collection of caller-entered digits (ced) and customer database provided digits (cdpd) for a call from the network.

Related links

[CINFO](#) on page 86

Vectoring (G3V4 Advanced Routing)

Provides for Rolling Average Speed of Answer Routing, Expected Wait Time Routing, and VDN Calls Routing.

Vectoring (G3V4 Enhanced)

Allows the use of enhanced comparators, wildcards in digit strings for matching on collected digits and ANI or II-digits, use of Vector Routing Tables, multiple audio/music sources for use with wait-time command and priority level with the oldest-call-wait conditional.

Vectoring (Holidays)

Indicates if the Holiday Vectoring feature is enabled or disabled that simplifies vector writing for holidays.

Vectoring (Prompting)

Allows flexible handling of incoming calls based on information collected from the calling party or from an ISDN-PRI message.

System parameters customer options: page 8

Logged-In ACD Agents

The total number of ACD agents that can be logged in simultaneously.

The limit applies to ACD agents on ACD and EAS calls. Auto-Available Split (AAS) agent ports are counted when they are assigned. AAS split or skill members are also counted. If the port for an AAS split/skill member is logged out, (for example, when a ringing call is redirected) the logged-in agent count is not updated. These counts are updated only during administration.

Logged-In Advocate Agents

The total number of Business Advocate Agents logged in simultaneously. The number of logged-in Business Advocate agents counts towards the total number of logged-in ACD agents.

Logged-In IP Softphone Agents

The total number of IP Softphone agents that can be logged-in simultaneously.

VDN of Origin Announcement

Provides a short voice message to an agent indicating the city of origin of the caller or the service requested by the caller based on the VDN used to process the call.

VDN Return Destination

Allows an incoming trunk call to be placed back in vector processing after all parties, except the originator, drop.

VuStats

Puts call center statistics on agents, splits or skills, Vector Directory Numbers (VDNs), and trunk groups on telephone displays.

VuStats (G3V4 Enhanced)

Provides G3V4 VuStats enhancements including historical data and thresholds.

System parameters customer options: page 9 (QSIG OPTIONAL FEATURES)

Basic Call Setup

Provides basic QSIG services: basic connectivity and calling line ID number. Either ISDN-PRI or ISDN-BRI Trunks must be enabled for the system.

Related links

[ISDN-PRI](#) on page 734

[ISDN-BRI Trunks](#) on page 734

Basic Supplementary Services

Provides the following QSIG Supplementary Services:

- Name ID
- Transit Capabilities; that is, the ability to tandem QSIG information elements
- Support of Notification Information Elements for interworking between QSIG and non-QSIG tandemed connections
- Call Forwarding (Diversion) by forward switching. No reroute capabilities are provided
- Call Transfer by join. No path replacement capabilities are provided.
- Call Completion (also known as Automatic Callback)

Either ISDN-PRI or ISDN-BRI Trunks must be enabled for the system.

Related links

[ISDN-BRI Trunks](#) on page 734

[ISDN-PRI](#) on page 734

Centralized Attendant

Allows all attendants in one location to serve users in multi locations. All signaling is done over QSIG ISDN lines.

Interworking with DCS

Allows the following features to work between a user on a DCS-enabled media server or switch in a network and a QSIG-enabled media server or switch:

- Calling/Called/Busy/Connected Name
- Voice Mail/Message Waiting
- Leave Word Calling

Supplementary Services with Rerouting

Provides the following QSIG Supplementary Services:

- Transit Capabilities; that is, the ability to tandem QSIG information elements.
- Support of Notification Information Elements for interworking between QSIG and non-QSIG tandemed connections.
- Call Forwarding (Diversion) by forward switching. In addition, reroute capabilities are provided.
- Call Transfer by join. In addition, path replacement capabilities are provided.

Transfer Into QSIG Voice Mail

Allows transfer directly into the voice mail box on the voice mail system when a QSIG link connects Avaya Communication Manager and the voice mail system.

Value Added (VALU)

Provides additional QSIG functionality, including the ability to send and display calling party information during call alerting.

System parameters customer options: page 10 (ASAI FEATURES)

Agent States

This field provides proprietary information that is used by Avaya applications.

 Note:

This field applies only to the CTI links administered as type adj-ip.

Go to the Avaya Support site. <http://support.avaya.com/> for current documentation, product notices, and knowledge articles on proprietary information used by Avaya applications,

CTI Stations

Use the CTI Stations field to enable or disable any application using a CTI link and a CTI station to receive calls.

Phantom Calls

Indicates if phantom calls are enabled. This field only applies to links administered as type ADJ-IP or ASAI-IP.

System parameters customer options: page 11 (MAXIMUM IP REGISTRATIONS BY PRODUCT ID)

Limit

Valid Entry	Usage
1000 5000	Maximum number of IP registrations allowed, depending on server configuration.

Product ID

Identifies the product using the IP (internet protocol) registration.

These are just a few examples of valid Product IDs. The valid Product IDs for a system are controlled by the license file.

Product ID example	Description
IP_API_A	Application Enablement Services
IP_Agent	One-X agent
IP_eCons	Soft Console IP attendant
IP_Phone	IP telephones
IP_Soft	IP softphones
oneX_Comm	One-X Communicator

Rel

Valid Entry	Usage
*	IP endpoint can register with any release for the product ID.

Chapter 123: System parameters maintenance

This screen is described in *Maintenance Commands for Avaya Aura® Communication Manager, Branch Gateways and Servers*.

Chapter 124: System Parameters Media Gateway Automatic Recovery Rule

This screen defines rules for returning a fragmented network, where a number of Branch Gateways are being serviced by one or more Survivable Remote Server (Local Survivable Processors) automatically to the primary media server. The system displays a different warning message or time window grid depending on the option selected for the **Migrate H.248 MG to primary** field.

In the time window grid, an x or X can be specified for each hour. Leave blank to disable. This method helps with overlapping time issues between days of the week. There is no limit on intervals.

For more information on Auto Fallback for Branch Gateways, see *Administering Network Connectivity on Avaya Aura® Communication Manager*.

Example command: `change system-parameters mg-recovery-rule n`, where *n* is the assigned recovery rule number.

Migrate H.248 MG to primary

Use this field to indicate auto-fallback preferences.

Valid entry	Usage
immediately	The system accepts the first gateway registration that comes from the gateway, regardless of call count or time of day. This is the default.
0-active calls	The system accepts the first gateway registration reporting a 0-active call count.
time-day-window	The system accepts a valid gateway registration message received during any part of the interval. When you select this option, the system displays a grid for defining required hours or days for the time window.
time-window-OR-0-active-calls	The system accepts a valid gateway registration reporting a 0-active call count or a valid registration with any call count is received during the specified time-day intervals. When you select this option, the system displays a grid for defining required hours or days for the time window.

Minimum time of network stability

Valid Entry	Usage
3 to 15	Administers the time interval for stability in the H.248 link before auto-fallback is attempted. Default is 3.

Recovery Rule Number

The number of the recovery rule up to the server maximum.

Rule Name

The name for the recovery rule in alpha-numeric characters. The recovery rule is an aid for associating rules with gateways.

Chapter 125: Media Server Recovery Rules

Communication Manager is now integrated with Avaya Aura® Media Server (MS), a software-based media application platform where all the media is processed on software, not on dedicated hardware. Avaya Aura MS is another type of media resource provider besides Media Gateways (MG).

Communication Manager sends a status message to each Avaya Aura® MS on the Communication Manager reporting list. The status message indicates whether Communication Manager is active or inactive for the specific media server.

Avaya Aura® MS serves both as a media server and a status keeper. Each server, whether the main server or the survivable server, sends the status information to Avaya Aura® MS and requests status information about other servers. Avaya Aura® MS is a conduit that transmits the information to Communication Manager when requested. Based on the status information received, each server processes survivability actions.

Type the command **change system-parameters ms-recovery-rule**.

Auto Return

Use the **Auto Return** field to control the recovery or fallback to the main server.

Valid entry	Usage
yes	The fallback to the main server is automatically done as soon as possible, after the specified time delay.
no	The fallback to the main server is manual. You must type the command enable ms-return . If you prefer to start the recovery process in off-peak hours, you can set the field to no .
scheduled	The fallback to the main server is scheduled for the specified day and time. The recovery occurs at the next scheduled occurrence.

Day

Use the **Day** field to start the recovery of the main server on the specified day.

The **Day** field is available only when you set the **Auto Return** field to `scheduled`. You can set any day of the week to start the recovery.

Report Expiration (sec)

Use the **Report Expiration** field to set the duration for Avaya Aura® MS to retain the status message of Communication Manager.

A short report expiration facilitates a rapid failover. However, the expiration increases the risk of unintended failover due to temporary conditions, such as a server reboot.

You must type a number in seconds from 2 through 400. The default value is 180 seconds.

Report Interval (sec)

At each report interval, Communication Manager sends a status message to each Avaya Aura® MS on the Communication Manager reporting list. The status message indicates whether Communication Manager is active or inactive for the specific media server. A short report interval increases network traffic and the CPU load of Communication Manager and Avaya Aura® MS.

You must type a number in seconds from 1 through 300. The default value is 60 seconds.

Time

Use the **Time** field to start the recovery of the main server at the specified time.

The **Time** field is available only when you set the **Auto Return** field to `scheduled`.

You must type a number from 0 through 23 for hours and from 0 through 59 for minutes. Do not leave the field blank. Type values in both the hour and minute fields.

Time Delay

Use the **Time Delay** field to set the duration for the main server to connect to Avaya Aura® Media Server. The connection to the main server is established only when the network is stable enough to start the fallback.

The **Time Delay** field is available only when you set the **Auto Return** field to *yes*.

You must type a number in minutes from 0 through 120. The default value is 0. Do not leave the field blank.

Chapter 126: System parameters OCM call classification

Administers the country tone characteristics for Outbound Call Management (OCM) applications. It is not required for United States OCM applications. This screen defines the busy tone and cadence and can be administered with up to four on and off steps, which is four valid cycles to determine busy tone.

Use a minimum of two on and off steps to determine a valid busy tone. If the cadence is administered with one on and off step, any time the classifier hears the cadence it is BTM signal.

Available only if **Global Call Classification** is enabled for the system, or when **Enable Busy Tone Disconnect for Analog loop-start Trunks** is enabled for the system.

Example command: `change system-parameters ocm-call-classification`

Related links

[Global Call Classification](#) on page 733

[Enable Busy Tone Disconnect for Analog Loop-start Trunks](#) on page 722

System parameters OCM call classification: page 1

Cadence Classification After Answer

Valid Entry	Usage
y	Communication Manager puts call classifiers into modes that listen for call progress tones while listening the voice, with or without Answering Machine Detection (AMD).
n	No-cadence call classification modes detect live voice with or without AMD whenever Communication Manager receives a connect or answer supervision message from an outbound trunk. These modes do not detect any call progress tone cadence.

Global Classifier Adjustment (dB)

Valid Entry	Usage
0 to 15	Specifies the dB loss adjustment. where 0 is the least adjustment, and 15 is the most adjustment.

USA Default Algorithm

Enables or disables the use of the default United States tone detection.

USA SIT Algorithm

Valid Entry	Usage
y	Uses the United States (SIT) tone characteristics for SIT tone detection.
n	The system treats tones with the administered tone name intercept as if they were SIT VACANT, and treats tones with the administered tone name information as if they were SIT UNKNOWN.

System parameters OCM call classification: page 2

Cadence Step

Identifies the number of each tone cadence step and indicates whether the tone is on or off during this cadence step.

Duration Maximum

Valid Entry	Usage
75 to 6375	The upper limit in milliseconds of the tone duration in increments of 25.

Duration Minimum

Valid Entry	Usage
75 to 6375	The lower limit in milliseconds (msec) of the tone duration in increments of 25.

Instance

Valid Entry	Usage
1 to 8	The number distinguishes tones that have the same name but more than one definition of silence and tone-on characteristics.

Tone Continuous

Valid Entry	Usage
y	Indicates a continuous tone.
n	Indicates a non-continuous tone.

Tone Name

Valid Entry	Usage
busy information intercept reorder ringback	The name of the tone. Busy is required when Busy Tone Disconnect for Analog Loop-start Trunks is enabled or Global Call Classification is disabled for the system. Required for tone definition outside of the U.S. and Canada.

Related links

[Global Call Classification](#) on page 733

[Enable Busy Tone Disconnect for Analog Loop-start Trunks](#) on page 722

Chapter 127: System Parameters Special Applications

Use this screen to enable the Special Application features. A Special Application feature is a custom feature requested by the customer for a specific need. Special Application features are included in the Communication Manager software base, however additional administration is required. For more information about the Special Application features, see *Avaya Aura® Communication Manager Special Application Features*.

 Note:

- Communication Manager provides the UCID of the related held call ID to the new call ID when any one of the following conditions are satisfied:
 - Communication Manager receives the third-party Auto Dial request and the special-application SA8348 is selected as 'y'.
 - Communication Manager receives the third-party Auto Dial request and the special-application SA8702 and the field "Copy UCID for Station Conference/Transfer?" are both selected as 'y'.
- Communication Manager provides the ASAI-UUI of the related held call ID to the new call ID when:
 - Communication Manager receives the third-party Auto Dial request and the field "Copy ASAI UUI During Conference/Transfer?" is selected as 'y'.

Chapter 128: Telecommuting Access

Administers the extension which allows remote users to use the Telecommuting Access feature.

Example command: `change telecommuting-access`

Telecommuting Access Extension

An extension that allows only remote access to the Telecommuting Access feature. This extension must consist of one to 16 digits that conforms to the system dial plan and is not assigned to any other system object.

Chapter 129: Tenant

This screen defines tenants to the system. If your server running Communication Manager uses tenant partitioning, see *Tenant Partitioning in Avaya Aura® Communication Manager Feature Description and Implementation*.

Example command: `change tenant n`, where *n* is the tenant partitioning number.

Tenant: page 1

change tenant 1

Page 1 of 2

TENANT 1

Tenant Description: _____

Attendant Group: 1 SIP? y

Routing Digits: _____ Attendant Server: _____

Ext Alert Port (TAAS): _____

Night Destination: _____

Music Source: 1

Attendant Vectoring VDN: _____

DISTINCTIVE AUDIBLE ALERTING

Internal: 1 External: 2 Priority: 3

Attendant Originated Calls: external

COS Group: 1

Attendant Group

Valid Entry	Usage
digits	The attendant group assigned to the tenant. For the maximum allowed entry, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .

Attendant Vectoring VDN

Assigns the VDN extension for Attendant Vectoring to a console. Available only if Attendant Vectoring is enabled for the system and Tenant Partitioning is disabled.

Related links

[Attendant Vectoring](#) on page 730

[Tenant Partitioning](#) on page 738

COS Group

Valid Entry	Usage
digits	The Class of Service (COS) group assigned to the tenant. Available only if Tenant Partitioning is enabled for the system. For the maximum allowed entry, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .

Related links

[Tenant Partitioning](#) on page 738

Ext Alert Port (TAAS)

The Trunk Answer Any Station (TAAS) alert port assigned to this tenant, if any. The port type and the object type must be consistent. The port can be assigned to only one tenant.

Valid Entry	Usage
1 to 999	Gateway
V1 to V9	Module
01 to 31	Circuit

Ext Alert (TAAS) Extension

The extension of the external alert TAAS extension. A system installer can then use the Terminal Translation Initialization (TTI) feature from a telephone plugged into any port to assign this extension number to that port. Doing so makes that port the external alert TAAS port.

Available only if **Ext Alert Port (TAAS)** is administered without hardware.

Related links

[Ext Alert Port \(TAAS\)](#) on page 759

Music Source

Valid Entry	Usage
digits	The music or tone source for this tenant partition. For the maximum allowed entry, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .

Night Destination

The night service station extension, if night service is wanted for this tenant.

SIP

Use this field to assign a SIP attendant to a tenant without using SIP stations.

* Note:

If native attendant like type “302” or “console” is already configured in the same Attendant Group, then the Communication Manager will not allow you to enable SIP on Tenant form. Also, if SIP is enabled on the Tenant form, Communication Manager will not allow you to add a new attendant user of type “302” or “console” in the same Attendant Group. Before SIP is enabled, Attendant Vectoring VDN field must be blank.

Valid Entry	Usage
n	Disables SIP attendant. This is the default value.
y	Enables SIP attendant. When you change the value to y, then the following values appear: • Routing Digits • Attendant Server

Routing Digits

Access code to route calls to SIP attendant server. You can enter up to four digits only.

The Routing Digits field appears on the Tenant form, only if the Tenant Partition is enabled.

Valid Entry	Usage
digits from 0 to 9	Digits comprising the AAR or ARS access code. The default value is blank.
blank	Call routing to attendant server is not specified.

Attendant Server

Attendant server number. This field cannot be blank.

The Attendant Server field appears on the Tenant form, only if the Tenant Partition is enabled.

Valid Entry	Usage
digits from 0 to 9	You can enter up to 17 digits.

Tenant

Displays the tenant number.

Tenant Description

A description of the tenant that consists of up to 40 alpha-numeric characters.

DISTINCTIVE AUDIBLE ALERTING

Available only if **Tenant Partitioning** is enabled for the system.

Attendant Originated Calls

Indicates which type of ringing applies to attendant-originated calls.

Valid Entry	Usage
internal	Internal ringing applies to attendant-originated calls.
external	External ringing applies to attendant-originated calls. Default is external.
priority	Priority ringing applies to attendant-originated calls.

Distinctive Audible Alerting (Internal, External, Priority)

The number of rings for Internal, External, and Priority calls. For virtual stations, this applies to the mapped-to physical telephone. This is also known as Distinctive Ringing. Defaults are as follows:

1. Internal calls
2. External and attendant calls
3. Priority calls

 Note:

SIP messaging includes the ring types internal, external, intercom, auto-callback, hold recall, transfer recall, or priority. In Communication Manager, types intercom, auto-callback, hold recall, and transfer recall are treated as priority.

Valid Entry	Usage
1	1 burst, meaning one burst of ringing signal per period
2	2 bursts, meaning two bursts of ringing signal per period
3	3 bursts, meaning two bursts of ringing signal per period

Related links

[Distinctive Audible Alerting \(Internal, External, Priority\)](#) on page 264

Tenant: page 2

This screen is used to enable additional tenant-to-tenant calling permissions.

change tenant 1										Page 2 of 2
TENANT 1										
CALLING PERMISSION (Enter "y" to grant permission to call specified Tenant)										
1? <u>y</u>	11? <u>y</u>	21? <u>y</u>	31? <u>y</u>	41? <u>y</u>	51? <u>y</u>	61? <u>y</u>	71? <u>y</u>	81? <u>y</u>	91? <u>y</u>	
2? <u>y</u>	12? <u>y</u>	22? <u>y</u>	32? <u>y</u>	42? <u>y</u>	52? <u>y</u>	62? <u>y</u>	72? <u>y</u>	82? <u>y</u>	92? <u>y</u>	
3? <u>y</u>	13? <u>y</u>	23? <u>y</u>	33? <u>y</u>	43? <u>y</u>	53? <u>y</u>	63? <u>y</u>	73? <u>y</u>	83? <u>y</u>	93? <u>y</u>	
4? <u>y</u>	14? <u>y</u>	24? <u>y</u>	34? <u>y</u>	44? <u>y</u>	54? <u>y</u>	64? <u>y</u>	74? <u>y</u>	84? <u>y</u>	94? <u>y</u>	
5? <u>y</u>	15? <u>y</u>	25? <u>y</u>	35? <u>y</u>	45? <u>y</u>	55? <u>y</u>	65? <u>y</u>	75? <u>y</u>	85? <u>y</u>	95? <u>y</u>	
6? <u>y</u>	16? <u>y</u>	26? <u>y</u>	36? <u>y</u>	46? <u>y</u>	56? <u>y</u>	66? <u>y</u>	76? <u>y</u>	86? <u>y</u>	96? <u>y</u>	
7? <u>y</u>	17? <u>y</u>	27? <u>y</u>	37? <u>y</u>	47? <u>y</u>	57? <u>y</u>	67? <u>y</u>	77? <u>y</u>	87? <u>y</u>	97? <u>y</u>	
8? <u>y</u>	18? <u>y</u>	28? <u>y</u>	38? <u>y</u>	48? <u>y</u>	58? <u>y</u>	68? <u>y</u>	78? <u>y</u>	88? <u>y</u>	98? <u>y</u>	
9? <u>y</u>	19? <u>y</u>	29? <u>y</u>	39? <u>y</u>	49? <u>y</u>	59? <u>y</u>	69? <u>y</u>	79? <u>y</u>	89? <u>y</u>	99? <u>y</u>	
10? <u>y</u>	20? <u>y</u>	30? <u>y</u>	40? <u>y</u>	50? <u>y</u>	60? <u>y</u>	70? <u>y</u>	80? <u>y</u>	90? <u>y</u>	100? <u>y</u>	

Calling permissions

Establishes or blocks calling permission between the tenant being administered and any other tenant. The system default allows each tenant to call only itself and Tenant 1.

Tenant

Displays the tenant number.

Chapter 130: Terminal Parameters

Administers system-level parameters and audio levels for the 603 CALLMASTER telephones and the 4600-series, 6400-series, 8403, 8405B, 8405B+, 8405D, 8405D+, 8410B, 8410D, 8411B, 8411D, 8434D, and 2420/2410 telephones. Only authorized Avaya personnel can administer this screen.

 Note:

With the Multinational Locations feature enabled, terminal parameters can be administered per location, rather than system-wide.

Example command: `change terminal-parameters`

Base Parameter Set

Determines which default set of telephone options and levels will be used. This field corresponds to the country codes.

Related links

[Country options table](#) on page 718

Customize Parameters

Allows or disallows permission for the administrator to change one or more of the default parameters.

ADJUNCT LEVELS

Touch Tone Sidetone (dB)

Determines the touchtone volume fed back from the telephone when a user presses a button.

Voice Receive (dB)

Determines the volume of voice inbound to the adjunct.

Voice Sidetone (dB)

Determines the volume of voice fed back from the handset voice microphone to the user's ear.

Voice Transmit (dB)

Determines the volume of voice outbound from the adjunct.

OPTIONS

Display Mode

Determines how the # and ~ characters appear on the telephone's display.

Valid Entry	Usage
1	The # and ~ do not change.
2	The telephone displays a # as a British pound sterling symbol and a ~ as a straight overbar.

DLI Voltage Level

Determines whether DCP Line Voltage used by the telephones is forced high, forced low, or allowed to automatically adjust.

Handset Expander Enabled

Determines whether or not the telephone reduces background noise on the handset.

Volume for DCP Types

Determines what volume adjustments are retained. Allows the DCP telephone volume to be adjusted while the call is in-progress.

Valid Entry	Usage
default speaker, handset unchangeable	The speaker resets to the default settings while the adjusted handset setting is retained.

Table continues...

Valid Entry	Usage
default settings used to begin each call	No adjusted handset and speaker settings are retained.
retain handset and speaker between calls	The adjusted handset and speaker settings are retained
retain speaker, handset unchangeable	Only the adjusted speaker setting is retained.

Volume for IP Types

Determines what volume adjustments are retained. Allows the IP telephone volume to be adjusted while the call is in progress.

 Note:

Avaya recommends that values for PRIMARY LEVELS or BUILT-IN SPEAKER LEVELS are not changed.

Valid Entry	Usage
default speaker, handset unchangeable	The speaker resets to the default settings while the adjusted handset setting is retained.
default settings used to begin each call	No adjusted handset and speaker settings are retained
retain handset and speaker between calls	The adjusted handset and speaker settings are retained
retain speaker, handset unchangeable	Only the adjusted speaker setting is retained

Touch Tone Transmit (dB)

Determines the touchtone volume fed outbound from the telephone.

Voice Receive (dB)

Determines the volume of voice inbound to the adjunct.

Voice Transmit (dB)

Determines the volume of voice outbound from the adjunct.

PRIMARY LEVELS

The system displays the default setting from the Base Parameter Set for all fields. Also, these fields all require the same input; valid entries are from -44.0 db through +14.0 db in 0.5 increments. For example, -44.0, -43.5, -43.0 and so on.

Touch Tone Sidetone (dB)

Determines the touchtone volume fed back from the telephone when a user presses a button.

Touch Tone Transmit (dB)

Determines the touchtone volume fed outbound from the telephone.

Voice Receive (dB)

Determines the volume of voice inbound to the adjunct.

Voice Sidetone (dB)

Determines the volume of voice fed back from the handset voice microphone to the user's ear.

Voice Transmit (dB)

Determines the volume of voice outbound from the adjunct.

Chapter 131: Terminating Extension Group

Defines a Terminating Extension Group (TEG). Any telephone can be assigned as a TEG member; however, only a multi-appearance telephone can be assigned a TEG button with associated status lamp. The TEG button allows the telephone user to select a TEG call appearance for answering or for bridging onto an existing call.

The TEG members are assigned on an extension number basis. Call reception restrictions applicable to the group are specified by the group class of restriction (COR). The group COR takes precedence over an individual member's COR. The members could all be termination restricted but still receive calls if the group is not restricted.

An extension number can be assigned to more than one TEG but can have only one appearance of each group.

Example command: `change term-ext-group n`, where *n* is the assigned group number.

```
change term-ext-group 1                                     Page 1 of 1
TERMINATING EXTENSION GROUP

  Group Number: 1                                           Group Extension: 360-0005
    Group Name: _____ Coverage Path: _____
  Security Code: _____ COR: 1
                                     TN: 1
ISDN/SIP Caller Disp: _____ LWC Reception: none
  AUDIX Name: _____

GROUP MEMBER ASSIGNMENTS

  Ext      Name
1: _____
2: _____
3: _____
4: _____
```

AUDIX Name

The name of the AUDIX machine. Must be the same name as the IP Node name and administered *after* the IP Node is configured.

Related links

[IP Node Names](#) on page 413

[IP Node Names](#) on page 413

COR

Valid Entry	Usage
0 to 995	The class of restriction (COR) number that reflects the required restrictions.

Coverage Path

The call coverage path number for this group. A TEG cannot serve as a coverage point; however, calls to a TEG can redirect to coverage.

Valid Entry	Usage
digits	Path number. For the maximum allowed entry, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .
t1 to t999	Time of day table
blank	Not administered

Group Extension

The extension of the terminating extension group. The extension can consist of one to seven digits and cannot be a Vector Directory Number (VDN). This field cannot be left blank.

Group Name

The name used to identify the terminating extension group.

Group Number

Displays the terminating extension group number.

ISDN Caller Disp

Specifies whether the TEG group name or member name (member of TEG where call terminated) is sent to the originating user. Required if **ISDN-PRI** or **ISDN-BRI Trunks** are enabled for the system.

Valid Entry	Usage
grp-name	TEG group name is sent.
mbr-name	Member name is sent.
blank	Not administered.

Related links

[ISDN-BRI Trunks](#) on page 734

[ISDN-PRI](#) on page 734

LWC Reception

Use this field to specify the location where the system must store the LWC messages.

Valid entry	Usage
spe	Use this option to store the LWC messages on Switch Processor Element (SPE).
none	Use this option if you do not want to store the LWC messages.
audix	Use this option to store the LWC messages on the voice messaging system.

Related links

[AUDIX Name](#) on page 358

Security Code

Use this field to enter the four-digit station security code for the **Demand Print messages** feature.

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

GROUP MEMBER ASSIGNMENTS

Ext

The extension number of one to seven digits assigned to a group number. The extension cannot be a Vector Directory Number (VDN).

Name

Displays the name assigned to the group number.

Chapter 132: TFTP Server

The Trivial File Transfer Protocol screen allows specification of the TFTP server that Avaya Communication Manager uses to get download files.

Example command: `change tftp-server`

Filename in Memory

Displays the name of the file currently in Communication Manager memory.

File Size

Displays the number of bytes transferred.

File Status

Displays the download status as Download In Progress, Download Failed, File Not Found, or Download Completed.

File to Retrieve

The name of the file to retrieve using up to 32 alphanumeric, case sensitive, characters for identification.

Local Node Name

A previously-administered local node name. The node must be assigned to a procr.

Valid Entry	Usage
1 to 15	The node name.
procr	Avaya Solutions Platform Server

Related links

[Name](#) on page 413

Station Type

Displays the information related to the firmware, font, language, boot, and Digital Signal Processor (DSP) when a firmware, language file, font, DSP, or boot loader file download successfully.

TFTP Server Node Name

A previously-administered TFTP server node name from 1 to 15 characters.

Related links

[Name](#) on page 413

TFTP Server Port

Valid Entry	Usage
1 to 64500	The number for the remote TCP port.

Chapter 133: Time of Day Coverage Table

This screen allows administration of up to five different coverage paths, associated with five different time ranges, for each day of the week. Only one coverage path can be in effect at any one time.

Example command: `change coverage time-of-day n`, where *n* is the assigned time of day coverage table.

change coverage time-of-day 1

Page 1 of 1

TIME OF DAY COVERAGE TABLE: 1

	Act Time	Cvg Path	Act Time	Cvg Path	Act Time	Cvg Path	Act Time	Cvg Path	Act Time	Cvg Path
Sun	0:00	___	:_	___	:_	___	:_	___	:_	___
Mon	0:00	___	:_	___	:_	___	:_	___	:_	___
Tue	0:00	___	:_	___	:_	___	:_	___	:_	___
Wed	0:00	___	:_	___	:_	___	:_	___	:_	___
Thu	0:00	___	:_	___	:_	___	:_	___	:_	___
Fri	0:00	___	:_	___	:_	___	:_	___	:_	___
Sat	0:00	___	:_	___	:_	___	:_	___	:_	___

Act Time

Valid Entry	Usage
00:01– 23:59	Specifies the activation time of the associated coverage path. Information must be entered in 24-hour time format. If there are time gaps in the table, there will be no coverage path in effect during those periods. The first activation time for a day is set to 00:00 and cannot be changed. Activation times for a day must be in ascending order from left to right.

CVG Path

Valid Entry	Usage
1 to 9999 blank	The coverage path number.

Time of Day Coverage Table

Displays the Time of Day Coverage Table number.

Chapter 134: Time of Day Routing Plan

This screen is used to set up Time of Day Routing Plans.

- AAR and ARS calls can be routed based on the time of day each call is made.
- Up to eight Time of Day Routing Plans can be designed, each scheduled to change up to six times a day for each day in the week.
- The Time of Day Routing Plan PGN# is matched with the **PGN#** field on the Partition Routing Table for the correct route pattern.

Available only if **Automatic Route Selection (ARS)** or **Private Networking, AAR/ARS Partitioning**, and **Time of Day Routing** are enabled for the system.

Example command: `change time-of-day n`, where *n* is the time of day routing plans.

Related links

[PGN 1 \(through PGN 8\)](#) on page 550

[ARS](#) on page 729

[Private Networking](#) on page 737

[Time of Day Routing](#) on page 738

Act Time

Valid Entry	Usage
00:00 to 23:59	Specifies the time of day the route pattern, identified by PGN#, begins. Time is represented using a 24 hour clock. List times for the same day in increasing order. There must be at least one entry for each day.

PGN

The route pattern for the activation time listed. The PGN must match a previously-administered PGN and route pattern on the Partition Routing Table. There must be at least one entry for each day.

Valid Entry	Usage
1 to 8	The route pattern range when SA9050 is inactive.
1 to 32	The route pattern range when SA9050 is active. The range depends on the type of Communication Manager template. The default value is 8. (For more information on how SA9050 changes capacities, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .)

Time of Day Routing Plan

Displays the Time of Day Routing Plan number.

Chapter 135: Time of Day Station Lock Table

This screen administers the ability to lock stations automatically by a time of day schedule.

Example command: `change tod-station-lock n`, where *n* is the assigned time of day station lock table number.

Begin Time

Valid Entry	Usage
00:00 to 23:59	When the time of day station lock interval begins for each day of the week. Time is represented using a 24-hour clock.

End Time

Valid Entry	Usage
00:00 to 23:59	When the time of day station lock interval ends for each day of the week. Time is represented using a 24-hour clock.

INTERVAL 1, 2, 3

Three separate time of day station lock intervals can be administered. The system imposes validation of overlapping intervals or invalid blank entries.

Manual Unlock allowed

Indicates whether or not the user can manually unlock the TOD-locked station using either a **sta-lock** button or a Feature Access Code followed by an SSC.

Table Active

Activates or deactivates time of day lock for all stations associated with this table.

Chapter 136: Toll Analysis

This screen associates dialed strings to the system's Restricted Call List (RCL), Unrestricted Call List (UCL), and Toll List. Users can be forced to dial an account code if you associate dialed strings with CDR Forced Entry of Account Codes.

To maximize system security, toll calling areas must be restricted as much as possible through the use of the **RCL** (Restricted Call List) and **Toll List** fields on this screen.

 Note:

The Toll List field on this screen does not interact with or relate to the ARS Toll Table.

Example command: `change toll n`, where *n* is the dialed digits.

CDR FEAC

Valid Entry	Usage
x	Requires an account code from a call whose facility COR requires a Forced Entry of Account Code.

Dialed String

Communication Manager matches the dialed numbers with the entry in the **Dialed String** field that most closely matches the dialed number. You can enter up to 18 digits that the call-processing server analyzes. You can also enter the wildcard characters, `x` and `X`.

Location

Valid Entry	Usage
1 to 250 or 1 to 2000 Depending on your server configuration, see <i>Avaya Aura® Communication Manager System Capacities Table</i> .	The location of the endpoint that is dialing the digits. Available only if ARS and Multiple Locations are enabled for the system. See the Location sections in <i>Avaya Aura® Communication Manager Feature Description and Implementation</i> , for the other ways to assign a location, and for a list of features that use location.
all	Indicates that this Toll Analysis Table is the default for all locations.

Related links

[ARS](#) on page 729

[Multiple Locations](#) on page 736

Max

Use this field to specify the maximum number of user-dialed digits that the system must collect to match the dialed string.

Min

Use this field to specify the minimum number of user-dialed digits that the system must collect to match the dialed string.

Percent Full

This field displays the percentage of system memory resources used for the AAR and the ARS digit analysis. For more information about the system memory resources, see *Avaya Aura® Communication Manager System Capacities Table*.

RCL

Valid Entry	Usage
x	Assigns the Dialed String to the Restricted Call List (RCL). All entries marked with x and their associated dialed strings are referred to as the System's Restricted Call List. The RCL can be assigned to any COR.

Toll List

Valid Entry	Usage
x	Assigns the Dialed String to the Toll List.

Dialed String	Minimum	Maximum	Toll List
0	1	23	X
1	4	23	X
20	10	10	X
21	10	10	X
30	10	10	X
31	10	10	X
40	10	10	X
41	10	10	X
50	10	10	X
51	10	10	X
60	10	10	X
70	10	10	X
71	10	10	X
80	10	10	X
81	10	10	X
90	10	10	X
91	10	10	X

Unrestricted Call List

Valid Entry	Usage
x	Assigns the dialed string to one of the system's Unrestricted Call Lists (UCL).

Chapter 137: Tone Generation

This screen administers the tone characteristics that parties on a call hear under various circumstances.

 Note:

With the Multinational Locations feature enabled, tone generation can be administered per location, rather than system-wide.

Tone Generation: page 1

440Hz PBX-dial Tone

Valid Entry	Usage
y	The switch primary dial tone is changed to a continuous 440Hz-17 tone.
n	A customized tone defaults to the Base Tone Generation Set .

Related links

[Base Tone Generator Set](#) on page 783

440Hz Secondary-dial Tone

Valid Entry	Usage
y	The secondary local telephone company central office dial tone is changed to a continuous 440Hz-17 tone.
n	A customized tone defaults to the Base Tone Generation Set .

Related links

[Base Tone Generator Set](#) on page 783

Base Tone Generator Set

Valid Entry	Usage
1 to 50	The country code that identifies the base tone generation set to be used.

Related links

[Country options table](#) on page 718

Tone generation: page 2 (TONE GENERATION CUSTOMIZED TONES page)

Cadence Step

Displays the number of each tone cadence step.

Duration (msec)

Valid Entry	Usage
50 to 12750	The duration of this step in the tone sequence. Values are in increments of 50.

Step

The number of the cadence step for this **goto** command.

Tone (Frequency/Level)

Valid Entry	Usage
silence	No tone. A final step of silence with an infinite duration will be added internally to any tone sequence that does not end in a goto.
goto	Repeats all or part of the sequence, beginning at the specified cadence step.

Table continues...

Valid Entry	Usage
350/-17.25	Specifies the frequency and level of the tone.
350+425/-4.0	
350+440/-13.75	
375+425/-15.0	
404/-11.0	
404/-16.0	
404+425/-11.0	
404+450/-11.0	
425/-4.0	
425/-11.0	
425/-17.25	
440/-17.25	
440+480/-19.0	
450/-10	
480/-17.25	
480+620/-24.0	
525/-11.0	
620/-17.25	
697/-8.5	
770/-8.5	
852/-8.5	
941/-8.5	
1000/0.0	
1000/+3.0	
1004/0.0	
1004/-16.0	
1209/-7.5	
1336/-7.5	
1400/-11.0	
1477/-7.5	
1633/-7.5	
2025/-12.1	

Valid Entry	Usage
2100/-12.1	
2225/-12.1	
2804/-16.0	

Tone Name

Valid Entry	Usage
1-call-wait	Indicates which of the individually administrable tones is modified.  Warning: Customization of “busy-verify” will also change the “warning” tone.
2-call-wait	
3-call-wait	
busy	
busy-verify	
call-wait-ringback	
conference	
confirmation	
disable-dial	
hold	
hold-recall	
immed-ringback	
intercept	
intrusion	
mntr/rec-warning	
PBX-dial	
recall-dial	
recall-dont-ans	
redirect	
reorder	
rep-confirmation	
reset-shift	
ringback	
secondary-dial	
special-dial	
whisper-page	
zip	

Chapter 138: Trunk Group

Trunk Group

Sets basic characteristics for every type of trunk group and assigns ports to the group. Many fields are dependent on the settings of other fields and only appear when certain values are entered in other fields on the screen. For example, the entry in **Group Type** might significantly change the content and appearance of the Trunk Group screen.

*** Note:**

For descriptions of the screens and fields that are unique to ISDN trunks, see *ISDN Trunk Group*.

Trunk Group: page 1

The screenshot displays the 'change trunk-group 1' screen, which is titled 'TRUNK GROUP' and is page 1 of 22. The screen contains the following fields and values:

Field	Value
Group Number	1
Group Name	Trunk Group 1
Direction	two-way
Dial Access?	n
Queue Length	0
Comm Type	voice
Prefix-1?	y
Group Type	co
COR	1
Outgoing Display?	n
Busy Threshold	255
Country	1
Incoming Destination	
Auth Code?	n
Trunk Flash?	n
CDR Reports	y
TN	1
TAC	
Night Service	
Digit Absorption List	
Toll Restricted?	y
Trunk Type	

Analog Gain

Reduces the strength of incoming signals on TN2199 ports if users regularly experience echo, distortion, or unpleasantly loud volume.

Available only if the country code is 15 and the **Trunk Type (in/out)** value is 2-wire-ac, 2-wire-dc, or 3-wire.

Valid Entry	Usage
a	Reduces the incoming signal by -3dB.
b	Reduces the incoming signal by -6dB.
c	Reduces the incoming signal by -8dB.
none	No reduction. This setting should not be changed unless the trunk group sound quality is unacceptable.

Related links

[Country](#) on page 791

[Trunk Type \(in/out\)](#) on page 804

Auth Code

The **Auth Code** field is available only for incoming or two-way trunk groups if you enable the **Authorization Codes** feature. If you enable the **Auth Code** field, Communication Manager performs an auth code check for the incoming trunk call that is routed over another trunk.

The following table provides the situations when the caller must enter an auth code.

Calling party	Called party	Facility Restriction Level (FRL) check?	Auth Code, required if
Station	Station	No	No
	Trunk	Yes, if Route Pattern accesses the trunk	the FRL of the calling station is less than the FRL of the outgoing route pattern preference
Incoming trunk	Station	No	the Auth Code field is enabled on the incoming trunk group
	Trunk	Yes, if Route Pattern accesses the trunk	the FRL of the incoming trunk is less than the FRL of the outgoing route pattern preference or the Auth Code field is enabled on the incoming trunk group

* Note:

The **Auth Code** field is unavailable if:

- the **Group Type** field is tandem
- the **Group Type** field is ISDN and the **Service Type** field is tandem.

In these situations, permissions of the caller are transmitted using Traveling Class Mark.

Related links

[Direction](#) on page 436

[Authorization Codes](#) on page 730

BCC

Generalized Route Selection uses the Bearer Capability Class (BCC) to select the appropriate facilities for routing voice and data calls. Far-end tandem servers/switches use the BCC to select outgoing routing facilities with equivalent BCC classes. The **BCC** entry is used to select the appropriate facilities for incoming ISDN calls. Communication Manager compares the **BCC** entry with the value of the Bearer Capability information element for the incoming call and routes the call over appropriate facilities.

Available only if all of the following conditions are true:

- **ISDN-BRI Trunks** or **ISDN-PRI** are enabled for the system.
- The trunk group type is access, co, fx, tandem, tie, or wats.
- The **Comm Type** is data, avd, or rbavd.

Valid Entry	Usage
0	Voice and voice-grade data
1	56 kbps synchronous data transmitted with robbed-bit signaling
2	Less than 19.2 kbps synchronous or asynchronous data
4	64 kbps data on unrestricted channels

Related links

[Comm Type](#) on page 790

[Group Type](#) on page 437

[ISDN-BRI Trunks](#) on page 734

[ISDN-PRI](#) on page 734

Busy Threshold

Valid Entry	Usage
1 to 255	The number of trunks that must be busy to alert attendants to control access to outgoing and two-way trunk groups during periods of high use. When the threshold is reached and the warning lamp for that trunk group lights, the attendant can activate trunk group control: internal callers who dial out using a trunk access code are connected to the attendant. Calls handled by AAR and ARS route patterns go out normally.

CDR Reports

Valid Entry	Usage
y	All outgoing calls on this trunk group generate call detail records. If Record Outgoing Calls Only is disabled for CDR, incoming calls on this trunk group also generate call detail records.
n	Calls over this trunk group do not generate call detail records.

Table continues...

Valid Entry	Usage
r (ring-intvl)	CDR records are generated for both incoming and outgoing calls. In addition, the following ringing interval CDR records are generated: • Abandoned calls: The system creates a record with a condition code of “H” indicating the time until the call was abandoned. • Answered calls: The system creates a record with a condition code of “G” indicating the interval from start of ring to answer. • Calls to busy stations: The system creates a record with a condition code of “I” indicating a recorded interval of 0.

Related links

[Record Outgoing Calls Only](#) on page 103

CESID I Digits Sent

For emergency 911 service, Communication Manager might send Caller’s Emergency Service Identification (CESID) information to the local telephone company central office or E911 tandem server or switch. This digit string is part of the E911 signaling protocol. Determine the correct entry for this field by talking to the E911 provider. Accepts from one to three digits.

Available only for a cama trunk group.

Related links

[Group Type](#) on page 437

Comm Type

Indicates whether the trunk group carries voice, data, or both.

 Note:

Comm Types of avd, rbavd, and data require trunk member ports on a DS1 port.

Valid Entry	Usage
avd	For applications that mix voice and Digital Communication Protocol data, such as video conferencing applications. The receiving end server discriminates voice calls from data calls and directs each to an appropriate endpoint. Neither originating nor terminating ends insert a modem pool for any calls. The Signaling Mode for the DS1 port must be set to either common-chan or CAS signaling.
data	All calls across the trunk group originate and terminate at Communication Manager digital data endpoints. Public networks don’t support data: supported by Avaya’s DCP protocol, this entry is used almost exclusively for the data trunk group supporting DCS signaling channels. The Signaling Mode for the DS1 port must be set to either robbed-bit or common-chan.

Table continues...

Valid Entry	Usage
rbavd	For digital trunk groups that carry voice and data with robbed-bit signaling. The Signaling Mode for the DS1 port must be set to robbed-bit unless mixed mode signaling is allowed on the DS1 port. In that case, the Signaling Mode must be isdn-ext or isdn-pri.
voice	For trunk groups that carry only voice traffic and voice-grade data (that is, data transmitted by modem). Analog trunk groups must use voice.

COR

Valid Entry	Usage
0 to 995	The Class of Restriction (COR) for the trunk group. Classes of restriction control access to trunk groups, including trunk-to-trunk transfers. Decisions regarding the use of Class of Restriction (COR) and Facility Restriction Levels (FRLs) should be made with an understanding of their implications for allowing or denying calls when AAR/ARS/WCR route patterns are accessed.

+ Tip:

Remember that FRLs are assigned to classes of restriction. Even if two trunk groups have classes of restriction that allow a connection, different facility restriction levels might prevent operations such as off-net call forwarding or outgoing calls by remote access users.

CO Type

Available only if the country code is 14. Used only by trunk group members administered on a DS1 circuit pack board.

Valid Entry	Usage
analog digital	Specifies whether the trunk group is connected to analog or digital facilities at the local telephone company central office (CO).

Related links

[Country](#) on page 791

Country

The country code that corresponds to the protocol used by the local telephone company central office (CO) where the trunk group terminates.

Available only for trunk groups that connect Communication Manager to a CO in the public network — CO, DID, DIOD, FX, and WATS trunk groups.

Valid Entry	Usage
1 to 25, except for 19	For a list of country codes, see the <i>Country code table</i> .

Table continues...

Valid Entry	Usage
11	Communication Manager can be administered for Public Network Call Priority (Call Retention and Re-ring).
15	Communication Manager can be administered for Public Network Call Priority (Intrusion and Re-ring).
18	Communication Manager can be administered for Public Network Call Priority (Mode of Release Control, Forced Disconnect, and Re-ring).
23	If the trunk Group Type is either CO or DID, Communication Manager can be administered for Block Collect Calls.

Related links

[Country options table](#) on page 718

[Trunk Termination](#) on page 821

Dial Access

Controls whether users can route outgoing calls through an outgoing or two-way trunk group by dialing its trunk access code. Allowing dial access does not interfere with the operation of AAR/ARS.

Security alert:

Calls dialed with a trunk access code over WATS trunks bypass AAR/ARS and are not restricted by facility restriction levels. For security, leave this field disabled unless dial access is needed to test the trunk group.

Valid Entry	Usage
y	Users can gain access to the trunk group by dialing its access code.
n	Users cannot gain access to the trunk group by dialing its access code. Attendants can still select this trunk group with a Trunk Group Select button.

Digit Absorption List

Valid Entry	Usage
0 to 4 blank	Assigns a digit absorption list, when used, to a trunk group that terminates at a local telephone company central office.

Note:

In a DCS network, DCS features that use the **remote-tgs** button (on telephones at a remote end) do not work when the incoming trunk group at your end deletes or inserts digits on incoming calls. To manipulate digits in a DCS network (for example, to insert an AAR feature access code), do it on the outgoing side based on the routing pattern.

Direction

The direction of the traffic on this trunk group.

Available for all trunk groups except DID and CPE.

Valid Entry	Usage
incoming	Traffic on this trunk group is incoming.
outgoing	Traffic on this trunk group is outgoing
two-way	Traffic on this trunk group is incoming and outgoing. Required for Network Call Redirection .

Related links

[Disconnect Supervision-Out](#) on page 447

[Answer Supervision Timeout](#) on page 445

[Receive Answer Supervision](#) on page 558

Group Name

A unique name that provides information about the trunk group. Accepts up to 27 characters.

This field should contain names that identify the vendor and function of the trunk group rather than the group type (DID, WATS).

* Note:

Supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, 1600-Series, 9600-Series IP Deskphones, 9600-Series H.323 Deskphones, and Avaya J100 Series IP Phones. Unicode is also an option for the 2420J telephone when the **Display Character Set** is katakana. For more information about 2420J, see *2420 Digital Telephone User's Guide*.

Related links

[Display Character Set](#) on page 722

Group Number

The trunk group number.

Group Type

The type of trunk group. The fields that are displayed and available might change according to the trunk group type selected.

Valid Entry	Usage
Access	Used to connect satellite servers to the main switch in Electronic Tandem Networks (ETN). Access trunks do not carry traveling class marks (TCM) and, therefore, allow satellite callers unrestricted access to out-dial trunks on the main server. This entry allows Inband ANI.
APLT	Advanced Private Line Termination (APLT) trunks. Used in private networks. This entry allows Inband ANI.
CAMA	Used to route emergency calls to the local community's Enhanced 911 systems.

Table continues...

Valid Entry	Usage
CO	Typically used to connect Communication Manager to the local telephone company central office, but can also connect adjuncts such as external paging systems and data modules.
CPE	Used to connect adjuncts, such as paging systems and announcement or music sources, to the server running Communication Manager.
DID	Used to direct callers directly to individuals within an organization without going through an attendant or some other central point. This entry allows Inband ANI.
DIOD	Two-way trunks that are used to transmit dialed digits in both directions. In North America, tie trunks are used for applications that require two-way transmission of dialed digits. This entry allows Inband ANI.
DMI-BOS	Digital Multiplexed Interface - Bit-Oriented Signaling (DMI-BOS) trunks allow communication with systems using DMI-BOS protocol. This entry also allows Inband ANI.
FX	A local telephone company central office (CO) trunk that connects the server running Communication Manager directly to a CO outside the local exchange area. Used to reduce long-distance charges if the organization averages a high volume of long-distance calls to a specific area code.
ISDN	Used when digital trunks are needed that can integrate voice, data, and video signals and provide the bandwidth needed for applications such as high-speed data transfer and video conferencing. ISDN trunks can also efficiently combine multiple services on one trunk group. Also used for Network Call Transfer.  Note: Available only if ISDN-PRI, ISDN-BRI Trunks, or both have been enabled for the system. For descriptions of the screens and fields that are unique to ISDN trunks, see "ISDN Trunk Group".
RLT	Used with Centralized Attendant Service in a private network.
SIP	Used to connect a server running Communication Manager to a Session Manager (SM), or to connect two Communication Manager servers.  Note: The Automatic CallBack, Priority Calling, and Whisper Page features work incorrectly if each of the call's parties is using a SIP endpoint administered on and managed by a different instance of Communication Manager.
Tandem	Used to connect tandem nodes in a private network. This entry allows Inband ANI.
Tie	Used to connect a server running Communication Manager to a local telephone company central office or to another server or switch in a private network. Tie trunks transmit dialed digits with both outgoing and incoming calls. This entry also allows Inband ANI.

Table continues...

Valid Entry	Usage
WATS	Used to reduce long-distance bills when your organization regularly places many calls to a specific geographical area in North America. With outgoing WATS service, calls can be sent to certain areas ("WATS band") for a flat monthly charge. Incoming WATS trunks allow toll-free calling to customers and employees.

Related links

[ISDN-PRI](#) on page 734

[ISDN-BRI Trunks](#) on page 734

[SBS](#) on page 456

[Local Country Code](#) on page 275

[International Access Code](#) on page 275

[Supplementary Service Protocol](#) on page 449

[Carrier Medium](#) on page 434

[Path Replacement](#) on page 462

[Call Still Held](#) on page 808

[Call Still Held](#) on page 554

Incoming Budget

* Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

This field is available only when you set the **ASAC Enabled** field on the Multiple Level Precedence & Preemption Parameters screen to *y*.

Incoming Budget is the number of incoming calls that a trunk group can allow based on the total call limit.

Communication Manager counts the number of active incoming calls and compares the total number of incoming calls with the incoming call budget. If the total number equals or exceeds the budget, Communication Manager replaces established low-precedence calls with new high-precedence calls and blocks new low-precedence calls.

Entire trunk budget (Number of Members) = Incoming budget + Outgoing budget

Valid Entry	Usage
0–255	The number of incoming calls that the trunk group can receive. The total of the Incoming Budget and the Outgoing Budget fields must not be greater than the total number of trunk group members.
Blank	This field is blank by default.

Incoming call budget high threshold

*** Note:**

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Communication Manager sends an SNMP trap when the system reaches the incoming call budget high threshold.

This field is available only when you set the **ASAC Enabled** field on the Multiple Level Precedence & Preemption Parameters screen to *y*.

Valid Entry	Usage
0–255	The call limit at which Communication Manager must send an SNMP trap.
Blank	The field is blank by default.

Incoming call budget low threshold

*** Note:**

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Communication Manager sends an SNMP trap to clear a budget threshold alarm when the system reaches the incoming call budget low threshold.

This field is available only when you set the **ASAC Enabled** field on the Multiple Level Precedence & Preemption Parameters screen to *y*.

Valid Entry	Usage
0–255	The call limit at which Communication Manager must send an SNMP trap.
Blank	The field is blank by default.

Incoming Destination

Sets the destination for all incoming calls on trunk groups such as CO, FX, and WATS that must terminate at a single destination. The destination entered here is also the default night service destination. Available only for incoming or two-way trunk groups.

Valid Entry	Usage
<i>Valid extension number</i>	Calls go to this extension number. You can enter any type of extension, though typically the extension entered here identifies a VDN, a voice response unit, or a voice messaging system. Night service overrides this setting when it is active.  Note: If you enter a shortened extension of the multilocation dial plan, the system does not perform certain administration and validation tasks. Therefore, the system might not display the resultant warnings or submittal denials.
attd	Calls go to the attendant and are recorded as Listed Directory Number (LDN) calls on call detail records.
blank	Leave this field blank if Trunk Type (in/out) is not administered as auto/....

Related links

[Direction](#) on page 436

[Night Service](#) on page 797

[Trunk Type](#) on page 821

ITC

Determines the line coding the Generalized Route Selection feature uses for comparison to select appropriate routes for voice and data calls.

Available only when the **Comm Type** is data, avd, or rbavd and a **BCC** value is administered.

Valid Entry	Usage
rest(tricted)	Restricted trunks use ami-basic or ami-zcs line coding and can carry only restricted calls.
unre(stricted)	Unrestricted trunks use b8zs, hdb3, or cmi line coding and can carry restricted or unrestricted calls. A trunk group with an unrestricted ITC can have only unrestricted trunks as members.

Related links

[Comm Type](#) on page 790

Night Service

Sets the destination for incoming calls when **Night Service** is operating. If a night service destination is administered for an individual trunk that is a member of this group, that entry overrides the group destination for that trunk. CPE, DID, and DIOD trunk groups do not support night service.

Tip:

Whenever possible, use a night service destination on your switch; otherwise, some features do not function correctly over a DCS network.

Valid Entry	Usage
An extension number (can be a VDN)	The extension of your night service destination. * Note: If you enter a shortened extension of the multilocation dial plan, the system does not perform certain administration and validation tasks. Therefore, the system might not display the resultant warnings or submittal denials.
attd	Calls go to the attendant and are recorded as Listed Directory Number (LDN) calls on call detail records.
blank	Leave this field blank if Trunk Type (in/out) is not administered as auto/....

Related links

[Trunk Type \(in/out\)](#) on page 804

Number of Members

The system displays this field SIP and H.323 trunk groups.

In Communication Manager Release 8.0 and later, the number of trunk members that can be administered on a SIP trunk group is increased from the current value of 255 to 9999.

Valid entry	Usage
0 to 9999	The number of trunks that are members of the Trunk group. You cannot administer more than 255 trunk members, only if the Number Assignment Method field is set to <code>auto</code> .

Related links

[Group Type](#) on page 437

Outgoing Budget* **Note:**

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

This field is available only when you set the **ASAC Enabled** field on the Multiple Level Precedence & Preemption Parameters screen to `y`.

Outgoing Budget is the number of outgoing calls a trunk group can allow within the total limit.

Communication Manager counts the number of active outgoing calls, and compares the total number of outgoing calls with the outgoing call budget. If the total number equals or exceeds the budget, Communication Manager replaces established low-precedence calls with new high-precedence calls and blocks new low-precedence calls.

Valid Entry	Usage
0–255	The number of outgoing calls that the trunk group can make. The total of the Incoming Budget and the Outgoing Budget fields must not be greater than the total number of trunk group members.
Blank	The field is blank by default.

 Note:

If the total number of **Incoming Budget** and **Outgoing Budget** is less than **Number of Members**, the system displays the following message:

WARNING: Total budget allocated less than total number of trunk group members.

Outgoing call budget high threshold

 Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Communication Manager sends an SNMP trap when the system reaches the outgoing call budget high threshold.

This field is available only when you set the **ASAC Enabled** field on the Multiple Level Precedence & Preemption Parameters screen to *y*.

Valid Entry	Usage
0–255	The call limit at which Communication Manager must send an SNMP trap.
Blank	The field is blank by default.

Outgoing call budget low threshold

 Note:

This field complies with the Unified Capabilities Requirements (UCR) 2008 Change 3 requirements and is approved by Joint Interoperability Test Command (JITC). This field is available only for the USA Department of Defense (DoD) and approved Federal government customers.

Communication Manager sends an SNMP trap to clear a budget threshold alarm when the system reaches the outgoing call budget low threshold.

This field is available only when you set the **ASAC Enabled** field on the Multiple Level Precedence & Preemption Parameters screen to *y*.

Valid Entry	Usage
0–255	The call limit at which Communication Manager must send an SNMP trap.
Blank	The field is blank by default.

Outgoing Display

With Outgoing Display, telephones can show the name and number of the trunk group used for an outgoing call before the call is connected.

Valid Entry	Usage
y	Displays the trunk group name and number.
n	Displays the digits the caller dials.

Prefix-1

If enabled, the prefix “1” is added to the beginning of the digit string for outgoing calls. Use this field for outgoing and two-way trunk groups handling long distance service. Do not enable for trunk groups in AAR or ARS route patterns.

Available only for CO, FX, and DIOD trunk groups.

Protocol Type

The type of line signaling protocol used by the local telephone company central office for DID and DIOD trunk groups.

Available only if the **Country** code is 15. Used only by trunk group members.

Valid Entry	Usage
inloc	Incoming local. Only the inloc protocol provides ANI.
intol	Incoming toll.

Related links

[Country](#) on page 791

[Country options table](#) on page 718

Queue Length

Available only for outgoing or two-way trunk groups.

Valid Entry	Usage
1 to 100	The number of outgoing calls that can wait in queue when all trunks in a trunk group are busy. Calls wait in queue in the order in which they were made. If a queue is administered, a caller hears a confirmation tone when no trunk is available for the outgoing call. The caller can then disconnect and wait; when a trunk becomes available, Communication Manager calls the extension that placed the original call. Communication Manager remembers the number the caller dialed and automatically completes the call.
0	Callers receive a busy signal when no trunks are available. Use for DCS trunks.

Related links

[Direction](#) on page 436

Service Type

The service for which this trunk group is dedicated. In addition to the predefined services or features listed as valid entries, any previously administered user-defined Network Facility **Facility Type** of 0 (feature) or 1 (service) is allowed.

Valid Entry	Usage
access	A tie trunk giving access to an Electronic Tandem Network.
accunet	ACCUNET Switched Digital Service — part of ACI (AT&T Communications ISDN) phase 2.
cbc	Call-by-Call service — provides different dial plans for different services on an ISDN trunk group. Indicates this trunk group is used by the Call-By-Call Service Selection feature.
dmi-mos	Digital multiplexed interface — message-oriented signaling.
i800	International 800 Service — A subscriber can receive international calls without a charge to the call originating party.
inwats	INWATS — provides OUTWATS-like pricing and service for incoming calls.
lds	Long-Distance Service — part of ACI (AT&T Communications ISDN) phase 2.
megacom	MEGACOM Service — an AT&T communications service that provides unbanded long-distance services using special access (switch to 4ESS switch) from an AT&T communications node.
mega800	MEGACOM 800 Service — an AT&T communications service that provides unbanded 800 service using special access (4ESS switch to switch) from an AT&T communications node.
multiquest	AT&T MULTIQUEST Telecommunications Service — dial 700 service. A terminating-user's service that supports interactive voice service between callers at switched-access locations and service provides directly connected to the AT&T Switched Network (ASN).
operator	Network Operator — provides access to the network operator.

Table continues...

Valid Entry	Usage
outwats-bnd	OUTWATS Band — WATS is a voice-grade service providing both voice and low speed data transmission capabilities from the user location to defined service areas referred to as bands; the widest band is 5.
public-ntwrk	Public network calls — It is the equivalent of CO (outgoing), DID, or DIOD trunk groups. If Service Type is public-ntwrk, Dial Access can be enabled.
sddn	Software Defined Data Network — provides a virtual private line connectivity via the AT&T switched network (4ESS switches). Services include voice, data, and video applications. These services complement the SDN service. Do not use for DCS with Rerouting.
sdn	Software Defined Network (SDN) — an AT&T communications offering that provides a virtual private network using the public switched network. SDN can carry voice and data between customer locations as well as off-net locations.
sub-operator	Presubscribed Common Carrier Operator — provides access to the presubscribed common carrier operator.
tandem	Tandem tie trunks integral to an ET.
tie	Tie trunks — general purpose.
wats-max-bnd	Maximum Banded Wats — a WATS-like offering for which a user's calls are billed at the highest WATS band subscribed to by users.

Related links

[Facility Type](#) on page 540

Signaling Group

Available only for sip trunk groups.

Valid Entry	Usage
1 to 999	The number of the SIP signaling group associated with this trunk group.

Related links

[Group Type](#) on page 437

[Group Number](#) on page 614

TAC

The trunk access code (TAC) that must be dialed to access the trunk group. A different TAC must be assigned to each trunk group. CDR reports use the TAC to identify each trunk group. The characters “*” and “#” can be used as the first character in a TAC. Accepts a one- to four-digit number.

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Toll Restricted

If enabled, restricts toll-restricted users from using a trunk access code to make restricted outgoing calls over this trunk group.

Trunk Flash

Enables or disables multifunction telephones to access local telephone company central office (CO) customized services that are provided by servers at the far-end or CO. These CO customized services are electronic features, such as conference and transfer, that are accessed by a sequence of flash signal and dial signals from the Communication Manager telephone on an active trunk call.

Trunk Signaling Type

Controls the signaling used by members in private network trunk groups, mainly in Italy, Brazil, and Hungary. This field also controls the signaling used by members in public network digital trunk groups.

Available only for access, apIt, rIt, tandem, or tie trunk groups.

E&M trunks in Italy, Brazil, and Hungary can use either continuous or discontinuous signaling. Each entry specifies a set of signals and available timers used in the process of setting up and releasing connections. The type of signaling must match the signaling type administered on the far-end server.

Valid Entry	Usage
cont	Continuous
dis	Discontinuous

The following entries are for tie trunks in Main-Satellite/Tributary networks. Use these values only when all trunk group members are assigned to a TN497 circuit pack.

Valid Entry	Usage
tgu	For outgoing trunks, tgu at the main server running Communication Manager administers a tie trunk group connected to a satellite server. (This same group should be administered as tge at the satellite.)
tge	For incoming trunks, tge at a satellite server administers a tie trunk group connected to the main server running Communication Manager. (This same group should be administered as tgu at the main server.)
tgi	For internal trunks, tgi administers a two-way tie trunk group between two satellites or between the main server and a satellite. (This trunk group should be administered as tgi on both servers.)

DIOD trunks support pulsed and continuous E&M signaling in Brazil and discontinuous E&M signaling in Hungary. The following entries are for DIOD trunks. Use these values only when all trunk group members are assigned to a TN464F (or later version) or TN2464 circuit pack.

Valid Entry	Usage
cont	Continuous E&M signaling.
pulsed	Pulsed E&M signaling.
discont	Discontinuous E&M signaling.
blank	Leave blank for R2 signaling.

Related links

[Group Type](#) on page 437

Trunk Type (in/out)

Controls the seizure and start-dial signaling used on this trunk group. Settings might differ for incoming and outgoing trunks.

Valid Entry	Usage
auto	Used for immediate connection to a single preset destination (incoming local telephone company central office trunks, for example). No digits are sent, because all calls terminate at the same place.
cont	Continuous signaling is used with Italian E&M tie trunks. The server/switch seizes a trunk by sending a continuous seizure signal for at least the duration specified by the Incoming Seizure Timer.
delay	The sending switch does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end switch, indicating that it is ready to receive the digits.
disc	Discontinuous signaling is used with Italian tie trunks that use E&M signaling. The Avaya server can seize a trunk by sending a single, short signal for the duration specified by the Normal Outgoing Seize Send value. However, with the Three-Way Seizure option, the calling end can also send routing information to the called end by sending one or a series of brief seizure signals.
wink	The sending server or switch does not send digits until it receives a wink start (momentary off-hook) signal from the far-end server or switch, indicating that it is ready to receive the digits.
immed	The sending server or switch sends digits without waiting for a signal from the far-end server or switch.
2-wire-ac 2-wire-dc 3-wire	Used with local telephone company central office (CO) trunks in Russia. Select the type of connection to the CO. Check with the network service provider for the type of connection. To use these entries, the Country code must be 15 and the CO trunks must use ports on a TN2199 circuit board.

+ Tip:

When incoming trunks use the setting immed/immed, the far-end server seizes the trunk and sends digits without waiting for acknowledgment from the receiving end. When traffic is heavy, the receiving server or switch might not immediately attach a Touch Tone Receiver to a call and, therefore, lose digits. Use wink-start trunks or increase the dial-guard timer value on the far-end server or switch to avoid this problem.

Related links

[Country](#) on page 791

[Normal Outgoing Seize Send \(msec\)](#) on page 844

Version

Adjusts the signaling on multi-country local telephone company central office (CO) trunk circuit packs. Entries adjust signaling characteristics on these circuit packs to match the signaling characteristics of the public network in a specific country.

Available only for CO, FX, and WATS trunk groups when the **Country** code is 5, 16, or 23.

If the **Country** code is 5, this field controls only TN2147 ports.

Valid Entry	Usage
a	Uses standard signaling for the Netherlands public network.
b	Uses country 1 (U.S.) signaling. This value is appropriate if Communication Manager is connected to a CO using an Ericsson AXE-10 switch.

If the **Country** code is 16 or 23, this field sets the input impedance value and only controls TN465C (vintage 2 or later) ports.

Valid Entry	Usage
a	Sets input impedance to 600 Ohms.
b	Sets input impedance to 900 Ohms. This value is appropriate in Brazil.

Related links

[Country](#) on page 791

Trunk Group: page 2

change trunk-group 1		Page 2 of 22
Group Type: co	Trunk Type:	
TRUNK PARAMETERS		
Outgoing Dial Type: <u>tone</u>	Cut-Through? <u>n</u>	
Trunk Termination: <u>rc</u>	Disconnect Timing(msec): <u>500</u>	
Auto Guard? <u>n</u>	Call Still Held? <u>n</u>	Sig Bit Inversion: <u>none</u>
Analog Loss Group: <u>6</u>	Digital Loss Group: <u>11</u>	
Trunk Gain: <u>high</u>		
Disconnect Supervision - In? <u>y</u> Out? <u>y</u>		
Answer Supervision Timeout: <u>10</u>	Receive Answer Supervision? <u>n</u>	
Administer Timers? <u>y</u>		
XOIP Treatment: <u>auto</u>		

Administer Timers

Enables or disables administration of timers on this trunk group. The default for the ISDN trunk group type is disabled. All other trunk group types are enabled by default.

Available for all trunk group types *except* cpe, h.323, and sip.

Related links

[Group Type](#) on page 437

Analog Loss Group

Determines which administered two-party row in the loss plan applies to this trunk group if the call is carried over an analog signaling port in the trunk group.

Valid Entry	Usage
1 to 17	The index into the loss plan and tone plan. If values are administered other than in between 6 and 10 or 15 and 17, the system displays a warning message indicating that the loss group may not be appropriate for this trunk group.

Answer Supervision Timeout

Valid Entry	Usage
0 to 250	The number of seconds Communication Manager waits before it acts as though answer supervision has been received from the far-end. During a cut-through operation, timing begins after each outgoing digit is sent and timing ceases after the far-end sends answer supervision. On senderized operation, the timer begins after the last digit collected is sent.

 Note:

This field's setting does not override answer supervision sent from the network.

Related links

[Receive Answer Supervision](#) on page 558

[Administer Timers](#) on page 445

Auto Guard

Enables or disables Auto Guard, which prevents repeated seizures of a defective trunk. Communication Manager does a maintenance busy-out on these trunks. This field controls ports only on TN438B, TN465B, and TN2147 circuit packs. TN438B ports have hardware support for detecting a defective trunk. TN465B and TN2147 ports use a trunk defective if no dial tone is detected on an outgoing call, and the **Outputpulse Without Tone** feature is disabled for the system.

Available only for co or fx trunk groups.

Related links

[Group Type](#) on page 437

[Outputpulse Without Tone](#) on page 279

Bit Rate

Available when the **Comm Type** is avd or rbavd. Also available if the **Comm Type** is data, but only if **ISDN-PRI** is enabled for the system.

Valid Entry	Usage
300	The baud rate to be used by pooled modems.
1200	
2400	
4600	
9600	
19200	

Related links

[Comm Type](#) on page 790

[ISDN-PRI](#) on page 734

Caller ID for Service Link Call to H.323 1xC

This field is available only when you set the **Direction** field to `outgoing` or `two-way` and the trunk group type is `ISDN` or `SIP`.

Valid entry	Usage
original-calling-number	Before sending the call to the service link, Communication Manager changes the calling party number to the original calling party number. When the user receives an incoming call, the number that the mobile displays is the original caller ID and not the extension number.
station-extension	Communication Manager does not change the calling party number. The default value is station-extension.

Call Still Held

If enabled, the system prevents glare by extending the Incoming Glare Guard timer and delaying an outgoing seizure of a trunk for at least 140 seconds after it is released from an incoming call. This field is used when the receiving end media server or switch initiates the disconnection of incoming calls. This field affects only TN438B, TN465B, and TN2147 ports and is used primarily when the **Country** code is 2.

Available only for co or fx trunk groups.

Related links

[Group Type](#) on page 437

Cut-Through

Available only if the **Outgoing Dial Type** field is rotary or tone.

Security alert:

Enabling this field reduces the ability to prevent toll fraud.

Valid Entry	Usage
y	Users get dial tone directly from the far end of the trunk. If you set the Cut-Through field to y, some administered restrictions are bypassed.
n	Users receive switch dial tone. Instead of digits being sent to the central office, they are collected, and sent all at once when the user finishes dialing.

Related links

[Outgoing Dial Type](#) on page 816

Cyclical Hunt

Controls the starting point Communication Manager uses to search for an available trunk when a call is sent to the trunk group. Cyclical hunts can be enabled or disabled at any time, however, all the trunks in the group must be idle or busied out.

Available only for two-way loop-start trunks.

Valid Entry	Usage
y	Communication Manager starts its search from the last trunk seized. This method is faster, and therefore suited for high-traffic trunk groups.
n	Communication Manager starts each search at member 1 (the first trunk administered as a group member).

Related links

[Direction](#) on page 436

[Trunk Type](#) on page 821

Delay Call Setup When Accessed Via IGAR

The system displays this, when the **Group Type** field is isdn or sip. If a user in one network region accesses an ISDN or SIP trunk in another network region through IGAR, the **Delay Call Setup When Accessed Via IGAR** field determines whether the IGAR trunk and the outgoing trunk are initiated in sequence or parallel.

Valid Entry	Usage
y	The outgoing trunk call waits until the IGAR trunk call is active.
n	The two trunk calls are set up in parallel. The default value is n.

Dial Detection

Indicates whether digit pulses are detected by observing the A-wire (default) or the B-wire only. Applies only to TN2199 ports. The **Country** code must be 15.

Related links

[Country](#) on page 791

Digital Loss Group

Valid Entry	Usage
1 to 19	Determines which administered two-party row in the loss plan applies to this trunk group if the call is carried over a digital signaling port in the trunk group. If values other than 18 or between 11 and 15 are administered, the system displays a warning message indicating that the loss group may not be appropriate for this trunk group.

Digits

If the **Digit Treatment** is absorption, this field specifies how many digits are deleted. If the **Digit Treatment** is insertion, this field identifies the specific digits that are added.

Valid Entry	Usage
1 to 5	The number of digits to be deleted (absorbed).
Up to 4 digits, including * and #	The actual digits to be added (inserted).
blank	This field can be blank only if Digit Treatment is not administered.

Related links

[Digit Treatment](#) on page 810

Digit Treatment

Modifies an incoming digit string by adding or deleting digits. This is required if the number of digits received does not match the dial plan.

Requires administration of the **Digits** to add or delete.

Valid Entry	Usage
absorption	Deletes digits, starting at the beginning of the string.
insertion	Adds digits, starting at the beginning of the string.
blank	The incoming digit string is not changed.

Related links

[Digits](#) on page 809

Disconnect Supervision-In

Indicates whether Communication Manager receives disconnect supervision for incoming calls over this trunk group.

Available only for incoming or two-way trunk groups.

Valid Entry	Usage
y	Allows trunk-to-trunk transfers involving trunks in this group. The far-end server or switch sends a release signal when the calling party releases an incoming call, and the far-end server or switch is responsible for releasing the trunk. Enhances Network Call Redirection.
n	The far-end server or switch does not provide a release signal, the hardware cannot recognize a release signal, or timers are preferred for disconnect supervision on incoming calls. Prevents trunk-to-trunk transfers involving trunks in this group.

 Caution:

In general, U.S. local telephone company central offices provide disconnect supervision for incoming calls but not for outgoing calls. Public networks in most other countries do not provide disconnect supervision for incoming or outgoing calls. Check with the network services provider.

Related links

[Trunk Direction](#) on page 558

[Direction](#) on page 436

Disconnect Supervision-Out

Indicates whether Communication Manager receives disconnect supervision for outgoing calls over this trunk group. Available for outgoing or two-way trunk groups.

Valid Entry	Usage
y	Allows trunk-to-trunk transfers involving trunks in this group. The far-end sends a release signal when the called party releases an outgoing call, and the far-end is responsible for releasing the trunk. Enhances Network Call Redirection. Available only if Answer Supervision Timeout is 0 and Receive Answer Supervision is enabled.
n	The far-end server or switch does not provide a release signal, the hardware cannot recognize a release signal, or timers are preferred for disconnect supervision on outgoing calls. Prevents trunk-to-trunk transfers involving trunks in this group.

 Caution:

Verify that the far-end server or switch provides answer supervision and disconnect supervision. Most public networks do not provide disconnect supervision over analog trunks. Check with the network services provider.

Related links

[Direction](#) on page 436

[Answer Supervision Timeout](#) on page 445

[Receive Answer Supervision](#) on page 558

Disconnect Timing (msec)

Valid Entry	Usage
140 to 2550 ms in increments of 10	Specifies the minimum time in milliseconds that the local telephone company central office or far-end server requires to recognize that this server has disconnected from a call. Settings on those two timers override this field. The default of 500 is an industry standard and should not be changed. If this field is set too high, the server or switch does not disconnect sometimes when it should. If this field is set too low, the server or switch disconnects when it should not.

Disconnect Type

Indicates which side or user controls the disconnect. A refers to the calling party, and B refers to the called party.

Available only if the **Country** code is 15 and the **Trunk Type** is 2-wire-ac, 2-wire-dc, or 3-wire. Applies only to the TN2199 port.

Valid Entry	Usage
AandB	Both parties control the disconnect.
AorB	Either party controls the disconnect.

Related links

[Country](#) on page 791

[Trunk Type](#) on page 821

Drop Treatment

Valid Entry	Usage
intercept busy silence	Determines what the calling party hears when the called party terminates an incoming call. For security reasons, you must apply a tone; silence can provide an opening for hackers. Applies only to DID trunks.

*** Note:**

In Italy, **Drop Treatment** must be administered as intercept for all DID trunk groups.

Duplex

Available if the **Comm Type** is avd or rbavd. Also available if the **Comm Type** is data, but only if **ISDN-PRI** is enabled for the system.

*** Note:**

Even if the trunk group supports full-duplex transmission, other equipment in a circuit might not.

Valid Entry	Usage
full	Allows simultaneous two-way transmission, which is most efficient.
half	Supports only one transmission direction at a time.

Related links

[Comm Type](#) on page 790

[ISDN-PRI](#) on page 734

End-to-End Signaling

Available only for a cpe (customer-provided equipment) trunk group.

Valid Entry	Usage
60 to 360 ms in increments of 10	The duration of the touch tone signal sent to auxiliary equipment. Equipment such as paging equipment and music sources might be connected to Communication Manager by auxiliary trunks. Communication Manager might send DTMF signals (touch tones) to these devices. This field sets the duration of these tones.

Related links

[Group Type](#) on page 437

Expected Digits

Valid Entry	Usage
1 to 18	The number of digits that the far-end server sends for an incoming connection. If the end is absorbing digits on this trunk group, the entry in this field must be larger than the value administered for the number of digits to be absorbed.
blank	Required if Digit Treatment is insert and Digits is a feature access code (for example, AAR or ARS) followed by digits. In this case, the number of digits expected are set on the AAR and ARS Digit Analysis Table and AAR and ARS Digit Conversion Table. If left blank, digit absorption cannot be administered.

Related links

[Digits](#) on page 809

[Digit Treatment](#) on page 810

Extended Loop Range

Available only for a DID trunk group.

Valid Entry	Usage
y	The distance between the local telephone company central office (CO) and the server is greater than the required distance.
n	The distance between the CO and the server is not greater than the required distance.

Format

Specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number.

Available only if the **Send Calling Number** or **Send Connected Number** field is enabled or restricted.

Valid Entry	Usage
public	Indicates that the number plan according to CCITT Recommendation E. 164 is used.
unknown	Indicates the Numbering Plan Indicator is unknown.
private	Indicates the Numbering Plan Indicator is PNP.
unk-pvt	Determines the type of number from the private numbering format, but the Numbering Plan Indicator is unknown.

Related links

[Send Calling Number](#) on page 457

[Send Connected Number](#) on page 458

Group Type

Displays the type of trunk group.

Related links

[Group Type](#) on page 437

[ISDN-BRI Trunks](#) on page 734

[ISDN-PR](#) on page 734

Incoming Calling Number - Delete

Available only for incoming or two-way trunk groups.

Valid Entry	Usage
1 to 15	The number of digits, if any, to delete from the calling party number for all incoming calls on this trunk group.
all	
blank	

Related links

[Direction](#) on page 436

Incoming Calling Number - Format

The TON/NPI encoding applied to CPN information modified by the CLI Prefix feature. This encoding does not apply to calls originating locally.

If this field is blank, Communication Manager passes on the encoding received in the incoming setup message. If the incoming setup message excludes CPN information and digits are added, the outgoing message contains these digits. If a numbering format is not administered in this case, the value defaults to pub-unk. If the numbering format is administered as unknown, the trunk group is modified to unk-unk encoding of the TON/NPI. Therefore, this field also must contain a value other than unknown.

The values for this field map to the Type of Numbering (TON) and Numbering Plan Identifier (NPI) values shown below.

Valid Entry	Type of Numbering (TON)	Numbering Plan Identifier (NPI)
blank	incoming TON unmodified	incoming NPI unmodified
natl-pub	national(2)	E.164(1)
intl-pub	international(1)	E.164(1)
locl-pub	local/subscriber(4)	E.164(1)
pub-unk	unknown(0)	E.164(1)
lev0-pvt	local(4)	Private Numbering Plan - PNP(9)
lev1-pvt	Regional Level 1(2)	Private Numbering Plan - PNP(9)
lev2-pvt	Regional Level 2(1)	Private Numbering Plan - PNP(9)
unk-unk	unknown(0)	unknown(0)

Related links

[Format](#) on page 813

[Format](#) on page 453

Incoming Calling Number - Insert

Valid Entry	Usage
0 to 9 all blank	Up to 15 digits added to the beginning of the digit string of incoming calls when the calling party is a member of this trunk group.

Incoming Dial Tone

Indicates whether or not the server running Communication Manager gives dial tone in response to far-end seizures of the trunk group.

Valid Entry	Usage
y	Used if the incoming trunk group transmits digits. For example, this option is used for two-way, dial-repeating tie trunks that users select by dialing a trunk access code.
n	Used for trunks that are not sending digits, such as tandem or incoming local telephone company central office trunks.

Incoming Dial Type

Indicates the type of pulses required on an incoming trunk group. This value should match what the local telephone company central office provides.

Available for Access, APLT, DID, DIOD, DMI-BOS, FX, RLT, Tandem, or WATS trunk groups. Also available for tie trunk groups when the **Trunk Signaling Type** is blank, cont, or dis.

Valid Entry	Usage
tone	Use this entry for Dual Tone Multifrequency (DTMF) addressing, also known as “touchtone” in the U.S. With this entry, trunk group supports both DTMF and rotary signals. Use this entry for the Inband ANI feature. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use tone or mf.
rotary	Allows only the dial pulse addressing method used by non-touch tone telephones. Though the tone entry supports rotary dialing as well, it is inefficient to reserve touch tone registers for calls that do not use DTMF.
mf	Use this option if the Trunk Signaling Type field is blank. The system displays this field only if you set the Multifrequency Signaling field on the System Parameters Customer-Options screen to y. You cannot use this option if the Used for DCS field is y. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use tone or mf.
mf2/6	Use this entry for signaling in Spain. This signaling type provides normal public network service and a virtual private network digital trunk service called IBERCOM. Use this entry if the Trunk Signaling Type field is blank. The system displays this entry only if you enable Multifrequency Signaling for the system. The system does not display this entry if the trunk is used for DCS.

Related links

[Group Type](#) on page 437

Incoming Rotary Timeout (sec)

Valid Entry	Usage
5 to 99 blank	Sets the maximum time to wait to receive all incoming digits from the far-end switch.

Related links

[Incoming Dial Type](#) on page 815

Line Length

Available only for trunk groups with a **Trunk Signaling Type** of tge, tgi, or tgu.

Valid Entry	Usage
short long	The line length.

Related links

[Group Type](#) on page 437

[Trunk Signaling Type](#) on page 803

Outgoing Dial Type

Sets the method used to transmit digits for an outgoing call. Usually, this method should match what the local telephone company central office provides.

DIOD trunks support pulsed and continuous E&M signaling in Brazil and discontinuous E&M signaling in Hungary.

Available for Access, APLT, CO, DIOD, DMI-BOS, FX, RLT, and WATS trunk groups. Also available for Tie trunk groups when the **Trunk Signaling Type** is blank, cont, or dis.

Valid Entry	Usage
tone	Uses Dual Tone Multifrequency (DTMF) addressing, also known as “touchtone” in the U.S. Allows the trunk group to support both DTMF and rotary signals.
rotary	Allows only the dial pulse addressing method used by non-touch tone telephones. For example, this value is appropriate for an internal full touch tone system and for a connection to a local telephone company central office that only supports rotary dialing.
automatic	For tie trunks if the Trunk Signaling Type is not administered. This provides “cut-through” operation to outgoing callers who dial a trunk access code, connecting them directly to local telephone company central office dial tone and bypassing any toll restrictions administered on Communication Manager.

Table continues...

Valid Entry	Usage
mf	Used if a Trunk Signaling Type is not administered. Available only if Multifrequency Signaling is enabled on the System-Parameters Customer-Options screen. Not available if this trunk is used for DCS. For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use tone or mf.
mf2/6	Used if a Trunk Signaling Type is not administered. Available only if Multifrequency Signaling is enabled on the System-Parameters Customer-Options screen. Not available if this trunk is used for DCS.
r1mf	Enter r1mf for CAMA trunk groups. It is the only outgoing dial type allowed on CAMA trunk groups. Enter r1mf to allow Russian MF Packet Signaling on outgoing trunks. Russian MF Packet Signaling carries calling party number and dialed number information. Group type field must be set to co.

Related links

[Group Type](#) on page 437

[Multifrequency Signaling](#) on page 735

[Trunk Signaling Type](#) on page 803

Preferred Minimum Session Refresh Interval (sec)

The system displays this field only for the SIP trunk groups that do not support Seamless Converged Communications Across Network (SCCAN) calls.

Use this field to set the value of the session refresh timer of non-SCCAN applications. The timer starts after Communication Manager establishes a session with the far-end SIP entities. After every timer interval, Communication Manager sends a session refresh request as a Re-INVITE or an UPDATE message to the SIP entities. If the SIP entities do not receive the request before the interval period expires, the session ends.

Valid entry	Usage
90 to 64800	Defines the session refresh timer. The default value is 600.

* Note:

If the SIP trunk is a member of a trunk group connected to Session Manager 6.3.6 or earlier, ensure that the value of the field is not more than 1800. If the SIP trunk is not a member, configure Session Manager with an adaptation module for Communication Manager, and set the **SessionTimeout** parameter on System Manager to a value greater than the value in the **Preferred Minimum Session Refresh Interval** field.

Related links

[Group Type](#) on page 437

[SCCAN](#) on page 818

Receive Answer Supervision

If enabled, the Communication Manager provides answer supervision for a trunk group. For Outbound Call Management applications, use for trunks supporting network answer supervision. For trunks that do not receive a real answer, this field determines when the CallVisor Adjunct-Switch Application Interface (ASAI) connect event is sent.

Related links

[Answer Supervision Timeout](#) on page 445

[Administer Timers](#) on page 445

Receive Release Ack

If enabled, the **Receive Release Ack** field is used to set the receive forward and backward release ack options on TN2140 ports (used for Italian and Hungarian tie trunks). Available only if the **Trunk Signaling Type** is cont or dis.

Related links

[Trunk Signaling Type](#) on page 803

Redirect on OPTIM failure

Valid Entry	Usage
250 to 32000 milliseconds	Calls are redirected by ROOF (Redirect on OPTIM Failure) procedures if a call terminates to an Extension to Cellular user cell phone (of any application type) that does not have an associated office telephone (or other method of terminating to a physical telephone), or the office telephone is out of order and Communication Manager receives a disconnect prior to the call being answered. The procedure redirects the call so that the caller does not hear ringback forever.

SCCAN

If enabled, this trunk group provides support for incoming SCCAN calls. Available only for sip type trunk groups when **Enhanced EC500** is enabled for the system.

Related links

[Enhanced EC500](#) on page 732

[Group Type](#) on page 437

Send Answer Supervision

If enabled, Communication Manager signals the calling server when an incoming call is answered. Available only if the **Trunk Signaling Type** is cont or dis. The field applies only to TN2140 ports. Available only for incoming or two-way trunks.

Related links

[Direction](#) on page 436

[Trunk Signaling Type](#) on page 803

Send Release Ack

Indicates whether a release acknowledgment is sent in response to a forward or backward release signal. Available only if the **Trunk Signaling Type** is cont or dis. This field applies only to TN2140 ports (used for Italian and Hungarian tie trunks).

Related links

[Trunk Signaling Type](#) on page 803

Sig Bit Inversion

Indicates which bits in bit-oriented signaling should be inverted, if any. This field inverts the A- and B-bits as necessary so that the far-end server or switch can understand seizure and release signals from Communication Manager. If the far-end server, such as a local telephone company central office, on this trunk group interprets the A- and B-bits differently from the default, invert one or both bits — to change “1” to “0” and vice-versa in the A-bit.

Valid Entry	Usage
A	For the MM710 media modules.
B	
A&B	
none	
A and none	For the Japanese 2Mbit trunk.

Supplementary Service Protocol

Available only for ISDN trunk groups.

Valid Entry	Usage
a	Allows ASAI Flexible Billing. AT&T, Telcordia Technologies, Nortel. When the Country code for the DS1 media module is 1A, SSA selects AT&T custom supplementary services. When the Country code for the DS1 media module is 1B, SSA selects Telcordia Technologies Supplementary Services. When the Country code for the DS1 media module is 1C, SSA selects Nortel Proprietary Supplementary Services.
b	QSIG; also used for SBS signaling trunk groups when full QSIG functionality is needed.
c	ETSI; used for Network Call Deflection.
d	ECMA QSIG
e	Allows ASAI Flexible Billing. Allows DCS with rerouting when the trunk is used for DCS and DCS with Rerouting is enabled.
f	Feature Plus
g	ANSI. Available only if ISDN-PRI or ISDN-BRI Trunks is enabled for the system, or the trunk is used for DCS. Used for Network Call Transfer.

Related links

[DCS with Rerouting](#) on page 731

[ISDN-BRI Trunks](#) on page 734

[ISDN-PRI](#) on page 734

[Used for DCS](#) on page 838

Synchronization

Available only if the **Group Type** is :

- dmi-bos or isdn
- access, co, fx, tandem, tie, or wats; and the **Comm Type** is avd or rbavd
- access, co, fx, tandem, tie, or wats; the **Comm Type** is data; and **ISDN-PRI** or **ISDN-BRI Trunks** is enabled for the system

 Caution:

Do not change this field without the assistance of Avaya or your network service provider.

Valid Entry	Usage
async sync	Determines whether the trunk group uses synchronous or asynchronous communications.

Related links

[Comm Type](#) on page 790

[Group Type](#) on page 437

[ISDN-BRI Trunks](#) on page 734

[ISDN-PRI](#) on page 734

Trunk Hunt

Defines the trunk hunt search order. Communication Manager performs a trunk hunt when searching for available channels within a facility in an ISDN trunk group. The search can be administered per ISDN-PRI trunk group, but it infers the direction of search within all ISDN-PRI facilities (or portions of those facilities) administered within the trunk group.

Valid Entry	Usage
ascend	Enables a linear trunk hunt search from the lowest to highest numbered channels. All trunks within an ISDN trunk group are selected without regard to the order in which trunks are administered within the trunk group.
cyclical	Enables a circular trunk hunt based on the sequence the trunks were administered within the trunk group. When using ISDN-BRI interfaces, only cyclical is supported. The cyclical option cannot be set if the trunk group using ISDN-PRI interfaces is to be used for Wideband operations.

Table continues...

Valid Entry	Usage
descend	Enables a linear trunk hunt search from the highest to lowest numbered channels. All trunks within an ISDN trunk group are selected without regard to the order in which trunks are administered within the trunk group.

Related links

[Wideband Support](#) on page 461

Trunk Termination

Adjusts the impedance of the trunk group for optimal transmission quality.

Valid Entry	Usage
600ohm	The distance to the local telephone company central office (CO) or to the server at the other end of the trunk is less than 3,000 feet.
rc	The distance to the CO or to the server at the other end of the trunk is more than 3,000 feet.

Trunk Type

Controls the seizure and start-dial signaling used on this trunk group. Entries in this field vary according to the function of the trunk group and must match the corresponding setting on the far-end server or switch.

Available only for CO, DID, FX, and WATS trunk groups.

Valid Entry	Usage
ground-start	Use ground-start signaling for two-way trunks whenever possible. Ground-start signaling avoids glare and provides answer supervision from the far end.
loop-start	In general, loop-start signaling is used only for one-way trunks. Loop-start signaling is susceptible to glare and does not provide answer supervision.
auto/auto auto/delay auto/immed auto/wink	The term before the slash tells Communication Manager how and when it receives incoming digits. The term after the slash tells Communication Manager how and when it should send outgoing digits. • auto — Used for immediate connection to a single preset destination (incoming central office trunks, for example). No digits are sent, because all calls terminate at the same place. • delay — The sending server running Communication Manager does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end server or switch, indicating that it is ready to receive the digits. • immed — The sending server running Communication Manager sends digits without waiting for a signal from the far-end server or switch. • wink — The sending server running Communication Manager does not send digits until it receives a wink start (momentary off-hook) signal from the far-end server or switch, indicating that it is ready to receive the digits.

Table continues...

Valid Entry	Usage
2-wire-ac 2-wire-dc 3-wire	These entries are used with local telephone company central office (CO) trunks in Russia. The specific CO should match one of these values. Available only if the Country code is 15 and the CO trunks use ports on a TN2199 circuit board.

Related links

[Country](#) on page 791

Unicode Name

Used to determine whether to send Name1 (legacy name) or Name2 (Unicode name). The value for this field is only examined for calls to SIP stations Available only for sip trunk groups.

 Note:

Avaya BRI deskphones support only ASCII characters because non-ASCII characters, such as Eurofont and Kanafont, show up incorrectly.

Valid Entry	Usage
y	Uses the Unicode name.
n	Uses the name as specified on the station form.  Note: Any non-ASCII characters in the name do not appear correctly on a SIP phone.
auto	The choice to use the station name or the Unicode name is automatically determined based on the called phone's capability and the user's preference.

Related links

[Group Type](#) on page 437

Wink Timer (msec)

Sets the wink timer as follows:

- Sets the maximum duration of the wink signal (wait-for-wink-to-end) when **Trunk Type (in/out)** is administered as .../wink.
- Sets the maximum interval after trunk seizure for the wink to begin (wait-for-wink-to-start) when **Trunk Type (in/out)** is administered as .../delay.

Requirements for the United States domestic network specify that the wink signal for wink-start trunks must begin within 5 seconds after a trunk is seized. For trunks with a delay-dial start, the wink must not last longer than 5 seconds.

Available only for wink-type trunks.

Related links

[Trunk Type \(in/out\)](#) on page 804

XOIP Treatment

Valid Entry	Usage
auto modem fax tty	Identifies treatment of an incoming trunk call. When set to modem, fax, or tty, the serving digital signal processor algorithms is placed in immediate pass-through mode. Use this field when the incoming trunk is transporting the XOIP traffic. By identifying this external equipment through administration, VoIP firmware uses this field to determine if it is necessary to immediately attempt to put a call in pass-through mode so that the DSPs do not miss identification of the initial tone events. Default is auto. * Note: • This feature does not apply to IP trunks (SIP and H.323 trunks). • This field is intended for exception cases only. For the majority of trunk groups, use the default setting of auto.

Trunk Group: page 3



Caution:

Do not change fields on this page without assistance from Avaya or your network service provider.

```

change trunk-group 1                                     Page 3 of 22
TRUNK FEATURES
  ACA Assignment? n                                Measured: none
                                                    Maintenance Tests? y
  Data Restriction? n
  Abandoned Call Search? n
  Suppress # Outpulsing? n
  Charge Conversion: 1
  Decimal Point: none
  Currency Symbol:  
  Charge Type: units                                Receive Analog Incoming Call ID: disabled
                                                    Per Call CPN Blocking Code:  
                                                    Per Call CPN Unblocking Code:  
  Outgoing ANI:                                  Dsl Echo Cancellation? n

```

Abandoned Call Search

Indicates whether this trunk group conducts an Abandoned Call Search to identify ghost calls. Abandoned Call Search is designed to work with analog ground-start local telephone

company central office (CO) trunks that do not provide disconnect supervision. The CO must support Abandoned Call Search for the feature to work properly. If the CO provides disconnect supervision, the Abandoned Call Search feature is not needed.

Available only for ground-start type trunks.

Related links

[Trunk Type](#) on page 821

ACA Assignment

Indicates whether Automatic Circuit Assurance (ACA) measurements are taken for this trunk group.

Charge Conversion

Available only for outgoing or two-way CO, DIOD, FX, and WATS trunk groups. Also available for isdn trunk groups, if the **Charge Advice** field is set to a value other than none.

Valid Entry	Usage
1 to 64 500	Communication Manager multiplies the number of charge units by the value of this field and displays it as a currency amount. Without a value in this field, Communication Manager displays the number of charge units without converting it to currency.

Related links

[Direction](#) on page 436

[Trunk Direction](#) on page 558

Charge Type

Text string used to describe charges related to a telephone call. These words or characters appear on telephone displays after the charge amount. Typically uses either the currency symbol or the charge type, but not both. Accepts up to seven characters. Embedded spaces count as characters.

Available only for outgoing or two-way CO, DIOD, FX, and WATS trunk groups.

Related links

[Direction](#) on page 436

Connected to CO

Enables or disables overlap sending to a local telephone company central office (CO).

Available only for tie trunk groups.

Related links

[Group Type](#) on page 437

Currency Symbol

The telephone displays the symbol before the charge amount. Accepts from one to three characters. Leading and embedded spaces count as characters.

Available only for outgoing or two-way CO, DIOD, FX, and WATS trunk groups. Also available for ISDN trunk groups, if the Charge Advice field is set to a value other than none.

Related links

[Direction](#) on page 436

[Trunk Direction](#) on page 558

Data Restriction

Enables or disables data restriction that is used to prevent tones, such as call-waiting tones, from interrupting data calls. Data restriction provides permanent protection and cannot be changed by the telephone user. Cannot be assigned if **Auto Answer** is administered as all or acd. If enabled, whisper page to this station is denied.

Decimal Point

The appropriate representation for a decimal point as telephone displays it. Available only with outgoing or two-way CO, DIOD, FX, and WATS trunk groups. Also available for isdn trunk groups, if the **Charge Advice** field is set to a value other than none.

* Note:

If the received charge contains no decimals, no decimal point is displayed (that is, the administered decimal point is ignored for charge information received with no decimals). On a QSIG trunk group, unlike other trunk groups, the **Decimal Point** field is independent whether the calling display shows a decimal point. Instead, it tells what symbol should be displayed if the QSIG AOC received has a 1/10 or 1/100 or 1/1000 Multiplier.

Valid Entry	Usage
comma	If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a comma as the decimal point. Divides the charge value by 100.
period	This is the default. If the received charge contains decimals, the charge is displayed at the calling endpoint's display with a period as the decimal point. Divides the charge value by 100.
none	No decimal point is displayed.

Related links

[Direction](#) on page 436

[Charge Advice](#) on page 435

DS1 Echo Cancellation

Enables or disables echo cancellation on a per port basis. If enabled, reduces voice call echo.

 Note:

Changes to the DS1 Echo Cancellation field is incomplete until one of the following occurs:

- Port is busied-out or released.
- Trunk group is busied-out or released.
- SAT command test trunk group is performed.
- Periodic maintenance runs.

DSN Term

Enables or disables the trunk group as a DSN termination telephone. The default is disabled.

Glare Handling

Determines the reaction of Communication Manager to glare.

The following circuit packs can detect glare:

- TN767 (all releases)
- TN760C (or later releases)
- TN464C (or later releases)

Available only for two-way trunks when the outgoing side of the **Trunk Type** is either .../wink or .../delay.

Valid Entry	Usage
control	Communication Manager seizes the trunk and proceeds with call setup. The other switch finds another trunk.
backoff	The other server or switch seizes the trunk and proceeds with call setup. The server or switch finds another trunk.
none	Not administered.

Related links

[Direction](#) on page 436

[Trunk Type](#) on page 821

Hold/Unhold notification

Use this field to enable Communication Manager to send a notification to the far end SIP endpoint when the far end SIP endpoint is put on hold or unhold.

When you upgrade to Communication Manager Release 8.1.3, the **Hold/Unhold Notifications** field is not visible for all SIP trunks and the existing values for all sip trunks are reset to n. For the **Hold/Unhold Notifications** field to be visible for all SIP trunks, you must enable the **SA9143 Hold/Unhold Notifications for SIP Trunks** field on the system-parameters special-applications screen. .

The default value of **Hold/Unhold Notifications** field on the trunk group for SIP trunks is n. For example, if you add a new SIP trunk when **SA9143 Hold/Unhold Notifications for SIP Trunks** field is already enabled, the value in the **Hold/Unhold Notifications** field appears as n.

The **Hold/Unhold Notifications** field appears for all SIP trunks, only if the **SA9143 Hold/Unhold Notification for SIP Trunks** field on the system-parameters special-applications screen is set to y. Also, if the **Hold/Unhold Notifications** field for any SIP trunk-group is set to y, then you cannot disable the **SA9143 Hold/Unhold Notifications for SIP Trunks** field on the system-parameters special-applications screen. For more information, see *Avaya Aura® Communication Manager Special Application Features* document.

Valid entry	Usage
y	Communication Manager sends the reINVITE to the far end SIP endpoint to notify the far end SIP endpoint that the endpoint is put on hold or unhold. The media direction in this reINVITE is sendonly when Music On Hold is enabled and is inactive when Music On Hold is disabled. When far end SIP endpoint is put on unhold then reINVITE with media direction sendrecv is send.
n	Communication Manager does not send the reINVITE to the far end SIP endpoint. The default value is n.

Incoming Tone (DTMF) ANI

Digits received through Automatic Number Identification (ANI) are printed on a CDR record, passed to the Intuity Audix and ASAI interfaces, and displayed on the telephone (and on tandem calls if the outgoing trunk requires ANI). Then the digits are sent to the outgoing trunk.

Available only if the **Incoming Dial Type** is tone.

Valid Entry	Usage
*ANI*DNIS*	If 555-3800 calls extension 81120, the trunk group receives *55538000*81120*. The telephone displays Call from 555-3800.
ANI*DNIS*	If 555-3800 calls extension 81120, the trunk group receives 55538000*81120*. The telephone displays Call from 555-3800.
no	Not administered.

Related links

[Incoming Dial Type](#) on page 815

Internal Alert

Indicates if internal ringing and coverage is used for incoming calls.

Invoke ID for USNI Calling Name

Communication Manager uses the **Invoke ID for USNI Calling Name** field to select an Invoke ID each time a calling name is sent over the ISDN trunk group using the USNI protocol.

The system displays the **Invoke ID for USNI Calling Name** field only when:

- The **Group Type** field is set to ISDN.

- The **Carrier Medium** field is set to PRI/BRI.
- The **Supplementary Service Protocol** field is set to a.

However, the valid entries, `variable` and `fixed-1`, have an impact only when the following conditions are met:

- On the Feature-related system parameters screen, the **USNI Calling Name for Outgoing Calls** field is set to `y`.
- On the Trunk group screen, the **Send Name** field is set to `y`.

Valid entry	Usage
variable	Communication Manager selects a new Invoke ID each time a calling name is sent over the ISDN trunk group. This entry is the default value.
fixed-1	Communication Manager selects an Invoke ID with the fixed value 1 each time a calling name is sent over the ISDN trunk group.

Long Holding Time (hours)

Available only if Automatic Circuit Assurance measurements are taken for the trunk group.

Valid Entry	Usage
0 to 10	The number of hours that the system considers a long holding time. A value of 0 indicates that the system does not support long holding calls.

Related links

[ACA Assignment](#) on page 450

Maintenance Tests

Enables or disables hourly maintenance tests on this trunk group.

Available only for `aplt`, `isdn`, `sip`, or `tie` trunk groups.

Related links

[Group Type](#) on page 437

Measured

Indicates if the system transmits data for this trunk group to the Call Management System.

Valid Entry	Usage
internal	Sends the data to the Basic Call Management System (BCMS), the VuStats data display, or both. Available only if BCMS (Basic) or VuStats is enabled for the system.
external	Sends the data to the CMS.

Table continues...

Valid Entry	Usage
both	Collects data internally and sends it to the CMS. Available only if BCMS (Basic) or VuStats is enabled for the system.
none	Trunk group measurement reports are not required.

Related links

[VuStats](#) on page 743

[BCMS \(Basic\)](#) on page 739

MF Tariff Free

If enabled, Communication Manager generates an MFC Tariff-Free Backward Signal during call setup instead of the “free” signal. This aids local telephone company central office billing.

Available only for Access, APLT, DID, DIOD, DMI-BOS, and Tandem trunk groups when the **Incoming Dial Type** is mf or for tie trunk groups when the **Trunk Signaling Type** is blank, cont, or dis, and the **Incoming Dial Type** is mf.

Related links

[Group Type](#) on page 437

[Incoming Dial Type](#) on page 815

[Trunk Signaling Type](#) on page 803

Modify Tandem Calling Number

The system displays this field for the SIP and ISDN trunk groups. This field is available with outgoing or two-way trunks.

Valid Entry	Usage
natl-intl-prefix	Adds the national or international prefixes from the Locations Parameters screen when the calling party number is appropriate.
tandem-cpn-form	Modifies the calling party number IE in the previously administered format specified for the Tandem Calling Party Number.
no	Does not modify the calling party number.

Related links

[Modify Tandem Calling Number](#) on page 454

Numbering Format

Available only for sip trunk groups.

Valid Entry	Usage
public	Indicates that the number plan according to CCITT Recommendation E. 164 is used.
private	Indicates the Numbering Plan Indicator is PNP.

Outgoing ANI

The digit string sent in place of normal ANI. Overrides the normal ANI if this trunk group is used for an outgoing call with ANI. The ANI is sent exactly as administered, except for the normal truncation to seven digits for Russian ANI. This ANI override works both for calls originated in Communication Manager and calls tandemed through it. Accepts up to 15 digits.

Available only for CO, DIOD, FX, and WATS trunk groups.

Path Replacement Method

Available only if the **Group Type** is ISDN, the **Supplementary Service Protocol** is b or e, and **Supplementary Services with Rerouting** or **DCS with Rerouting** is enabled for the system. Unavailable if **Path Replacement with Retention** is enabled.

Valid Entry	Usage
always	Use any QSIG (SSB) trunk group as the replacement trunk group. A new call is always originated, even when the original trunk group is determined to be the replacement trunk group.
BR (better route)	Route pattern preferences help determine trunk group path replacement. The original trunk group is retained if Path Replacement with Retention is enabled. Path replacement fails (and the original trunk group is retained) if Path Replacement with Retention is disabled.

Related links

[Group Type](#) on page 437

[Supplementary Service Protocol](#) on page 449

[Supplementary Services with Rerouting](#) on page 744

[DCS with Rerouting](#) on page 731

[Path Replacement with Retention](#) on page 463

Path Replacement with Retention

Available only if the **Group Type** is ISDN, the **Supplementary Service Protocol** is b or e, and **Supplementary Services with Rerouting** or **DCS with Rerouting** is enabled for the system.

Valid Entry	Usage
y	Retains the original trunk group.
n	Provides path replacement according to settings for the Path Replacement Method .

Related links

[Supplementary Service Protocol](#) on page 449

[Supplementary Services with Rerouting](#) on page 744

[DCS with Rerouting](#) on page 731

[Group Type](#) on page 437

PBX ID

Valid Entry	Usage
1 to 63 blank	Identifies the remote switch in the network with which the trunk communicates on a DCS signaling link. Available only for trunks that are used for DCS.

Related links

[Used for DCS](#) on page 838

Per Call CPN Blocking Code

A one- to four-digit number used for access, APLT, CO, DIOD, FX, tandem, tie, and WATS trunk groups only. The characters * and # can be used as the first digit.

Per Call CPN Unblocking Code

A one- to four-digit number used for access, APLT, CO, DIOD, FX, tandem, tie, and WATS trunk groups only. The characters * and # can be used as the first digit.

Precedence Incoming

Available only when the trunk group is a DSN termination telephone and the trunk group type is tie.

Valid Entry	Usage
digit	Precedence level for dual-tone multifrequency (DTMF) or tone trunks is received as digits (rotary pulses).
dtmf (a-d)	Precedence level for dual-tone multifrequency (DTMF) or tone trunks is received as DTMF signals (touchtones).

Related links

[DSN Term](#) on page 452

[Group Type](#) on page 437

Precedence Outgoing

Available only when the trunk group is a DSN termination telephone and the trunk group type is tie.

Valid Entry	Usage
digit	Precedence level for dual-tone multifrequency (DTMF) or tone trunks is sent as digits (rotary pulses).
dtmf (a-d)	Precedence level for dual-tone multifrequency (DTMF) or tone trunks is sent as DTMF signals (touchtones).

R2 MFC Signaling

Available only if:

- **Multinational Locations** is enabled for the system, and either of the following:
 - **Outgoing Dial Type** is mf.
 - **Incoming Dial Type** or **Outgoing Dial Type** is rotary, and **Country** code is 15 (Russia).

Valid Entry	Usage
1 to 8	The MFC signaling parameters set used by this trunk group.

Related links

[Country](#) on page 791

[Multinational Locations](#) on page 736

[Incoming Dial Type](#) on page 815

[Outgoing Dial Type](#) on page 816

Receive Analog Incoming Call ID

Enables or disables the collection of incoming call ID information on analog trunks. Fifteen characters of name and number information associated with an incoming call (ICLID, or incoming call line identification information) is stored and displays.

Available for CO, DID, and DIOD trunk groups when **Analog Trunk Incoming Call ID** is enabled. The trunk must be incoming or two-way.

Valid Entry	Usage
Bellcore	Collects ICLID information in the U.S.
NTT	Collects ICLID information in Japan.
disabled	Stops the collection of ICLID information on analog trunks.
V23–Bell	For Telcordia Technologies protocol with V.23 modem tones. Used in Bahrain and similar countries.

Related links

[Analog Trunk Incoming Call ID](#) on page 728

[Direction](#) on page 436

Replace Unavailable Numbers

If enabled, replaces unavailable numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. Applies to BRI/PRI, H.323, and SIP trunks. Also applies to analog trunks if **Analog Trunk Incoming Call ID** is enabled for the system and **Receive Analog Incoming Call ID** for the trunk is set to any value except disabled.

Available only if the group type is isdn or sip.

Related links

[Analog Trunk Incoming Call ID](#) on page 728

[Group Type](#) on page 437

[Receive Analog Incoming Call ID](#) on page 832

Request Category

Indicates if Communication Manager should request a call category from the local telephone company central office (CO).

Available only if the **Country** code is 15 and MF shuttle signaling is enabled.

Related links

[Country](#) on page 791

[Shuttle](#) on page 837

Replace Restricted Numbers

The Replace Restricted Numbers field indicates whether to replace restricted numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. If the **Replace Restricted Numbers** field is enabled, Communication Manager replaces the display irrespective of the service type of the trunks such as BRI, PRI, H.323, and SIP trunks. The **Replace Restricted Numbers** field is available if the **Group Type** is **ISDN** or **SIP**.

Related links

[Group Type](#) on page 437

Seize When Maintenance Busy

Indicates whether this server generates an outgoing seizure when a trunk in this trunk group is maintenance busied and whether the far-end server or switch is administered to do likewise. This supports the Electronic Tandem Network Busyout feature, which is intended to prevent a far-end server or switch from reporting problems with a trunk that has been removed from service. This field does not affect the behavior of the far-end server or switch. It controls the behavior of your server and defines the expected far-end behavior.

This field only affects ports on TN760C (or later release), TN767, and TN464C (or later release) circuit packs. For DIOD trunks using TN464F (or later release) or TN2464 circuit packs, available only for diod trunk groups when the **Trunk Signaling Type** is pulsed, cont, or dis.

Valid Entry	Usage
near-end	Communication Manager generates an outgoing seizure when a trunk is maintenance busied, but the far-end server or switch does not. The seizure is maintained until the maintenance busyout is released.
far-end	The far-end server or switch generates an outgoing seizure when a trunk is maintenance busied, but this server running Communication Manager does not.
both-ends	Both this server running Communication Manager and the far-end server or switch generate an outgoing seizure when a trunk is maintenance busied.

If a server generates an outgoing seizure when a trunk is busied out, the seizure will probably cause alarms at the far-end server or switch, perhaps leading to a far-end maintenance busy out, unless the far-end server or switch is administered to expect this behavior.

If the administered value of this field is either far-end or both-ends, any abnormally long incoming seizure (including failure to drop from a completed call) is the result of a far-end maintenance busy condition. This assumption might be incorrect, since the abnormally long seizure might actually be due to failure of the trunk circuit.

Related links

[Group Type](#) on page 437

[Trunk Signaling Type](#) on page 803

Send Called/Busy/Connected Number

Specifies if the dialed number, whether called (ringing), busy (busy tone), or connected (answered) is sent on incoming or tandemed ISDN calls.

Available only if **QSIG Value-Added** is enabled for the trunk group.

Valid Entry	Usage
y	The dialed number is sent on incoming or tandemed ISDN calls. This field must be enabled in order for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation. If enabled, the Numbering - Public/Unknown Format is accessed to construct the actual number sent, or the Numbering - Private Format is used.
n	Disables the sending of the dialed number on incoming or tandemed ISDN calls.
r	Restricted. The connected number is sent "presentation restricted".

Related links

[Numbering — Public/Unknown Format](#) on page 546

[QSIG Value-Added](#) on page 463

[Numbering-Private Format](#) on page 543

Send Calling Number

Specifies whether the calling party's number is sent on outgoing or tandemed ISDN calls.

* Note:

The Numbering - Public/Unknown Format can override the Send Calling Number administration

Valid Entry	Usage
y	The calling party's number is sent on outgoing or tandemed ISDN calls. If enabled, the Numbering - Public/Unknown Format is accessed to construct the actual number sent, or the Numbering - Private Format is used.

Table continues...

Valid Entry	Usage
n	Disables the sending of the calling party's number on outgoing or tandemed ISDN calls. If disabled, an incoming number is not tandemed out again. This applies to all Supplementary Service Protocols.
r	Restricted. The calling number is sent "presentation restricted". If set to restricted, an incoming number is marked restricted when it is tandemed out again. This applies to all Supplementary Service Protocols.

Related links

[Numbering-Private Format](#) on page 543

[Incoming Number Format](#) on page 93

[Numbering — Public/Unknown Format](#) on page 546

Send Connected Number

Specifies if the connected party's number is sent on incoming or tandemed ISDN calls.

Available only if **QSIG Value-Added** is disabled for the trunk group.

Valid Entry	Usage
y	The connected party's number is sent on outgoing or tandemed ISDN calls. If enabled, the Numbering - Public/Unknown Format is accessed to construct the actual number sent, or the Numbering - Private Format is used. This field must be enabled for the Calling Party Number of an incoming ISDN call to display at the transferred-to station after a QSIG transfer operation.
n	Disables the sending of the connected party's number on outgoing or tandemed ISDN calls. If disabled, an incoming number is not tandemed out again. This applies to all Supplementary Service Protocols.
r	Restricted. The connected number is sent "presentation restricted". If this field is set to r, an incoming number is marked restricted when it is tandemed out again. This applies to all Supplementary Service Protocols.

* Note:

The AT&T Switched Network Protocol does not support restricted displays of connected numbers. Therefore, if you administer the 1a country-protocol/ protocol-version combination for the media module, you should not administer the Send Connected Number as restricted, as this causes display problems. The Numbering - Public/Unknown Format overrides the Send Connected Number administration for any administrable block of extensions.

Related links

[Numbering-Private Format](#) on page 543

[QSIG Value-Added](#) on page 463

[Numbering — Public/Unknown Format](#) on page 546

Send EMU Visitor CPN

Controls which calling party identification (extension of the primary telephone or extension of the visited telephone) is used when a call is made from a visited telephone. There are areas where public network trunks disallow a call if the calling party information is invalid. In this case, there can be instances where the extension of the primary telephone is invalid and the extension of the visited telephone must be used.

Valid Entry	Usage
y	Sends calling party identification information on the extension of the EMU user's telephone.
n	Sends calling party identification information on the primary telephone.

Send Name

Specifies whether the calling, connected, called, or busy party's administered name is sent to the network on outgoing or incoming calls. Available only for isdn or sip trunk groups.

Valid Entry	Usage
y	When the Supplementary Service Protocol is e (DCS with Rerouting), only values of y and n are permitted.
n	When the Supplementary Service Protocol is e (DCS with Rerouting), only values of y and n are permitted.
r (restricted)	The calling/connected name will be sent by Communication Manager, but will be marked "presentation restricted". This value is valid only if the Supplementary Service Protocol is a (national supplementary service), b (for called/busy only), or d for the QSIG Global Networking Supplementary Service Protocol.

* Note:

If name information is not administered for the calling station or the connected, called, or busy station; the system sends the extension number instead of the name.

Related links

[Group Type](#) on page 437

[Supplementary Service Protocol](#) on page 819

Short Holding Threshold

Available only if Automatic Circuit Assurance measurements are taken for this trunk group.

Valid Entry	Usage
0 to 30	The number of times the system will record a short holding call before alerting an attendant to the possibility of a faulty trunk. If 0 is entered, no short holding calls are recorded.

Related links

[ACA Assignment](#) on page 450

Short Holding Time (seconds)

Available only if Automatic Circuit Assurance measurements are taken for this trunk group.

Valid Entry	Usage
0 to 160	The length of time that the system considers as being a short holding time. If 0 is entered, the system does not support short holding calls.

Related links

[ACA Assignment](#) on page 450

Show ANSWERED BY on Display

Available only for isdn pri/bri, isdn h.323, and sip trunk groups.

Valid Entry	Usage
y	The words “ANSWERED BY” display in addition to the connected telephone number on calls over this trunk. This is the default.  Note: Based on display language settings for stations, “ANSWERED BY” is translated into and displayed in the appropriate language.
n	Only the connected telephone number displays. This might be preferred when outgoing calls are over a trunk that might be redirected.

Shuttle

Enables or disables MF shuttle signaling.

Available only for co, fx, or wats trunk groups, with a **Country** code of 15, when the **Outgoing Dial Type** is rotary.

Related links

[Country](#) on page 791

[Group Type](#) on page 437

[Outgoing Dial Type](#) on page 816

Signaling Group

The signaling group number.

Start B Signal

Indicates which B-signal should be used to start a call. The value administered in this field must be coordinated with the local telephone company central office.

Available only when the **Country** code is 15 and MF shuttle signaling is enabled.

Valid Entry	Usage
1	Start calls with signal B1 (first digit).
2	Start calls with signal B2 (next digit).
3	Start calls with signal B3 (previous digit).

Related links

[Country](#) on page 791

[Shuttle](#) on page 837

Start Position

Available only when the **Country** code is 15 and MF shuttle signaling is enabled.

Valid Entry	Usage
1 to 9	Indicates which digit in the digit string is the previously sent digit. The value administered in this field must be coordinated with the local telephone company central office.

Related links

[Country](#) on page 791

[Shuttle](#) on page 837

Suppress # Outpulsing

Indicates whether or not to suppress the final “#” in cases where the system would normally outpulse it. Used if end-to-end signaling begins with (and includes) “#”. This field should be enabled when the local telephone company central office or any other facility treats “#” as an error.

Time (sec) to Drop Call on No Answer

Available only for co or diod trunk groups when the **Outgoing Dial Type** is mf, or for co, diod, fx, or wats trunk groups when the **Country** code is 15.

Valid Entry	Usage
0 to 1200	The duration in seconds that Communication Manager should wait for outgoing calls to be answered. If the call is unanswered in the specified number of seconds, the call drops. If 0 is entered, the timer is not set and no calls drop.

Related links

[Country](#) on page 791

[Group Type](#) on page 437

[Outgoing Dial Type](#) on page 816

Used for DCS

If enabled, this trunk group sends and receive messages on a DCS signaling link.

 Note:

This field cannot be enabled if the trunk group number is greater than 255 or if the Trunk Access code is more than 3 digits long.

Related links

[TSC Supplementary Service Protocol](#) on page 626

[DCS with Rerouting](#) on page 731

[ISDN-BRI Trunks](#) on page 734

[ISDN-PRI](#) on page 734

[Group Type](#) on page 614

Used Only for Paging

Indicates whether or not this trunk is used only for paging. Default is disabled.

Related links

[Group Type](#) on page 437

UII Treatment

UII treatment field specifies whether the User-to-User Information (UII) is shared.

Valid Entry	Usage
shared	The trunk is connected to the server.
service-provider	When forwarding the other data as MSI, Communication Manager forwards the ASAI user data in a non-shared codeset 0 UII.

Voice Paging Timeout (sec)

Available only if the trunk is used only for paging.

Valid Entry	Usage
10 to 6000	The number of seconds before a paged trunk call drops. Default is 10.

Related links

[Used Only for Paging](#) on page 839

Wideband Support

 Note:

This feature is not supported on the DS1 interfaces on branch gateways (G700).

Enables or disables wideband switching on this trunk group.

Available only if **Wideband Switching** is enabled for the system.

 Note:

Wideband trunk calls are treated as a single trunk call when Automatic Circuit Assurance (ACA) measurements are taken. This way, if an ACA referral call is generated (for short or long holding time), the wideband call only triggers a single referral call using the lowest B-channel trunk member associated with the wideband channel.

Related links

[Wideband Switching](#) on page 738

Administrable Timers

The system might not display this screen for all trunk group types.

 Caution:

Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

change trunk-group 1

Page 4 of 22

ADMINISTRABLE TIMERS

Send Incoming/Outgoing Disconnect Timers to TN465 Ports? n

Incoming Glare Guard(msec): 1500

Outgoing Dial Guard(msec): 1600

Outgoing Glare Guard(msec): 1500

Ringing Monitor(msec): 5200

Incoming Seizure(msec): 500

Outgoing End of Dial(sec): 10

Outgoing Seizure Response(sec): 5

Programmed Dial Pause(msec): 1500

Flash Length(msec): 540

END TO END SIGNALING

Tone (msec): 350 Pause (msec): 150

OUTPULSING INFORMATION

PPS: 10 Make (msec): 40 Break (msec): 60 PPH? n

Answer Send (msec)

Available only for incoming or two-way trunk groups when the **Trunk Type** is dis. Only TN2140 and TN2199 ports receive this timer.

Valid Entry	Usage
10 to 2550 in increments of 10	The duration of the answer signal pulse.

Related links

[Direction](#) on page 436

[Trunk Type](#) on page 821

Busy Tone Disconnect

If enabled, Communication Manager recognizes a busy tone signal as a disconnect on this trunk group.

Available only if **Enable Busy Tone Disconnect for Analog loop-start Trunks** is enabled for the system country options.

Related links

[Enable Busy Tone Disconnect for Analog Loop-start Trunks](#) on page 722

Cama Outgoing Dial Guard (msec)

Available only for cama trunk groups (the trunk group type used for emergency 911 service).

Valid Entry	Usage
25 to 6375 in increments of 25	Minimum interval between the seizure acknowledgment on the receiving server or switch and the outpulsing of digits by this server.

Related links

[Group Type](#) on page 437

Cama Wink Start Time (msec)

Available only for cama trunk groups.

Valid Entry	Usage
20 to 5100 in increments of 20	The duration (the wait-for-wink-to-end time) for a wink-start CAMA trunk. The wink must begin before the Outgoing Seizure Response timer expires.

Related links

[Group Type](#) on page 437

Disconnect Signal Error (sec)

The system displays this field only for ground-start trunk groups.

Valid entry	Usage
1 to 255 in increments of 1	The maximum interval that Communication Manager waits to receive a disconnect signal from the far-end after the local party, such as a telephone or tie trunk, goes on-hook. If the timer expires, the disconnect signal fails.

Flash Length (msec)

Valid Entry	Usage
10 to 2550 in increments of 10	The duration of a flash signal generated toward the local telephone company central office. This timer is sent to TN436B, TN459B, TN464C (or later), TN465B (or later), TN753 (if Country code is 23), TN2146, TN2147, TN2184, and TN2199 circuit boards.

Glare

Available only for two-way or outgoing trunk groups when the **Trunk Type** is cont. Only TN2140 ports receive this timer.

Valid Entry	Usage
40 to 100 in increments of 10	The minimum acceptable interval between the moment your server running Communication Manager sends an outgoing seizure and the moment it receives a seizure acknowledgment. If acknowledgment is received before the timer expires, glare is assumed.

Related links

[Direction](#) on page 436

[Trunk Type](#) on page 821

Incoming Dial Guard (msec)

Valid Entry	Usage
10 to 2550 in increments of 10	The minimum acceptable interval between the detection of an incoming seizure and the acceptance of the first digit. Communication Manager does not accept digits before this timer expires. This timer is never sent to TN429 ports.

Incoming Disconnect (msec)

Available only for incoming or two-way trunk groups when the **Trunk Type** is blank or cont.

Valid Entry	Usage
50 to 2550 in increments of 10	The minimum valid duration of a disconnect signal for an incoming call. Communication Manager does not recognize shorter disconnect signals. This field cannot be blank. For Brazil pulsed E&M signaling, use 600.

Related links

[Direction](#) on page 436

[Trunk Type](#) on page 821

Incoming Disconnect Send (msec)

Available only for incoming or two-way trunk groups when the **Trunk Type** is dis. Only TN2140 ports receive this timer.

Valid Entry	Usage
500 to 1200 in increments of 100	The duration of the backward release signal the server running Communication Manager sends at the end of an incoming call.

Related links

[Direction](#) on page 436

[Trunk Type](#) on page 821

Incoming Glare Guard (msec)

Available only for two-way trunk groups.

Valid Entry	Usage
100 to 25500 in increments of 100	The minimum interval that must elapse between a trunk's release from an incoming call and its seizure for an outgoing call. This delay gives the far-end time to release all equipment after the trunk is released. This field cannot be blank.

Related links

[Direction](#) on page 436

Incoming Partial Dial (sec)

Available only if the **Incoming Dial Type** is rotary. This timer is never sent to TN429 ports.

Valid Entry	Usage
5 to 255 in increments of 1	The maximum time allowed between incoming rotary digits.

Related links

[Incoming Dial Type](#) on page 815

Incoming Seizure (msec)

Available only for incoming or two-way trunk groups, and, when applicable, the **Trunk Type** is cont. Only TN429, TN438 (any release), TN 447, TN464C (or later), TN465 (any release), TN767, TN2138, TN2140, TN2147, TN2184, and TN2199 ports receive this timer. For DID trunks, only TN2199 and TN429D (or later) receive this timer.

Valid Entry	Usage
20 to 2550 in increments of 10	The duration of the shortest incoming seizure signal the server running Communication Manager can recognize. For ICLID, set this field to 120. The field cannot be blank.

Related links

[Direction](#) on page 436

[Trunk Type](#) on page 821

Normal Outgoing Seize Send (msec)

Available only for two-way or outgoing trunk groups when the **Trunk Type** is dis. Only TN2140 ports receive this timer.

Valid Entry	Usage
10 to 990 in increments of 10	The duration of the signal the server running Communication Manager sends for an outgoing seizure.

Related links

[Direction](#) on page 436

[Trunk Type](#) on page 821

Outgoing Dial Guard (msec)

Valid Entry	Usage
100 to 25500 in increments of 100	The minimum interval between seizure acknowledgment of a trunk and the outpulsing of digits. For trunks that do not provide seizure acknowledgment, the timer specifies the minimum time between seizure and the outpulsing of digits. Any digit the caller dials after lifting the receiver, but before the timer expires, is not outpulsed until the timer expires. This field cannot be blank.

Outgoing Disconnect (msec)

Valid Entry	Usage
50 to 2550 in increments of 10	The minimum valid duration of a disconnect signal for an outgoing call. Communication Manager does not recognize shorter disconnect signals. This timer begins timing when a disconnect signal is detected on an outgoing call and resets when the signal is no longer detected. If the timer expires, the trunk drops. This field cannot be blank. For Brazil pulsed E&M signaling, use 600.

Outgoing Disconnect Send (msec)

Available only for two-way or outgoing trunk groups when the **Trunk Type** is dis. Only TN2140 ports receive this timer.

Valid Entry	Usage
100 to 9900 in increments of 100	The duration of the forward release signal the server running Communication Manager sends at the end of outgoing calls.

Related links

[Direction](#) on page 436

[Trunk Type](#) on page 821

Outgoing End of Dial (sec)

Available for outgoing or two-way trunk groups when the network does not provide answer supervision.

Valid Entry	Usage
1 to 254 in increments of 1	The maximum time Communication Manager waits to receive answer supervision for outgoing calls on the ports controlled by firmware timers. Controls firmware answer supervision timers on media module that have them. During a cut-through operation, timing begins after Communication Manager sends each outgoing digit and ceases when answer supervision is received. If the timer expires, Communication Manager acts as if it has received answer supervision. On senderized operation, the timer begins after the switch sends the last digit collected. The timer ceases when answer supervision is received. If the timer expires, Communication Manager acts as if it has received answer supervision. For Brazil pulsed E&M signaling, use 40.

Related links

[Direction](#) on page 436

[Receive Answer Supervision](#) on page 558

Outgoing Glare Guard (msec)

Available only for outgoing and two-way trunk groups.

Valid Entry	Usage
100 to 25500 in increments of 100	The minimum interval that must elapse between a trunk's release from an outgoing call and its seizure for another outgoing call. This delay gives the far-end time to release all equipment after the outgoing trunk is released. This field cannot be blank.

Related links

[Direction](#) on page 436

Outgoing Last Digit (sec)

Available only for two-way or outgoing trunk groups when the **Trunk Type** is dis or cont. Only TN497 and TN2140 ports receive this timer.

Valid Entry	Usage
1 to 40 in increments of 1	The maximum time that Communication Manager waits for the next digit dialed. After the timer expires, no more digits are accepted by the circuit pack.

Related links

[Direction](#) on page 436

[Trunk Type](#) on page 821

Outgoing Rotary Dial Interdigit (msec)

Available only if:

- The trunk **Group Type** is access, aplt, co, diod, dmi-bos, fx, rlt, tandem, or wats, and the **Outgoing Dial Type** is rotary.
- The trunk **Group Type** is tie, the **Trunk Type** is blank, cont, or dis, and the **Outgoing Dial Type** is rotary.
- The trunk **Group Type** is tie, and the **Trunk Type** is tge, tgi, or tru.

Valid Entry	Usage
150 to 2550 in increments of 10	The minimum time between outpulsed digits on outgoing rotary trunks.

Related links

[Group Type](#) on page 437

[Outgoing Dial Type](#) on page 816

[Trunk Type](#) on page 821

Outgoing Seizure Response (sec)

Valid Entry	Usage
1 to 255 in increments of 1	The maximum interval that Communication Manager waits after sending a seizure signal to receive seizure acknowledgment from the far-end. If the acknowledgment is not received within this time, a seizure failure response is uplinked. For Brazil pulsed E&M signaling, use 255.

Programmed Dial Pause (msec)

Valid Entry	Usage
100 to 25500 in increments of 100	The exact duration of the pause used during abbreviated dialing, ARS outpulsing, and terminal dialing operations. This timer is administrable for all outgoing and two-way trunk groups. All central office (CO) media module that accept administrable timers accept this timer.

Release Ack Send (msec)

Available only for incoming and two-way trunk groups when the **Trunk Type** is dis. Only TN2140 ports receive this timer.

Valid Entry	Usage
500 to 1200 in increments of 100	The duration of the signal the server running Communication Manager sends for a forward release acknowledgment. After the server running Communication Manager receives a forward release signal, it must send a forward release acknowledgment signal.

Related links

[Direction](#) on page 436

[Trunk Type](#) on page 821

Ringing Monitor (msec)

The minimum time that Communication Manager requires to determine if a trunk disconnects.

You cannot leave the field blank.

Valid entry	Usage
200 to 51000 in increments of 200	If the ringing signal disappears for a duration longer than the time specified in this field, you can be sure that the call is disconnected.

Seize Ack Delay (msec)

Available only for incoming or two-way trunk groups if the **Trunk Type** is dis. Only TN2140 ports receive this timer.

Valid Entry	Usage
40 to 120 in increments of 10	The maximum interval the server running Communication Manager waits after receipt of an incoming seizure to send seizure acknowledgment.

Related links

[Direction](#) on page 436

[Trunk Type](#) on page 821

Seize Ack Send (msec)

Available only for incoming or two-way trunk groups if the **Trunk Type** is dis. Only TN2140 ports receive this timer.

Valid Entry	Usage
10 to 990 in increments of 10	The duration of the seizure acknowledgment signal the server running Communication Manager sends in response to an incoming seizure.

Related links[Direction](#) on page 436[Trunk Type](#) on page 821

Send Incoming/Outgoing Disconnect Timers to TN465 Ports

If enabled, the incoming disconnect and outgoing disconnect timer values are sent to the trunk group ports that are on a TN465 board. Available only for a co, fx, or wats trunk group.

END TO END SIGNALING

Pause (msec)

Available only if the **Trunk Type** is blank.

Valid Entry	Usage
20 to 2550 in increments of 10	The interval (pause) between DTMF tones sent from a hybrid telephone. All CO, DIOD, and tie media module that accept administrable timers accept this timer.

Related links[Trunk Type](#) on page 821

Tone (msec)

Available only if the **Trunk Type** is blank.

Valid Entry	Usage
20 to 2550 in increments of 10	The duration of the DTMF tone sent when a button on a hybrid telephone is pressed. All CO, DIOD, and tie media module that accept administrable timers accept this timer.

Related links[Trunk Type](#) on page 821

OUTPULSING INFORMATION

Break (msec)

The duration of the break interval (the pulse duration) while the system is outpulsing digits using dial pulse signaling. The field cannot be blank.

Valid Entry	Usage
20 to 80 in increments of 5	If the PPS (pulses per second) value is 10, the sum of the Make (msec) and Break (msec) values must equal 100.
10 to 40 in increments of 5	If the PPS (pulses per second) value is 20, the sum of the Make (msec) and Break (msec) values must equal 50.

Related links

[Make \(msec\)](#) on page 849

[PPS](#) on page 849

Make (msec)

The duration of the make interval (the pause between pulses) while the system is outputting digits using dial pulse signaling. The field cannot be blank.

Valid Entry	Usage
20 to 80 in increments of 5	If the PPS (pulses per second) value is 10, the sum of the Make (msec) and Break (msec) values must equal 100.
10 to 40 in increments of 5	If the PPS (pulses per second) value is 20, the sum of the Make (msec) and Break (msec) values must equal 50.

Related links

[Break \(msec\)](#) on page 848

[PPS](#) on page 849

PPM

Determines if Periodical Pulse Metering (PPM) pulses should be collected from the public network to determine call cost. For CO, DIOD, FX, PCOL, and WATS trunks. Available only for outgoing or two-way trunk groups.

Related links

[Direction](#) on page 436

[Direction](#) on page 436

PPS

Available only for cama trunk groups.

Valid Entry	Usage
10 20	The rate (pulses per second) at which outgoing rotary pulses are sent over this trunk group.

Related links

[Break \(msec\)](#) on page 848

[Make \(msec\)](#) on page 849

Far-End Test No

The access number dialed to reach the terminating test line (TTL). Accepts up to 16 digits.

Trunk Contact

The name or telephone number of someone from the trunk vendor who can be contacted if there are problems with the trunks. Accepts up to 25 alphanumeric characters.

Trunk Length

This field is not required. Since noise on a trunk increases with the length of the trunk, however, this information might be useful.

Valid Entry	Usage
0 to 4 k	The length of the trunk group in kilometers.
0 to 4 m	The length of the trunk group in miles.

TTL Vendor

The name of the vendor supplying the terminating test line (TTL). Accepts up to 22 alphanumeric characters.

Trunk Vendor

The name of the vendor providing service over this trunk group that needs to be notified in the event of problems with the trunks in this trunk group. Accepts up to 22 alphanumeric characters.

TTL Contact

The name or telephone number of someone from the TTL vendor who can be contacted in the event of problems with the terminating test line. Accepts up to 25 alphanumeric characters.

TTL Type

Type of terminating test line (TTL) selected for testing trunks. The TTL type determines what ATMS tests can be completed and which threshold values need to be administered.

Valid Entry	Usage
105-w-rl	105 with return loss. Full range of 18 measurements or some defaults for return loss used (56A).
105-wo-rl	105 without return loss. Cannot return default values for far-end return loss.
high-lts	High-level tone source. Sends a fixed sequence of tones at 0 dBm.
low-lts	Low-level tone source. Sends a fixed sequence of tones at -16dBm.
100	100 type. Up to 5 measurements that sends a 1004 Hz tone, then a quiet termination.
102	102 type. One measurement that sends a 1004 Hz tone.

MARGINAL / UNACCEPTABLE

Allow ATMS Busyout, Error Logging and Alarming

Enables or disables ATMS error logging and alarming.

Marginal Threshold --Dev - 404 Hz Loss

Valid Entry	Usage
0 to 9	The maximum negative deviation of measured loss at 404 Hz from the 1004 Hz test tone noise level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold -+Dev - 404 Hz Loss

Valid Entry	Usage
0 to 9	The maximum positive deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold --Dev - 2804 Hz

Valid Entry	Usage
0 to 9	The maximum negative deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold -+Dev - 2804 Hz

Valid Entry	Usage
0 to 9	The maximum positive deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as out of tolerance. Smaller dB values are more restrictive.

Marginal Threshold - Max - 1004 Hz Loss

Valid Entry	Usage
0 to 21	The maximum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as out of tolerance. A smaller dB value is more restrictive.

Marginal Threshold - Maximum C Message Noise

Valid Entry	Usage
15 to 55	The maximum C-message noise telephone as measured within the voice band frequency range (500 to 2500 Hz) allowed before reporting a trunk as out of tolerance. Smaller values are more restrictive.

Marginal Threshold - Maximum C Notched Noise

Valid Entry	Usage
34 to 74	The maximum C-notched signal dependent noise interference in dBmC allowed before reporting a trunk as out of tolerance. Smaller values are more restrictive.

Marginal Threshold - Min -1004 Hz Loss

Valid Entry	Usage
-2 to 21	The minimum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as out of tolerance. A larger dB value is more restrictive.

Marginal Threshold - Minimum ERL

Valid Entry	Usage
0 to 40	The minimum low-frequency echo return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Marginal Threshold - Minimum SRL-HI

Valid Entry	Usage
0 to 40	The minimum high-frequency signaling return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Marginal Threshold - Minimum SRL-LO

Valid Entry	Usage
0 to 40	The minimum low-frequency signaling return loss in dB allowed before reporting a trunk as out of tolerance. Larger values are more restrictive.

Maximum Percentage of Trunks Which Can Be Removed From Service by ATMS

Available only if **Allow ATMS Busyout**, **Error Logging**, and **Alarming** are enabled.

Valid Entry	Usage
0, 25, 50, 75, 100	The highest percentage of trunks from the trunk group that can be removed from service at one time because of unacceptable transmission measurement results.

Related links

[Allow ATMS Busyout, Error Logging and Alarming](#) on page 851

Unacceptable Threshold --Dev - 404 Hz

Valid Entry	Usage
0 to 9	The maximum negative deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold -+Dev - 404 Hz

Valid Entry	Usage
0 to 9	The maximum positive deviation of measured loss at 404 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold --Dev - 2804 Hz

Valid Entry	Usage
0 to 9	The maximum negative deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold -+Dev - 2804 Hz

Valid Entry	Usage
0 to 9	The maximum positive deviation of measured loss at 2804 Hz from the 1004 Hz test tone loss level (in dB) allowed before reporting a trunk as unacceptable. Smaller dB values are more restrictive.

Unacceptable Threshold - Max - 1004 Hz Loss

Valid Entry	Usage
0 to 21	The maximum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as unacceptable. A smaller dB value is more restrictive.

Unacceptable Threshold - Maximum C Message Noise

Valid Entry	Usage
15 to 55	The maximum C-message noise interference in dBmC above reference noise terminating on a telephone as measured within the voice band frequency range (500 to 2500 Hz) allowed before reporting a trunk as unacceptable. Smaller values are more restrictive.

Unacceptable Threshold - Maximum C Notched Noise

Valid Entry	Usage
34 to 74	The maximum C-notched signal dependent noise interference in dBmC allowed before reporting a trunk as unacceptable. Smaller values are more restrictive.

Unacceptable Threshold - Min - 1004 Hz Loss

Valid Entry	Usage
-2 to 21	The minimum signal loss allowed for a 1004 Hz test tone (in dB) before a trunk is reported as unacceptable. A larger dB value is more restrictive.

Unacceptable Threshold - Minimum ERL

Valid Entry	Usage
0 to 40	The minimum low-frequency echo return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Unacceptable Threshold - Minimum SRL-HI

Valid Entry	Usage
0 to 40	The minimum high-frequency signaling return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Unacceptable Threshold - Minimum SRL-LO

Valid Entry	Usage
0 to 40	The minimum low-frequency signaling return loss in dB allowed before reporting a trunk as unacceptable. Larger values are more restrictive.

Trunk Group: Group Member Assignments

For SIP trunks, the group member-assignment pages are not individually administered. The system automatically populates and displays these fields based on the number of members of SIP trunk groups previously specified.

change trunk-group 1 Page 6 of 22

TRUNK GROUP

Administered Members (min/max): 1/10
Total Administered Members: 10

GROUP MEMBER ASSIGNMENTS

	Port	Code	Sfx	Name	Night	Mode	Type	Ans Delay
1:	_____			_____	_____	?	?	_____
2:	_____			_____	_____	?	?	_____
3:	_____			_____	_____	?	?	_____
4:	_____			_____	_____	?	?	_____
5:	_____			_____	_____	?	?	_____
6:	_____			_____	_____	?	?	_____
7:	_____			_____	_____	?	?	_____
8:	_____			_____	_____	?	?	_____
9:	_____			_____	_____	?	?	_____
10:	_____			_____	_____	?	?	_____
11:	_____			_____	_____			_____
12:	_____			_____	_____			_____
13:	_____			_____	_____			_____
14:	_____			_____	_____			_____
15:	_____			_____	_____			_____

Administered Members (min/max)

The minimum and maximum member numbers that have been administered for this trunk group.

Ans Delay

Caution:

Customers should not attempt to administer this field. Go to the Avaya Support website at <http://support.avaya.com> for current documentation, product notices, knowledge articles on administering this field.

Valid Entry	Usage
20 to 5100 in increments of 20	The length of time the server running Communication Manager waits before it sends answer supervision for incoming calls on tie trunks.
blank	Same as setting the field to zero.

Code

The type of media module physically installed or logically administered at the location to which this member is assigned. If no media module is installed or administered at the port address, the field is blank.

Name

The name of the trunk group member. The name should identify the trunk unambiguously. Accepts up to 10 characters.

Example

- The telephone number assigned to incoming trunks
- The Trunk Circuit Identification number assigned by the service provider

Night

The night service destination for this trunk group member if different from the night service destination administered for the trunk group. Incoming calls are routed to this destination when the system is placed in night service mode.

Valid Entry	Usage
<i>A valid extension</i>	The extension of the night destination for the trunk.
attd	Calls go to the attendant when night service is active.
blank	Not administered

Related links

[Night Service](#) on page 797

Port

Valid Entry	Usage
1 to 999	Gateway
V1 to V9	Module
01 to 31	Circuit

Sfx

The model suffix for the type of media module physically installed at the location to which this member is assigned. If no media module is installed at the port address, the field is blank.

Total Administered Members

The total number of members administered in the trunk group.

Trunk Group: Protocol Variations

Available only for sip trunk groups.

change trunk-group 2 Page 4 of 4

PROTOCOL VARIATIONS

Mark Users as Phone? n
 Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
 Send Transferring Party Information? n
 Network Call Redirection? n

 Send Diversion Header? n
 Support Request History? y
 Telephone Event Payload Type: ____

 Convert 180 to 183 for Early Media? n
 Always Use re-INVITE for Display Updates? n
 Resend Display UPDATE Once on Receipt of 481 Response? n
 Identity for Calling Party Display: P-Asserted-Identity
 Block Sending Calling Party Location in INVITE? n
 Accept Redirect to Blank User Destination? n
 Enable Q-SIP? n
 Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
 Request URI Contents: may-have-extra-digits

Related links

[Group Type](#) on page 437

Accept Redirect to Blank User Destination

The following table represents the way in which this field works.

Valid entry	User destination in the Refer-To domain of the SIP request	Result
y	present	Communication Manager places the call to the user destination.

Table continues...

Valid entry	User destination in the Refer-To domain of the SIP request	Result
y	missing	Communication Manager places the call over the signaling group specified in the Proxy Sel Rte Pat field of the Locations screen.
n	present	Communication Manager places the call to the user destination.
n	missing	Communication Manager rejects the SIP request.

Always Use re-INVITE for Display Updates

The system displays this field when the **Group Type** field is SIP.

Valid entry	Usage
y	In SIP messages, Communication Manager sends the re-invite message to display update.
n	In SIP messages, Communication Manager sends the update message to display update. The default value is n.

Resend Display UPDATE Once on Receipt of 481 Response

The value in the **Resend Display UPDATE Once on Receipt of 481 Response** field determines whether Communication Manager should resend display update message on receiving 481 response to Session Manager, instead of dropping the call immediately.

When Communication Manager receives 481 as a response for display update message from Session Manager (only when the display update message is sent after the first non 100 response), then Communication Manager resends display update message to Session Manager.

Note:

The display update message must be after the first 18x response. If not, the display update message is not resent.

If Session Manager responds with 481 again, then Communication Manager drops the call. This field does not apply to all other update messages except display update message.

Valid Entry	Usage
y	If Communication Manager gets a 481 response for a display update message from Session Manager, Communication Manager resends the display update message once.
n	The Communication Manager does not re-send any update message (even if it is 481 response). By default, the value is n.

Block Sending Calling Party Location in INVITE

Based on the , Communication field value, Manager selects the IP address that must be included in the Via header in an outgoing call. The system displays this field if the value of the **Group Type** field in the trunk group screen is SIP.

Valid Entry	Usage
y	Communication Manager does not send the IP address of the caller in the INVITE.
n	Communication Manager sends an IP address that identifies the location of the caller. This IP address is included in the lower-most Via header in the INVITE. Based on the type of endpoint of the caller, Communication Manager determines the IP address to be sent. • For calls made from H.323 endpoints, Communication Manager sends the IP address of the endpoint. • For calls made from TDM endpoints or TDM trunks, Communication Manager sends the IP address of the media processor used in the call. This value is the default value.

Build Refer-To URI of REFER From Contact For NCR

The system displays this field only for SIP trunk groups for which **Network Call Redirection** is enabled. Communication Manager uses the REFER message during a call transfer.

Valid entry	Usage
y	Communication Manager uses the address in the Contact header to create the URI of the Refer-To header in a REFER message.
n	Communication Manager uses the address in the From header to create the URI of the Refer-To header in a REFER message.

Convert 180 to 183 for Early Media

It is used for early media and direct media cut-through. When SDP answer is returned by Communication Manager in 18x messages, some entities have a problem receiving the SDP in a 180 Ringing message. If the **Convert 180 to 183 for Early Media** field is set to y, Communication Manager puts the SDP into a 183 Session Progress message. The default value is n.

Enable Q-SIP

The system displays Enable Q-SIP, only when the **Group Type** field is sip.

Valid Entry	Usage
y	Enables the QSIG over SIP (Q-SIP) feature for the trunk group. If trunk members are already assigned to this trunk group, you must not change the value of Enable Q-SIP field. If you change the value of Enable Q-SIP field, the system displays an error message and prompts you to remove all assigned members before enabling Q-SIP.
n	The QSIG over SIP feature for the trunk group is disabled. By default, the value is n.

Identity for Calling Party Display

The system displays this field, when the **Group Type** field is sip.

This field determines which header to retrieve display information for the calling party when both the From and P-Asserted-Identity headers are available.

Valid Entry	Usage
P-Asserted-Identity	If a call is terminating at SIP, H.323, or DCP stations, or at outgoing H.323 or SIP trunk, Communication Manager displays the name and Calling Party number from the PAI header instead of the FROM header. This is the default value.
From	If a call is terminating at SIP, H.323, or DCP stations, or at outgoing H.323 or SIP trunk, Communication Manager displays the name and Calling Party number from the FROM header instead of the PAI header.

Interworking of ISDN Clearing with In-Band Tones

The setting of this field is for call-clearing of the tandem calls in which the incoming trunk to Communication Manager is a SIP trunk and the outgoing trunk from Communication Manager is an ISDN trunk. The far end at the ISDN PRI trunk ends the call.

If you select the **keep-channel-active** option, Communication Manager keeps the call active so that the caller can hear an in-band tone or announcement played over the ISDN trunk.

If you select the **drop-with-sip-error** option, Communication Manager detects the cause of call clearing, maps ISDN Cause Value to a SIP error response message, and sends the response message to the caller. With the **drop-with-sip-error** option, an adjunct placing outgoing calls is informed that the call has ended and the user can disconnect the call immediately instead of waiting for the ISDN network to end the call after the tone or announcement is complete.

Note:

An in-band tone is an announcement or a tone that provides call-clearing information.

Valid entry	Usage
keep-channel-active	Communication Manager sends a SIP 183 Progress message to the caller. The ISDN network plays an in-band tone that the caller hears. The default value is keep-channel-active.

Table continues...

Valid entry	Usage
drop-with-sip-error	Communication Manager sends a SIP error response message, which specifies the cause of call-clearing to the caller. The called party plays an in-band tone. The caller does not hear the in-band tone or announcement.

Mark Users as Phone

Enables or disables the encoding of URIs in call control signaling messages originated at the gateway with the “user=phone” parameter. No subscription messages are encoded with the “user=phone” parameter, even when the field is set to y. Default is n.

Note:

Do not change the default of n for this field unless you are sure that every recipient of SIP calls using this trunk can accept and properly interpret the optional “user=phone” parameter. Enterprise users without support for “user=phone” in their SIP endpoints will experience adverse effects, including rejected calls.

Network Call Redirection

If enabled, Network Call Redirection (NCR) service is signaled over this trunk group. NCR only works on trunk groups connected to Service Providers that support NCR.

Prepend '+' to Calling/Alerting/Diverting/Connected Number

The system displays this field only for SIP trunk groups.

If you set this field to y, when a call is routed over a trunk group, Communication Manager inserts a leading plus (+) sign, if absent, in outgoing calling, alerting, diverting, or connected numbers. The default value is n.

To prevent Communication Manager from inserting a leading plus (+) sign in the calling number when a call is routed from Communication Manager to Session Manager, set the **Numbering Format** field to private and the **Prepend '+' to Calling/Alerting/Diverting/Connected Number** field to n.

Related links

[Group Type](#) on page 437

QSIG Reference Trunk Group

The system displays this when the **Group Type** field is sip, and the **Enable Q-SIP** is set to y.

Valid Entry	Usage
1 to 2000	Assigns a number for the QSIG trunk group. If trunk members are already assigned to this trunk group, do not change the value of the QSIG Reference Trunk Group field. If you change the value of this field, the system displays an error message and prompts you to remove all assigned members before enabling Q-SIP. Applicable for Avaya supported servers. For the Avaya supported servers, see <i>Avaya Aura® Communication Manager Hardware Description and Reference</i> .
1 to 99	(For Avaya Solutions Platform S8300 Server) Assigns a number for the QSIG trunk group. If trunk members are already assigned to this trunk group, do not change the value of the QSIG Reference Trunk Group field. If you change the value of this field, the system displays an error message and prompts you to remove all assigned members before enabling Q-SIP.
blank	No QSIG trunk group is assigned. By default, the value is blank.

Request URI Contents

Use this field to configure Communication Manager to allow or restrict the called-party digits in the Request URI of the SIP INVITE message, REFER message, or 3xx redirect response for INVITE message.

Valid entry	Usage
may-have-extra-digits	Communication Manager routes an incoming Request URI without considering the total number of digits. To route the call, Communication Manager might use an entry that matches fewer digits than the total number in the Request URI.
called number-only	Communication Manager considers the total number of digits when routing an incoming Request URI. The routing fails if a match that incorporates all digits is unavailable.

Send Diversion Header

This field is available only when the **Service Type** field is set to public-ntwrk. If you enable the **Send Diversion Header** field, the SIP diversion header is sent on the public SIP trunks. This header allows service providers to use the ten-digit number of the forwarding party for functions such as billing. Customers can administratively enable or disable the use of the SIP diversion header on any public SIP trunk. The default value is n.

Related links

[Service Type](#) on page 441

Send Transferring Party Information

Enables or disables sending the transferring party information on a transferred call. Default is disabled.

Support Request History

You can activate the **Support Request History** field for a SIP trunk to capture the request location that was sent to a proxy. This information is used to send diversion information and notify the origin of the call to the far end equipment. If this field is disabled, History-Info header is not included in any requests.

 Note:

This feature does not apply to Off PBX SIP station trunk groups for which the Communication Manager never suppresses the History-Info header.

Telephone Event Payload Type

Valid Entry	Usage
96 to 127 blank	The default payload type offered by Communication Manager for SIP trunks. The payload type number encoding for originating (offering) the RFC 2833 RTP “telephone-event” payload format is based on the administered number from this field. This value is used for Communication Manager originations (outgoing offers). Default is 127.

Chapter 139: Uniform Dial Plan Table

The **UDP** provides a common dial plan length — or a combination of extension lengths — that can be shared among a group of Avaya servers. Additionally, **UDP** can be used alone to provide uniform dialing between two or more private switching systems without ETN, DCS, or Main/Satellite/Tributary configurations.

Available only if **Uniform Dialing Plan** is enabled for the system.

Example command: `change uniform-dialplan  $n$` , where n is the uniform dial plan length.

[illegible]

Related links

[Uniform Dialing Plan](#) on page 738

Conv

Enables or disables additional digit conversion.

Del

Valid Entry	Usage
0 to 9	The number of digits to delete before routing the call. This number must be less than or equal to the number of user dialed digits the system collects to match to the matching pattern.

Related links

[Len](#) on page 865

Insert Digits

Use this field to specify the digits or the number of digits of the location prefix that Communication Manager inserts in the dialed string before routing a call.

Valid entry	Usage
0 to 9	The digits that replace the deleted portion of the dialed number. The deleted portion is the number of digits that Communication Manager deletes specified in the Del field. You can enter up to 10 digits. Leave the field blank to delete the digits specified in the Del field.
Lx	The variable x is a number from 1 to 11 and represents the number of leading digits of the administered location prefix. Communication Manager prepends the digits of the location prefix to the dialed string. The value of x must be less than the value of the Prefix field on the Location screen.

Related links

[Locations](#) on page 482

Len

Valid Entry	Usage
1 to 18	The number of user-dialed digits the system collects to match to this Matching Pattern. This number must be greater than or equal to the Matching Pattern value. The value 2 can be used only when Insert Digits contains an Lx value.

Related links

[Insert Digits](#) on page 865

[Matching Pattern](#) on page 866

Matching Pattern

Communication Manager matches the value of the **Matching Pattern** field to the dialed numbers.

Valid entries	Usage
0 to 9	Communication Manager matches the value with the dialed numbers. You can enter up to 16 digits.
blank	Leave the field blank if you do not want to specify a pattern.

Net

Valid Entry	Usage
aar ars enp ext	The server or switch network used to analyze the converted number. The converted digit-string is routed either as an extension number or through its converted AAR address, its converted ARS address, or its ENP node number. For enp, the ENP node number must be administered, digit conversion must be disabled, and the Insert Digits must not be administered.

Related links

[Conv](#) on page 864

[Insert Digits](#) on page 865

[Node Num](#) on page 866

Node Num

Valid Entry	Usage
1 to 999	The ENP (Extension Number Portability) Node Number.

Percent Full

Displays the percentage (0 to 100) of the memory resources allocated for the uniform dial plan data that are currently being used. For details on the system memory resources, see *Avaya Aura® Communication Manager System Capacities Table*.

Chapter 140: User Profile

This screen is described in *Maintenance Commands for Avaya Aura® Communication Manager, Branch Gateways and Servers*. For more information on administering user profiles and logins, see AAA Services in *Avaya Aura® Communication Manager Feature Description and Implementation*.

Chapter 141: Variables for Vectors

This screen creates variables and defines the necessary parameters for each variable type. Specifies the variable type, the name to use for the variable, the size of the variable, how the variable gets set or assigned, and whether the variable is local or global. Up to 702 variables can be supported using A to Z and AA to ZZ rows.

Example command: `change variables`

change variables

Page 1 of 39

VARIABLES FOR VECTORS							
Var	Description	Type	Scope	Length	Start	Assignment	VAC
A							
B							
C							
D							
E							
F							
G							
H							
I							
J							
K							
L							
M							
N							
O							
P							
Q							
R							

Assignment

A number to pre-assign to the variable. This field displays the current value for global values. Administration is optional. Available only when the type is value or collect G.

Related links

[Type](#) on page 869

Description

An identifying name or description of the vector variable. Administration is optional. Default is blank. Accepts up to 27 characters.

Length

Valid Entry	Usage
1 to 16	The maximum number of digits from the data to assign to the variable. This field does not apply to the do, dow, and tod variables. This entry is required for all types to which it applies.

Scope

Valid Entry	Usage
G	The variable is used globally.
L	The variable is used locally.

Start

Valid Entry	Usage
1 to 96	Specifies the beginning character position of the data digits string to be used for assigning to the variable. The combination of the Start position and maximum length of the digits string defines what is to be assigned to the variable. If the number of digits to be used is less than the maximum length specified, only that portion is assigned to the variable.

Type

Use this field to specify the vector variable type. The following are the valid entries:

- ani
- asaiuuu
- collect

- dow
- doy
- stepcnt
- tod
- value
- vdn
- vdntime
- agent

VAC

Valid Entry	Usage
1 to 9 blank	The Vector Variable Feature Access Code (FAC) to use for changing the value.

Var

Valid Entry	Usage
A to Z, AA to ZZ	The letter identifying the row of a specific vector variable.

Chapter 142: Vector Directory Number

Defines vector directory numbers (VDN) for the Call Vectoring feature. A VDN is an extension number used to access a call vector. Each VDN is mapped to one call vector. VDNs are software extension numbers (that is, not assigned to physical equipment). A VDN is accessed through direct dial local telephone company central office trunks mapped to the VDN (incoming destination or night service extension), DID trunks, and LDN calls. The VDN can be the Night Destination for a LDN.

An asterisk (*) next to a field indicates that the field follows VDN override rules when the system changes the active VDN for a call.

Example command: `add vdn n`, where *n* is the VDN extension number.

Vector Directory Numbers: page 1

Host: drccd User: ljm Dir: /usr/add-on/defty-ws1/ljm

add vdn next Page 1 of 3

VECTOR DIRECTORY NUMBER

Extension: 20043 Unicode Name? n

Name*:

Destination: Vector Number 1

Attendant Vectoring? n

Meet-me Conferencing? n

Allow VDN Override? n

COR: 1

TN*: 1

Measured: none Report Adjunct Calls as ACD*? n

VDN of Origin Annc. Extension*:

1st Skill*:

2nd Skill*:

3rd Skill*:

SIP URI:

* Follows VDN Override Rules

1st/2nd/3rd Skill

This field follows VDN override rules when the system changes the active VDN for a call. Available only if **Expert Agent Selection (EAS)** is enabled and **Meet-me Conferencing** is disabled.

Valid entry	Usage
1 to 8000	The skill number for each skill.
blank	Not administered. This is the default.

Related links

[Expert Agent Selection \(EAS\)](#) on page 740

[Meet-me Conference](#) on page 873

Acceptable Service Level (sec)

Available only if **BCMS/VuStats Service Level** is enabled for the system and the VDN is measured by BCMS.

Valid Entry	Usage
0 to 9999	The number of seconds within which calls to this VDN should be answered. This allows BCMS to report the percentage of calls that were answered within the specified time.
blank	Not administered. This is the default.

Related links

[BCMS/VuStats Service Level](#) on page 739

[Measured](#) on page 873

Allow VDN Override

Available only if **Meet-me Conferencing** is disabled for that particular VDN

Valid Entry	Usage
y	Allows the system to change the routed-to VDN to the “active” VDN for a call. The “active” VDN is the VDN to be used for parameters associated with the call such as VDN name, skills, tenant number, BSR application, VDN variables, and so on. The field follows VDN override rules when the system changes the “active” VDN for a call.
n	The routed-to VDN does not become the active VDN. The parameters of the original VDN are used. This is the default.

Related links

[Meet-me Conference](#) on page 873

Attendant Vectoring

Indicates whether or not Attendant Vectoring is selected on this VDN. Attendant Vectoring does not support Call Center features.

Related links

[Attendant Vectoring](#) on page 730

COR

Valid Entry	Usage
0 to 995	The class of restriction (COR) of the VDN consisting of a one- or two-digit number. This field cannot be blank.

Destination

Specifies if the calls are routed using a Vector Number or Policy Routing Table.

Extension

The extension number of the VDN. The extension is a number that starts with a valid first digit and length as defined by the system's dial plan.

Measured

Enables measurement data collection for this VDN. Data can be collected for reporting by BCMS or the Call Management System. Available only if **Meet-me Conferencing** is disabled. **BCMS** must be enabled for the system for this field to be set to internal or both. In addition, the appropriate CMS release must be administered for the system if this field is external or both.

Valid Entry	Usage
internal	Data is measured internally by BCMS.
external	Data is measured internally by CMS.
both	Data is measured internally by both BCMS and CMS.
none	Data is not measured. This is the default.

Related links

[Meet-me Conference](#) on page 873

[BCMS \(Basic\)](#) on page 739

Meet-me Conference

Enables or disables Meet-me Conferencing for this VDN. Determines if the VDN is a Meet-me Conference VDN.

*** Note:**

If the VDN extension is part of your DID block, external users are able to access the conference VDN. If the VDN extension is not part of the DID block, only internal callers on the network (including DCS or QSIG) or remote access callers can access the conference VDN.

Available only if **Enhanced Conferencing** is enabled for the system. If **Enhanced Conferencing** is enabled, but no other vectoring options are enabled, only Meet-me Conference vectors can be assigned.

Related links

[Enhanced Conferencing](#) on page 732

Name

The name associated with the VDN. When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call. Accepts up to 27 alphanumeric characters.

The name might be truncated on agents’ displays depending on the application. When information is forwarded with an interflowed call, only the first 15 characters are sent.

*** Note:**

Supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW, 1600-Series, 9600-Series IP Deskphones, 9600-Series H.323 Deskphones, and Avaya J100 Series IP Phones. Unicode is also an option for the 2420J telephone when the **Display Character Set** is katakana. For more information about 2420J, see *2420 Digital Telephone User's Guide*.

Related links

[Display Character Set](#) on page 722

[Meet-me Conference](#) on page 873

Number

Valid Entry	Usage
1 to 8000	The number of the vector or the Policy Routing Table (PRT) to which the calls are allocated.

Service Objective

Use this field to assign a service level to the VDN. The system displays this field only if you set the **Dynamic Advocate** field to y. The range of the acceptable value is between 1 to 9999. The default value is 20. For more information on **Service Objective**, see *Avaya Business Advocate User Guide*.

Related links

[BCMS/VuStats Service Level](#) on page 739

[Dynamic Advocate](#) on page 740

SIP URI

Use this field to place a call using the URI. URI is an alphanumeric string such as letters, numbers, and special characters. You can enter up to 70 characters, including handle and domain name. For example:

help.desk@example.com

helpdesk123@example.com

helpdesk@example.com

Following are not supported:

- Unicode characters
- Eurofont characters
- Using same URIs across hunt groups, VDNs, and stations within the same Communication Manager.

TN

Use this field to specify a tenant number. You can enter a value from 1 to 250.

Unicode Name

This read-only field shows the assignment Unicode Name to the VDN.

Valid entry	Usage
n	Unicode Name is not assigned to the VDN.
y	Unicode Name is assigned to the VDN.

For more information on Unicode Language Display, see *Administering Avaya Aura® Communication Manager*.

VDN of Origin Annc. Extension

The extension number of the VDN of Origin announcement. A VDN of Origin announcement is a short recording that identifies something about the call originating from the VDN. The agent hears the recording just prior to the delivery of the call. This field follows VDN override rules when the system changes the “active” VDN for a call.

Available only if VDN of **Origin Announcement** is enabled for the system and **Meet-me Conferencing** is disabled.

Related links

[Meet-me Conference](#) on page 873

[VDN of Origin Announcement](#) on page 743

Vector Number

An identifying number that specifies the call vector that is accessed through the VDN. This field cannot be blank.

Vector Directory Number: page 2 (Meet-me Conference disabled)

add vdn 3600006

Page2 of 3

VECTOR DIRECTORY NUMBER

AUDIX Name:

Return Destination*:

VDN Timed ACW Interval*: After Xfer or Held Call Drops*? n

BSR Application*:

BSR Available Agent Strategy*: 1st-foundUsed for BSR Polling? n

BSR Tie Strategy*: system

Observe on Agent Answer? n

Send VDN as Called Ringing Name Over QSIG? n

Display VDN for Route-To DAC*? n

VDN Override for ASAI Messages*: no

BSR Local Treatment*? n

Reporting for PC or POM Calls? n

Pass Prefixed CPN to VDN/Vector*? system

* Follows VDN Override Rules

Related links

[Meet-me Conference](#) on page 873

After Xfer or Held Call Drops

Use this field to put an auto-in agent into the Timed ACW mode when an ACD or DAC call on hold drops or the call is transferred and the Timed ACW feature is active. This replaces the Available mode.

Valid Entry	Usage
y	Communication Manager places an agent in the Timed ACW state if a caller on hold disconnects or the call is transferred, in addition to the existing conditions for Timed ACW.
n	Communication Manager continues with the existing operation. This is the default value.

AUDIX Name

The name of the AUDIX machine if this VDN is associated with the AUDIX vector.

BSR Application

Available only if **Lookahead Interflow (LAI)** and **Vectoring (Best Service Routing)** are enabled for the system. When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call.

Valid Entry	Usage
1 to 511 blank	A one- to three-digit number that specifies an application plan for the VDN when using multi-site Best Service Routing with this VDN.

Related links

[Lookahead Interflow \(LAI\)](#) on page 740

[Vectoring \(Best Service Routing\)](#) on page 741

BSR Available Agent Strategy

This field determines how Best Service Routing (BSR) identifies the best split or skill in an agent surplus situation to route a call. The system displays this field only if **Vectoring (Best Service Routing)** is enabled. If **Meet-me Conferencing** is disabled, the **BSR Available Agent Strategy** field follows VDN override rules when the system changes the active VDN for a call.

Valid entry	Usage
1st-found	The call is routed to the first available agent.
UCD-LOA	The call is routed to the least occupied agent, without regard to skill level. This option is available only if Least Occupied Agent (LOA) or Business Advocate is enabled.
UCD-MIA	The call is routed to the most idle agent, without regard to skill level. This option ensures a high degree of equity in agent workloads even when call-handling times vary.
EAD-LOA	The call is routed to the highest skill level agent with the lowest occupancy. This option is available only if Expert Agent Selection , and Least Occupied Agent (LOA) or Business Advocate are enabled.
EAD-MIA	The call is routed to the most idle agent with the highest skill level. This option is available only if Expert Agent Selection (EAS) is enabled.

Related links

[Business Advocate](#) on page 739

[Expert Agent Selection \(EAS\)](#) on page 740

[Least Occupied Agent](#) on page 740

[Vectoring \(Best Service Routing\)](#) on page 741

BSR Local Treatment

Enables or disables local server control of BSR treatment. In a multi-site BSR configuration, a call that arrives at a local communication server can be rerouted to a remote server located in a different part of the world. This feature maintains control at the local server and allows the server to provide local audio feedback for IP and ISDN calls, or to take back the call while the call waits in queue on a remote server. When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call.

 Note:

This field must be enabled on both the local and remote vdns, or else call interflow attempts might result in dropped calls.

BSR Tie Strategy

Available only if **Vectoring (Best Service Routing)** is enabled for the system. When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call.

Valid Entry	Usage
system	System-wide settings apply.
1st-found	BSR uses the previously selected best choice as the best skill or location. This is the default setting.
alternate	Alternates the BSR selection algorithm when a tie in EWT or available agent criteria occurs. Every other time a tie occurs for calls from the same VDN, the step with the tie is selected to send the call instead of the first selected split, skill, or location. This helps balance the routing when the cost of routing remotely is not a concern.

Related links

[Vectoring \(Best Service Routing\)](#) on page 741

Call Origin

The **Call Origin** field is applicable only if you enter a VDN extension in the **Return Destination** field. Use this field to administer the types of calls that will be redirected to the extension in the **Return Destination** field when the destination is released.

Valid Entry	Usage
external	Applies VDN Return Destination only to incoming external trunk calls directly to the VDN. This is the default setting.
internal	Applies VDN Return Destination only to internal calls that have not been through vector processing. Internal calls for this purpose include trunk calls that are transferred, forwarded, or covered to the VDN or ASAI routed.
both	Applies VDN Return Destination to both internal and external calls.

Display VDN for Route-To DAC

Enables or disables the display of the VDN when a call arrives as a result of a route-to at an agent terminal under the following conditions:

- A route-to number with coverage is enabled, or route-to digits with coverage is enabled and the vector command routes a call to an agent as an EAS direct agent call.
- Adjunct routing routes a direct agent call to the agent.

When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call.

Available only if **Expert Agent Selection (EAS)** is enabled for the system.

Related links

[Expert Agent Selection \(EAS\)](#) on page 740

Observe on Agent Answer

Enables or disables a service observer to start observing a call to the VDN when the call is delivered to the agent or station. The ability to observe VDNs when the call is delivered to an agent/station is activated by setting the **Observe on Agent Answer** field to *y*. The observer sees the name of the VDN, agent, or trunk as each is accessed in sequence by the VDN. For example, during vector processing the VDN name is displayed, but when the call connects to an agent, the agent name is displayed. When the observer connects to a call in vector processing, the system maintains the connection until the call is disconnected or the observer disconnects, even if the call is routed or transferred externally. If the observer does not disconnect after one observed call is disconnected, the observer is connected to another call on the same VDN. Observing is listen only as long as the call is in vector processing. Once the call is out of vector processing, an observer with listen/talk capability can talk as well as listen.

* Note:

When the **Observe on Agent Answer** field is set to *y*, service observers of the VDN are not connected to a call until the call is delivered to an agent. The default setting of this field is *n*. With the default setting, the non-by location observer of the VDN can hear the call throughout the vector processing and delivery to the agent. For information on **Observe on Agent Answer**, see *Avaya Aura® Call Center Elite Feature Reference*.

Pass Prefixed CPN to VDN/Vector

This feature is used to optionally pass the CPN prefix to VDNs and vectors as part of the CPN. This option can be set on a system wide basis, and overridden, if required, on an individual VDN screen. Accordingly, international call handling can be improved by prioritizing and routing more intelligently, such as by automatically routing calls to agents who speak the pertinent language. For example, if a customer from Germany calls in to a call center in Spain, based on the international code of Germany (0049 - international prefix + country code), the call can be routed to a German speaking agent. Also, an international call might be given priority over a national call to save toll charges.

PC Predictive Reports Skill

The **PC Predictive Reports Skill** field is applicable only if the **ASAI Link Plus Capabilities** or the **Computer Telephony Adjunct Links** field is enabled on page 3 of the System-Parameters Customer-Options screen, and if you enable the **Reporting for PC or POM Calls** field on page 2 of the Vector Directory Number screen.

Valid Entry	Usage
1 to 99	Skill number for Avaya Solutions Platform S8300 switches. VDN option applies to PC reporting.
1 to 8000	Skill number for all other platforms. VDN option applies to PC reporting.
blank	VDN option applies to Proactive Outreach Manager (POM) calls.

Related links

[Reporting for PC or POM Calls](#) on page 880

[ASAI Link Plus Capabilities](#) on page 729

[Computer Telephony Adjunct Links](#) on page 731

Reporting for PC or POM Calls

The **Reporting for PC or POM Calls** field indicates that the VDN is used for POM or Proactive Contact (PC) outbound calling. If you have switch-classified outbound calling with PC, use this field and the associated skill field to activate the improved integration with PC feature. If you have POM, use the field to activate this VDN for indication to reporting of a Proactive Outreach Manager call to the VDN. Valid entries are y and n. The default is n.

* Note:

You can administer the field only after you set the **Measured** field on page 1 of the Vector Directory Number screen to either external or both.

For more information on how to use the feature, see Proactive Contact outbound calling improved reporting in the *Avaya Aura® Call Center Feature Reference* document.

Related links

[Measured](#) on page 873

Return Destination

The VDN extension number to which an incoming trunk call is routed if it returns to vector processing after the agent drops the call. When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call.

Send VDN as Called Ringing Name Over QSIG

Valid Entry	Usage
y	If the ISDN trunk call is made, receiver can view the VDN name when the telephone is ringing.
n	The receiver cannot view the VDN name when the telephone is ringing. The default value is n.

Used for BSR Polling

Use the **Used for BSR Polling** field to prevent Communication Manager from sending messages to CMS or Avaya IQ regarding unmeasured calls for unmeasured VDNs that route to vectors with the **BSR reply-best** command.

The **Used for BSR Polling** field is applicable to unmeasured VDNs assigned to a vector that has consider steps terminated by the reply-best step and no-queuing step. The trunk groups that are used to route to the VDNs are also unmeasured.

For **Used for BSR Polling** to function, set the **Measured** field to none.

Valid entry	Usage
y	Communication Manager does not send messages for unmeasured calls to CMS or Avaya IQ.
n	Communication Manager sends messages, even for unmeasured calls, to CMS or Avaya IQ. The default value is n.

VDN Override for ASAI Messages

Determines if the active VDN is sent as the called number for ISDN Trunk ASAI messages. When **Meet-me Conferencing** is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call. The “active” VDN is the VDN receiving the call and will be changed to a routed-to VDN if **Allow VDN Override** is enabled. Available only if **ASAI Link Core Capabilities** is enabled for the system.

Valid Entry	Usage
n	The “Called Number” information is sent for the “Call Offered”, “Alerting”, “Queued”, and “Connect” ASAI event notification messages. The adjunct-request message is always the called VDN extension in the Called Number IE sent in the incoming ISDN SETUP message or the local call’s called number and does not change after routing to the called VDN and subsequent routed-to VDNs.
ISDN Trunk	When an incoming ISDN trunk call is routed to this VDN, the “Called Number” information sent in the ASAI event and “Adjunct Route Request” ASAI messages is the “active VDN” extension. This extension becomes associated with the call based on the VDN Override rules. This option does not apply to local/internal calls.

Table continues...

Valid Entry	Usage
all	The active VDN is used for the called number for all types of calls to the VDN, including local/internal calls as well as external incoming ISDN trunk calls.

Related links

[Allow VDN Override](#) on page 872

[ASAI Link Core Capabilities](#) on page 729

VDN Timed ACW Interval

Valid Entry	Usage
1 to 9999 blank	Sets the length of the timed ACW Interval. When a value is entered in this field, an agent in auto-in work mode who receives a call from this VDN is automatically placed into After Call Work (ACW) when the call drops. When the administered time is over, the agent automatically becomes available. This field takes precedence over the hunt group Timed ACW Interval . When Meet-me Conferencing is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call.

Related links

[Timed ACW Interval \(sec\)](#) on page 354

Vector Directory Number: page 2 (Meet-me Conference enabled)

change vdn 601501	Page 2 of 3
VECTOR DIRECTORY NUMBER	
MEET-ME CONFERENCE PARAMETERS:	
Conference Access Code:	
Conference Controller:	
Conference Type:	local

Related links

[Meet-me Conference](#) on page 873

Conference Access Code

! Security alert:

To ensure conference security, always assign an access code to a Meet-me Conference VDN. The Meet-Me Conference VDN access code. Accepts up to six digits.

Conference Controller

Controls which user is allowed to change the access code for a Meet-me Conference VDN using a feature access code. This can be a local user or someone dialing in from remote access trunks.

Valid Entry	Usage
<i>Extension number</i>	Only a user at this extension can change the access code for that VDN using a feature access code.
blank	Any station user that is assigned with console permissions can change the access code for that VDN using a feature access code.

Conference Type

The conference type that is appropriate for the call.

Valid Entry	Usage
local	For six or fewer participants. Default is local. Prior to Release 8.1.2, valid entry was <code>6-party</code>. In Release 8.1.2, the valid entry is changed to <code>local</code>.  Note: For 12 parties to participate in a conference, you must enable the 12-party Conferences field in the Feature-Related System-Parameters screen.
expanded	Enables the Expanded Meet-me Conference feature. For a conference with more than six participants.

Route-to Number

The ARS or AAR Feature Access Code (FAC) followed by the routing digits. Alternately, the unique UDP extension. Accepts up to 16 digits. Allows administration of the routing digits (the ARS/AAR Feature Access Code with the routing digits and the Conference ID digits for the VDN). Available only for expanded conference types.

 Note:

The Route-to Number must be unique across all Expanded Meet-me Conference VDNs.

Related links

[Conference Type](#) on page 884

Vector Directory Number: page 3

add vdn 3600006 Page 3 of 3

VECTOR DIRECTORY NUMBER

VDN VARIABLES*

Var	Description	Assignment
V1		
V2		
V3		
V4		
V5		
V6		
V7		
V8		
V9		

VDN Time-Zone Offset*: + 00:00
 Daylight Saving Rule*: system
 Use VDN Time Zone For Holiday Vectoring*? n
 Apply Ringback for Auto Answer calls*? y

* Follows VDN Override Rules

VDN VARIABLES

When **Meet-me Conferencing** is disabled, the system displays an asterisk (*) next to the heading, indicating that variables V1 through V9 follow VDN override rules when the system changes the “active” VDN for a call.

Related links

[Meet-me Conference](#) on page 873

Assignment

Valid Entry	Usage
0 to 9	Assigns an up to 16-digit unvalidated decimal number to each of the VDN variables V1 through V5.
blank	No decimal number is assigned to the VDN variable.

Description

A description of the VDN variable. Accepts up to 15 characters.

Var

The number assigned to the VDN variable.

Apply Ringback for Auto Answer calls

The **Apply Ringback for Auto Answer calls** field is applicable only if you enable the **Auto Answer** field on either page 2 of the Station screen or on page 1 of the Agent Login ID screen. You can use the field to not apply ringback to the callers for calls delivered to agents in the auto answer mode when application of ringback is undesirable. This field follows VDN Override rules and the setting on the active VDN for the call is applicable.

Valid Entry	Usage
y	Applies audible ringback to the callers during any delay experienced for calls delivered to agents in the auto answer mode. This is the default value.
n	Prevents returning audible ringback to the callers for calls delivered to agents in the auto answer mode.

Related links

[Auto Answer](#) on page 42

[Auto Answer](#) on page 668

Daylight Saving Rule

Defines the daylight saving time rule. The daylight saving time rule is applied to `goto time-of-day` commands in the vector that is assigned to the VDN. The time-of-day calculations are based on the local time of the receiving call's VDN. The assigned rule number applies start and stop rules that are administered for the system for that rule.

Valid Entry	Usage
system	The system uses the same daylight saving time rule as the system clock.
0	No daylight saving rule is applied. If the system time has a daylight saving rule specified, this rule is removed before evaluating the <code>goto if time-of-day</code> conditional.
1 to 15	The Daylight Saving Rule number. When you use a number other than 0, the rule associated with the main server clock display time and the main server offset are not used. The offset and rule assigned to the active VDN for the call are applied to the operating system standard time so that local time for the VDN is used to test the time-of-day step.

Related links

[Daylight Saving Rules](#) on page 183

[VDN Time Zone Offset](#) on page 887

Use VDN Time Zone For Holiday Vectoring

Valid Entry	Usage
y	Communication Manager uses the VDN time zone offset and DST adjustment for processing the date and time assigned in the Holiday Table screen.

Table continues...

Valid Entry	Usage
n	Communication Manager uses the system time for processing the date and time assigned in the Holiday Table screen. This is the default value.

VDN Time Zone Offset

Valid Entry	Usage
+, - 00-23 - hour 00-59 - minute	This field is applied against the switch clock when a time of day vector command is executed. Daylight saving time changes are handled by the switch clock using the existing operation. When Meet-me Conferencing is disabled, this field follows VDN override rules when the system changes the “active” VDN for a call. The valid entries are based on a syntax of +HH:MM. The default is +00:00. When the default is set, the system switch time is used without modification.

Related links

[Meet-me Conference](#) on page 873

Chapter 143: Video Bridge

Used to configure available ad-hoc conferencing resources. For more detailed information on Avaya Video Telephony, see *Avaya Video Telephony Solution Networking Guide, 16-601423*.

Example command: `add video-bridge n`, where *n* is the video bridge number.

Bridge ID

The ID number for this video bridge.

Call Rate

The maximum allowable call rate for the conference. Available only when **Far End Resource Info** is disabled and **Type** is not administered.

Related links

[Far End Resource Info](#) on page 888

[Type](#) on page 890

Far End Resource Info

Enables or disables far end resource tracking and reporting.

Valid Entry	Usage
y	The far end tracks port usage and provides updates on resource availability.
n	No resource information is provided from the far end.

ID Range Start/End

A range of conference IDs that this video bridge can use. There must be enough IDs so that all ports can be used — one ID for every six ports. Accepts up to nine digits.

The default is blank.

Available only for h.323 trunk groups.

Related links

[Group Type](#) on page 437

Max Ports

Valid Entry	Usage
3 to system max	The maximum number of video conferencing ports for this video bridge. Default is none.

Name

The name that identifies this video bridge. Accepts up to 30 alphanumeric characters.

Priority Factory Number

Priority versus Standard factory number depends on who creates the conference; if a user with Priority Video permissions creates the conference, the **Priority Factory Number** is used. **Priority Factory Number** can have a dedicated video bridge or a bridge with bandwidth more than the **Standard Factory Number**. Available only for h.323 or sip trunk groups when the **Far End Resource Info** is enabled.

Valid Entry	Usage
1 to 9 digits (0,9)	At least one of Priority Factory Number or Standard Factory Number must be filled in. Standard and Priority factory numbers can be the same.
blank	Priority calls can use the bridge, but prefer a bridge with a priority factory. This is the default.

Related links

[Far End Resource Info](#) on page 888

[Group Type](#) on page 437

[Standard Factory Number](#) on page 890

Standard Factory Number

Priority versus Standard factory number depends on who creates the conference; if a user with Priority Video permissions creates the conference, the **Priority Factory Number** is used. **Priority Factory Number** has a dedicated video bridge or a bridge with better bandwidth than the **Standard Factory Number**. Available only for h.323 or sip trunk groups. For h.323, the **Far End Resource Info** must be enabled.

Valid Entry	Usage
1 to 9 digits (0,9)	At least one of Priority Factory Number or Standard Factory Number must be filled in. Standard and Priority factory numbers can be the same.
blank	Non-priority conferences are unable to use this video bridge. Default is blank.

Related links

[Priority Factory Number](#) on page 889

[Far End Resource Info](#) on page 888

[Group Type](#) on page 437

Trunk Groups

Valid Entry	Usage
1 to 2000 blank	Assigns trunk groups to this video bridge. At least one incoming and one outgoing trunk, or a two-way trunk, must be assigned. All trunks on a given video bridge must be the same type; H.323 and SIP trunks cannot be mixed. The default is blank.

Related links

[Group Type](#) on page 437

Type

The type of video bridge. Can be administered for existing or new video bridges. Available only if **Far End Resource Info** is disabled.

Valid Entry	Usage
Exact	All participants in the conference must use the exact rate specified or be in the audio-only mode.
Maximum	Call Rate is enabled, and any call rate up to the configured rate is allowed in a conference. Default is Maximum.
Region	The initial entry for an existing video bridge. Call Rate is unavailable.

Related links

[Call Rate](#) on page 888

[Far End Resource Info](#) on page 888

Chapter 144: Resources

Communication Manager documentation

The following table lists the documents related to Communication Manager. Download the documents from the Avaya Support website at <http://support.avaya.com>.

Title	Description	Audience
Design		
<i>Avaya Aura® Communication Manager Overview and Specification</i>	Provides an overview of the features of Communication Manager.	Sales Engineers, Solution Architects
<i>Avaya Aura® Communication Manager Security Design</i>	Describes security-related issues and security features of Communication Manager.	Sales Engineers, Solution Architects
<i>Avaya Aura® Communication Manager System Capacities Table</i>	Describes the system capacities for Avaya Aura® Communication Manager.	Sales Engineers, Solution Architects
<i>LED Descriptions for Avaya Aura® Communication Manager Hardware Components</i>	Describes the LED for hardware components of Avaya Aura® Communication Manager.	Sales Engineers, Solution Architects
<i>Avaya Aura® Communication Manager Hardware Description and Reference</i>	Describes the hardware requirements for Avaya Aura® Communication Manager.	Sales Engineers, Solution Architects
<i>Avaya Aura® Communication Manager Survivability Options</i>	Describes the system survivability options for Avaya Aura® Communication Manager.	Sales Engineers, Solution Architects
<i>Avaya Aura® Core Solution Description</i>	Provides a high level description for the solution.	Sales Engineers, Solution Architects
Maintenance and Troubleshooting		
<i>Avaya Aura® Communication Manager Reports</i>	Describes the reports for Avaya Aura® Communication Manager.	Sales Engineers, Solution Architects, Implementation Engineers, Support Personnel
<i>Maintenance Procedures for Avaya Aura® Communication Manager, Branch Gateways and Servers</i>	Provides procedures to maintain Avaya servers and gateways.	Sales Engineers, Solution Architects, Implementation Engineers, Support Personnel

Table continues...

Title	Description	Audience
<i>Maintenance Commands for Avaya Aura® Communication Manager, Branch Gateways and Servers</i>	Provides commands to monitor, test, and maintain Avaya servers and gateways.	Sales Engineers, Solution Architects, Implementation Engineers, Support Personnel
<i>Avaya Aura® Communication Manager Alarms, Events, and Logs Reference</i>	Provides procedures to monitor, test, and maintain Avaya servers and describes the denial events listed on the Events Report form.	Sales Engineers, Solution Architects, Implementation Engineers, Support Personnel
Administration		
<i>Administering Avaya Aura® Communication Manager</i>	Describes the procedures and screens for administering Communication Manager.	Sales Engineers, Implementation Engineers, Support Personnel
<i>Administering Network Connectivity on Avaya Aura® Communication Manager</i>	Describes the network connectivity for Communication Manager.	Sales Engineers, Implementation Engineers, Support Personnel
<i>Avaya Aura® Communication Manager SNMP Administration and Reference</i>	Describes SNMP administration for Communication Manager.	Sales Engineers, Implementation Engineers, Support Personnel
<i>Administering Avaya Aura® Communication Manager Server Options</i>	Describes server options for Communication Manager.	Sales Engineers, Implementation Engineers, Support Personnel
<i>Avaya Aura® Communication Manager Data Privacy Guidelines</i>	Describes how to administer Communication Manager to fulfill Data Privacy requirements.	Sales Engineers, Implementation Engineers, Support Personnel
Implementation and Upgrading		
<i>Deploying Avaya Aura® Communication Manager in Virtualized Environment</i>	Describes the implementation instructions while deploying Communication Manager on VMware.	Implementation Engineers, Support Personnel, Solution Architects
<i>Deploying Avaya Aura® Communication Manager in Software-Only and Infrastructure as a Service Environments</i>	Describes the implementation instructions while deploying Communication Manager on a software-only environment and Amazon Web Service, Microsoft Azure, and Google Cloud Platform.	Implementation Engineers, Support Personnel, Solution Architects

Table continues...

Title	Description	Audience
<i>Upgrading Avaya Aura® Communication Manager</i>	Describes instructions while upgrading Communication Manager.	Implementation Engineers, Support Personnel, Solution Architects
Understanding		
<i>Avaya Aura® Communication Manager Feature Description and Implementation</i>	Describes the features that you can administer using Communication Manager.	Sales Engineers, Solution Architects, Support Personnel
<i>Avaya Aura® Communication Manager Screen Reference</i>	Describes the screens that you can administer using Communication Manager.	Sales Engineers, Solution Architects, Support Personnel
<i>Avaya Aura® Communication Manager Special Application Features</i>	Describes the special features that specific customers request for their specific requirement.	Sales Engineers, Solution Architects, Avaya Business Partners, Support Personnel

Finding documents on the Avaya Support website

Procedure

1. Go to <https://support.avaya.com>.
2. To log in, click **Sign In** at the top of the screen and then enter your login credentials when prompted.
3. Click **Product Support > Documents**.
4. In **Search Product**, start typing the product name and then select the appropriate product from the list displayed.
5. In **Select Release**, select the appropriate release number.
This field is not available if there is only one release for the product.
6. **(Optional)** In **Enter Keyword**, type keywords for your search.
7. From the **Select Content Type** list, select one or more content types.
For example, if you only want to see user guides, click **User Guides** in the **Select Content Type** list.
8. Click  to display the search results.

Accessing the port matrix document

Procedure

1. Go to <https://support.avaya.com>.
2. To log in, click **Sign In** at the top of the screen and then enter your login credentials when prompted.

3. Click **Product Support > Documents**.
4. In **Select Release**, select the appropriate release number.
This field is not available if there is only one release for the product.
5. From the **Select Content Type** list, select one or both of the following options:
 - **Application & Technical Notes**
 - **Design, Development & System Mgt**

Avaya Documentation Center navigation

For many programs, the latest customer documentation is available on the Avaya Documentation Center website at <https://documentation.avaya.com>. Some functionality is only available when you log in to the Avaya Documentation Center. The available functionality depends on your role.

Important:

If the documentation you are looking for is not available on the Avaya Documentation Center, you can find it on the [Avaya Support website](#).

While navigating through the Documentation Center, you can click the **Avaya Documentation Center** logo at the top of the screen to return to the home page anytime. On the Avaya Documentation Center, you can do the following:

- Click **Avaya Links** in the top menu bar to access other Avaya websites, including the Avaya Support website.
- Click **Languages** () in the top menu bar to change the display language and view localized documents.
- In the **Search Documentation** field, search for keywords and click **Filter** to filter by solution category, product, or user role.
You can select multiple items in each filter category. For example, you can select a product and multiple user roles.
- Click **Library** in the top menu bar to access the complete library of documents. Use the filtering options to refine your results.
- After performing a search or accessing the library, you can sort content on the search results page. When you find the item you want to view, click it to open it.
- Use the table of contents in a document for navigation. You can also click < or > next to the document title to navigate to the previous topic or the next topic.
- Click **Share** () to share a topic by email or copy the URL.
- Download a PDF of the current topic in a document, the topic and its subtopics, or the entire document.
- Print the section you are viewing.
- Add content to a collection by clicking **Add to My Topics** (). You can add the topic and its subtopics or add the entire publication.

- View the topics in your collections. To access your collections, click your name in the top menu bar and then click **My Topics**.

You can do the following:

- Create, rename, and delete a collection.
 - Set a collection as the default or favorite collection.
 - Save a PDF of the selected content in a collection and download it to your computer.
 - Share content in a collection with others through email.
 - Receive collections that others have shared with you.
- Click **Watch** (👁) to add a topic to your watchlist so you are notified when the content is updated or removed.
 - View and manage your watchlist by clicking **Watchlist** from the top menu with your name.

You can do the following:

- Enable **Email notifications** to receive email alerts.
 - Unwatch the selected content or all topics.
- Send feedback for a topic.

Training

The following courses are available on the Avaya Learning website at <http://www.avaya-learning.com>. After logging in to the website, enter the course code or the course title in the **Search** field and press **Enter** or click > to search for the course.

Course code	Course title
70380W	What's New with Avaya Aura® 10.2
70390W	Upgrading to Avaya Aura® 10.2
70410W	Migrating to ASP R6.0.x (KVM on RHEL 8.10) Hypervisor
71301V	Integrating Avaya Aura® Communications Applications
72301V	Supporting Avaya Aura® Communications Applications
20460W	Virtualization and Installation Basics for Avaya Team Engagement Solutions
71201V	Integrating Avaya Aura® Core Components
72201V	Supporting Avaya Aura® Core Components
61131V	Administering Avaya Aura® System Manager
61451V	Administering Avaya Aura® Communication Manager

Viewing Avaya Mentor videos

Avaya Mentor videos provide technical content on how to install, configure, and troubleshoot Avaya products.

About this task

Videos are available on the Avaya Support website, listed under the video document type, and on the Avaya-run channel on YouTube.

- To find videos on the Avaya Support website, go to <https://support.avaya.com/> and do one of the following:
 - In **Search**, type `Avaya Mentor Videos`, click **Clear All** and select **Video** in the **Select Content Type**.
 - In **Search**, type the product name. On the Search Results page, click **Clear All** and select **Video** in the **Select Content Type**.

The **Video** content type is displayed only when videos are available for that product.

In the right pane, the page displays a list of available videos.

- To find the Avaya Mentor videos on YouTube, go to www.youtube.com/AvayaMentor and do one of the following:
 - Enter a keyword or keywords in the **Search Channel** to search for a specific product or topic.
 - Scroll down Playlists, and click a topic name to see the list of videos available. For example, Contact Centers.

Note:

Videos are not available for all products.

Support

Go to the Avaya Support website at <https://support.avaya.com> for the most up-to-date documentation, product notices, and knowledge articles. You can also search for release notes, downloads, and resolutions to issues. Use the online service request system to create a service request. Chat with live agents to get answers to questions, or request an agent to connect you to a support team if an issue requires additional expertise.

Using the Avaya InSite Knowledge Base

The Avaya InSite Knowledge Base is a web-based search engine that provides:

- Up-to-date troubleshooting procedures and technical tips.
- Information about service packs.

- Access to customer and technical documentation.
- Information about training and certification programs.
- Links to other pertinent information.

If you are an authorized Avaya Partner or a current Avaya customer with a support contract, you can access the Knowledge Base without extra cost. You must have a login account and a valid Sold-To number.

Use the Avaya InSite Knowledge Base for any potential solutions to problems.

1. Go to <https://support.avaya.com>.
2. To log in, click **Sign In** at the top of the screen and then enter your login credentials when prompted.
3. Click **Product Support > Products**.
4. In **Search Product**, start typing the product name and then select the appropriate product from the list displayed.
5. Select the release number, if applicable.
6. Click the **Technical Solutions** tab to view articles for resolving technical issues.

Appendix A: PCN and PSN notifications

PCN and PSN notifications

Avaya issues a product-change notice (PCN) for any software update. For example, a PCN must accompany a service pack or an update that must be applied universally. Avaya issues a product-support notice (PSN) to alert Avaya Direct, Business Partners, and customers of a problem or a change in a product. A PSN can also be used to provide a work around for a known problem, steps to recover logs, or steps to recover software. Both these notices alert you to important issues that directly impact Avaya products.

Viewing PCNs and PSNs

About this task

To view PCNs and PSNs, perform the following steps:

Procedure

1. Go to the Avaya Support website at <https://support.avaya.com> and log in.
2. On the top of the page, in **Search Product**, type the product name.
The Avaya Support website displays the product name.
3. Select the required product name.
4. In the **Choose Release** field, select the specific release from the drop-down list.
5. On the product page, click **Product Documents**.
6. In the Latest Support, Service and Product Correction Notices section, click **View All Notices**.
7. Select the appropriate filters as per your search requirement.

For example, if you select Product Support Notices, the system displays only PSNs in the documents list.

You can apply multiple filters to search for the required documents.

Signing up for PCNs and PSNs

About this task

Manually viewing PCNs and PSNs is helpful, but you can also sign up for receiving notifications of new PCNs and PSNs. Signing up for notifications alerts you to specific issues you must be aware of. These notifications also alert you when new product documentation, new product patches, or new service packs are available. The Avaya Notifications process manages this proactive notification system.

To sign up for notifications:

Procedure

1. Go to <https://support.avaya.com> and search for “Guide to Managing Your Avaya Access Profile for Customers and Partners”.

Under the Search Results section, click Guide to Managing Your Avaya Access Profile for Customers and Partners.

2. Set up e-notifications.

For detailed information, see the **Subscribe to E-Notifications** procedure.

Index

Special Characters

;Ethernet Options;	393
(Calls Warning) Port	339
(Time Warning) Port	345
% of Gateways in Network Region with Hyperactive Registration Alarms	419

Numerics

01–1000	586
1-Step Clearing	684
1st/2nd/3rd Skill	
Vector Directory Number	872
2-Party Loss Plan/Tone Loss Plan	492
3PCC H.323 Override SIP Station Activation/Deactivation	225
440Hz PBX-dial Tone	783
440Hz Secondary-dial Tone	783
6400/8400/2420J LINE APPEARANCE LED SETTINGS	308
7103A Button Assignment	
Abbreviated Dialing	33
7405ND Numeric Terminal Display	258
7434ND	258
802.1P/Q Parameters	401

A

A/D Grp/Sys List Dialing Start at 01	728
AACC Conference Modes	
Feature Access Codes	228
AAR and ARS Digit Conversion Table	
ANI Req'd	23
AAR Digit Analysis Table	18
AAR Digit Conversion Table	23
AAR/ARS Access	258
AAR/ARS Access Code	149 , 187
AAR/ARS Dial Tone Required	238
AAR/ARS Internal Call Prefix	193
AAR/ARS Internal Call Total Length	193
AAR/ARS Partitioning	116
AAS	347
Agent Login ID	41
Abandoned Call Search	823
Abbreviated Dial	213
Abbreviated Dial Programming by Assigned Lists	239
abbreviated dialing	154
Abbreviated Dialing	244
Enhanced list	26
Group list	26
Personal list	26
System list	26
ABBREVIATED DIALING	66

Abbreviated Dialing Enhanced List	728
Abbreviated Dialing List 1, List 2, List 3	681
Abbreviated Dialing List1 Access Code	212
Abbreviated Dialing List2 Access Code	212
Abbreviated Dialing List3 Access Code	213
abbreviated dialing lists	154
Abort Conference	266
Abort Transfer	266
Absorption Treatment Assignment	199
Absorption Treatment Information	199
AC	37
Authorization Code COR Mapping	74
ACA Assignment	450 , 824
ACA Long Holding Time Originating Extension	239
ACA Referral Calls	239
ACA Referral Destination	239
ACA Remote PBX Identification	240
ACA Short Holding Time Originating Extension	240
Accept Redirect to Blank User Destination	857
Acceptable Service Level (sec)	872
Access Code	677
Access Code 2	213
access endpoint	34
Access Endpoint	34
Communication Type	34
Extension	35
Name	35
TN	36 , 444 , 802
Width	36
Access Security Gateway (ASG)	728
Access to MCT	111
accessing port matrix	894
ACD	338 , 739
ACD Login Identification Length	291
Act Time	468 , 773 , 775
Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point	715
Activation Code	237
Active	
Coverage Path	165
Active Station Ringing	667
Actual %	562
ACW Agent Considered Idle	
Agent Login ID	41
ACW Agents Considered Idle	287
ACW Forced Logout Reason Code	299
Ad hoc Video Conferencing	135
Add Agent Skill Access Code	229
add announcement	52
add attendant	59
add audio-group	71
Add/Remove Agent Skills	112
Address Digits Include End-of-Digits Signal	520

Adjunct CTI Link		Always Use (<i>continued</i>)	
Hunt Group	347	Attendant Console	63
Adjunct Supervision	669	Station	667
Administer Timers	445, 806	Always Use re-INVITE for Display Updates	
Administered Connection	37	Trunk Group	858
Administered Members (min/max)	364, 471, 855	AMD Treatment	634
Administrable Timers	465, 840	Analog Busy Auto Callback	267
AE Services Server	432	Analog Gain	787
AES host	176	Analog Line Transmission	
After Call Work Access Code	229	Location Parameters	486
After Xfer or Held Call Drops	876	Analog Loss Group	553, 806
AGENT AND CALL SELECTION	287	Analog Ringing Cadence	
Agent Login ID	41	Location Parameters	486
Agent Login ID page 1	41	Analog Trunk Incoming Call ID	728
Agent Login ID page 2	47	ANI Available	523
Agent States	745	ANI Not Available	523
AGL	408	ANI Prefix	513
Alarm Threshold		ANI Req'd	
Administered Connection	39	AAR and ARS Digit Analysis Table	18
Alarm Type		ANI Source for Forwarded & Covered Calls	521
Administered Connection	40	ANI/II-Digits	85
alarms		Annc Name	
hyperactive registration	419	Announcements Audio Sources	53
alarms and detection	418	Annc Type	
Alert Incoming SIP Crisis Calls	608	Announcements Audio Sources	53
Signaling Group	608	Announcement Access Code	213
Alert Pager	171	Announcement Extension	601
ALERT PAGER	171	Announcement Ports	330
ALERT STATION	171	Announcement Server IP Address	417
Alerting (sec)	155	Announcement Storage Path Name	417
Alias Set Type		Announcement Type	331
Alias Station	50	Announcements	267, 532
Alias Station	50	Announcements Audio Sources	52
Allocation Plan Number	469	Ans Delay	855
Allow ANI Restriction on AAR/ARS	259	Answer Back Access Code	213
Allow ATMS Busyout, Error Logging and Alarming	851	answer groups	161
Allow Bridge DM Answer		Answer Send (msec)	840
Feature-related system parameters	311	Answer Supervision by Call Classifier	728
Allow Conference via Flash	277	Answer Supervision Timeout	445, 554, 807
Allow Crisis Alert Across Tenants	277	APLT	112
Allow Direct-IP Multimedia	385	Appl	574
Allow H.248 Gateways	393	Appl.	631
Allow H.323 Endpoints	393	application	690
Allow Night Service Calls To Forward	149	Application	
Allow Ringer-off with Auto-Answer	293	Stations With Off-PBX Telephone Integration	690
Allow SIP URI Conversion	407	Apply Intercept Locally	281
Allow Trunk Seizure On Busy Indicator	123	Apply Local Ringback	450
Allow VDN Override	872	Apply MCT Warning Tone	255
allowed with exclusion		Apply Ready Indication Tone To Which Parties In The	
service observing	285	Call	268
single step conference (SSC)	285	Apply Ringback for Auto Answer calls	886
Alpha-name		Apply Ringback upon Receipt of	143
Alias Station	51	ARS	214, 729
Alphanumeric Dialing Table	51	ARS Digit Analysis Table	18
Alternate FRL Station	149	ARS Digit Conversion Table	23
Alternate Route Timer	608	ARS FAC	482
Always Use		ARS Prefix 1 Required for 10-Digit NANP Calls	483

ARS Toll Table	58	Audix Name	667
ARS/AAR Dialing without FAC	729	AUDIX Name	358, 768, 877
ARS/AAR Partitioning	729	Audix Name for Messaging	
As-needed Inactivity Time-out (min)	631	Agent Login ID	42
ASAC Enabled	532	Audix Names	73
ASAI	320, 466	Audix-MSA Node Names	73
ASAI Call Classification	289	Auth Code	433, 788
ASAI Link Core Capabilities	729	Authoritative Domain	398
ASAI Link Plus Capabilities	729	Authorization Code Cancellation Symbol	252
ASAI Routing	86	Authorization Code COR Mapping	74
ASAI Uses Station Lock	119	Authorization Code Length	252
ASCII mode	329	AUTHORIZATION CODE PARAMETERS	251
Assigned Member — Ext	686	Authorization Code Required	582
Assigned Member — Name	686	Authorization Codes	730
Assigned Members	377	Authorization Codes Enabled	252
Assigned VDN	204	Auto Abbreviated/Delayed Transition Interval (rings)	240
Assignment	868, 885	Auto Alternate Routing	214
Assist Access Code	229	Auto Answer	123
Assistance Call	157	Agent Login ID	42
Associated Signaling	609	Attendant Console	59
Asynch. Transfer Mode (ATM) PNC	729	Media link calls	119
Asynch. Transfer Mode (ATM) Trunking	730	Station	668
At End of Member List	364	Auto Guard	807
ATM WAN Spare Processor	730	Auto Hold	259
ATMS	730	Auto Inspect on Send All Calls	256
Att. Call Waiting Indication	666	Auto Reserve Agents	287
Attd FAC	483	Auto Restoration	
Attendant Access Code	213	Administered Connection	40
Attendant Console	59	auto return	750
display module button assignment	70	day	751
feature button assignment	68	time	751
Softconsole IP Attendant	63	time delay	752
VIS feature options	65	Auto Route Selection (ARS) Access Code 1	214
attendant console group parameters	149	Auto Select Any Idle Appearance	669
Attendant Console X	59	Auto Start	259
Attendant Diversion Timing (sec)	534	Attendant Console	65
Attendant Group	758	Auto-A/D	681
Attendant Group Name	149	Auto-answer IP Failure AUX Reason Code	297
Attendant Lockout	150	Auto-In Access Code	229
Attendant Originated Calls	264, 761	Auto?	
Attendant Queue Announcement	533	IP Interfaces	396
Attendant Time Out Flag	251	Automatic Callback	131
Attendant Tone;Auto Hold	259	Automatic Callback — No Answer Timeout Interval	
Attendant Vectoring	86, 730, 873	(rings)	240
Attendant Vectoring VDN	150, 759	Automatic Callback Activation	214
Audible Message Waiting	668, 730	Automatic Charge Display	112
Audible Notification	313	Automatic Circuit Assurance; Call Park Timeout;DID/Tie/	
Audio 802.1p Priority	401	ISDN/SIP Treatment	240
Audio Codec	380	Automatic Exclusion	131
Audio Group	71	Automatic Exclusion by COS	304
Audio PHB Value	401	Automatic Exclusion Coverage/Hold	304
Audio Resource Reservation Parameters	402	Automatic Exclusion with Whisper Page	305
Audio Source Location		Automatic Moves	668
Audio Group	71	Automatic Selection of DID Numbers	331, 687
Audix		Automatic Wakeup	474
Agent Login ID	42	Automatic Wakeup Call Access Code	232
AUDIX Mach ID	506	Aux Work	580

Aux Work Access Code	229	Board Location	370
AUX work mode	42	Brazil Collect Call Blocking	120
Aux Work Reason Code Type	296	Break (msec)	848
Agent Login ID	42	Bridge ID	888
Auxiliary Board for Announcement	331	Bridged Appearance Origination Restriction	179, 678
Available Agent Adjustments for BSR	282	Bridged Call Alerting	669
Avaya Aura Media Server	750	Bridged Calls	693
Avaya Aura MS	751	Bridged Idle Line Preference	670
Avaya InSite Knowledge Base	897	Bridging Tone	259
Avaya Media Server	503	BSR	86
Avaya support website	897	BSR Application	877
B		BSR Available Agent Strategy	
Backup Alerting	150	Vector Directory Number	877
Backup MFSS CCA-ID	315	BSR Local Treatment	878
Backup Servers in Priority Order	406	BSR Reply-best DISC Cause Value	450
Backward Cycle Timer (sec)	513	BSR Tie Strategy	282, 878
Band	591	Build Refer-To URI of REFER From Contact For NCR	
Barge-In Tone	143	Trunk Group	859
Barrier Code	582	Building	
Barrier Code Length	583	Station	682
Base Parameter Set	763	bulletin board	79
Base Tone Generator Set	783	Bulletin Board text lines	79
Basic		Business Advocate	739
Call Vector	86	Busy	
Basic Call Setup	743	Coverage Path	166
Basic Mode Activation	234	Busy Auto Callback without Flash	669
Basic Supplementary Services	743	Busy Indicator for Call Parked on Analog Station Without	
basic vectoring	741	Hardware?	155
Bcc		Busy Indicator lamp	155
Attendant Console	65	Busy Threshold	434, 789
BCC	645, 789	Busy Tone Disconnect	841
BCC Value	592	Busy Tone Disconnect for Analog Loop-start Trunks	722
BCIE	592	Busy, Not Equipped	533
BCMS (Basic)	739	Busy/Reorder/Intercept Tone Timeout (Seconds)	302
BCMS/VuStats Abandon Call Timer (seconds)	291	BUTTON ASSIGNMENTS	684
BCMS/VuStats LoginIDs	291	button labels	475
BCMS/VuStats Measurement Interval	291	Button Type Customization Restriction	80
BCMS/VuStats Service Level	739	BW Management Option	423
Bearer	645	Bypass If IP Threshold Exceeded	609
Bearer Capability Information Element	592	C	
Begin Time	777	CA-TSC Request	
Beginning Station	317	Route Pattern	593
Best Service Routing	76, 741	Cable	682
Bit Rate		Cadence Classification After Answer	753
ISDN Trunk Group	445	Cadence Step	754, 784
Trunk Group	807	call appearance buttons	475
Block CMS Move Agent Events	177	Call Appearance Display Format	671
Block Enhanced Conference Transfer Display		Call Appearance Selection for Origination	143
Class of Restriction	120	Call Category for Vector ii-digits	521
Block Hang-up by Logged-In Auto Answer Agents	288	Call Center Elite	300
Block Sending Calling Party Location in INVITE	859	Call Center Release	739
Block Service Observe Across Tenants	277	call clearing	
Block Transfer Displays	120	tandem calls	860
Blocked Precedence Level	533	Call Control 802.1p Priority	402
Board	426	Call Control PHB Value	401

Call Counts	562	Calling Party Number Conversion for Tandem Calls	90
Call Coverage/Call Forwarding	708	Calling Party Number to INTUITY AUDIX	358
Call Forward Override	714	Calling Party Restriction	112
Call Forwarding Activation Busy/DA	214	CALLING PERMISSION	128
Call Forwarding All Calls	131	Calling permissions	762
Call Forwarding Busy/DA	132	Callr-info Display Timer (sec)	293
Call Forwarding Enhanced	135	Calls Allowed	647, 694
Call Forwarding Enhanced Activation/Deactivation	214	Calls In Queue Warning	150
Call Forwarding Enhanced Status	214	Calls Share IP Signaling Connection	609
Call Handling Preference		Calls to Hunt Group — Record	97
Agent Login ID	47	Calls Used	583
call identifiers	476	Calls Warning Extension	338
Call Limit		Calls Warning Threshold	339
Stations With Off-PBX Telephone Integration	694	CAMA Numbering format	63
Call Log Notify		CAMA Numbering Format	95
Configuration Set	144	Cama Outgoing Dial Guard (msec)	841
CALL MANAGEMENT SYSTEMS	290	Cama Wink Start Time (msec)	841
Call Origin	878	Can Be a Service Observer	113
Call Park Access Code	214	Can Be Picked Up By Directed Call Pickup	113
Call Park Timeout	250	Can Be Service Observed	113
Call Park Timeout Interval (minutes)	240	Can Change Coverage	114
Call Pickup	313	Can Use Directed Call Pickup	114
Call Pickup Access Code	215	Cancel Do-Not-Disturb for Wakeup Calls	335
Call Pickup Alerting	313	capacity profile max sessions	504
Call Pickup for call to Coverage Answer Group	313	capacity profile MPU estimate	504
Call Pickup on Intercom Calls	249, 313	Carrier Medium	434
CALL PROCESSING OVERLOAD MITIGATION	248	CAS	150
Call Rate	888	CAS Back-Up Extension	151
Call Selection Measurement	288	CAS Branch	730
Call Still Held	554, 808	CAS Main	730
Call Type	189	CAS Remote Hold/Answer Hold-Unhold Access Code	215
Survivable ARS Analysis Table	697	CBC Service Trunk Group Allocation Plan Assignment	
Call Type AAR only		Schedule	468
AAR and ARS Digit Analysis Table	19	CBC Trunk Group Usage Allocation	469
Call Type Analysis Behavior When Extension Not		CC	690
Assigned		Stations With Off-PBX Telephone Integration	691
Call Type Digit Analysis	194	CCA Enabled	315
Call Type ARS only		CCSA	112
AAR and ARS Digit Analysis Table	19	CDR	497
Call Type Digit Analysis Table	82	CDR Account Code	215
Call Vector	84	CDR Account Code Length	98
Call Waiting Indication	670	CDR Date Format	98
Call Work Codes	739	CDR FEAC	779
Call-Type Ordering Within Priority Levels?	157	CDR for Calls to EC500 Destination	145
Called Len	366	CDR for Origination	145
Called Number	367	CDR Privacy	670
called number-only	862	CDR Reports	435, 551, 789
Called Party Restriction	112	CDR retention	98
Caller ID calls	241	CDR system parameters	97
Caller ID for Service Link Call to H.323 1xC	808	Cell Phone Number	645
caller ID format for Extension to Cellular telephones	144	Cellular Voice Mail Detection	
Caller ID Message Waiting Indication	646	Configuration set	145
Caller ID on Call Waiting Delay Timer (msec)	276	Centralized Attendant	744
Caller ID on Call Waiting Parameters	276	Centralized Attendant Service (CAS)	149, 150
Caller's Emergency Service Identification (CESID)	95	Centralized Automatic Message Accounting (CAMA)	95
Calling Number Style	144	CESID	95
Calling Number Verification	144	CESID I Digits Sent	790

Chained Call Forwarding	302	Clear Callr-info	293
Chan Port	629	Clear VuStats Shift Data	292
change aar analysis	18	Clear-channel Mode	385
change aar digit-conversion	23	Client Room	132
change abbreviated-dialing group	28	Client Room Class of Service	60
change abbreviated-dialing personal	29	Client Room Coverage Path Configuration	327
change abbreviated-dialing system	31	Clock Time Forced Logout Reason Code	299
change administered-connection	37	Cluster ID	699
change agent-loginID		Cluster Preferred CM List	137
auto	41	cluster Session Manager	139
change alias station	50	clustered	610
Change all board location translations from board	372	CMS appl mis	
change alphanumeric-dial-table	51	Feature-related system parameters	290
change ars analysis	18	CO Type	791
change ars digit-conversion	23	coach on single step conference (SSC)	285
change ars toll	58	Code	471 , 856
change authorization-code	74	Code Calling IDs	141
change best-service-routing	76	Code Calling Playing Cycles	497
change bulletin-board	79	Code Calling, COR	497
change button restriction	80	Code Calling, TAC	498
change calltype analysis	82	Code Calling, TN	498
Change COR by FAC	731	Code-Pattern Choice Assignments	590
Change Coverage Access Code	215	Codec Set	381 , 400
Change Day (Start)	184	codec-set	408
Change Day (Stop)	184	CODES	590
change media-server	503	Codeset to Send Display	446
Change Station Extension	108	Codeset to Send National IEs	446
change vector	84	Collect All Digits Before Seizure	514
changes to platform support	15	Collected Digits	467
Character Set for QSIG Name	461	collection	
Charge Advice	435 , 445	delete	895
Charge Conversion	451 , 554 , 824	edit	895
Charge Display Update Frequency (seconds)	277	generating PDF	895
Charge Type	554 , 824	sharing content	895
Check-In status function	132	Comm Type	790
Check-Out status function	132	command parameters	690
chime code extensions	141	Commercial Cost Avoidance (CCA) Related System	
chime codes	141	Parameters	315
CINFO	86 , 741	Communication Interface Processor Channels	574
Circuit Type		Communication Manager	503 , 600 , 751
Signaling Group	610	Communication Manager software	16
City/Area	484	Community Size	699
class of restriction		Community String	421
calling permission	127	Companding Mode	486
Service Observing Permission	127	Computer Telephony Adjunct Links	731
Class of Restriction	111 , 125	Condition Code 'T' for Redirected Calls	98
Class of service	130	Conf/Trans On Primary Appearance	671
class of service page 2		Conference Access Code	883
Ad hoc Video Conferencing	134	Conference Controller	883
Call Forwarding Enhanced	134	conference parties	259
Masking CPN/Name Override	134	Conference Tone	260
Priority Ip Video	134	conference tone on service observing	285
Class of service: page 1		Conference Type	884
Automatic Callback	131	Conference/Transfer	265
Automatic Exclusion	131	configuration set	690
Call Forwarding Busy/D Client Room	131	location for routing incoming overlap calls	147
Console Permissions	131	Configuration Set	142 , 646 , 691

Configuration Set Description	146	COR	176, 435, 570, 583, 646, 791, 873
Confirmed Answer	146	Access Endpoint	35
Connect		Agent Login ID	43
Signaling Group	610	Announcements Audio Sources	54
CONNECT Reliable When Call Leaves ISDN	446	Attendant Console	60, 66
Connected to CO	824	Authorization Code COR Mapping	74
Connection Number		console parameters	151
Administered Connection	37	Group Paging Using Speakerphone	321
Connectivity Time	704	Hunt Group	339
Connectivity Timer	430	Terminating Extension Group	768
Console parameters	149	COR Description	114
Console Parameters		COR Number	114
abbreviated dialing	154	COR to Use for DPT	254
common shared extensions	155	COR/FRL check for Covered and Forwarded Calls	709
incoming call reminders	155	Cord Length	682
timing	156	COS	570, 583
Console parameters: page 3	157	Access Endpoint	35
Console Permissions	132	Attendant Console	60, 66
Console Type		console parameters	151
Attendant Console	59	MOC Control	135
Contact Closure Activation	132	Station	646
Contact Closure Close Code	215	COS Group	132, 759
Contact Closure Open Code	216	COS Name	132
Contact Closure Pulse Code	216	Count	155
content		Country	555, 791
publishing PDF output	895	Country code for CDR	487
searching	895	country options table	718
sharing	895	Country Protocol	610, 687
sort by last updated	895	Coverage - Caller Response Interval (seconds)	708
watching for updates	895	Coverage After Forwarding	670, 714
Contiguous	470, 572	Coverage Answer Group	161
Continue Daily Until Completed	317	Coverage Module	687
Continuous		Coverage Msg Retrieval	670
Administered Connection	38	Coverage Of Calls Redirected Off-Net Enabled	716
Controlled Outward Restriction Intercept Treatment	249	coverage path	215
Controlled Restrictions Configuration	327	Coverage Path	163, 339, 551
Controlled Station to Station Restriction	250	Agent Login ID	43
Controlled Termination Restriction	250	Terminating Extension Group	768
Controlled Toll Restriction Intercept Treatment	252	Coverage Path 1 or Coverage Path 2	645
Controlled Toll Restriction Replaces	253	Coverage Path Description	164
Controlling Adjunct		Coverage Path Number	163
Hunt Group	347	CPN Prefix	92, 547
Conv	864	CPN, ANI for Dissociated Sets	246
AAR and ARS Digit Conversion Table	24, 568	CPN/ANI/ICLID Parameters	274
Converse Data Return Code	232	CPN/ANI/ICLID Replacement for Restricted Calls	274
Converse First Data Relay	283	CPN/ANI/ICLID Replacement for Unavailable Calls	274
Converse Second Data Relay	283	Create Universal Call ID (UCID)	257
Converse Signaling Pause	283	Crisis Alert Code	171
Converse Signaling Tone	283	crisis alert notification	256
Conversion to Full Public Number - Delete	405	Crisis Alert System Parameters	170
Conversion to Full Public Number - Insert	405	Crisis Alert to a Digital Pager	171
Convert 180 to 183 for Early Media	859	Criteria for Logged Off/PSA/TTI Stations	712
Convert First Digit End-of-Dial To	514	CTI link	179
Convert ISDN Public Numbers to International for		CTI Link	175, 176
Display	487	CTI Link Number	316
Location Parameters	487	CTI Stations	
Copy ASAI UUI During Conference/Transfer	288	System Parameter Customer Options	745

Currency Symbol	451 , 555 , 825	Default Reason Code	580
Custom Selection of VIP DID Numbers	332 , 687	Default Route Digit	534
Customer Telephone Activation (CTA) Enabled	246	Default RTCP Report Period (secs)	417
Customizable Labels	646	Default Server IP Address	418
Customize	492	Default Server Port	418
Customize Parameters	763	Default Service Domain	534
Cut-Through	808	Definition for Rooms in State 1 through 6	335
Cvg Enabled for VDN Route-To Party	164	Del	865
Cvg Of Calls Redirected Off-net	731	AAR Digit Conversion Table	24 , 568
CVG Path	774	ARS Digit Conversion Table	24 , 568
Cyclical Hunt	808	Incoming Call Handling Treatment	367
D		Delay	
dadmin login	732	feature related system parameters	281
Daily Wakeup	332	Delay Call Setup When Accessed Via IGAR	809
Data Extension		Delay for USNI Calling Name for Analog Caller ID	
Attendant Console	66 , 684	Phones (seconds)	269
Data Item	105	Delay Sending Release	255
Data Module		Delete	92
Attendant Console	60	Call Type Digit Analysis Table	82
Station	648	Deluxe Paging and Call Park Timeout to Originator	250
Data Option	687	Deny	
Data Origination Access Code	217	Survivable ARS Analysis Table	697
Data Privacy	132	Description	538 , 869 , 885
Data Privacy Access Code	217	Holiday Table	324
Data Restriction	451 , 552 , 671 , 825	Service Hours Table	606
Date		Desktop Video Conferencing	64 , 676
Bulletin Board	79	Dest # IP address	422
Date (Start)	184	Dest. Digits	631
Date (Stop)	184	Destination	873
Date and Time	180	Administered Connection	37
Date Format on Terminals	278 , 487	Destination Code Control	236 , 565
Day of the Month	180	Destination Node	426 , 575
Day of the Week	181	Destination Port	575
Daylight Saving Rule	181 , 886	detection and alarms	418
Daylight Saving Rules	183	Dial Access	436 , 792
DC	374	Dial Code	
DCC Activation/Deactivation	135	Enhanced List	27 , 28 , 30 , 31
DCP Terminal-parameters Plan	488	Personal List	27 , 28 , 30 , 31
DCP/Analog Bearer Capability	611	System List	27 , 28 , 30 , 31
DCS (Basic)	731	DIAL CODE	
DCS Call Coverage	731	7103A Button Assignment	33
DCS to QSIG TSC Gateway	187	Dial Detection	809
DCS with Rerouting	731	Dial Plan Analysis Table	189
DCS/QSIG Intw	593	Dial Plan Parameters	193
Deactivation Code	237	Dial Plan Transparency in Survivable Mode	406
Decimal Point	452 , 555 , 825	dial prefix	690
dedicated Voip channel licenses	504	Dial Prefix	647 , 691
Default ANI	514	Dial Tone Validation Timer (sec)	721
Default Announcement Extension	332	Dialed String	190
Default Call Appearance Display Format	200	AAR and ARS Digit Analysis Table	20 , 565 , 698 , 779
Default COR	246	AAR and ARS Digit Conversion Table	20 , 565 , 698 , 779
Default Coverage Path for Client Rooms	327	Dialed String length (Min, Max)	82
DEFAULT DIALING	67	Dialed String Match	
Default Dialing Abbreviated Dialing Dial Code	684	Call Type Digit Analysis Table	82
Default Multimedia Outgoing Trunk Parameter Selection	244	DID Busy Treatment	260
		DID to Attendant	158
		DID-LDN Only to LDN Night Ext	151

DID/Tie/ISDN/SIP Intercept Treatment	241	Display Language (<i>continued</i>)	
Diffserv/TOS Parameters	401 , 404	Attendant Console	60
Digit Absorption	199	Display Mode	764
Digit Absorption List	792	Display Module Button Assignments	70
Digit Handling (in/out)	447	Display Notification for a locked Station	302
Digit to Insert/Delete	332	Display Notification for Call Forward	303
Digit Treatment	810	Display Notification for Do Not Disturb	303
Digital Loss Group	447 , 556 , 809	Display Notification for Enhanced Call Forward	302
Digital Loss Plan Modification	731	Display Notification for Limit Number of Concurrent Calls	303
Digits	809	Display Notification for Posted Messages	303
Digits to Record for Outgoing Calls	98	Display Notification for Send All Calls	303
Direct Agent Announcement Extension	282	Display Parameters	200
Direct Agent Calling	114	Display PBX data on telephone	222
direct agent calls	43	Display Room Information in Call Display	332
Direct Agent Skill		Display Text	274
Agent Login ID	48	Display VDN for Route-To DAC	879
Direct Agents Calls First		Distinctive Audible Alert	
Agent Login ID	43	Station	672
Direct Dial Access-Trunk	508	Distinctive Audible Alerting	264 , 761
Direct Inside Access	508	Diversion by Reroute	462
Direct IP-IP Audio Connections	611	Diverted Party Identification	709
Attendant Console	63 , 305 , 311 , 672	DLI Voltage Level	764
Direct Trunk Group Select Button Assignments		DND/SAC/Go to Cover	
Attendant Console	62	All	166
direct-WAN	408	Coverage Path	166
Directed Call Pickup	113 , 114 , 313	Don't Answer	166
Directed Call Pickup Access Code		Do Not Disturb Restriction	250
Feature Access Code page 2	217	Do Not Send Group B Signals to CO	521
Directed Group Call Pickup Access Code	217	Do not share port with unnamed registration	678
Direction	436 , 792	document changes	17
Directory Search Sort Order	721	documentation	
Disable call classifier for CCRON over ISDN trunks	716	Communication Manager	892
Disable call classifier for CCRON over SIP trunks	716	documentation center	895
Disable Following a Security Violation	583	finding content	895
Disconnect Information in Place of FRL	99	navigation	895
Disconnect on No Answer by Call Type	721	documentation portal	895
Disconnect Signal Error		Domain	210
Trunk Group	841	Download Flag	421
Disconnect Supervision-In	556 , 810	Download Set Type	317
Disconnect Supervision-Out	447 , 810	Download Station Type	317 , 772
Disconnect Timing (msec)	490 , 811	Drop Parking User From the Call After Timeout	250
Disconnect Type	811	Drop Treatment	812
Disp Client Redir		DS1 Echo Cancellation	452 , 732 , 825
Attendant Console	60	DS1 MSP	732
Disp Parm	483	DSN Number	688
Display Authorization Code	253	DSN Term	452 , 826
Display between Locations with these Display		DST	485
Parameters	200	dst rgn	409
Display Caller ID	648	DTMF Duration - Tone (msec)	171
Display Calling Number for Room to Room Caller ID		DTMF Duration On	509
Calls	241	DTMF Feedback Signals For VRU	739
Display Cartridge	648	DTMF over IP	611
Display Character Set	722	DTMF Tone Feedback	260
Display Client Redirection	671	Dual Wakeup	333
Display Connected Name/Number for ISDN DCS Calls	269	Duplex	
Display Information With Bridged Call	308	IP Interfaces	396
Display Language	648	ISDN Trunk Group	448

Duplex (<i>continued</i>)		Enable Reachability	
Trunk Group	812	Station Domain	685
Duplicate Station	203	Enable Session	206
Duplicate Vector	204	Enable SIP Agent Reachability?	300
Duration		Enable Voice/Network Stats	416
Administered Connection	38	Enabled	428 , 431
Duration (msec)	784	Signaling Group	631
Duration Maximum	754	Survivable Processor	702
Duration Minimum	754	encrypted SRTCP	381
Duration of Call Timer Display	305	End Day	324
DVC	64 , 676	End Hour	325
Dyn CAC	409	End Min	325
Dynamic Advocate	740	End Month	325
Dynamic Percentage Adjustment	348	End OCM After Answer (msec)	488
Dynamic Queue Position	348	End of OCM Intercept Extension	488
Dynamic Threshold Adjustment	348 , 740	End Time	777
E		End to end signaling	466 , 848
EAS	41 , 87	End-to-End Signaling	812
EAS-PHD	49 , 740	End-to-End total loss (dB) in a n-party conference	492
EC500 Self Administration Access Code	217	Ending Station	318
EC500 state	672	Endpoint	254
EC500delay deactivation state	146	Endpt ID	649
Echo Digits Dialed	65	Endpt Init	649
ECM	385	Enforce PNT-to-PNT Restrictions	281
Edge	702	Enforce SIPS URI for SRTP	612
Edit Dialing on 96xx H.323 Terminals	278	Enhanced Abbreviated Dial Length	244
Emergency Access	158	Enhanced Abbreviated Dialing	475
Emergency Access Redirection Extension	251 , 254	Enhanced Attendant Notification	154
Emergency Access to Attendant	732	Enhanced Call Pickup Alerting	313
Emergency Access To Attendant Access Code	217	Enhanced Call Pickup Delay Timer (sec.) Display	314
Emergency Location Ext	672	Enhanced Callr-Info display for 1-line display phones	678
Attendant Console	63	Enhanced Conferencing	
emergency location extension	108	System parameters customer options	732
Emergency Location Extension	109 , 378	Enhanced EC500	732
EMU Inactivity Interval for Deactivation	248	Enhanced EC500 Activation	217
EMU Login Allowed	649	Enhanced EC500 Deactivation	218
Enable	575	Enhanced List	
Administered Connection	37	Abbreviated Dialing	26
Enable 'dadmin' Login	732	Enhanced Mode Activation	234
Enable Busy Tone Disconnect for Analog Loop-start		Enhanced PSA Location/Display Information Enabled	247
Trunks	722	Enterprise	702
Enable CDR Storage on Disk	99	Enterprise Mobility User Activation	218
Enable Command Logging	495	Enterprise Mobility User Deactivation	218
Enable Detection and Alarms	418	Enterprise Survivable Server	732
Enable Dial Plan Transparency in Survivable Mode	255	Enterprise Wide Licensing	733
Enable Enbloc Dialing without ARS FAC	276	EPSCS	112
Enable Ethernet Interface	394 , 396	Erase 24xx/14xx/94xx User Data Upon:	
Enable Failover Event Package Subscription	629	Dissociate or unmerge this phone	123
Enable Inter-Gateway Alternate Routing	255	EMU login or logoff at this phone	124
Enable Layer 3 Test	612	ESS Administration	732
Enable on Survivable Processors (ESS and LSP)	628	Established	631
Enable on the Main Processor(s)	628	ETA Node Number	194
Enable PE for H.248 Gateways	700	ETA Routing Pattern	194
Enable PE for H.323 Endpoints	700	Ethernet Link	394 , 397
Enable Q-SIP	859	ETSI CCBS Support	613
		Event Minimization	178
		Every User Responds	171

Expand ISDN Numbers to International for 1XCES	312	Facility Busy Indicators	226
Expansion Module	649	Facility Coding	540
Expected Call Handling Time (sec)	349	Facility Test Calls Access Code	219
Expected Digits	813	Facility Type	540
Expert Agent Selection	41	Failover Group Domain Mapping	210
Expert Agent Selection (EAS)	740	Failover/failback Signaling-group Pair	629
Expert Agent Selection (EAS) Enabled	282	Far End Resource Info	888
Expert Agent Selection-Preference Handling Distribution ...	49	Far End Test Line No.	436
Expiration Date	584	Far-end Domain	613
Ext	141, 560, 561	Far-end Listen Port	614
Attendant Console	67	Far-end Network Region	614
Coverage Answer Group	162	Far-end Node Name	
Group Paging Using Speakerphone	321	Signaling Group	614
Hunt Group	365	Far-End Test No	849
Intercom Group	374	Fast Connect on Origination	
Listed Directory Numbers	481	Configuration Set	147
Terminating Extension Group	770	FAX Mode	
Ext Alert (TAAS) Extension	152, 759	IP codec set	385
Ext Alert Port (TAAS)	151, 759	FB Timer	386
Ext Code	543, 547	FEAC	115
CAMA Numbering Format	95	Feature Access Code	212
Ext Len	544, 548	Feature Access Code (FAC):Refresh Terminal	
CAMA Numbering Format	95	Parameters Access Code	223
Extended Call Fwd Activate All	218	Feature Access Code page 2	
Extended Call Fwd Activate Busy D/A	218	Directed Call Pickup Access Code	217
Extended Cvg/Fwd Admin	732	FEATURE BUTTON ASSIGNMENTS	68
Extended Forwarding All	133	Feature Invocation by In-Call DTMF Code	
Extended Forwarding B/DA	133	Configuration Set	147
Extended Group Call Pickup	251, 314	Feature Module	688
Extended Group Call Pickup Access Code	219	Feature Plus Ext	270
Extended Group Number	207, 561	Feature-related system parameters	238
Extended Loop Range	813	File Size	771
Extended Pick-Up Group	207	File Status	771
Extension	209, 570, 873	File to Retrieve	771
Announcements Audio Sources	54	Filename in Memory	771
Attendant Console	61	finding content on documentation center	895
CTI Link	176	finding port matrix	894
Intra-Switch CDR	377	Firmware Station Download	317
Station	648	First Announcement Delay (sec)	359
EXTENSION DISPLAY FORMATS	196, 200	First Announcement Extension	359
Extension only label for Team button on 96xx H.323		Fixed	469
terminals	274	Fixed TEI	649
Extension to Cellular call security	143	Flash	160, 534
Extension to Receive Failed Wakeup LWC Messages	333	Flash Access Code	219, 236
Extensions Administered to have an MCT-Control Button	208	Flash Length (msec)	842
extensions for chime codes	141	Flash Override	160, 534
Extensions to Call Which Activate Features by Name	209	Flash Override Access Code	236
External Coverage	508	Flashhook Interval	490
External Coverage Treatment for Transferred Incoming		Flexible Billing	733
Trunk Calls	712	Floor	
External Device Alarm Admin	733	Station	682
External Ringing for Calls with Trunks	266	Force Entry of Acct Code for Calls Marked on Toll	
		Analysis Form	99
F		Force Phones and Gateways to Active LSPs	418
Facilities Restriction Level	115	Forced Agent Aux Work by Location Reason Code	300
Facility Access Trunk Test	114	Forced Agent Aux Work by Skill Reason Code	300
		Forced Agent Logout by Location Reason Code	300

Forced Agent Logout by Skill Reason Code	300
Forced Agent Logout Time	
Agent Login ID	43
Forced Entry of Account Codes	115 , 733
Forced Entry of Stroke Counts or Call Work Codes	356
Forced SIP Agent Logout for Unreachable Reason Code	300
Format	453 , 813
Forward Cycle Timer (sec)	514
Forward Disconnect Timer (msec)	490
Forward PMS Message to INTUITY Lodging	328
Forwarded Destination	679
Frames Per Pkt	382
FRL	115 , 593
FROM / TO	493
From IP Address	378
Fully Restricted Service	115

G

G3 Version	724
G3V4 Adv Route	87
G3V4 Enhanced	87
Gatekeeper Priority	394
Gateway	426
Glare	842
Glare Handling	826
Global Call Classification	733
Global Classifier Adjustment (dB)	754
Group	
Attendant Console	61
Group Control Restrict Activation/Deactivation	219
Group Controlled Restriction	116
Group Extension	321 , 340 , 768
Hunt Group	364
Group II Called Party Category	515
Group II Category For MFC	116
Group List	
Abbreviated Dialing	28
Group List	29
Group Member Assignments	471 , 561 , 855
Group Name	321 , 437 , 552 , 793
Audio Group	72
Coverage Answer Group	162
Hunt Group	340
Group Number	321 , 437 , 552 , 561 , 769 , 793
Coverage Answer Group	162
Hunt Group	340 , 364
Intercom Group	374
Signaling Group	614
Group Page	
Zip Tone Timer (secs)	322
Group Paging Using Speakerphone	320
alert	320
Group Port	
Announcements Audio Sources	54
Group Timeout (secs)	322
Group Type	437 , 448 , 552 , 793 , 813

Group Type (<i>continued</i>)	
Hunt Group	340 , 364
Signaling Group	614
Grp No	593
Gtway to	575

H

H.235 Annex H Required	615
H.245 DTMF Signal Tone Duration (msec)	615
H.248 Gateways	393
H.320 Conversion	
Attendant Console	61 , 673
H.320 Desktop Video Conferencing	64 , 676
H.323 IP Endpoints	403
H.323 Link Bounce Recovery	403
H.323 Security Profiles	406
H.323 Station Outgoing Direct Media	615
H0	471 , 572
H11	471 , 572
H12	471 , 572
Handset Expander Enabled	764
Headset	682
Hear Zip Tone Following VOA?	278
Held Call UCID	467
Hide Post-Dialing DTMF On List Trace	601
High-available cluster type	504
Holiday After Coverage	165
Holiday Coverage	164
Holiday Table	164
Holidays	87 , 742
Home	679
Homing Policy for Users	124
Hop Lmt	594
Hospitality	60 , 327 , 733
Hospitality (G3V3 Enhancements)	733
Hot Desking Enhancement Station Lock	247
HOT LINE DESTINATION	67
Hour	181
Housekeeper Information Configuration	328
Housekeeping Status (Client Room) Access Code	232
Housekeeping Status (Station) Access Code	233
Howler After Busy	723
Hundreds Select	63
Hundreds Select Button Assignments	
Attendant Console	63
hunt group	337
Hunt Group Busy Activation/Deactivation	219
Hunt Group Location	710
Hunt Group:Service Level Target (% in sec)	352
Hunt-to Station	649
hyperactive registration alarms	419
Hyperactive Registration Window (minutes)	419

I

IAA	241

IAS (Branch)	152	Incoming Partial Dial (sec)	843
IAS Att. Access Code	152	Incoming Rotary Timeout (sec)	816
IAS Tie Trunk Group No.	152	Incoming Seizure (msec)	843
IC Adjunct Routing	178	Incoming Tone (DTMF) ANI	827
ICHT	366	Incoming/Outgoing	523
ID Range Start/End	889	Incomplete Dial Timer (sec)	515
Identity for Calling Party Display	860	Increment (Start)	185
Identity When Bridging		Index	562
Feature-related system parameters	275	Individual Attendant Access	158
Idle	308	Inflow Threshold (sec)	349
Idle Appearance Preference	688	Information Transfer Capability	595
Idle Code	616	Initial IP-IP Direct Media	616
Idle Traffic Interval (seconds)	403	Insert	92
Idle/Active Ringing (Callmaster)	688	Call Type Analysis Table	83
IGAR	255 , 409	Incoming Call Handling Treatment	367
IGAR Over IP Trunks	255	Insert Digits	
Ignore Network Answer Supervision	716	Uniform Dial Plan Table	865
Ignore Rotary Digits	673	Inserted Digits	506 , 594
Immediate	160 , 535	Instance	754
Immediate Access Code	236	integrated announcement	55
Immediate Redirection on Receipt of PROGRESS		Integrated Announcement Boards	370
Inband Information	712	Integrated Announcement Extension	333
IMS Enabled	616	Integrated Announcement Translations	372
in-band tone	860	Inter-Exchange Carrier (IXC) Codes	376
In-VDN Time	467	Inter-Gateway Alternate Routing	255 , 405
INADS	603	Inter-Location	202
Inc Attd Call Record	99	Inter-location Loss Group	492
Incoming	520	Inter-PBX Attendant Service (IAS)	149
Incoming Backward Signal Types (Tones to CO)	526	Inter-region IP-IP Direct Audio	400
Incoming Backward Signal Types (Tones to CO), Group		Inter-System IP DTMF Transmission Mode	419
A	526	Intercept Treatment on Failed Trunk Transfers	278 , 306
Incoming Backward Signal Types (Tones to CO), Group		Intercom Group	374
B	527	Interdigit Pause	723
incoming budget	795	Interface	617
Incoming call budget high threshold	796	Interface Channel	575
Incoming call budget low threshold	796	Interface Companding	617
Incoming Call Handling Treatment	366	Interface Link	576
Incoming Call Type	515	Interflow VDN	
Incoming Calling Number - Delete	814	Best Service Routing	76
Incoming Calling Number - Format	439 , 814	Interflow-qpos EWT Threshold	284
Incoming Calling Number - Insert	815	Internal Alert	827
Incoming Calling Number Insert	439	Internal Auto-Answer of Attd-Extended/Transferred Calls	241
Incoming Destination	796	Internal Coverage	508
Incoming Dial Guard (msec)	842	Internal Users	124
Incoming Dial Tone	815	International Access Code	275 , 488
Incoming Dial Type	815	International Call Routing	275
Incoming Dialog Loopbacks	616	International CPN Prefix	270
Incoming Disconnect (msec)	842	Interposition	158
Incoming Disconnect Send (msec)	843	Interruptible	580
Incoming Forward Signal Types (Tones from CO)	524	Interruptible Aux	356
Incoming Forward Signal Types (Tones from CO), Group		Interruptible Aux Deactivation Threshold	
I	524	Hunt group	356
Incoming Forward Signal Types (Tones from CO), Group		Interruptible Aux Notification Timer (sec)	293
II	525	Interruptible Aux threshold	356
Incoming Glare Guard (msec)	843	Interruptible Aux Threshold	
Incoming LDN Extension	406	Hunt group	356
Incoming Number Format	93	INTERVAL 1, 2, 3	777

Interval between OPTIONS messages (seconds)	630	ISDN-PRI	734
Interval For Applying Periodic Alerting Tone	268	Italian DCS Protocol	280
Intervening-regions	409	Italian Protocol Enabled	281
Interworking Feat-flag	100	ITC	595 , 797
Interworking Message	617	Access Endpoint	35
Interworking of ISDN Clearing with In-Band Tones		ITC (Information Transfer Capability)	650
Trunk Group Protocol Variations	860	IXC	595
Interworking with DCS	744	IXC Access Number	376
Intra-Location	202	IXC Code Format	376
Intra-region IP-IP Direct Audio	400	IXC codes	376
Intra-Switch CDR	100 , 377	IXC Name	376
Intra-System IP DTMF Transmission Mode	420	IXC Prefix	376
Intrusion Tone	261		
Invalid Number Dialed Display	307	J	
Invalid Number Dialed Intercept Treatment	307	Jack	683
Invoke ID for USNI Calling Name		Journal/Schedule Endpoint	328
Trunk Group	827		
inward restriction override	117	K	
IP Address	73 , 413	KB	
Survivable Processor	700	Support site	897
IP Address Mapping	378	Keep Bridged Information on Multiline Displays During	
IP Attendant Consoles	733	Calls	314
IP codec Set	380	Keep Held SBA at Coverage Point	713
IP DTMF transmission mode	420	Keep-Alive Count	403
IP DTMF Transmission Mode	419	Keep-Alive Interval (seconds)	403
IP Interfaces	393		
IP Media Parameters	380	L	
IP network region	398 , 403	Label Language	
IP Network Region	398	System List	32
IP network region page 3	405	Labels	696
IP Node Names	413	LABELS FOR 2420/4620 STATIONS	
IP parameter emergency location	108	System List	32
IP Parameter Emergency Location	109	LAI	87
IP Phone Group ID	679	Language translations	474
IP Routing	425	Language Translations	
IP Services	428	Automatic Wakeup	474
IP Softphone	649	button labels	475
IP Softphones		Enhanced Abbreviated Dialing	475
video capable	727	Leave Word Calling	476
IP Stations	734	Malicious Call Trace	476
IP Trunks	734	miscellaneous call identifiers	476
IP Video	618 , 650	Miscellaneous features	477
IP Video Softphone	650	Property Management Interface	477
IPEI	680	Self Administration	477
IPv4 Server Port	404	Softkey Labels	478
IPv6 Server Port	404	Transfer Conference	478
IQ (appl ccr)	290	View Buttons	479
ISDN Access Code	220	Vustats	479
ISDN Caller Disp	343 , 769	LAR	595
ISDN Feature Plus	734	Last Number Dialed Access Code	220
ISDN Network Call Redirection	735	Layer 1 timer	723
ISDN Network Facilities	540	Least Occupied Agent	740
ISDN Parameters	269	Leave Word Calling	42 , 476
ISDN Precedence Call Timeout (sec)	535		
ISDN Protocol	19		
ISDN Trunk Group	433		
ISDN-BRI Trunks	734		

Leave Word Calling Cancel A Message	221	Location for Covered and Forwarded Calls	709
Leave Word Calling Message Retrieval Lock	220	Location for Routing Incoming Calls	618
Leave Word Calling Message Retrieval Unlock	220	Location Name	
Leave Word Calling Send A Message	221	Best Service Routing	77
Len	93, 865	Location Parameters	486
Length	869	Long Distance Access Code	489
CDR System Parameters	107	Time Format on Terminals	489
Length of Dial Code	375	Location Parameters:Off-PBX Feature Name Extension	
Length of Time to Remain Connected to Announcement	333	Set	489
Level 1 Code	272	Location-Based Call Type Analysis	
Level 1 Threshold (sec)	349	Call Type Digit Analysis	197
Level 2 Code	273	Locations	482
Level 2 Threshold (sec)	349	Lock	
Level of Tone Detection	278	Best Service Routing	77
Limit	745	Call Vector	88
Limit Number of Concurrent Calls Activation/Deactivation	221	Lock Messages	650
Line Appearance Conferencing	267	Log CTA/PSA/TTI Transactions	496
Line Intercept Tone Timer	261	Log Data Values	495
Line Length	816	Log IP Registrations and Events	496
Line Load Control	120	Log PMS/AD Transactions	496
Line Load Control Restriction Level	535	Logged off/PSA/TTI	
Lines Per Page	254	Coverage Path	167
Link Failure	414, 418	Logged-In ACD Agents	742
Link Loss Delay Timeout (minutes)	414	Logged-In Advocate Agents	742
Link Loss Delay Timer (minutes)	415	Logged-In IP Softphone Agents	742
Link Loss Delay Timer (sec)	618	logging all submission failures	495
Linkage	165	Logging Levels	495
List Number	199	Login	206
List1	66	IP Options	417
List1, List2, List3	154	Login Access Code	229
listed directory number (LDN)	151	Login Administration	494
Listed Directory Numbers	481	Login ID	
Listen Only Talk Path on Auto Answered Intercom Calls	125	Agent Login ID	44
Live Stream Source	54	LoginID for ISDN Display	44
LLC	488, 493	Logout	581
Load-sharing cluster type	504	Logout Access Code	230
Loc Number	484	Logout Reason Code Type	296
Loc Parm	483	Agent Login ID	44
Local Agent Preference	344	Long Distance Access Code	489
Local Call Preference		Long Hold Recall Timer	261
Agent Login ID	48	Long Holding Time (hours)	828
Local Country Code	275	Lookahead Interflow	740
Local Cvg Subsequent Redirection/CFWD No Ans		Loss Group	651
Interval (rings)	709	Loss Plans	492
Local E.164 Country Code	489	Loudspeaker Paging	497
Local Ext	632	Lower Bound (msec)	490
Local Facility	422	LRQ Required	618
Local Node	429, 703	LSC CCA-ID	316
Local Node Name	772	LSP	700
Local Node Number	195	LWC	42
Local Port	429, 703	LWC Activation	673
Local Survivable Processor	735	LWC Log External Calls	673
Location	399, 498, 588, 650, 695, 780	LWC Reception	
AAR Digit Conversion Table	20, 24, 191	Agent Login ID	44, 360, 673, 769
ARS Digit Conversion Table	20, 24, 191		
Call Type Digit Analysis Table	83		
System Parameters Customer Options	724		

M

Mach ID	187 , 576 , 632	Maximum Agent Occupancy Parameters	298
Machine ID	507	Maximum Agent Occupancy Percentage	298
Maid status function	132	Maximum Auto Reserve Agents	350
Main Number	172	Maximum Call Rate for Direct-IP Multimedia IP Codec Set	386
Maintain SBA At Principal	713	Maximum Call Rate for Priority Direct-IP Multimedia IP Codec Set	386
Maintenance Call Type	515	Maximum Concurrently Registered IP eCons	727
Maintenance Tests	453 , 571 , 828	Maximum Concurrently Registered IP Stations	727
Maintenance-related system parameters	501	Maximum Entries	544
Make (msec)	849	Maximum G700 VAL Sources	727
Malicious Call Trace	111 , 476	Maximum Length	270
Malicious Call Trace Activation	222	Maximum Number of Avaya Media Server VoIP Channels	727
Malicious Call Trace parameters	735	Maximum Number of Call Forwarding Hops	717
MALICIOUS CALL TRACE PARAMETERS	255	Maximum Number of Digits for Directed Group Call Pickup	314
Manual Unlock allowed	778	Maximum Number of DS1 Boards with Echo Cancellation	727
Manual-In Access Code	230	Maximum Number of Expanded Meet-me Conference Ports	727
Map-to Station	689	Maximum Number of External Calls Logged Per Station	245
Mapped String Alias Station	51	Maximum Number of Messages Per Station	245
Mapping Mode	651 , 695	Maximum Number of Trunks to Use for IGAR	406
Marginal / Unacceptable	851	Maximum Off-PBX Telephones	725
Marginal Threshold --Dev - 2804 Hz	851	Maximum Off-PBX Telephones - EMX	726
--Dev - 404 Hz Loss	851	Maximum Off-PBX Telephones - OPS	725
--+Dev - 2804 Hz	852	Maximum Off-PBX Telephones - PVFMC	726
--+Dev - 404 Hz Loss	851	Maximum Off-PBX Telephones - SCCAN	726
Max - 1004 Hz Loss	852	Maximum Off-PBX Telephones — PBFMC	725
Maximum C Message Noise	852	Maximum Percentage of Trunks Which Can Be Removed From Service by ATMS	853
Maximum C Notched Noise	852	Maximum Ports per Expanded Meet-me Conf	266
Min -1004 Hz Loss	852	Maximum Precedence Level	120
Minimum ERL	852	Maximum Resend Requests	516
Minimum SRL-HI	852	Maximum Size of UII IE Contents	453
Minimum SRL-LO	853	Maximum Stations	725
Mark Users as Phone	861	Maximum Suppression Time Best Service Routing	77
Mask CPN/Name for Internal Calls	125	Maximum Time Agent in ACW before Logout (sec.)	299
Masking CPN/Name Override	135	Maximum time agent in ACW before logout (sec) Agent Login ID	45
Match BCA Display to Principal Class of Service	135	Maximum TN2501 VAL Boards	728
Matching Pattern AAR Digit Conversion Table	24 , 568	Maximum TN2602 Boards with 320 VoIP Channels	728
ARS Digit Conversion Table	24 , 568	Maximum TN2602 Boards with 80 VoIP Channels	728
Uniform Dial Plan Table	866	Maximum Video Capable IP Softphones	727
Max AAR Digit Conversion Table	21 , 24 , 566 , 568 , 780	Maximum Video Capable Stations	727
ARS Digit Conversion Table	21 , 24 , 566 , 568 , 780	Maximum XMOBILE Stations	725
Max Message Size to Send	448	may-have-extra-digits	862
Max number of CA TSC	619	MCSNIC	135
Max number of NCA TSC	619	MCT group extensions, 1 to 100	208
Max Ports	889	MCT Voice Recorder Trunk Group	256
Max# Chan	469	mct-control	208
Maximum Administered Ad-hoc Video Conferencing Ports	726	Measured	350 , 454 , 828 , 873
Maximum Administered IP Trunks	726	Media Complex Ext	652
Maximum Administered Remote Office Trunks	726	Media Encryption	382 , 619
Maximum Administered SIP Trunks	726	Media Encryption Over IP	735
Maximum Agent Occupancy AUX Reason Code	298	Media Parameters	400

Media Server	503	MM Early Answer	344
media server recovery rule		Mobile Call (CTI) Extension	549
Communication Manager	750	Mobility Trunk Group	653
Media-Gateway	502	Mode	576
Meet-me Conf	88	Mode Code for Centralized Voice Mail	735
Meet-me Conference	873	Mode Code Interface	261
meet-me Conference Access Code Change	222	Mode code related system parameters	508
Member Assignment Method	440	Mode Codes (From Switch to VMS)	508
Member Range Allowed	364	Model	653
Message Center	360	Modem Mode	
Message Center AUDIX Name	360	IP codec set	387
Message Center ID	507	Modem Relay Preferred (NoAudio)	389
Message Center MSA Name	360	Modified Circuit ID Display	100
message lamp	108	Modify Tandem Calling Number	454
Message Lamp	109	Trunk Group	829
Message Lamp Ext	651	Modulation Mode	389
Message Sequence Trace (MST) Disable	222	MOH Group	511
Message Server Name	689	MOH Group Name	512
Message Waiting Configuration	328	MOH Source Location	511
Message Waiting Indication for External Calls	245	Month	181
Message Waiting Indication Subscriber Number Prefixes	506	Month (Start)	185
Message Waiting Indicator	652	Month (Stop)	185
Message Waiting Lamp Indicates Status For	282	More Members Exist	365
Message Waiting Type	653	More VDN's	205
Messaging Server Name for Messaging		Mounting	683
Agent Login ID	45	MSA Names	73
Metric	427	Mtce	410
MF ANI Prefix	116	Multifrequency Signaling	735
MF Incoming Call Trace	120	Multifrequency-Signaling-Related Parameters	
MF Interdigit Timer (sec)	490		513, 520, 523, 528
MF Signaling Intercept Treatment - Incoming	516	Multifrequency-Signaling-Related Parameters page 1	513
MF Signaling Intercept Treatment - Outgoing	516	Multimedia	88
MF Tariff Free	829	Multimedia Appl. Server Interface (MASI)	735
MFE Type	516	Multimedia Call Access Code	234
MGR1	603	Multimedia Call Handling (Basic)	735
MIA Across Skills		Multimedia Call Handling Enhanced	736
Agent Login ID	45	Multimedia Data Conference Activation	234
MIA Splits or Skills	289	Multimedia Data Conference Deactivation	234
Migrate H.248 MG to primary	748	Multimedia Early Answer	673
Milliseconds Before PMS Link Acknowledgment Timeout	329	Multimedia IP SIP Trunking	736
MIM Mtce/Mgt	652	Multimedia Multi-Address Access Code	234
MIM Support (Management Information Message		Multimedia Parameter Access Code	235
Support)	652	Multinational Locations	736
Min		Multiple Call Handling	351, 740
AAR Digit Conversion Table	21, 25, 566, 569, 780	Multiple Call Handling (Forced)	740
ARS Digit Conversion Table	21, 25, 566, 569, 780	Multiple Level Precedence	121
Min# Chan	470	Multiple Level Precedence & Preemption (MLPP)	
Minimum Agent-LoginID Password Length	46, 282	Parameters	532
Minimum Digit Length	276	Multiple Level Precedence and Preemption	736
Minimum Station Security Code Length	604	Multiple Level Precedence and Preemption (MLPP)	235
Minimum time of network stability	749	Multiple Locations	736
Minute	181	Music (or Silence) On Transferred Trunk Calls	241
Miscellaneous Call	158	Music on Hold	116
Miscellaneous features	477	Music Source	759
Miscellaneous Parameters	39	Music Sources	537
Misoperation Alerting	278	MUSIC/ANNOUNCEMENTS IP-CODEC	
MLPP Service Domain	121	PREFERENCES	423

Music/Tone on Hold	241
Mute Button Enabled	687
MWI - Number of Digits per Voice Mail	270
MWI Served User Type	45 , 674

N

Name	471 , 481 , 484 , 540 , 560–562 , 571 , 856 , 874 , 889
Administered Connection	38
Agent Login ID	46
Attendant Console	61 , 66 , 67
Call Vector	89
Coverage Answer Group	162
CTI Link	177
Duplicate Vector	205
Group Paging Using Speakerphone	322
Holiday Table	325
Hunt Group	365
Intercom Group	375
IP Network Region	399
IP Node Names	413
Network Region Group	541
Signaling Group	620
Station	653
Terminating Extension Group	770
National CPN Prefix	270
nc Trk Call Splitting	99
NCA-TSC Trunk Member	455
Near End Establishes TCP Signaling Socket	407
Near End TCP Port Max	408
Near End TCP Port Min	407
Near-end Listen Port	620
Near-end Node Name	620
Net	569 , 866
AAR Digit Conversion Table	25
ARS Digit Conversion Table	25
NET	603
Net Redir	
Best Service Routing	77
Network (Japan) Needs Connect Before Disconnect	455
Network Bits	427
Network Call Redirection	455 , 597 , 861
Network Call Transfer	620
Network Facilities	540
Network Feedback During Tone Detection	279
Network Level	273
Network region	394
Network Region	379 , 588
Network Region Group	541
Network uses 1's for Broadnet Addresses	394
Next ANI Digit	520
Next Path Number	165
Night	472 , 856
night bells	227
Night Destination	481 , 760
Night Serv	368
Night Service	59 , 797

Night Service Act. Ext.	152
Night Service Destination	344
Night Service Disconnect Timer	261
nite-serv	59
No Answer Timeout (sec)	155
No Dial Tone Conferencing	265 , 266
No Hold Conference Timeout	266
No. Del. Digits	596
No. Dgts Subaddress	596
No. of Calls	584
Node Name	394 , 397 , 588
Survivable Processor	700
Node Num	866
Node Number	542
AAR Digit Analysis Table	21
ARS Digit Analysis Table	21
Node Number (Local PBX ID)	100
Node Number Routing	542
non-SCCAN applications	817
Normal Outgoing Seize Send (msec)	844
notification using crisis alert	256
notify	
Hold notification	826
Unhold notification	826
notify sip user using crisis alert	256
NPA	597
NR Group	
IP Network Region	398
Num	
Best Service Routing	78
Number	563 , 874
Best Service Routing	78
Call Vector	89
Holiday Table	325
Service Hours Table	607
Number of Codes Administered	75
Number of Digits from PMS	333
Number of Digits in PMS Coverage Path	334
Number of Emergency Calls Allowed in Attendant Queue	251
Number of Housekeeper ID Digits	329
Number of Incoming ANI Digits	521
Number of Members	440
Trunk Group	798
Number of OPTIONS message failure before failover	630
Number of OPTIONS message successes before failback	630
Number of Outgoing ANI Digits	521
Number of Pings Per Measurement Interval	416
Number of Recordings	370
Number of Registrations within the Window	418
Number of Rings	167
Numbering — Public/Unknown Format	546
Numbering Format	829
Route Pattern	597
Numbering-Public/Unknown format	63
NxDS0	471
NXDS0	573

O

Observe on Agent Answer	879
Off	509
Off Premises Station	654
Off-Hook Alert	133
Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings)	711
Off-PBX Feature Name Extension Set	489
Off-PBX telephone configuration set	142
Off-PBX telephone feature name extensions	209
Off-PBX Telephone Mobile Feature Extensions	549
Off-PBX telephone station-mapping	690
Off-Premises Tone Detect Timeout Interval	242
Offline Call Logging	680
Station	680
On-hook Dialing on 607/2400/4600/6400/8400 Terminals	279
One-Step Recording	268
one-X Server Access	125
Options	300 , 301
OPTIONS Request Parameters	630
Originating Auto Restoration	571
Originating Extension	172 , 601
Originator	
Administered Connection	38
Other LAI Information	467
Other Related Parameters	509
Other Stations When Call Is Active	308
Other Stations When Call Is Put On-Hold	309
Outg Attd Call Record	100
Outg Trk Call Splitting	101
Outgoing	520
Outgoing ANI	830
Outgoing Backward Signal Types (Tones from CO)	529
Outgoing Backward Signal Types (Tones from CO), Group A	529
Outgoing Backward Signal Types (Tones from CO), Group B	530
Outgoing Budget	798
Outgoing call budget high threshold	799
Outgoing call budget low threshold	799
Outgoing Call Type	517
Outgoing Channel ID Encoding	455
Outgoing Dial Guard (msec)	844
Outgoing Dial Type	557 , 816
Outgoing Disconnect (msec)	844
Outgoing Disconnect Send (msec)	844
Outgoing Display	440 , 553 , 800
Outgoing End of Dial (sec)	845
Outgoing Forward Signal Absent Timer (sec)	517
Outgoing Forward Signal Present Timer (sec)	517
Outgoing Forward Signal Types (Tones to CO)	528
Outgoing Forward Signal Types (Tones to CO), Group I	528
Outgoing Forward Signal Types (Tones to CO), Group II	529
Outgoing Glare Guard (msec)	845
Outgoing II by COR	521
Outgoing Last Digit (sec)	845

Outgoing Number Format	94
Outgoing Rotary Dial Interdigit (msec)	846
Outgoing Seizure Response (sec)	846
Outgoing Shuttle Exchange Cycle Timer (sec)	491
Outgoing Start Timer (sec)	517
Outgoing Trunk Alerting Timer	125
Outgoing Trunk Disconnect Timer	121 , 125
Outgoing Trunk Groups	
Calling Party Number Conversion for Tandem Calls	94
Outpulse Without Tone	279
Outpulsing Information	848
Overlap Sending on Link-to-Link Tandem Calls	518
Overview timer to Group Queue	156

P

PA	
Agent Login ID	48
Packet Loss (%)	416
Packet Resp Timer	430 , 704
Packet Size	387
Packet Size (ms)	384
Pager Number	172
Parameters for creating QSIG selection numbers	272
Parameters for Media Gateway Alarms	419
Parameters for Network Region Registration (NR-REG) Alarms	419
Partition Routing Table	550
Partitioned Group Number	116 , 542
Pass Prefixed CPN to ASAI	271
Pass Prefixed CPN to VDN/Vector	879
Passive Signalling Station	655
Passphrase	621
Password	206 , 431
Agent Login ID	46
IP Options	417 , 421
Password enter again	
Agent Login ID	46
Password to Change COR by FAC	305
PASTE (Display PBX Data)	740
PASTE Access Code	222
PASTE Display PBX Data on telephone	
Class Of Restriction	117
Path Replacement	462
Path Replacement Method	455 , 463 , 830
Path Replacement While in Queue/Vectoring	271
Path Replacement with Measurements	271
Path Replacement with Retention	463 , 830
Pattern Choices	590
Pattern Name	597
Pattern Number	598
Pause (msec)	172 , 466 , 848
Pause Duration	634
PBX ID	831
PC Non-Predictive Reports Skill	293
PC Predictive Reports Skill	880
PCN notification	899

PCOL/TEG Call Alerting	689	PMS Log Endpoint	329
Peer Detection Enabled	621	PMS Protocol Mode	330
Peer Server	621	PMS Sends Prefix	334
Per Button Ring Control	674	Point1, Point2, ...	167
Per Call CPN Blocking Code	831	Policy Routing Table	562
Per Call CPN Blocking Code Access Code		polling feature	300
Feature access codes	222	Port	472, 498, 856
Per Call CPN Unblocking Code	831	Attendant Console	61
Per Call CPN Unblocking Code Access Code		Feature-Related System Parameters	242
Feature access codes	222	Station	654
Per Call CPN/BN	367	Port #	
Per Station CPN - Send Calling Number	674	IP Options	422
Percent Full	191, 866	Port Board Security	603
AAR Digit Conversion Table	21, 25, 566, 780	Port Board Security Notification Interval	604
ARS Digit Conversion Table	21, 25, 566, 780	Port Extension	
Percentage IPBO Blocked	566	Agent Login ID	46
Period	563	port matrix	894
Periodic Registration Timer (min)	415	Post Connect Dialing Options	148
Permanently Disable	584	Posted Messages	223, 736
Personal CO Line Group	551, 560	PPM	849
Personal List		PPS	849
Abbreviated Dialing	29	Precedence Call Timeout (sec)	535
List Number	30	Precedence Call Waiting	675
Personal List	30	Precedence Calling Access Code	235
Personal Station Access		Precedence Calling-Dialed Digit Assignment	533
Processor Ethernet	737	Precedence Incoming	831
Personal Station Access (PSA)	133, 736	Precedence Outgoing	831
Personal Station Access (PSA) Associate Code	222	Precedence Routing Digit Analysis Table	565
Personal Station Access (PSA) Dissociate Code	223	Precedence Routing Digit Conversion Table	568
Personalized Ringing Pattern	654	Preempt Emergency Call	535
PGN #	775	Preempt Method	566
PGN 1 (through PGN 8)	550	Preemptable	121
PGN/TN/COR for Covered and Forwarded Calls	711	Pref-CM-List Name	138
Phantom Calls	745	Pref-CM-List Number	138
phone number	690	Prefer H.323 over SIP for Dual-Reg station 3PCC Make	
Phone Number		Call	294
Stations With Off-PBX Telephone Integration	692	Prefer use of G.711 by Announcement Sources	423
Pickup Group	561	Prefer use of G.711 by IP Endpoints Listening to	
Pickup Group Number	207	Announcements	423
Pickup Number	207	Prefer use of G.711 by IP Endpoints Listening to Music	423
Pickup on Transfer	309	Prefer use of G.711 by Music Sources	418, 423
PIN Checking for Private Calls	314	Preferred CM List	126
PIN Checking for Private Calls Access Code	225	Preferred Minimum Session Refresh Interval sec	
PIN Checking for Private Calls Using AAR Access Code		Trunk group	817
Feature Access Codes	225	Prefix	484, 865
PIN Checking for Private Calls Using ARS Access Code		Prefix Mark	598
Feature Access Codes	225	Prefix-1	558, 800
Pin Number	172	Prepend '+' to Calling/Alerting/Diverting/Connected	
Ping Test Interval (sec)	416	Number	
pings per measurement interval	416	Trunk Group	861
Plan #	469	Prepend '+' to Outgoing Calling/Alerting/Diverting/	
Platform		Connected Public Numbers	621
System Parameters Customer Options	724	Prgm Group List Access Code	213
Platform Maximum Ports	725	PRI Endpoint	570
PMS Endpoint	330	Primary	
PMS Link Maximum Retransmission Requests	329	Hunt Group	361
PMS Link Maximum Retransmissions	329	Primary D Channel	622

PRIMARY LEVELS	766
Primary MFSS CCA-ID	316
primary output endpoint	101
Primary Output Format	101
Primary Search Time (seconds)	415
Priority	160 , 535
Administered Connection	40
Priority Access Code	236
Priority Calling	133
Priority Calling Access Code	223
Priority Factory Number	889
Priority Ip Video	136
Priority On Intraflow	351
Priority Queuing	117
Priority Ring	126
Priority Score	700
Priority Video	622
Priority with respect to Media Servers	703
Privacy — Attendant Lockout	150
Privacy — Digits to Hide	102
Private Group II Permissions and Public Interworking	518
Private Networking	737
Private Prefix	544
Privileged	
Enhanced List	27 , 29 , 32
Personal List	27 , 29 , 32
System List	27 , 29 , 32
Proc Chan	576
Processor and System MSP	737
Processor Channel Assignment/ Communication	
Interface processor	574
Processor Ethernet Network	700
Product ID	745
Program Access Code	223
Program Ext	
Group List	29
Programmed Dial Pause (msec)	465 , 846
Prohibit Bridging Onto Calls with Data Privacy	245
Prompting	89
Prompting Timeout (secs)	284
propagation delay	417
Property Management Interface	477
Protected	
Announcements Audio Sources	55
Protocol for Caller ID Analog Terminals	242
Protocol Type	800
Protocol Variations	857
Protocol Version	622
Provide forced local ringback for EC500	
Configuration Set	148
Provide Ringback	361
Proxy Sel Rte Pat	485
PSA display information	247
PSN notification	899
Public (PSTN) Number	689
public network trunks conference call	261
Pull Transfer	279

Q

Q-SIP	623
QSIG and SIP Diverted Calls Follow Diverted to Party's	
Coverage Path	711
QSIG Call Offer Originations	133
QSIG CAS Number	153
QSIG Path Replacement Extension	271
QSIG Reference Trunk Group	861
QSIG selection numbers, parameters for creating	272
QSIG Signaling Group	623
QSIG to DCS TSC Gateway	578
QSIG Trunk Group Options	461
QSIG VALU Coverage Overrides QSIG Diversion with	
Rerouting	713
QSIG Value-Added	463
QSIG-Value Coverage Encoding	464
QSIG/ETSI TSC Extension	271
Queue	55
Hunt Group	344
Queue Length	440 , 800
Announcements Audio Sources	56
Queue Limit	344
queue priorities	157

R

R Balance Network	
Station	655
R2 MFC Signaling	832
R2-MFC signaling trunks	116
Rate	56 , 371
RCL	781
Re-try	305
Reachability for domain control SIP Stations	
disable	300
enable	300
Reason Code Names	580
Reason Codes	740
Recall from VDN	262
Recall Rotary Digit	305 , 675
Recall Timing	490
Receive Analog Incoming Call ID	832
Receive Answer Supervision	558 , 818
Receive Release Ack	818
Receive Unencrypted from IP Endpoints	604
Received Signal Gain (dB)	519
Record Agent ID on Incoming	102
Record Agent ID on Outgoing	102
Record Call-Assoc TSC	102
Record Called Vector Directory Number Instead of Group	
or Member	102
Record Length	107
Record Non-Call-Assoc TSC	103
Record Outgoing Calls Only	103
Recording Delay Timer	268
Recovery Rule Number	749

Redirect Notification	675	Reporting for PC or POM Calls	880
Redirect on IP/OPTIM Failure to VDN	357	Request Call Category at Start of Call	522
Redirect on No Answer (rings)	357	Request Category	833
Redirect on No Answer to VDN	357	Request CPN at Start of Call	522
Redirect on OPTIM failure	818	Request Incoming ANI (non-AAR/ARS)	519
Redirected Call	158	Request URI contents	862
Redirected DID Call	159	Require Mutual Authentication if TLS	680
Redirection on No Answer Aux Work Reason Code	297	Resend display UPDATE once on receipt of 481 response	858
Redundancy		Reserved Slots for Attendant Priority Queue	251
IP Codec Set	388	Reset Shift Timer	262
Referral Destination	601	Restart ANI from Caller Category	522
Refresh MW Lamp	509	Restrict Call Forward Off Net	737
Refresh Terminal Parameters Access Code	223	Restrict Call Fwd-Off Net	134
Region	399	Restrict Call Joining	228
registration alarms	419	Restrict Calls	248
Rel	746	Restrict Customization Of Button Types	80
Release Ack Send (msec)	847	Restrict Customization Of Labels For the Following Button Types	80
Reliable Protocol	431, 704	Restrict First Consult Activation Deactivation Feature Access Codes	228
Remote Access	582	Restrict Last Appearance	675
Remote Access Dial Tone	584	Restrict Second Agent Consult	121
Remote Access Extension	585	Restrict Second Call Consult Class of Restriction	121
Remote Attendant Route String	536	Restrict Second Consult Activation Deactivation Feature access codes	228
Remote Call Coverage Table	586	Restricted Call List	117
Remote Logout of Agent	121	Restricted Number Dialed Display	307
Remote Logout of Agent Access Code	230	Restricted Number Dialed Intercept Treatment	307
Remote Node	429, 703	Restriction	493
Remote Office	588, 624, 737	Restriction - Controlled	227
Remote Office Phone	655	Restriction Override	117
remote office trunks	726	Retries	173
Remote Port	429, 703	Retry ARS/AAR Analysis if All-Location Entry Inaccessible	195
Remote Send All Calls	223	Retry Interval Administered Connection	40
Remote Soft Phone Emergency Calls	675	Retry Interval (sec)	173
Remote Softphone Emergency Calls Attendant Console	64	Retry upon RSVP Failure Enabled	402
Remote VMS Extensions - Second	510	Return Call	159
Remote VMS Extensions- First	509	Return Call Timeout (sec)	156
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers	623	Return Destination	880
Remove '+' from SIP Numbers		Reverse Star	284
CDR System Parameters	103	RFC 3389 Comfort Noise	624
CTI Link	178	Rg	683
Remove # From Called Number	103	RHNPA Table	590
Remove Agent Skill Access Code	230	RHNPA TABLE	590
Remove Inactive BCMS/VuStats Agents	292	Ringer Off/Auto Answer	293
Repetitive Call Waiting Interval (sec)	279	Ringing	309
Repetitive Call Waiting Tone	279	Ringing Monitor Trunk Group	847
Replace International Access Code with '+'	491	RL Agent Login ID	48
Location Parameters	491	RLT Trunk Group No.	153
Replace Restricted Numbers	456, 833	RMS Feature Enabled	604
Replace Unavailable Numbers	456, 832	Rng	168
Replacement String			
AAR Digit Conversion Table	25, 569		
ARS Digit Conversion Table	25, 569		
report expiration			
Avaya Aura MS	751		
report interval	751		
Reporting for PC Non-Predictive Calls	294		

ROOF Failure and Unreachable Aux Work Reason Code	297	Secondary Output Endpoint	103
Room		Secondary Output Format	104
Station	683	Seconds Before PMS Link Idle Timeout	330
Room Activated Wakeup with Tones	334	Secure	
Room Change/Swap status function	132	Enable Session	206
Roundtrip Propagation Delay (ms)	417	Secure SIP	599
Route Number		Secure Terminal Equip	658
IP Routing	427	Security Code	553 , 656
Route Pat	542	Agent Login ID	46 , 345 , 769
Route Pattern	211 , 591	Attendant Console	62
AAR Digit Analysis Table	21	Security Code for Terminal Self Administration Required	604
ARS Digit Analysis Table	21	Security violation notification	601 , 604
Route Selection	685	Security violation notification parameters	601 , 603
Route-to Number	884	Seize Ack Delay (msec)	847
Route-to VDN	563	Seize Ack Send (msec)	847
Routine	536	Seize When Maintenance Busy	833
Routine Access Code	236	Select Last Used Appearance	677
Routing Digits	507	Selected Survivable Processor Node Names	628
Routing Digits (e.g. AAR/ARS Access Code)	361	Self Administration	477
Routing Extension on Unavailable Voice Synthesis	334	Self Station Display Activation	224
RRQ Required	624	Self-Station Display Enabled	242
RSVP Enabled	402	Send All Call Options	256
RSVP Profile	402	Send All Calls Activation/Deactivation	224
RSVP Refresh Rate (secs)	402	Send All Calls and Call Forwarding (SAC/CF) Override	122
RSVP unreserved (BBE) PHB Value	403	Send All Calls Applies to	256
RTCP Monitor Server	417	send all calls on ringing bridge	257
RTCP Monitor Server Parameters;	404	send all calls on ringing bridge leaves call ringing on	
RTCP Report Period (secs)	404	other bridge	257
RTCP Reporting to Monitor Server Enabled	404	Send ANI for MFE	118
Rule	185	Send Answer Supervision	818
Rule Name	749	Send Called/Busy/Connected Number	457 , 834
Russian signaling trunks	116	Send Calling Number	457 , 834
S		Send Codeset 6/7 LAI IE	458
SAC Notification	154	Send Connect Event to ASAI For Announcement Answer	294
SAC/CF Override	179 , 681	Send Connected Number	458 , 835
SAC/CF Override by Priority Call and Dialing	126	Send Custom Messages Through QSIG?	271
SAC/CF Override by Team Btn		Send Disconnect Event for Bridged Appearance	178
Class Of Restriction	126	Send Diversion Header	862
SAC/CF Override Protection for Priority Call and Dialing	127	Send DTMF Over Telecommuter Link	
SAC/CF Override Protection for Team Btn	127	Feature-related system parameters	279
SBS	456 , 464 , 624	Send EMU Visitor CPN	836
SCCAN	599 , 818	Send Incoming/Outgoing Disconnect Timers to TN465	
SCCAN Parameters	276	Ports	848
Schedule Download	318	Send ISDN Trunk Group Name on Tandem Calls	271
Scheduled	469	Send Name	459 , 836
Scope	869	Send Non-ISDN Trunk Group Name as Connected Name	272
Scroll Status messages Timer (sec.)	303	Send QSIG Message Center ID	507
SDP Capability Negotiation for SRTP	312	Send Release Ack	819
searching for content	895	Send Reroute Request	362
Second	182	Send Transferring Party Information	863
Second Announcement Delay (sec)	361	Send UCID	459
Second Announcement Extension	362	Send UCID to ASAI	290
Second Announcement Recurring	362	Send UII IE	459
Secondary Alert on Held Reminder Calls?	156	Sending Delay	510
Secondary Data Module	737	Serial Call	159
		Serial_Number	700
		Server ID	432 , 701

server IP address	418	Signaling Type	657
Server IPv4 Address	405	Signaling-group of Primary SM BW Mgr	424
Server IPv6 Address	405	Signaling-group of Secondary SM BW Mgr	424
server port	418	signing up	
Service Hours Table	606	PCNs and PSNs	900
Service Interruption	533	Silence Suppression	384
Service Level Algorithm for SLM	295	Simultaneous Calls	571
Service Level Interval	351	SIP	
Service Level Maximizer	741	Actual	644
Service Level Maximizer Algorithm	289	attendant	644
Service Level Supervisor	352	console parameters	153, 760
Service Level Supervisor Call Selection Override	289	SIP 64K Data	388
Service Level Target (% in sec)	352	SIP Endpoint Managed Transfer	312
Service Link Mode		SIP Reference Trunk Group	464
Attendant Console	64, 676	SIP Signaling Group	625
Service Objective	353, 874	SIP URI	345, 686, 875
Agent Login ID	49	SIT Ineffective Other	634
service observing	285	sit no circuit	635
Service Observing	113	SIT Reorder	635
Service Observing (Basic)	741	SIT Treatment for Call Classification	633
Service Observing (Remote/By FAC)	741	SIT Vacant Code	636
Service Observing (VDNs)	741	Site Data	589, 637
Service Observing by Recording Device	121	Size	
Service Observing Listen Only Access Code	230	Enhanced List	27, 29, 30, 32
Service Observing Listen/Talk Access Code	230	Personal List	27, 29, 30, 32
Service Observing Next Call Listen Only Access Code	231	System List	27, 29, 30, 32
Service Observing No Talk Access Code	231	Skill	
SERVICE OBSERVING PERMISSION	129	Hunt Group	353
Service Type	429–431, 441, 703, 704, 801	SL	
Service/Feature	368, 470, 599, 632	Agent Login ID	49
Session - Local/Remote	577	SLM Count Abandoned Calls	354
Session Connect Message Cntr	431, 704	SN	
Session Establishment Timer (min)	624	Agent Login ID	49
Session Manager		SNMP notification	173
primary, secondary, third, fourth	685	emergency call	174
session refresh timer		Softkey Labels	478
non-SCCAN applications	817	Software Package	724
Set Color	683	sort documents	895
Set Layer 1 timer T1 to 30 seconds	723	Source	538
Sfx	371, 472, 856	Announcements Audio Sources	54
Share Level	696	SOURCE ADDRESSES	422
Shared UI Feature Priorities	466	Source File	318
sharing content	895	Source No	538
Short Holding Threshold	836	SPDU Cntr	431, 705
Short Holding Time (seconds)	837	Speaker	683
Short Interdigit Timer	262	Speakerphone	657
Short/Prefixed Registration Allowed	415, 656	speakerphone group paging	320
Show ANSWERED BY on Display	459, 837	Special Application	756
Shuttle	837	Special Character for Restricted Number	178
Sig Bit Inversion	819	Special Dial Tone	262, 301
Sig Grp	187, 473, 578	Special Dial Tone for Digital/IP Stations	302
Signaling group	608	Special Dialing Option	659
Signaling Group	442, 503, 571, 802, 837	SPECIAL DIALING OPTION	
signaling group:Incoming Dialog Loopbacks	616	Attendant Console	66
signaling group:RFC 3389 Comfort Noise	624	SPID — (Service Profile Identifier)	658
Signaling Mode	625	split supervisor	229
signaling trunks, Russian and R2-MFC	116	SPRT Retransmissions	390

SPRT TC0 Payload Size	390	Station When Call is Active	310
SPRT TC1 Payload Size	391	Station-Button Display of UI IE Data	122
SPRT TC1 Window Size	391	Stations With Off PBX Telephone Integration	695
SPRT TC2 Payload Size	391	stations with Off-PBX telephone integration	690
SPRT TC2 Window Size	391	Stations With Off-PBX Telephone Integration	
SPRT TC3 Payload Size	391	Dual Mode	692
SPRT Timer TC1-TA01	391	Stations With System-wide Retrieval Permission	245
SPRT Timer TC1-TA02	391	Status	432
SPRT Timer TC1-TR03	392	status functions	132
SPRT Timer TC2-TA01	392	Status Poll VDN	
SPRT Timer TC2-TA02	392	Best Routing Service	78
SPRT Timer TC2-TR03	392	Step	784
src rgn	410	Stop Date/Time	318
SSE Inter-Packet Interval	392	Store to disk	703
SSE Repetition Redundancy (Packets)	392	Store VDN Name in Station's Local Call Log	284
Stand alone cluster type	504	Stub Network Region	399
Standard Factory Number	890	Subnet Bits	379
Start	869	Subnet Mask	394 , 397
Start B Signal	837	Subnet Trunking	58
Start Date/Time	318	Subscriber Number	578
Start Day	325	Suite Check-in	336
Start Days (Sun through Sat)		Supervisor Extension	354
Administered Connection	39	Supplementary Service Protocol	449 , 819
Start Hour	326	Supplementary Services with Rerouting	744
Start Min	326	support	897
Start Month	326	Support Request History	863
Start Position	838	Supported Set Type	
Start Time		Alias Station	50
Administered Connection	39	Suppress # Outpulsing	460 , 838
Start/End		Suppress CDR for Ineffective Call Attempts	104
Service Hours Table	607	Survivable COR	657
Starting Extension	155	Survivable GK Node Name	658
Starting Port		Survivable License Grace Period	701
Access Endpoint	36	Survivable Processor	699
Station		Survivable Trunk Dest	658
Change Station Extension	110	SVN Authorization Code Violation Notification Enabled	602
Station and Trunk MSP	737	SVN Login	602
Station as Virtual Extension	737	SVN Remote Access Violation Notification Enabled	602
Station Call Transfer Recall Timer	263	SVN Station Security Code Violation Notification Enabled	604
Station Coverage Path For Coverage After Forwarding		Switch Hook Query Response Timeout	297
System Parameters Call Coverage Call Forwarding	717	Switch Name	255
station extension	690	Switch Node	78
Station Extension	692	Switchhook Flash	677
Station Firmware Download Access Code	224	Sync	410
Station Hunt Before Coverage	714	Synchronization	449 , 820
Station Lock	122	Synchronization over IP	311
Station Lock Activation/Deactivation	226	System CESID Default	96
Station Lock COR	122	System ID	659
station name	108	System In Day Service	509
Station Name		System In Night Service	509
Change Station Extension	110	System List	
Station Putting Call On-Hold	310	Abbreviated Dialing	31
Station Security Code Change	226	System Management Data Transfer	737
Station Security Code verification	604	System Manager	600
Station Tone Forward Disconnect	280	System parameters	
Station User Admin of FBI Assign	226	multifrequency signaling	513
Station User Button Ring Control	226	Security violation notification	601 , 603

System parameters call coverage/ call forwarding: page 1	708 , 715
System Parameters Media Gateway Automatic Recovery Rule	748
System parameters OCM call classification	753
System parameters, mode code	508
System Preferred	701
SYSTEM PRINTER PARAMETERS	254
System Updates Time On Station Displays	280

T

T2 (Backward Signal) Activation Timer (secs)	306
T303 Timer (sec)	625
Table Active	778
TAC	442 , 553 , 802
Take Down Link for Lost Messages	330
Talk Duration	636
Target %	563
Target Socket Load	395
Target socket load and Warning level	395
TCP Signaling Link Establishment for Avaya H.323 Endpoints	407
TDD/TTY Mode	389
Team Btn Display Name	127
Team Btn Silent if Active	127
Team Pick Up by Going Off Hook	127
TEI Station	659
Telcordia Technologies NI-2 Operator Service Access	597
Telecommuting Access	757
Telecommuting Access Extension	757
Telephone Event Payload Type	863
Temporary Bridged Appearance on Call Pickup	315
Tenant	758 , 760 , 762
attendant group	758
Tenant Description	760
tenant number	47
Tenant Partitioning	738
Terminal Dial-Up Access Code	226
Terminal Parameters	763
Terminal Trans. Init. (TTI)	738
Terminal Translation Initialization Enabled	247
Terminal Translation Initialization Merge Code	226
Terminal Translation Initialization Separation Code	227
Terminate to Coverage Pts. with Bridged Appearances	169
Terminating Extension Group	767
TestCall BCC	442
Testcall ITC	443
Testcall Service	443
Tests Station	659
Text lines on Bulletin Board	79
TFTP Server	771
TFTP Server Node Name	772
TFTP Server Port	772

Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls	712
Tie Call	159
Time (sec) to Drop Call on No Answer	838
Time (Start)	185
Time (Stop)	186
Time Before Off-Hook Alert	251
Time Format on Terminals Location Parameters	489
Time In Queue Warning (sec)	156
Time Interval	602
Time of Day Chart	118
Time of Day Coverage Table	773 , 774
Time of Day Lock Table	660
Time of Day Routing	738
Time of Day Routing Plan	775 , 776
Time of Day Station Lock Table	777
Time of Scheduled Emergency Access Summary Report	334
Time of Scheduled Wakeup Activity Report	334
Time of Scheduled Wakeup Summary Report	335
Time Remaining	371
Time Reminder on Hold (sec)	156
Time to Login	206
Time Warning Extension	345
Time Warning Threshold	346
Timed ACW	741
Timed ACW Interval (sec)	354
Timezone Offset	485
TN	57 , 62 , 66 , 346 , 481 , 571 , 585 , 660 , 770 , 875
Agent LoginID	47
Group Paging Using Speakerphone	322
TN2501 VAL Boards	728
TN2602 Boards with 320 VoIP Channels	728
TN2602 Boards with 80 VoIP Channels	728
to board Integrated Announcement Translations	373
To IP Address	379
toll	58
Toll Analysis	779
Location	780
Toll List	600 , 781
Toll Restricted	803
Tone (Frequency/Level)	784
Tone (msec)	466 , 848
Tone Continuous	755
Tone Detection Mode System parameters country options	723
Tone Generation	783
Tone Generation Plan	490
Tone Name	755 , 786
Topology	702
Total Administered	544
Total Administered Members	365 , 473 , 857
Total CPN Len	548
Total Len	544
Total Length	192 , 698
CAMA Numbering Format	96

Totals	564	two observers	285
Touch Tone Sidetone (dB)	763 , 766	Two-Digit Aux Work Reason Codes	178 , 297
Touch Tone Transmit (dB)	765 , 766	Type	564 , 890
training	896	Attendant Console	62
Transfer	597	Call Type Analysis Table	83
Transfer Conference	478	CTI Link	177
Transfer Into QSIG Voice Mail	744	Date and Time	182
Transfer to Voice Mail Access Code	227	Feature-Related System Parameters	243
Transfer Upon Hang-Up	267	IP Interfaces	395
Transferred Ring Pattern	280	Station	660
Translation-ID Number Mismatch Interval (days)	603	Survivable Processor	702
transmission mode for inter-system IP DTMF	419	Type (Column)	538
transmission mode for Intra-System IP DTMF	420	Type (field)	538
Transmitted Signal Gain (dB)	519	Type of 3PCC Enabled	686
Transport Method	625		
Trk Grp(s)	544 , 548	U	
Trk-to-Trk Restriction Override		UCID Network Node ID	257
Class Of Service	134	UDP Extension Search Order	196
Truncate Station Number in ANI	522	UDP Port Range	401
Trunk	124	UDP Port Range Max	401
trunk access codes	62	UDP Port Range Min	401
Trunk Alerting Timer	127	UDP-ARS Calls Considered Offnet	195
Trunk Alerting Tone Interval	263	Unacceptable Threshold	
Trunk Answer Any Station Access Code	227	--Dev - 2804 Hz	853
Trunk Contact	850	--Dev - 404 Hz	853
Trunk COR	263	--Dev - 2804 Hz	854
Trunk Direction	558	--Dev - 404 Hz	853
Trunk Flash	803	Max - 1004 Hz Loss	854
Trunk Gain	558	Maximum C Message Noise	854
Trunk Group	787	Maximum C Notched Noise	854
Format	453	Min - 1004 Hz Loss	854
Trunk Group for Channel Selection	625	Minimum ERL	854
Trunk Groups	890	Minimum SRL-HI	854
Trunk Grp No	698	Minimum SRL-LO	855
Trunk Hunt	449 , 820	Unanswered DID Call Timer	263
Trunk Length	850	Unauthorized Precedence Level	533
Trunk Port	559	Unhold	267
trunk selection	690	Unicode	60
Trunk Selection		Unicode Name	822
Stations With Off-PBX Telephone Integration	693	Agent Login ID	47
Trunk Signaling Type	803	Station login	665
Trunk Termination	560 , 821	VDN	875
Trunk Type	450 , 559 , 821	Uniform Dial Plan Table	864
Trunk Type (in/out)	804	Uniform Dialing Plan	738
Trunk Vendor	850	Universal Call ID	257 , 468
Trunk-to-Trunk Transfer	243	Unknown Numbers Considered Internal for AUDIX	272
TSC		Unnamed Registrations and PSA for IP Telephones	248
Route Pattern	600	Unrestricted Call List	118 , 782
TSC Index	188 , 578	Upper Bound (msec)	491
TSC Method for Auto Callback	465	US NI Delayed Calling Name Update	460
TSC per MWI Interrogation	362	USA Default Algorithm	754
TSC Supplementary Service Protocol	626	USA SIT Algorithm	754
TTI Security Code	247	Usage Alloc	444
TTI State	247	Usage Allocation Enhancements	738
TTL Contact	850	Use COR for All Group II Responses	519
TTL Type	850	Use COR for Calling Party Category	519
TTL Vendor	850		

Use Default Server Parameters	405	Verify Wakeup Announcement Access Code	233
Use Enhanced Formats	104	Version	805
Use ISDN Layouts	105	Video (Norm)	411
Use Legacy CDR Formats	105	Video (Prio)	411
Use Time Adjustments from Location	607	Video (Shr)	411
Use VDN Time Zone For Holiday Vectoring	886	Video 802.1p Priority	402
Used		Video Bridge	888
System Parameters Customer Options page 1	725	video capable IP Softphones	727
System Parameters Customer Options page 2	727	video capable stations	727
Used for BSR Polling		Video PHB Value	401
Vector Directory Number	881	videos	897
Used for DCS	838	View Buttons	479
Used for SIP Stations	600	viewing	
Used Only for Paging	839	PCNs	899
User Control Restrict Activation/Deactivation	227	PSNs	899
User Guidance Display	275	VIP Caller	136
User Profile	867	VIP Wakeup	335
user to user information	839	VIP Wakeup Reminder Call	159
USNI Calling Name for Outgoing Calls	272	VIP Wakeups Per 5 Minutes	335
UUI IE Treatment	460	Virtual Channel Identifier	626
UUI Treatment	839	Virtual Path Identifier	627
V		VIS FEATURE OPTIONS	65
V.32	390	VLAN	379 , 396
V.34	390	VMS Hunt Group Extension	510
V.42 Error Correction	392	VOA messages	116
V.90	390	Voice Coverage Message Retrieval Access Code	227
V.92	390	Voice Do Not Disturb Access Code	233
VAC	870	Voice Mail Extension	363
Vacant Code	533	Voice Mail Handle	363
Validate BCMS/VuStats Login IDs	292	Voice Mail Hunt Group Ext	267
Value Added (VALU)	744	Voice mail Number	684
Var	870 , 885	Voice Mail Number	188 , 363
Variables for Vectors	868	Voice Paging TAC	
VDN Name	468 , 564	Loudspeaker Paging	499
VDN of Origin Annc. Extension	875	Voice Paging Timeout (sec)	499 , 839
VDN of Origin Announcement	116 , 743	Voice Paging, COR	499
VDN Override for ASAI Messages	881	Voice Paging, TN	500
VDN Return Destination	743	Voice Principal Message Retrieval Access Code	227
VDN Time Zone Offset	887	Voice Receive (dB)	764 – 766
VDN Timed ACW Interval	882	Voice Sidetone (dB)	764 , 766
VDN Variables	885	Voice Transmit (dB)	764 – 766
Vector	205	Voip Channel License Limit	503
Hunt Group	346	Volume for DCP Types	764
Vector Directory Number	871	Volume for IP Types	765
Send VDN as Called Ringing Name Over QSIG	881	Vustats	479
Vector Disconnect Timer (min)	280	VuStats	292 , 743
Vector Number	876	VuStats (G3V4 Enhanced)	743
Vector Variable x	232	VuStats Objective	355
Vectoring (ANI/II-Digits Routing)	741	W	
Vectoring (Basic)	741	WAE	34
Vectoring (Best Service Routing)	741	Wait Answer Supervision Timer	280
Vectoring (CINFO)	741	Wait time before fallback (seconds)	630
Vectoring (G3V4 Advanced Routing)	742	wakeup announcement	233
Vectoring (Holidays)	742	WAN-BW limits (Total)	411
Vectoring (Prompting)	742	WAN-BW-limits (Units)	412

warning tone on service observing	285
watchlist	895
Whisper Page	308
Whisper Page Activation Access Code	227
Whisper Page Tone Given To	308
Wideband Access Endpoint	34
Wideband Support	461 , 839
Wideband Support Options	470 , 572
Wideband Switching	738
Width	572
Wink Timer (msec)	822
Wireless	739
Without Flash	268
WNDP Emergency 911 Route String	536
WNDP PRECEDENCE ACCESS CODES	236
Worldwide Numbering Dial Plan Active	536

X

X-Mobility/Wireless Type	627
XID	666
XMOBILE Type	666
XMT	386
XOIP Endpoint type	666
XOIP Treatment	823

Y

Year	182

Z

Zip Tone Burst For Callmaster Endpoints	295