

Administering Avaya Aura® Call Center Elite

© 2019-2023, Avaya Inc. All Rights Reserved.

Notice

While reasonable efforts have been made to ensure that the information in this document is complete and accurate at the time of printing, Avaya assumes no liability for any errors. Avaya reserves the right to make changes and corrections to the information in this document without the obligation to notify any person or organization of such changes.

Documentation disclaimer

"Documentation" means information published in varying mediums which may include product information, operating instructions and performance specifications that are generally made available to users of products. Documentation does not include marketing materials. Avaya shall not be responsible for any modifications, additions, or deletions to the original published version of Documentation unless such modifications, additions, or deletions were performed by or on the express behalf of Avaya. End User agrees to indemnify and hold harmless Avaya, Avaya's agents, servants and employees against all claims, lawsuits, demands and judgments arising out of, or in connection with, subsequent modifications, additions or deletions to this documentation, to the extent made by End User.

Link disclaimer

Avaya is not responsible for the contents or reliability of any linked websites referenced within this site or Documentation provided by Avaya. Avaya is not responsible for the accuracy of any information, statement or content provided on these sites and does not necessarily endorse the products, services, or information described or offered within them. Avaya does not guarantee that these links will work all the time and has no control over the availability of the linked pages.

Warranty

Avaya provides a limited warranty on Avaya hardware and software. Refer to your sales agreement to establish the terms of the limited warranty. In addition, Avaya's standard warranty language, as well as information regarding support for this product while under warranty is available to Avaya customers and other parties through the Avaya Support website: https://support.avaya.com/helpcenter/getGenericDetails?detailld=C20091120112456651010 under the link "Warranty & Product Lifecycle" or such successor site as designated by Avaya. Please note that if You acquired the product(s) from an authorized Avaya Channel Partner outside of the United States and Canada, the warranty is provided to You by said Avaya Channel Partner and not by Avaya.

"Hosted Service" means an Avaya hosted service subscription that You acquire from either Avaya or an authorized Avaya Channel Partner (as applicable) and which is described further in Hosted SAS or other service description documentation regarding the applicable hosted service. If You purchase a Hosted Service subscription, the foregoing limited warranty may not apply but You may be entitled to support services in connection with the Hosted Service as described further in your service description documents for the applicable Hosted Service. Contact Avaya or Avaya Channel Partner (as applicable) for more information.

Hosted Service

THE FOLLOWING APPLIES ONLY IF YOU PURCHASE AN AVAYA HOSTED SERVICE SUBSCRIPTION FROM AVAYA OR AN AVAYA CHANNEL PARTNER (AS APPLICABLE), THE TERMS OF USE FOR HOSTED SERVICES ARE AVAILABLE ON THE AVAYA WEBSITE, HTTPS://SUPPORT.AVAYA.COM/LICENSEINFO UNDER THE LINK "Avaya Terms of Use for Hosted Services" OR SUCH SUCCESSOR SITE AS DESIGNATED BY AVAYA, AND ARE APPLICABLE TO ANYONE WHO ACCESSES OR USES THE HOSTED SERVICE. BY ACCESSING OR USING THE HOSTED SERVICE, OR AUTHORIZING OTHERS TO DO SO, YOU, ON BEHALF OF YOURSELF AND THE ENTITY FOR WHOM YOU ARE DOING SO (HEREINAFTER REFERRED TO INTERCHANGEABLY AS "YOU" AND "END USER"), AGREE TO THE TERMS OF USE. IF YOU ARE ACCEPTING THE TERMS OF USE ON BEHALF A COMPANY OR OTHER LEGAL ENTITY, YOU REPRESENT THAT YOU HAVE THE AUTHORITY TO BIND SUCH ENTITY TO THESE TERMS OF USE. IF YOU DO NOT HAVE SUCH AUTHORITY,

OR IF YOU DO NOT WISH TO ACCEPT THESE TERMS OF USE, YOU MUST NOT ACCESS OR USE THE HOSTED SERVICE OR AUTHORIZE ANYONE TO ACCESS OR USE THE HOSTED SERVICE.

Licenses

THE SOFTWARE LICENSE TERMS AVAILABLE ON THE AVAYA WEBSITE, HTTPS://SUPPORT.AVAYA.COM/LICENSEINFO. UNDER THE LINK "AVAYA SOFTWARE LICENSE TERMS (Avaya Products)" OR SUCH SUCCESSOR SITE AS DESIGNATED BY AVAYA, ARE APPLICABLE TO ANYONE WHO DOWNLOADS, USES AND/OR INSTALLS AVAYA SOFTWARE, PURCHASED FROM AVAYA INC., ANY AVAYA AFFILIATE, OR AN AVAYA CHANNEL PARTNER (AS APPLICABLE) UNDER A COMMERCIAL AGREEMENT WITH AVAYA OR AN AVAYA CHANNEL PARTNER. UNLESS OTHERWISE AGREED TO BY AVAYA IN WRITING AVAYA DOES NOT EXTEND THIS LICENSE IF THE SOFTWARE WAS OBTAINED FROM ANYONE OTHER THAN AVAYA, AN AVAYA AFFILIATE OR AN AVAYA CHANNEL PARTNER; AVAYA RESERVES THE RIGHT TO TAKE LEGAL ACTION AGAINST YOU AND ANYONE ELSE USING OR SELLING THE SOFTWARE WITHOUT A LICENSE. BY INSTALLING, DOWNLOADING OR USING THE SOFTWARE, OR AUTHORIZING OTHERS TO DO SO, YOU, ON BEHALF OF YOURSELF AND THE ENTITY FOR WHOM YOU ARE INSTALLING, DOWNLOADING OR USING THE SOFTWARE (HEREINAFTER REFERRED TO INTERCHANGEABLY AS "YOU" AND "END USER"), AGREE TO THESE TERMS AND CONDITIONS AND CREATE A BINDING CONTRACT BETWEEN YOU AND AVAYA INC. OR THE APPLICABLE AVAYA AFFILIATE ("AVAYA").

Avaya grants You a license within the scope of the license types described below, with the exception of Heritage Nortel Software, for which the scope of the license is detailed below. Where the order documentation does not expressly identify a license type, the applicable license will be a Designated System License as set forth below in the Designated System(s) License (DS) section as applicable. The applicable number of licenses and units of capacity for which the license is granted will be one (1), unless a different number of licenses or units of capacity is specified in the documentation or other materials available to You. "Software" means computer programs in object code, provided by Avaya or an Avaya Channel Partner, whether as stand-alone products, pre-installed on hardware products, and any upgrades, updates, patches, bug fixes, or modified versions thereto. "Designated Processor" means a single stand-alone computing device. "Server" means a set of Designated Processors that hosts (physically or virtually) a software application to be accessed by multiple users. "Instance" means a single copy of the Software executing at a particular time: (i) on one physical machine; or (ii) on one deployed software virtual machine ("VM") or similar deployment.

License type(s)

Concurrent User License (CU). End User may install and use the Software on multiple Designated Processors or one or more Servers, so long as only the licensed number of Units are accessing and using the Software at any given time as indicated in the order, Documentation, or as authorized by Avaya in writing. A "Unit" means the unit on which Avaya, at its sole discretion, bases the pricing of its licenses and can be, without limitation, an agent, port or user, an e-mail or voice mail account in the name of a person or corporate function (e.g., webmaster or helpdesk), or a directory entry in the administrative database utilized by the Software that permits one user to interface with the Software. Units may be linked to a specific, identified Server or an Instance of the Software.

Heritage Nortel Software

"Heritage Nortel Software" means the software that was acquired by Avaya as part of its purchase of the Nortel Enterprise Solutions Business in December 2009. The Heritage Nortel Software is the software contained within the list of Heritage Nortel Products located at https://support.avaya.com/LicenseInfo under the link "Heritage Nortel Products" or such successor site as designated by Avaya. For Heritage Nortel Software, Avaya grants Customer a license to use Heritage Nortel Software provided hereunder solely to the extent of the authorized activation or authorized usage level, solely for the purpose specified in the Documentation, and solely as embedded in, for execution on, or for communication with Avaya equipment.

Charges for Heritage Nortel Software may be based on extent of activation or use authorized as specified in an order or invoice.

Copyright

Except where expressly stated otherwise, no use should be made of materials on this site, the Documentation, Software, Hosted Service, or hardware provided by Avaya. All content on this site, the documentation, Hosted Service, and the product provided by Avaya including the selection, arrangement and design of the content is owned either by Avaya or its licensors and is protected by copyright and other intellectual property laws including the sui generis rights relating to the protection of databases. You may not modify, copy, reproduce, republish, upload, post, transmit or distribute in any way any content, in whole or in part, including any code and software unless expressly authorized by Avaya. Unauthorized reproduction, transmission, dissemination, storage, and or use without the express written consent of Avaya can be a criminal, as well as a civil offense under the applicable law.

Virtualization

The following applies if the product is deployed on a virtual machine. Each product has its own ordering code and license types. Unless otherwise stated, each Instance of a product must be separately licensed and ordered. For example, if the end user customer or Avaya Channel Partner would like to install two Instances of the same type of products, then two products of that type must be ordered.

Third Party Components

"Third Party Components" mean certain software programs or portions thereof included in the Software or Hosted Service may contain software (including open source software) distributed under third party agreements ("Third Party Components"), which contain terms regarding the rights to use certain portions of the Software ("Third Party Terms"). As required, information regarding distributed Linux OS source code (for those products that have distributed Linux OS source code) and identifying the copyright holders of the Third Party Components and the Third Party Terms that apply is available in the products, Documentation or on Avaya's website at: https:// support.avaya.com/Copyright or such successor site as designated by Avaya. The open source software license terms provided as Third Party Terms are consistent with the license rights granted in these Software License Terms, and may contain additional rights benefiting You, such as modification and distribution of the open source software. The Third Party Terms shall take precedence over these Software License Terms, solely with respect to the applicable Third Party Components to the extent that these Software License Terms impose greater restrictions on You than the applicable Third Party Terms.

The following applies only if the H.264 (AVC) codec is distributed with the product. THIS PRODUCT IS LICENSED UNDER THE AVC PATENT PORTFOLIO LICENSE FOR THE PERSONAL USE OF A CONSUMER OR OTHER USES IN WHICH IT DOES NOT RECEIVE REMUNERATION TO (i) ENCODE VIDEO IN COMPLIANCE WITH THE AVC STANDARD ("AVC VIDEO") AND/OR (ii) DECODE AVC VIDEO THAT WAS ENCODED BY A CONSUMER ENGAGED IN A PERSONAL ACTIVITY AND/OR WAS OBTAINED FROM A VIDEO PROVIDER LICENSED TO PROVIDE AVC VIDEO. NO LICENSE IS GRANTED OR SHALL BE IMPLIED FOR ANY OTHER USE. ADDITIONAL INFORMATION MAY BE OBTAINED FROM MPEG LA, L.L.C. SEE HTTP://WWW.MPEGLA.COM.

Service Provider

THE FOLLOWING APPLIES TO AVAYA CHANNEL PARTNER'S HOSTING OF AVAYA PRODUCTS OR SERVICES. THE PRODUCT OR HOSTED SERVICE MAY USE THIRD PARTY COMPONENTS SUBJECT TO THIRD PARTY TERMS AND REQUIRE A SERVICE PROVIDER TO BE INDEPENDENTLY LICENSED DIRECTLY FROM THE THIRD PARTY SUPPLIER. AN AVAYA CHANNEL PARTNER'S HOSTING OF AVAYA PRODUCTS MUST BE AUTHORIZED IN WRITING BY AVAYA AND IF THOSE HOSTED PRODUCTS USE OR EMBED CERTAIN THIRD PARTY SOFTWARE, INCLUDING BUT NOT LIMITED TO MICROSOFT SOFTWARE OR CODECS, THE AVAYA CHANNEL PARTNER IS REQUIRED TO INDEPENDENTLY OBTAIN ANY APPLICABLE LICENSE AGREEMENTS, AT THE AVAYA CHANNEL PARTNER'S EXPENSE, DIRECTLY FROM THE APPLICABLE THIRD PARTY SUPPLIER.

WITH RESPECT TO CODECS, IF THE AVAYA CHANNEL PARTNER IS HOSTING ANY PRODUCTS THAT USE OR EMBED THE H.264 CODEC OR H.265 CODEC, THE AVAYA CHANNEL PARTNER ACKNOWLEDGES AND AGREES THE AVAYA CHANNEL PARTNER IS RESPONSIBLE FOR ANY AND ALL RELATED FEES AND/OR ROYALTIES. THE H.264 (AVC) CODEC IS LICENSED UNDER THE AVC PATENT PORTFOLIO LICENSE FOR THE PERSONAL USE OF A CONSUMER OR OTHER USES IN WHICH IT DOES NOT RECEIVE REMUNERATION TO: (I) ENCODE VIDEO IN COMPLIANCE WITH THE AVC STANDARD ("AVC VIDEO") AND/OR (II) DECODE AVC VIDEO THAT WAS ENCODED BY A CONSUMER ENGAGED IN A PERSONAL ACTIVITY AND/OR WAS OBTAINED FROM A VIDEO PROVIDER LICENSED TO PROVIDE AVC VIDEO. NO LICENSE IS GRANTED OR SHALL BE IMPLIED FOR ANY OTHER USE. ADDITIONAL INFORMATION FOR H.264 (AVC) AND H.265 (HEVC) CODECS MAY BE OBTAINED FROM MPÈG LÁ, L.L.C. SEE HTTP:// WWW.MPEGLA.COM.

Compliance with Laws

You acknowledge and agree that it is Your responsibility for complying with any applicable laws and regulations, including, but not limited to laws and regulations related to call recording, data privacy, intellectual property, trade secret, fraud, and music performance rights, in the country or territory where the Avaya product is used.

Preventing Toll Fraud

"Toll Fraud" is the unauthorized use of your telecommunications system by an unauthorized party (for example, a person who is not a corporate employee, agent, subcontractor, or is not working on your company's behalf). Be aware that there can be a risk of Toll Fraud associated with your system and that, if Toll Fraud occurs, it can result in substantial additional charges for your telecommunications services.

Avaya Toll Fraud intervention

If You suspect that You are being victimized by Toll Fraud and You need technical assistance or support, call Technical Service Center Toll Fraud Intervention Hotline at +1-800-643-2353 for the United States and Canada. For additional support telephone numbers, see the Avaya Support website: https://support.avaya.com or such successor site as designated by Avaya.

Security Vulnerabilities

Information about Avaya's security support policies can be found in the Security Policies and Support section of https://support.avaya.com/security.

Suspected Avaya product security vulnerabilities are handled per the Avaya Product Security Support Flow (https://support.avaya.com/css/P8/documents/100161515).

Downloading Documentation

For the most current versions of Documentation, see the Avaya Support website: https://support.avaya.com, or such successor site as designated by Avaya.

Contact Avaya Support

See the Avaya Support website: https://support.avaya.com for product or Hosted Service notices and articles, or to report a problem with your Avaya product or Hosted Service. For a list of support telephone numbers and contact addresses, go to the Avaya Support website: https://support.avaya.com (or such successor site as designated by Avaya), scroll to the bottom of the page, and select Contact Avaya Support.

Trademarks

The trademarks, logos and service marks ("Marks") displayed in this site, the Documentation, Hosted Service(s), and product(s) provided by Avaya are the registered or unregistered Marks of Avaya, its affiliates, its licensors, its suppliers, or other third parties. Users are not permitted to use such Marks without prior written consent from Avaya or such third party which may own the Mark. Nothing contained in this site, the Documentation, Hosted Service(s) and product(s) should be construed as granting, by implication, estoppel, or otherwise, any license or right in and to the Marks without the express written permission of Avaya or the applicable third party.

Avaya is a registered trademark of Avaya Inc.

All non-Avaya trademarks are the property of their respective owners.

Avaya, the Avaya logo, Avaya one-X® Portal, Communication Manager, Application Enablement Services, Modular Messaging, and Conferencing are either registered trademarks or trademarks of Avaya Inc. in the United States of America and/or other jurisdictions.

All non-Avaya trademarks are the property of their respective owners.

Contents

Cł	napter 1: Introduction	. 10
	Purpose	10
	Change history	10
Cł	napter 2: Screen reference	. 11
	Agent LoginID screen	. 11
	Agent LoginID administration commands	11
	Agent LoginID field descriptions	12
	BCMS/VuStats Login ID screen	. 22
	BCMS/VuStats Login ID administration commands	. 23
	BCMS/VuStats field descriptions	24
	Best Service Routing Application Plan screen	24
	BSR Application Plan administration commands	. 24
	BSR Application Plan field descriptions	. 25
	Call Vector screen	. 27
	Call Vector administration commands	. 27
	Call Vector field descriptions	27
	Class of Restriction screen	30
	COR administration commands	
	COR field descriptions	. 30
	Duplicate Vector screen	
	Duplicate Vector administration command	
	Duplicate Vector field descriptions	
	Feature-Related System Parameters screen	
	Feature-Related System Parameters administration commands	
	Agent and Call Selection field descriptions	
	ASAI field descriptions	
	Call Center Miscellaneous field descriptions	
	Call Management System field descriptions	
	EAS field descriptions	
	Forced Agent Logout/Aux Parameters field descriptions	
	Maximum Agent Occupancy Parameters field descriptions	
	Reason Codes field descriptions	
	Redirection on IP Failure field descriptions	
	Service Observing field descriptions	
	SIP Station Reachability Checking Options field descriptions	
	SIP Transfer VDN Counted-Calls Options field descriptions	
	Vectoring field descriptions	
	Holiday Table screen	
	Holiday Table administration commands	
	Holiday Table field descriptions	55

Contents

	Hunt Group screen	55
	Hunt Group administration commands	. 55
	Hunt Group field descriptions	. 56
	Off-PBX Feature-Name-Extensions screen	74
	Off-PBX Feature-Name-Extensions administration commands	74
	Off-PBX Feature-Name-Extensions field descriptions	75
	Policy Routing Table screen	
	Policy Routing Table administration commands	
	Policy Routing Table field descriptions	
	Reason Code Names screen	
	Reason Code Names administration commands	78
	Reason Code Names field descriptions	78
	Service Hours Table screen	
	Service Hours Table administration commands	79
	Service Hours Table field descriptions	79
	SIT Treatment for Call Classification screen	
	SIT Treatment for Call Classification administration commands	80
	SIT Treatment for Call Classification field descriptions	80
	Station screen	
	Station administration commands	
	Station field descriptions	81
	Vector Directory Number screen	
	VDN administration commands	. 85
	Vector Directory Number field descriptions	. 85
	Vector Routing Table screen	
	Vector Routing Table administration commands	
	Vector Routing Table field descriptions	97
	VuStats Display Format screen	
	VuStats Display Format administration commands	98
	VuStats Display Format field descriptions	99
	VuStats Display Format data types	
Ch	apter 3: Administering features	121
	AAS	
	Abandoned Call Search	
	Add/Remove Skills	
	Adding skills	
	Removing skills	
	Changing skills from a voice terminal	
	Agent Call Handling	
	Caller information	
	Agent/Caller Disconnect Tones	
	Agent Mobility	
	Automatic Call Distribution	127

Avaya IQ measurements	128
Basic Call Management System	128
Administering Business Advocate	129
Business Advocate screen reference	130
Best Service Routing	131
Singlesite BSR	131
Multisite BSR	132
BSR Local Treatment	133
BSR Tie Strategy	133
Administering BSR polling over IP without the B-channel	134
Call Prompting	139
Call Vectoring	140
Expert Agent Selection	142
EAS-related fields	143
Other parameters that support EAS Agent LoginID	144
Direct Agent Calling	146
Forced Agent Logout by Clock Time	149
Forced Agent Logout/Aux Work by Location/Skill	149
Forced Agent Logout from ACW	150
Forced Agent Logout from ACW	151
Tips for administering Forced Agent Logout from ACW	151
Inbound Call Management	152
Interruptible Aux	152
Intraflow and Interflow	153
Location Preference Distribution	153
Look-Ahead Interflow	154
Maximum Agent Occupancy	155
Multinational CPN Prefix	
Multiple Call Handling	156
Network Call Redirection	156
NCR for ISDN trunks	156
NCR for SIP trunks	157
NCR with the AT&T In-Band Transfer Connect service	158
Administering station or ASAI transfer, conference, or release	159
Percentage Allocation Routing	160
Queue Status Indications	160
Reason Codes	161
Redirection on IP Failure	162
Redirection on No Answer	163
Redirection on OPTIM Failure	164
Remote Logout of Agent	165
Reporting adjuncts on Communication Manager	165
Measured trunks versus unmeasured facilities	165

Contents

	Adding reporting adjunct nodes	
	Communication Manager to reporting adjunct interface	166
	Administering reporting adjuncts with Communication Manager	166
	Processor Channel Assignment	166
	Proactive Contact predictive calls	167
	Proactive Contact non-predictive calls	167
	Service Hours Table	168
	Service Level Maximizer	168
	Administering the ASL algorithm	169
	Service Observing	169
	Service Observing with Multiple Observers	170
	Universal Call ID	170
	Sending UCIDs over ISDN or SIP trunks	171
	User-to-User Information	172
	UUI Treatment for ISDN trunks	172
	UUI Treatment for SIP trunks	173
	UUI-info feature button	173
	VDN in a Coverage Path	174
	VDN of Origin Announcement	174
	VDN Return Destination	174
,	VDN Time-Zone Offset	175
	Voice Response Integration	
	VuStats	
	Vu-display feature button	
	Zip Tone Burst for Callmaster [®] Endpoints	
Cha	apter 4: Administering the 96X1 SIP agent deskphones	178
	Dedicating station and call routing SIP trunk groups	179
	Administering the Signaling Group screen	180
	Administering the Trunk Group screen	181
Cha	apter 5: Administering C and D tones for a VRU port	183
	Administering C and D tones for a VRU port	
	IVR or VRU ports	183
	Station field descriptions	
Cha	apter 6: Recorded announcements	189
	Recorded announcement types	190
	Analog line types	
	DS1 types	
	Auxiliary trunk types	
	Integrated types	
	When to use recorded announcements	
	Barge-in	
	Barge-in operational details	
	Nonharge-in operational details	193

Int	egrated and externally recorded announcements	194
	bw to record announcements	
	Announcement recording	195
	Announcement sessions	
	Recording the announcement	195
	Stop recording the announcement	196
	Playing back the announcement	196
	Deleting the announcement	197
Re	ecorded announcements for ACD and other call center features	
	Recorded announcements and automatic wakeup	197
Lo	cally sourced music and announcements	197
	Definitions	197
	Locally sourced music and announcements	198
	Audio groups	
	Selecting the most local source of an audio group	198
	Capabilities of locally sourced music and announcements	199
Chapt	ter 7: Time of Day Clock Synchronization	
	DD synchronization methods	
	How to use NTP/SNTP to administer direct switch synchronization	
	How to use Avaya Site Administration to schedule synchronization tasks	
Us	ing NTP/SNTP to synchronize Communication Manager to the UTC time	
	ow to use Avaya Site Administration to set up a TOD synchronization schedule	
	Prerequisites for using Avaya Site Administration to set up a TOD synchronization	
	schedule	203
	Things to know before you set up a synchronization schedule	203
	Designing a TOD clock synchronization schedule	205
	Setting up a TOD synchronization task schedule in Avaya Site Administration	209
NT	P or SNTP and Internet Time Servers	210
	SNTP on switch platforms that support direct synchronization	210
	Platforms that synchronize through an Avaya Site Administration client computer	211
Se	tting up ACD offset times for CMS reporting	
	Setting switch time zone offset values for CMS report times	212
Chapt	ter 8: Resources	213
Do	ocumentation	213
	nding documents on the Avaya Support website	
	cessing the port matrix document	
	aya Documentation Center navigation	
Tra	aining	216
	ewing Avaya Mentor videos	
	pport	
	Using the Avaya InSite Knowledge Base	
Gloss	ary	219

Chapter 1: Introduction

Purpose

This document contains information about how to perform Avaya Aura® Call Center Elite administration tasks including how to use Automatic Call Distribution (ACD) and Call Vectoring features.

This document is intended for system administrators and implementation engineers.

Change history

Issue	Date	Summary of changes
2	May 2023	Added the section SIP Transfer VDN Counted-Calls Options field descriptions on page 51.
1	December 2021	Initial 10.1 issue.

Chapter 2: Screen reference

Agent LoginID screen

Use the Agent LoginID screen to add or change agent login IDs and skill assignment for Expert Agent Selection (EAS).

Agent LoginID administration commands

Use the following commands to administer agent login IDs on the Agent LoginID screen:

Command name	Parameter	Qualifier	
add	agent-loginID	xxxxx or next	
change	agent-loginID	xxxx [auto]	
duplicate	agent-loginID	xxxxx [start nnn] [count x]	
remove	agent-loginID	xxxxx	
list	agent-loginID	staffed [name] [AAS y or n]	
		unstaffed [name] [AAS y or n]	

auto indicates automatic logout and login of logged-in agents after you change agent skills.

• If you use auto, you will see the following warning message:

Agents must log in again before non-skill changes take effect.

• If you do not use auto, you will see the following warning message:

Agents must log in again for the changes to take effect.

xxxx is the extension.

nnn is the extension number of the first duplicate agent login ID in the number sequence.

[] indicates that the qualifier is optional.

Agent LoginID field descriptions

Name	Description
AAS	To use the extension as an Auto Available Split/Skill (AAS) port. Valid entries are y and n .
	If you select y , Communication Manager clears the password. To select n , use the remove agent-loginid xxx command.
	Use the field for adjunct equipment ports only, not for human agents.
ACW Agent Considered Idle	To include agents who are in the After Call Work (ACW) mode in the Most Idle Agent (MIA) queue. Communication Manager counts the time in ACW as idle time. Valid entries are system , n , and y .
	If you select n , Communication Manager excludes ACW agents from the MIA queue while they are in the ACW mode.
	If you select y , Communication Manager keeps or places agents in the MIA queue while they are in the ACW mode.
	If you select system , Communication Manager uses the systemwide field settings.
	If you change the value of the ACW Agent Considered Idle field from the system-parameters features screen, agents must log out and log back in for the change to be reflected.
AUX Agent Remains in LOA Queue	To include agents who are in the AUX mode in the Least Occupied Agent (LOA) queue. Valid entries are system , n , and y . The default value is system .
	If you select n , Communication Manager excludes AUX agents from the LOA queue while they are in AUX work. The n value matches the legacy Communication Manager functionality.
	If you select y , Communication Manager keeps or places agents in the LOA queue while they are in AUX work.
	If you select system , Communication Manager uses the systemwide field settings.
	If you change the value of the AUX Agent Remains in LOA Queue field from the system-parameters features screen, agents must log out and log back in for the change to be reflected.

Name	Description
AUX Agent Considered Idle (MIA)	To include agents who are in the AUX mode in the Most Idle Agent (MIA) queue. Communication Manager counts the time in AUX as idle time. Valid entries are system , n , and y . Default value is system .
	If you select n , Communication Manager excludes AUX agents from the MIA queue while they are in AUX work. The n value matches the legacy Communication Manager functionality.
	If you select y , Communication Manager keeps or places agents in the MIA queue while they are in AUX work.
	If you select system , Communication Manager uses the system-wide field settings.
	If you change the value of the AUX Agent Considered Idle (MIA) field from the system- parameters features screen, agents must log out and log back in for the change to be reflected.
Attribute	To enter a character string that represents a combination of characteristics of that agent defined by the call center management for use in reporting.
	The Attribute field is a 20-character alphanumeric field. The Attribute field can also be left blank. The contents of the Attribute field are sent to reporting, including CMS and Avaya IQ.
AUDIX	To use the extension as an Audio Information Exchange (AUDIX [™]) port.
	Valid entries are y and n .
	Note:
	Select y in the AAS field or the AUDIX field. Do not use both the fields together.
AUDIX Name for Messaging	To perform one of the following actions:
	Type the name of the messaging system that stores Leave Word Calling (LWC) messages.
	Type the name of the messaging system that is the coverage point for this login ID.
	You can also leave this field blank.

Name	Description
Auto-Answer	To determine agent settings.
	If you use Expert Agent Selection (EAS), the option in this field applies to the station where the agent logs in. If the field option on the Station screen is different from the field option in this field, the agent setting overrides the station setting.
	Valid entries are:
	all: Communication Manager sends all ACD and non-ACD calls to the agent. If you select y in the Allow Ringer-off with Auto-Answer field, agents can press ringer-off to prevent ringing.
	acd: Communication Manager sends only ACD and direct agent calls to the agent.
	none: This option is the default.
	station: Communication Manager uses the field option on the Station screen.
Aux Work Reason Code Type	To determine whether agents must type a reason code when changing the work mode to Auxiliary (AUX) work.
	forced: To ensure that an agent types a reason code.
	• none: To not use reason codes.
	requested: To request an agent to type a reason code.
	system: To use the system settings. This field option is the default.
	To use the forced and requested field options, you must select y in the Reason Codes and Expert Agent Selection (EAS) fields on the System- Parameter Customer-Options screen.

Name	Description
Call Handling Preference	To determine the call selection method for an agent during call surplus conditions.
	Valid entries are:
	greatest-need: To select the oldest, highest priority call for any assigned skill.
	percent-allocation: To select calls based on the target allocation for each skill that you assign to an agent. This field option is applicable when you use Business Advocate.
	skill-level: To select the oldest, highest priority call waiting for the highest skill level when calls are in a queue and an agent becomes available. The default is skill-level.
Check skill TN to match LoginID TN Associated field: Include Tenant Calling Permissions	To ensure that Communication Manager delivers calls based on Tenant Number (TN) assignments. Thus, agents within a tenant partition receive calls for skills in the same tenant partition.
	Valid entries are y and n . The default is n .
	If you select y , Communication Manager prevents skill assignment to a login ID when the tenant number of the login ID does not match the tenant number of the skill.
	To use this field, ensure that the Tenant Partitioning field on the System-Parameters Customer-Options screen is y .
	The associated field is available when the Check skill TN to match LoginID TN field value is y .
	Use the associated field for extended inter-tenant call delivery where agents can receive calls for more than one TN.
	If you grant tenant calling permissions to an agent, you can assign skills with TNs for which the agent can receive calls.
COR	To assign a Class of Restriction (COR) number to the login ID.
	Valid entries are 0 to 995. The default is 1.

Name	Description
Coverage Path	To assign a coverage path for calls to the login ID.
	Valid entries are:
	• Path number from 1 to 999.
	• Time of day table from t1 to t999.
	You can also leave this field blank.
	Communication Manager uses the coverage path when an agent is logged out of the system, that is, the agent is unstaffed, busy, or does not answer calls.
Direct Agent Calls First	To override the Percent Allocation call selection method to deliver direct agent calls before other ACD calls. Valid entries are y and n .
	The Direct Agent Calls First field replaces the Service Objective field when you select percent-allocation in the Call Handling Preference field.
Direct Agent Skill	To assign a skill number for handling direct agent calls. Valid entries are 1 to 8000 or blank.
Forced Agent Logout Time	To automatically log agents out of the system based on a timer.
	Valid entries for the hour field are 01 to 23. Valid entries for the minute field are 00, 15, 30, and 45.
	You can also leave this field blank.
Local Call Preference	To administer Location Preference Distribution for handling agent or call surplus conditions.
	To set up an algorithm for agent surplus conditions, administer the Local Agent Preference field on the Hunt Group screen.
	Valid entries are y or n . The default is n .
	You can select y if the Call Center Release field is 3.0 or later and the field option in the Multiple Locations Customer field is y .
Login ID	To view the identifier for a Logical Agent. This field is a display-only field.

Name	Description
LoginID for ISDN/SIP Display	To include the Agent LoginID Calling Party Number (CPN) and Name fields in ISDN and SIP messaging over network facilities.
	If you select y , Communication Manager sends the physical station extension CPN and the name.
	The Send Name on the ISDN Trunk Group screen prevents sending the calling party name and number if set to n and can prevent sending it if set to restricted .
Logout Reason Code Type	To determine whether agents must type a reason code when logging out of the system.
	Valid entries are:
	forced: To ensure that an agent types a reason code.
	none: To not use reason codes.
	requested: To request an agent to type a reason code.
	system: To use the system settings. This field option is the default.
	To use the forced and requested field options, you must select y in the Reason Codes and Expert Agent Selection (EAS) fields on the System- Parameter Customer-Options screen.
LWC Log External Calls	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.
LWC Reception	To determine the storage point for LWC messages.
	Valid entries are:
	• audix: To store LWC messages in an AUDIX [™] system.
	none: To not store LWC messages.
	msa: To store LWC messages on Messaging Server Adjunct (MSA).
	spe. To store LWC messages in the system or on Switch Processor Element (SPE). The default is spe.

Name	Description
Maximum time agent in ACW before logout (sec)	To determine the time in ACW after which the system logs an agent out.
	30-9999: To assign an ACW timeout. This field option takes precedence over the system settings.
	none: To not apply an ACW timeout.
	system: To use the system settings. This field option is the default.
Messaging Server Name for Messaging	To perform one of the following actions:
	Type the name of the messaging system that stores Leave Word Calling (LWC) messages.
	Type the name of the messaging system that is the coverage point for this login ID.
	You can also leave this field blank.
MIA Across Skills	To remove an agent from the MIA queue for all splits or skills that the agent is available in when the agent answers a call from any assigned splits or skills.
	Valid entries are system , n , and y .
	If you change the value of the MIA Across Skills field from the system-parameters features screen, agents must log out and log back in for the change to be reflected.
MWI Served User Type	Administers from which type of message center the user's message waiting indicator (MWI) is interrogated, or whether the MWI is to be audited.
	Blank: The user's MWI is not audited. The user's MWI is also not audited if the user is not a served by an fp-mwi or qsig-mwi message center.
	fp-mwl. The user is served by an fp-mwi message center.
	qsig-mwl. The user is served by a qsig-mwi message center.
	sip-adjunct. Use this option to audit the user's MWI.
Name	The name of this agent. The can be up to 27 characters. Any alphanumeric character is valid. By default, the name is blank.

Name	Description
Password	To assign up to nine digits as the password that an agent must enter when logging in to the system. Valid entries are from 0 to 9.
	Type the minimum number of digits in this field specified by the Minimum Agent-LoginID Password Length field on the Feature-Related System Parameters screen.
	This field is applicable if you select n in the AAS and AUDIX fields.
Password (enter again)	To ensure that an agent reenters the password.
PA	This option is displayed only when Call Handling Preference is set to percent-allocation (PA). This is the percentage for each of the agent skills. A valid entry is a number from 1 to 100 for each skill. Entries for all the agent skills together must add up to 100%. Do not use target allocations for reserve skills. Percent Allocation is available as part of the Avaya Business Advocate software.
Port Extension	To type the assigned extension for an AAS or AUDIX port. This extension cannot be a VDN or an agent login ID.
	The field is applicable if you enter y in the AAS field or the AUDIX field.

Name	Description
RL (Reserve Level)	To assign a reserve level to the agent login ID for a skill with Service Level Supervisor (SLS) or the type of interruption with the Interruptible Aux work feature.
	You can assign a reserve level of 1 or 2 , or an interruptible level of a , m , or n where a=Auto-In-Interrupt, m=Manual-In-Interrupt, and n = Notify-Interrupt. For no reserve or interruptible level, leave the field blank.
	Changes to this field take effect the next time the agent logs in to the system.
	You can use the values 1 or 2 if the Business Advocate field is active for the system. A skill level cannot be assigned with a reserve level setting. RL set to 1 or 2 defines the Expected Wait Time (EWT) threshold level for the agent to be added to the assigned skill as a reserve agent.
	When EWT for this skill reaches the corresponding threshold set on the Hunt Group screen, the system automatically adds this skill to logged-in agent skills. The system delivers calls from this skill to the agent, until the skill EWT drops below the assigned overload threshold for that level, or if Oldest Call Waiting (OCW) is used as a threshold.
	Use the Interruptible Aux feature when the service-level target is not being met.
Security Code	To enter the 4-digit security code or password for the Demand Print Messages feature. The default is blank.
Service Objective	To administer Service Objective on the Hunt Group and Agent LoginID screens.
	This field is applicable when the Call Handling Preference field is set to greatest-need or skill-level .
	Communication Manager selects the arriving ACD calls for staffed agents according to the ratio of Predicted Wait Time (PWT) or Current Wait Time (CWT) to the administered Service Objective for the skill.

Name	Description
SL (Skill Level)	To determine the skill level for each skill that you assign to an agent.
	If you select n in the Expert Agent Selection - Preference Handling Distribution (EAS-PHD) field, you can use two priority levels.
	If you select y in the Expert Agent Selection - Preference Handling Distribution (EAS-PHD) field, you can use 16 priority levels.
SN (Skill Number)	To assign skills to the agent login ID. Do not enter the same skill twice.
	If you select n in the Expert Agent Selection - Preference Handling Distribution (EAS-PHD) field, you can assign up to 4 skills.
	If you select y in the Expert Agent Selection - Preference Handling Distribution (EAS-PHD) field, you can assign up to 60 or 120 skills.
	Important:
	Assigning many skills to agents can affect the performance of your system. Review system designs with Avaya Sales Factory when a significant number of agents have more than 60 skills for each agent.
TN	To assign a Tenant Partition number.
	Valid entries are from 1 to 250.
Unicode Name	This is a display-only field that indicates whether a Unicode name is assigned to the agent. The options are "y" or "n". Changing the Name field does not change the Unicode name. You must use an external administration application such as Avaya Aura [®] System Manager to change the Unicode name.
	For more information about Unicode language display, see <i>Administering Avaya Aura</i> ® <i>Communication Manager</i> .

Name	Description
Work Mode on Login	To specify the work mode the agent uses when the agent logs in to Call Center Elite.
	Valid entries are:
	 system: Agents log in to Call Center Elite in the work mode that has been specified on the EAS section on the Feature-Related System Parameters screen. This is a systemwide field setting.
	 auto-in: Agents log in to Call Center Elite in the auto-in work mode and are available for an ACD call.
	 manual-in: Agents log in to Call Center Elite in the manual-in work mode and are available for an ACD call.
	 aux: Agents log in to Call Center Elite in the aux work mode and are unavailable for an ACD call.

You can use the list agent-loginid command to view the following details about an agent login ID.

Name	Description
Name	To view the name administered for the agent.
AAS/AUD	To view whether the login ID is an Auto-Available Split/Skill (AAS) or an AUDIX [™] port.
Agt Pr	To view the call handling preference of the login ID.
Dir Agt	To view the field option in the Direct Agent Skill field.
Extension	To view the physical extension. This field is blank if no agent is logged in to the system.
SkI/Lv	To view the skills and the skill levels of an agent.
COR	To view the Class of Restriction (COR) assignments for an agent login ID.
SO	To view the field option in the Service Objective field.

BCMS/VuStats Login ID screen

Use the BCMS/VuStats Login ID screen to add or change login IDs for Basic Call Management System (BCMS) tracking purpose.

You can administer this screen if you:

- Do not use Expert Agent Selection (EAS).
- Administer BCMS/VuStats Login ID on the Feature-Related System Parameters screen.

An agent name is optional. If you do not type an agent name, the system displays the following data from BCMS or VuStats: ID xxxxxxxxx, where xxxxxxxxx is an agent login ID.

Only agents using BCMS login IDs can log in to a BCMS-measured split or skill.

BCMS/VuStats Login ID administration commands

Use the following administration commands to administer the BCMS/VuStats Login ID screen.

Command name	Parameter	Qualifier
add	bcms-vustats loginIDs	No qualifier
change	bcms-vustats loginIDs	[login ID]
display	bcms-vustats loginIDs	[login ID]
list	bcms-vustats loginIDs	[login ID] count X

The screen displays only two pages a time, which is equivalent to 64 login IDs. If you want to add login IDs, you can execute the add command to use another two pages.

When you use the **change** or **display** commands, the system displays two pages of the login IDs beginning with the ID that you type with the commands. If you do not enter a login ID, the system displays two pages beginning with the first login ID.

The list command lists all the login IDs.

BCMS/VuStats field descriptions

Field title	Field description
Login ID	To assign a login ID with the same number of characters as the ID that you assigned in the ACD Login Identification Length field on the Feature-Related System Parameters screen.
	If the number of characters is not equal, the system displays an error message and places the cursor on the incorrect field.
	If you enter a duplicate login ID, the system displays an error message and places the cursor on the duplicated field.
	If you change the administered login ID to a different value, the system changes the allowed number of characters for all other IDs on the screen.
	If you do not adjust the number of characters, agents cannot log in to the system. You must change the ACD login identification length to match the existing login IDs or change the login IDs to match the ACD login identification length.
Name	To assign a name to the login ID. This field is optional.

Best Service Routing Application Plan screen

Use the Best Service Routing Application Plan screen to identify the remote locations used in each BSR application.

BSR Application Plan administration commands

Use the following administration commands to administer the BSR Application Plan screen.

Command name	Parameter	Qualifier
add	best-service-routing	xxx or next, where xxx is the BSR application number.
change	best-service-routing	xxx
display	best-service-routing	xxx
remove	best-service-routing	xxx

Command name	Parameter	Qualifier
list	best-service-routing	No qualifier as this command lists all the administered BSR applications.

[] indicates that the qualifier is optional. Single quotes ('') indicate that you must enter the text inside the quote exactly as shown. You can also enter an abbreviated screen of the word.



Note:

If you execute remove best-service-routing against a BSR application table with no name assigned, the system generates an error Identifier not assigned message. To resolve the problem, name the table and use the remove best-service-routing command again.

BSR Application Plan field descriptions

Field title	Field description
Interflow VDN	To assign a routing number including the dial access code that Communication Manager uses to gain access to the interflow Vector Directory Number (VDN) at the remote location. Valid entries can be up to 16 characters long and contain the following characters:
	• 0-9
	• * or #
	• p (pause)
	• w/W (wait)
	• m (mark)
	• s (suppress)
Location Name	To assign a name with up to 15 characters to each location.
Lock	To determine whether Communication Manager must send information to Call Management System (CMS).
	Valid entries are:
	• y : To prevent Communication Manager from sending information to CMS.
	n: To allow Communication Manager to send the information to CMS.

Field title	Field description
Maximum Suppression Time	To assign the maximum poll suppression time. Valid entries are 0 to 60 seconds.
	This field value applies when a subsequent consider command replaces a location as the best. For example, if the poll suppression time is 30 seconds, Communication Manager suppresses polling to a remote location for up to 30 seconds.
Name	To assign a name with up to 15 characters to the Best Service Routing (BSR) application plan.
Net Redir	To determine whether Communication Manager must redirect calls. Valid entries are y and n .
Num	To type a location number. Location numbers are identifiers and do not have to be in a sequential order. For example, you can assign locations with the identifiers 1, 3, 14 and 89 to one BSR application plan.
Number	To verify the identifying number of the BSR application plan. This display-only field is numbered from 1 to 255 or 511 based on the server type driving Communication Manager.
Status Poli VDN	To assign a routing number including the dial access code that Communication Manager uses to gain access to the Status Poll VDN at a remote location. Valid entries can be up to 16 characters long and contain the following characters:
	• 0-9
	• * or #
	• p (pause)
	• w/W (wait)
	• m (mark)
	• s (suppress)
Switch Node	To assign a Universal Call Identification (UCID) network node ID for each Communication Manager server. The range for valid network node IDs is from 1 to 32,767.
	This field is optional.

Call Vector screen

Use the Call Vector screen to write Call Vectoring commands that specify how to handle calls directed to a VDN.

For more information, see Programming Call Vectoring Features in Avaya Aura® Call Center Elite.

Call Vector administration commands

Use the following commands to administer the Call Vector screen.

Command name	Parameter	Qualifier
change	vector	1-MAX
display	vector	1-MAX ['print' or 'schedule']
list	vector	1-MAX ['count' 1-MAX] ['print' or 'schedule']

! Important:

Do not change a call vector while the vector is processing a call. You can add a new vector using the Call Vector screen and use the Vector Directory Number screen to point an existing VDN to the new vector.

Call Vector field descriptions

Field title	Field description
3.0 Enhanced	To view whether the Vectoring (3.0 Enhanced) field on the System-Parameters Customer-Options screen is active for the system.
	This field is a display-only field.
ANI/II-Digits	To view whether the system uses the Automatic Number Identification (ANI) and Information Indicator (II) Digits vector routing commands. The field is applicable if the Vectoring (G3V4 Enhanced) and Vectoring (ANI/II-Digits Routing) fields on the System-Parameters Customer-Options screen is active for the system. Calling Line Identification (CLID) also follows ANI rules. This field is a display-only field.
ASAI Routing	To determine whether to use Adjunct Switch Application Interface (ASAI) Routing.
	This field is applicable if you use CallVisor/ASAI Routing.

Field title	Field description
Attendant Vectoring	To determine whether call vectors must process all attendant-seeking calls.
	The field is applicable if the Attendant Vectoring field on the System-Parameters Customer-Options screen is active for the system. Valid entries are y and n .
	If the Basic Vectoring and Vector Prompting fields are administered as n , the default setting of the Attendant Vectoring field is y , which means that you cannot change the field settings.
	To associate VDNs and vectors for Attendant Vectoring, use the fields on both the VDN and the Call Vector screens. When you indicate Attendant Vectoring for VDNs and vectors, the system removes all call center-related fields, such as Skills and BSR .
Basic	To view whether the Vectoring (Basic) field on the System-Parameters Customer-Options screen is active for the system.
	This field is a display-only field.
BSR	To view whether the Vectoring (Best Service Routing) field on the System-Parameters Customer-Options screen is active for the system.
	This field is a display-only field.
CINFO	To view whether the Vectoring (CINFO) field on the System-Parameters Customer-Options screen is active for the system.
	This field is a display-only field.
EAS	To view whether the Expert Agent Selection field on the System-Parameters Customer-Options screen is active for the system.
	If Expert Agent Selection (EAS) is active for the system, all help and error messages that are associated with the screen reflect a terminology change from split to skill. In addition, the vector commands entered are affected by the terminology change. For example, check backup split becomes check backup skill.
	This field is a display-only field.

Field title	Field description	
G3V4 Adv Route	To view whether the G3V4 Advanced Vector Routing commands are available.	
	This field is a display-only field.	
G3V4 Enhanced	To view whether the G3V4 Enhanced Vector Routing commands and features are available.	
	This field is a display-only field.	
Holidays	To view if the Holiday Vectoring field on the System-Parameters Customer-Options screen is active for the system.	
	This field is a display-only field.	
LAI	To view if the Look-Ahead Interflow field on the System-Parameters Customer-Options screen is active for the system.	
	This field is a display-only field.	
Lines 01 through 99	To use vector commands up to the maximum allowed in your configuration.	
Lock	To control access to the vector from Call Management System (CMS) or Visual Vectors. Valid entries are y and n .	
	If you select n , CMS and Visual Vectors users can administer the vector through their client programs. If you select y , CMS and Visual Vectors users cannot administer the vector.	
	To view and administer locked vectors, use the System Administration Terminal (SAT) or a terminal emulator.	
	Note:	
	Always lock vectors that contain secure information. For example, access codes.	
Meet-me Conf	To view whether the Meet-me Conference field on the System-Parameters Customer-Options screen is active for the system.	
Multimedia	To determine whether the vector must receive early answer treatment for multimedia calls. The field is applicable only if the fields related to Multimedia Call Handling are active for the system.	
	If you select y , the call is answered at the start of vector processing and billing for the call starts at the same time.	

Field title	Field description
Name	To assign an alphanumeric name of up to 27 characters that identifies the vector. This is an optional field.
Number	To gain access to the screen using a change administration or a display administration command.
Prompting	To view whether the Vectoring (Prompting) field on the System-Parameters Customer-Options screen is active for the system. This field is a display-only field.
Variables	To view whether the Vectoring (Variables) field on the System-Parameters Customer-Options screen is active for the system. This field is a display-only field.

Class of Restriction screen

Use the Class of Restriction screen to determine levels of restriction for agents and supervisors.

COR administration commands

Use the following commands to administer levels of restriction on the Class of Restriction screen.

Command name	Parameter	Qualifier
change	cor	xxx
display	cor	xxx
list	cor	No qualifier. The system displays all the administered Class of Restriction (COR) numbers.
xxx is the COR number.		

COR field descriptions

Field title	Field description	
Add/Remove Agent Skills	To add or remove skills. Valid entries are y and n .	
Can Be A Service Observer	To determine whether supervisors with this Class of Restriction (COR) number can observe calls between an agent and a caller. Valid entries are y and n .	

Field title	Field description
Can Be Service Observed	To determine whether calls to agents with this COR number can be observed by a supervisor. Valid entries are y and n .
Can Force a Work State Change	To determine whether supervisors with this COR number can change the work mode of an agent. Valid entries are y and n .
Direct Agent Calling	To determine whether agents with this COR number can receive direct agent calls. Valid entries are y and n .
Hear System Music on Hold	To play music for calls on hold. Valid entries are y and n .
Hear VDN of Origin Annc.	To determine whether agents with this COR number can receive a message about the location of an incoming call or the type of service required by a caller. Valid entries are y and n .
Priority Queuing	To assign priority to calls in a hunt group queue for agents with this COR number. Valid entries are $\bf y$ and $\bf n$.
Remote Logout of Agent	To determine whether supervisors with this COR number can enter a Feature Access Code (FAC) from a remote location to log an agent out of the system. Valid entries are y and n .
Service Observing by Recording Device	To determine whether supervisors with this COR number can use audio recording devices. Valid entries are y and n .
Service Observing Permissions	To determine whether supervisors with this COR number can grant permissions to observe another COR number. Valid entries are y and n .
Station-Button Display of UUI IE Data	To determine whether agents with this COR number can press uui-info to see the Adjunct Switch Application Interface (ASAI)-related data. Valid entries are y and n .
Work State Change Can Be Forced	To determine whether the work mode of agents with this COR number can be changed by a supervisor. Valid entries are y and n .

Duplicate Vector screen

Use the duplicate vector command to duplicate vectors from an existing vector. You can configure one vector as a template and can reuse the template when creating similar vectors.

Duplicate Vector administration command

Use the following command to access the Duplicate Vector screen.

Command name	Parameter	Qualifier
duplicate	vector	master_vector [start nnnn] [count
		xx]

duplicate vector

Use the command to create up to 16 duplicate vectors.

master vector Is the

Is the number of the vector to be duplicated or to be used as a template.

[start nnnn]

Specifies the first vector number to be used as a duplicate. The parameter is optional. If you do not specify a start number, the first available vector after the master vector number is selected. Only one

vector is selected.

[count xx]

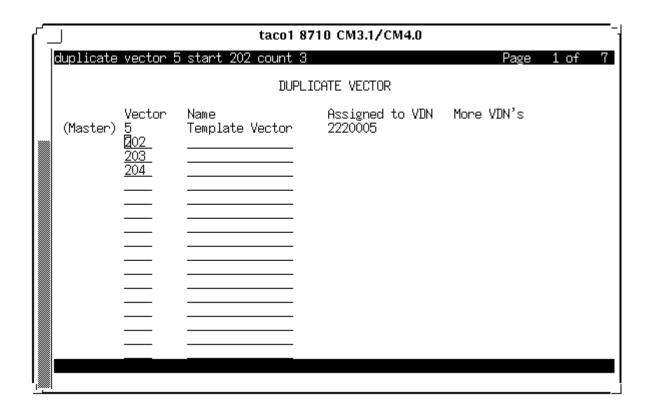
Specifies the number of duplicates to be created from the master vector. You can enter a number from 1 to 16. The parameter is optional. If you do not specify a count number, the first available vector after the master

vector is selected. Only one vector is selected.

Example

The following example creates vectors 202, 203, and 204 as exact duplicates of vector 5.

duplicate vector 5 start 202 count 3



Duplicate Vector field descriptions

Field title	Field description
Count	To view the number of duplicates created from the master vector.
More	To view if more than one VDN is assigned to the same vector. For example, if the system displays 5555 in the VDN Assigned to field and an asterisk (*) sign in the More field, the master vector you selected is
	already assigned to VDN 5555 and to other VDNs.
Name	To view the vector name. You can assign names to the duplicated vectors, but not steps. You can edit the vector name for any of the duplicated vectors.
	If you specify a used or out of range vector number, the system displays an error message. You cannot move to the next field until you enter an unused number.
VDN Assigned to	To view the VDN if you assign a VDN to the master vector.
Vector	To view the vector number of the master vector and each new vector that you create from the master vector.

Feature-Related System Parameters screen

Use the Feature-Related System Parameters screen to administer Call Center Elite features for the system.

Feature-Related System Parameters administration commands

Use the following commands to administer Call Center Elite features on the Feature-Related System Parameters screen.

Command name	Parameter	Qualifier
change	system-parameters features	_
display		

Agent and Call Selection field descriptions

Field title	Field description
ACW Agents Considered Idle	To determine whether to include agents who are in the After Call Work (ACW) mode to the Most-Idle Agent (MIA) queue. Valid entries are y and n .
	If you select y , Communication Manager adds the ACW time to the idle time and moves agents to the MIA queue.
AUX Agent Remains in LOA Queue	To include agents who are in the AUX mode in the Least Occupied Agent (LOA) queue. Valid entries are n and y .
	If you select n , Communication Manager excludes AUX agents from the LOA queue while they are in AUX work. The n value matches the legacy Communication Manager functionality.
	If you select y , Communication Manager keeps or places agents in the LOA queue while they are in AUX work.
	Note:
	Call Center Elite release must be 7.0 or later.

Field title	Field description
AUX Agent Considered Idle (MIA)	To include agents who are in the AUX mode in the Most Idle Agent (MIA) queue. Communication Manager counts the time in AUX as idle time. Valid entries are n , and y .
	If you select n , Communication Manager excludes AUX agents from the MIA queue while they are in AUX work. The n value matches the legacy Communication Manager functionality.
	If you select y , Communication Manager keeps or places agents in the MIA queue while they are in AUX work.
	Note:
	Call Center Elite release must be 7.0 or later.
Auto Reserve Agents	To determine whether to use auto reserve agents to meet service levels.
	Valid entries are:
	all: To place an agent on standby for all skills.
	none: To not place an agent on standby for any additional skills.
	secondary-only: To place an agent on standby only for secondary skills.
Block Hang-up by Logged-in Auto-Answer Agents	To prevent an agent in the auto-answer mode from accidently logging out or dropping an active call. Valid entries are y and n .
	Before you change the field options, verify if the Auto Answer field is administered on the Agent LoginID screen or the Station screen.
	This field is applicable you use Expert Agent Selection (EAS).

Field title	Field description
Call Selection Measurement	To determine how Communication Manager selects a call for an agent when the agent becomes available and there are calls in the queue.
	Valid entries are:
	current-wait-time: To select the oldest call waiting for any agent skill.
	predicted-wait-time: To use the time a call is predicted to wait in a queue instead of the time that the call has already waited. This field is applicable only if you use Business Advocate.
	Predicted Wait Time (PWT) is a Business Advocate feature and is the default setting if Business Advocate is active for the system. Current Wait Time (CWT) and PWT are mutually exclusive and are applied systemwide.
MIA Across Splits or Skills	To determine whether to keep an agent in the MIA queue even when the agent responds to a call from one of the assigned splits or skills. Valid entries are y and n .
	If you select y , Communication Manager removes the agent from the MIA queue for all the splits or skills that the agent is available (idle) in when the agent answers a call from any of the assigned splits or skills.
	If you select n , Communication Manager keeps the agent in the MIA queue for the other splits or skills when the agent answers a call from one of the assigned splits or skills. This field option is the default.

Field title	Field description
Service Level Maximizer Algorithm	To select an alternative algorithm for selecting agents and delivering calls in order to maximize service level targets. This field is applicable if Service Level Maximizer on the System-Parameter Customer-Options screen is active for the system.
	Valid entries are:
	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 an agent answers the call 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	To determine whether Communication Manager changes the call handling preferences when a skill using Service Level Supervisor (SLS) exceeds the Level 1 threshold. Valid entries are y and n .
	If you select y , Communication Manager overrides the administered call handling preferences and delivers calls for the skill that exceeds the Level 1 threshold.
	If you select n , Communication Manager delivers calls based on the administered call handling preferences.
	This field is applicable if you use Business Advocate and EAS.

ASAI field descriptions

Field title	Field description
Call Classification After Answer Supervision	To determine when the classifier is inserted in the connection. For use with Adjunct Switch Application Interface (ASAI) Outbound Call Management (OCM).
	Valid entries are:
	• y: To force the server on which Communication Manager is running to rely on the network to provide one of the following classifications to the server: answer, busy, or drop.
	After the agent answers the call, you can add a call classifier to perform answering machine, modem, and voice answering detection.
	 n: To always connect a classifier after call setup for determining call progress and answer. ISDN progress messages generally take precedence. This field option is the default.
	Communication Manager drops the call if you administer this field as n and no classifiers are available during the reservation phase.
Copy ASAI UUI During Conference/Transfer	To determine whether to copy the ASAI User-to User Information (UUI) data from the last-held call to the new call. The new call results from pressing the conference or transfer button. ASAI UUI is available for display on the phone the call is conferenced with or transferred to.
	Valid entries are y and n .
For ASAI Send DTMF Tone to Call Originator	To determine whether to send the Dual-Tone Multi-Frequency (DTMF) tones to all parties. Valid entries are y and n .
	The ASAI 3rd Party Send DTMF Digits feature lets all parties including the originator hear the DTMF tones.
Send Connect Event to ASAI For Announcement Answer	To determine whether to send a connect event for an announcement vector step or an announcement that the system plays for a collect digits vector step. Valid entries are y and n .
Send UCID to ASAI	To determine whether to send the Universal Call Identification (UCID) data to ASAI. Valid entries are y and n .

Call Center Miscellaneous field descriptions

Name	Description
Agent/Caller Disconnect Tones	To help supervisors identify if a caller or an agent disconnected first from an active ACD or direct agent call.
	Valid entries are y and n .
	• y: To play one of the two distinct disconnect tones.
	n: To play no distinct tone. This field option is the default.
	For measured trunks, Call Management System and Avaya IQ offer reports that indicate which party disconnected the call first.
Allow Ringer-off with Auto-Answer	To determine whether an agent can press ringer-off that prevents ringing on EAS auto-answer calls.
	Valid entries are y and n .
Callr-info Display Timer (sec)	To determine the display duration for call-related information when an agent presses callr-info .
	This field is applicable to softphones, H.323 phones, and one-line display phones with the Enhanced Callr-Info display for 1–line phones field on the Station screen administered as n .
	Valid entries are from 3 to 60 seconds. The default value is 10 seconds.
Clear Callr-info	To determine when Communication Manager clears call-related information from the phone display.
	leave-ACW: Communication Manager clears call- related information when an agent leaves the After Call Work (ACW) mode.
	next-call: Communication Manager clears call- related information when an agent receives the next call. This field option is the default.
	on-call-release: Communication Manager clears call-related information when an agent releases a call.

Name	Description
Interruptible Aux Notification Timer (sec)	To determine the number of seconds the endpoint Interruptible Aux notifications, flashing lamp, display, or tone are on before an auto-in-interrupt agent or a manual-in-interrupt agent is available.
	The delay ensures that an agent is not immediately made available when the agent presses interruptible-aux. The delay also provides a brief period to an agent already in the Interruptible Aux mode before the system makes the agent automatically available.
	Valid entries are from 1 to 9 seconds with the default value of 3 seconds.
PC Non-Predictive Reports Skill	To administer a skill hunt group for reporting of Avaya Proactive Contact (PC) non-predictive switch classified calls for each system. Reports are generated as if agent were in the ACD-OUT state.
	You can select a skill number from 1 to 8000.
	This field is applicable if you administer the Reporting for PC Non-Predictive Calls field as y.
Reporting for PC Non-Predictive Calls	To start or stop integration with Proactive Contact for non-switch classified outbound calling.
	This feature improves Call Management System (CMS) tracking of switch classified and non-switch classified calls, that is, agent classified, outbound calls that the Proactive Contact soft dialer places through Adjunct Switch Application Interface (ASAI).
	Valid entries are y and n .

Name	Description
Zip Tone Burst for Callmaster Endpoints	To apply a single burst of zip tone for calls to auto- answer ACD agents.
	The field option applies to:
	Zip tones for auto-answer ACD calls with the station/agent ID auto-answer field set to acd.
	Incoming Call Identification (ICI) tones for auto- answer non-ACD calls with the field set to all.
	The options are:
	double: To keep the current operation, which applies double bursts of zip tone for ACD calls and double bursts of ICI tone for non-ACD calls, on the Callmaster® series phones.
	single: To eliminate the second burst of zip tone or ICI tone to reduce:
	The time for an agent to begin a conversation with the caller.
	- The possibility of the agent and the caller hearing an <i>open mike</i> background noise between the first tone and second tone.
	Use this field option with a Callmaster [®] series phone when the agent hears the single burst of tone to recognize an incoming call.

Call Management System field descriptions

Name	Description
Reporting Adjunct Release	The field that determines the protocol used by the appl link.
	The CMS (appl mis) and IQ (appl ccr) parameters determine the Switch Protocol Interpreter (SPI) language protocol used for the CMS (mis) and Avaya IQ (ccr) links. You must administer the mis and ccr links on the Processor Channel Assignment screen.
	You can assign maximum two links of each type of parameter. For example, you can assign two mis links and two ccr links. If you activate Special Application SA9090, you can administer three to four links as application type mis.
	If you administer three links as appl type mis, you can administer only one Avaya IQ interface ccr link.
	If you administer all four links as appl type mis, you cannot administer the ccr links because the total number of mis and ccr links is four.

Name	Description
CMS (appl mis)	The option to select the release of Call Management System (CMS) to which you are connecting.
	The options are:
	• R16.1/R16.x/R17.0: These releases apply to R16.1, R16.2, R16.3, or R17.0.
	• R18: This release applies to CMS R18.0.x.
	• R18.1: This release applies to CMS R18.1.x.
	• R19.0: This release applies to CMS R19.0.x.
	Connection to the second CMS using mis2 link is optional.
	You can leave the field blank to indicate that CMS is not connected to the system. This option is the default.
IQ (appl ccr)	The option to select a release of Avaya IQ.
	The options are:
	• 5.1/5.2: Apply to 5.1.x, 5.2.0, 5.2.1, 5.2.2, 5.2.3, 5.2.4, and 5.2.5.
	• 5.2.6+: Applies to 5.2.6 and 5.3.x.
	You must administer Expert Agent Selection (EAS) and Universal Call ID (UCID) before establishing a connection with Avaya IQ.
	Connection to the second Avaya IQ using ccr2 link is optional.
	You can leave the field blank to indicate that Avaya IQ is not connected to the system. This option is the default.

The minimum release field entry combinations for the **CMS (appl mis)** and **IQ (appl ccr)** fields and the Switch Protocol Interpreter (SPI) language that Communication Manager uses to establish connections with the reporting adjuncts are:

CMS release	Avaya IQ release	SPI language
R16.1/R16.x/R17.0	5.1/5.2	23
R18	5.2.6+	24
	Note:	
	Selecting R5.2.6+ applies to R5.2.6 or R5.3.x.	
18.1	5.2.6+	24
	Note:	
	Selecting R5.2.6+ applies to R5.2.6 or R5.3.x.	

CMS release	Avaya IQ release	SPI language
19.0	5.2.6+	24
	Note:	
	Selecting R5.2.6+ applies to R5.2.6 or R5.3.x.	

To use other combinations of CMS and Avaya IQ releases, select the lowest release of one reporting adjunct. For example, for CMS or Avaya IQ, enter the supported release for the other reporting adjunct, and set the Communication Manager release on the respective reporting adjuncts to the lowest release supported by the lowest release of the reporting adjunct.

For example, to configure CMS R16.1 with Avaya IQ 5.2.6+, set:

- CMS (appl mis) to R18
- IQ (aapl ccr) to R5.2.6+



For backward compatibility, set the Communication Manager release on CMS or Avaya IQ to the latest Communication Manager release supported by the earlier release reporting adjunct. To connect CMS and Avaya IQ to Communication Manager, match the releases of the two reporting adjuncts and the assignment of the Communication Manager release on the adjuncts using the release settings that are required to run the same link interface protocol language, that is, the SPI language.

EAS field descriptions

Field title	Field description
Direct Agent Announcement Extension Associated field: Delay	To type the extension of the direct agent announcement.
Associated field. Delay	To determine how long must the caller hear ringback before the calling party hears a direct agent announcement.
	This field is applicable if the Expert Agent Selection or ASAI Link Core Capabilities fields on the System-Parameters Customer-Options screen are active for the system.
	You can assign a delay period from 0 to 99 seconds or leave the field blank for no delay.
Expert Agent Selection (EAS) Enabled	To use skill-based routing.
	You can administer this field if the system has Automatic Call Distribution (ACD) or Call Vectoring hunt groups.

Field title	Field description
Message Waiting Lamp Indicates Status For	To assign an indication for messages to stations or agent login IDs. The field is applicable if you use EAS.
	Valid entries are:
	station: The message waiting lamp on a phone indicates the message is for the phone extension.
	loginID: The message waiting lamp on a phone indicates the message is for an agent login ID.
	Setting the Message Lamp Ext field on the station to an extension other than the station default overrides the agent login ID system option for the station. The lamp does not track the logged-in agent.
Minimum Agent-LoginID Password Length	To determine the minimum number of digits that are required for a login ID password of an EAS agent. The field accepts up to nine digits and is applicable if you use EAS.
Work Mode on Login	To specify the work mode the agent uses when the agent logs in to Call Center Elite.
	Valid entries are:
	aux: Agents log into Call Center Elite in the aux work mode and are unavailable for an ACD call. This is the default option.
	auto-in: Agents log into Call Center Elite in the auto-in work mode and are automatically available for an ACD call.
	manual-in: Agents log in to Call Center Elite in the manual-in work mode and are automatically available for an ACD call.
ACD Login Identification Length	To assign the number of characters for an ACD agent login ID that identifies an ACD agent to CMS.
	The valid entries range from 1 to 9. This number must be equal to the number of characters in the agent logon ID.
	Note:
	This field is applicable if you are using BCMS/VuStats Login IDs.

Field title	Field description
BCMS/VuStats LoginIDs	To activate valid agent login IDs for agents to monitor call activity.
	BCMS and VuStats login IDs are available in addition to the EAS login IDs if you use EAS. BCMS and CMS use the same login ID for an agent.
BCMS/VuStats Measurement Interval	To select measurement intervals for polling and reporting data. Maximum 25 time slots are available for measurement intervals.
	You can change the interval at any time, but the change takes effect only after the current interval.
	If you do not use BCMS or VuStats, but you select an option for this field, the system displays the following error message: Cannot be used. Assign either BCMS or VuStats first.
	The valid entries are:
	half-hour
	• hour
Clear VuStats Shift Data	To determine when to clear the shift data for an agent.
	The valid entries are:
	on-login: To clear the shift data for an agent when the agent logs in.
	at-midnight: To clear the shift data for all agents at midnight.
Remove Inactive BCMS/VuStats Agents	To determine whether to delete agent details from a report when the agents have no staff time since the last 7 days.
	The valid entries are:
	• y
	• n
Validate BCMS/VuStats LoginIDs	To determine whether agents must type login IDs that you administer for BCMS.
	The valid entries are:
	• y
	• n
	If you select n, you can type any ACD login ID.

Forced Agent Logout/Aux Parameters field descriptions

Field title	Field description
ACW Forced Logout Reason Code	To assign a reason code when Communication Manager logs an agent out of the system due to a time out in the After Call Work (ACW) mode. Valid entries are from 0 to 9.
Clock Time Forced Logout Reason Code	To assign a reason code Communication Manager logs an agent out of the system due to clock time.
	Valid entries are from 0 to 9.
Maximum Time Agent in ACW before Logout (sec)	To assign a system-wide timer for agents in the ACW mode before Communication Manager logs the agents out of the system.
	Valid entries are from 30 to 9999. You can leave the field blank.
Forced Agent Logout for Unreachable Reason Code	To assign a reason code when Communication Manager logs an agent out of the system if the SIP agent is unreachable.
	Valid entries are from 0 to 9. The default value is 0.

Maximum Agent Occupancy Parameters field descriptions

Field title	Field description
Maximum Agent Occupancy AUX Reason Code	To assign a reason code when Communication Manager changes the agent work mode to Auxiliary (AUX) work due to Maximum Agent Occupancy (MAO).
	You can administer a reason code from 0 to 99. Do not use reason code 0.
Maximum Agent Occupancy Percentage	To assign the maximum percentage of time that an agent can take calls.
	You can administer an MAO percentage from 0 to 100. The default field option is 100%.

Reason Codes field descriptions

Field title	Field description
Aux Work Reason Code Type	To determine whether staffed agents must enter a numeric 1–digit or 2–digit code that describes the reason for changing the work mode to Auxiliary (AUX) work.
	Valid entries are:
	forced: To ensure that an agent is forced to enter a reason code when changing the work mode to AUX work. If an agent enters an invalid code or does not enter a code within the time out interval, Communication Manager denies the work mode change and the agent remains in the current work mode.
	• none: To not use reason codes.
	requested: To request an agent to enter a reason code when changing the work mode to AUX work. If an agent enters an invalid code or does not enter a code within the time out interval, Communication Manager changes the agent work mode to AUX work with the default code 0.
	The forced and requested field options are applicable if the Reason Codes and EAS fields on the System-Parameters Customer-Options screen are active for the system.
Logout Reason Code Type	To determine whether staffed agents must enter a numeric 1–digit or 2–digit code that describes the reason for logging out of the system.
	Valid entries are:
	forced: To ensure that an agent is forced to enter a reason code when logging out of the system.
	• none: To not use reason codes.
	requested: To request an agent to enter a reason code when logging out of the system.
	The forced and requested field options are applicable if the Reason Codes and EAS fields on the System-Parameters Customer-Options screen are active for the system.

Field title	Field description
Redirection on No Answer Aux Work Reason Code	To assign a reason code for reporting purpose when Communication Manager changes the agent work mode to AUX work due to Redirection on No Answer (RONA).
	Valid entries are:
	0-99: If you administer the Two-Digit Aux Work Reason Code field as y.
	• 0-9: If you administer the Two-Digit Aux Work Reason Code field as n .
Redirection on OPTIM Failure and Unreachable Aux Work Reason Code	To assign a reason code for reporting purpose when Communication Manager changes the agent work mode to AUX work due to Redirection on OPTIM Failure (ROOF) or when the agent is unreachable.
	Valid entries are:
	0-99: If you administer the Two-Digit Aux Work Reason Code field as y.
	• 0-9: If you administer the Two-Digit Aux Work Reason Code field as n .
	This AUX work reason code is applicable to ACD agents using SIP phones that you administer as Off-PBX Telephone Integration and Mobility (OPTIM) endpoints.
Two-Digit Aux Work Reason Code	To determine whether to use two-digit reason codes for agent work mode changes to AUX work.
	Valid entries are y and n .

Redirection on IP Failure field descriptions

Field title	Field description
IP Failure AUX Reason Code	To assign a reason code for reporting purpose when Communication Manager changes the agent work mode to AUX work due to Redirection on IP Failure (ROIF).
	Valid entries are:
	• 0-99: If you administer the Two-Digit Aux Work Reason Code field as y .
	• 0-9: If you administer the Two-Digit Aux Work Reason Code field as n .

Field title	Field description
Switch Hook Query Response Timeout	To determine the duration that call processing waits for a response from the switch hook query before Communication Manager triggers ROIF.
	Valid entries are from 500 to 5000 milliseconds (msec). You can leave the field blank to indicate that ROIF is inactive for the system.

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 Service Observing features.

Field title	Field description
Allow Two Observers in	Controls whether two observers can monitor the same call.
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 and 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 calls.
	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 for the system.
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 calls.
	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 for the system.
	The option is not applicable if you set the Service Observing: Warning Tone option to y .

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, either by 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.
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, either by COS or by manual activation of Exclusion.
	If you select b , Communication Manager allows both visible and invisible SSC parties a connection towards a station with Exclusion active, either by 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.
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, not any other parties on the call.
	Valid entries are y and n (default).
	If you select y , the supervisor can change to the 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 also 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 only available from Call Center Elite SIP devices.

SIP Station Reachability Checking Options field descriptions

Field title	Field description
Enable SIP Agent Reachability	To enable Communication Manager to monitor the reachability of logged in idle agents.
	Valid entries are ${\tt y}$ and ${\tt n}.$ The default value is ${\tt n}.$
Enable Reachability for Domain Control SIP Stations	To enable Communication Manager to poll the domain-controlled SIP stations and send the station reachability information to CTI applications that require to track the status of the station.
	Valid entries are disable-all, y and n. The default value is n.
	Note:
	Use the disable-all option to override all the station settings and disables all the domain control reachability.
SIP Station Reachability Attempts	To set the number of attempts that Communication Manager makes to reach the SIP station before logging out the station.
	Valid entries are from 1 to 5. The default value is 3.
SIP Reachability Polling Interval (minutes)	To set the approximate time interval used by Communication Manager to send a signal to check the reachability of the SIP station.
	Valid entries are from 5 to 30. The default value is 5.
SIP Unreachable Polling Period (minutes)	To set the period of time after unreachability was determined that Communication Manager continues to poll this station to aid in the synchronization for that station once it become available.
	Valid entries are from 0 to 1440. The default value is 60.

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 .

Vectoring field descriptions

Field title	Field description
Available Agent Adjustments for BSR	To allow Best Service Routing (BSR) adjustments for identification of the best split or skill to respond to calls in an agent surplus condition.
	The field is applicable if Vectoring (Best Service Routing) on the System-Parameters Customer-Options screen is active for the system.
BSR Tie Strategy	To determine the best split or skill selection strategy in an agent surplus condition.
	The field is applicable if Vectoring (Best Service Routing) on the System-Parameters Customer-Options screen is active for the system.
	Valid entries are:
	1st-found: Communication Manager routes the call to the first available agent. This option is the default.
	alternate: Communication Manager alternates the BSR selection algorithm during a tie between Expected Wait Time (EWT) or available agent criteria.
	Each time a tie occurs for calls from the same active VDN, Communication Manager sends calls based on the selection from the consider step with the tie instead of the first selected split/skill or location. The selection helps balance routing over the selected local splits or skills and remote locations when the cost of remote routing is not a concern.
Converse First Data Delay	To determine the number of seconds that data is prevented from being outpulsed, as a result of a converse vector step, from the system to Avaya IR. The delay commences when the IR port answers a call and prevents data outpulsing till IR is ready.
	Valid entries are from 0 to 9 seconds.
	The field is applicable if Vectoring (Best Service Routing) on the System-Parameters Customer-Options screen is active for the system.
Converse Second Data Delay	To determine the number of seconds used when Communication Manager must outpulse two groups of digits, as a result of a converse vector step, to Avaya IR. The delay commences when Communication Manager outpulses the first group of digits and prevents the second set from being outpulsed before IR is ready.
	The field is applicable if Vectoring (Best Service Routing) on the System-Parameters Customer-Options screen is active for the system.
	Valid entries are from 0 to 9 seconds.

Field title	Field description
Converse Signaling Pause	To determine the delay period in milliseconds (msec) between the digits being passed. The optimum timer settings for Avaya IR are 60 msec tone and 60 msec pause.
	Valid entries are from 40 to 2550 milliseconds with increments of 10 milliseconds.
	Communication Manager adjusts the field values based on 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 msecs tone rounds down to 125 msecs, a 70 msecs pause rounds up to 75 msecs for a total of 200 msecs for each tone.
	• TN464F, TN767E, or later suffix DS1 boards: Rounds up to the nearest 20 msecs. For example, a 130 msecs tone rounds up to 140 msecs, a 70 msecs pause rounds up to 80 msecs for a total of 220 msecs for each tone.
	If you use a circuit pack for end-to-end signaling to IR and then to send digits to a different destination, IR timers can stay in effect. To reset the timers to the system default, pull and reset the circuit pack.
	This field is applicable if the Vectoring (Basic) and DTMF Feedback Signals for VRU fields on the System-Parameters Customer-Options screen are active for the system.
Interflow-qpos EWT Threshold	To determine the number of seconds for the interflow-qpos EWT threshold. Communication Manager does not interflow calls predicted to be answered before the threshold value.
	The field is applicable if the Lookahead Interflow (LAI) field on the System-Parameters Customer-Options screen is active for the system.
	You can administer a threshold value from 0 to 9 seconds or leave the field blank to interflow calls.
Prompting Timeout (secs)	To determine the number of seconds before the Collect Digits command times out awaiting caller entry.
	The field is applicable if the Vectoring (Prompting) field on the System-Parameters Customer-Options screen is active for the system.
	Valid entries are 2 to 10 seconds with increments of 1 second.

Field title	Field description
Reverse Star/Pound Digit for Collect Step	To assign indicators for the collect vector step. The asterisk (*) sign is interpreted as a caller end-of-dialing indicator and the pound sign (#) is an indicator to clear all digits previously entered by the caller for the current collect vector step.
	Valid entries are y and n .
	If you select y , Communication Manager reverses the asterisk and pound digits for the collect vector step. Reversal of digits does not affect any other vector step or other non-ACD features, such as Alternate Route Selection (ARS).
	If you select n , the asterisk (*) and the pound (#) sign digit-processing is unchanged.
Store VDN Name in Station's Local Call Log	To enable message transmission from Communication Manager so that the following phones can store the VDN name or the calling party name in the station call log:
	• 2420
	• 4610
	• 4620
	• 4625
	• 9608CC H.323
	• 9611CC H.323
	• 9621CC H.323
	• 9641CC H.323

Holiday Table screen

Use the Holiday Table screen to determine when Communication Manager must use Holiday Vectoring.

The fields on this screen are applicable if Holiday Vectoring is active for the system.

Holiday Table administration commands

Use the following commands to administer the Holiday Table screen.

Command name	Parameter	Qualifier
change	holiday-table	1 to 999 or next
display		
remove		
list		none: Lists all administered holiday tables

Holiday Table field descriptions

Field title	Field description
Description	To type a description of the holiday table. This field is optional.
End	To determine the parameters on which vector processing for the holiday ends.
	Fill in the following details:
	• Month: 1 to 12
	• Day: 1 to 31
	Hour (Optional): 00 to 23
	Minute (Optional): 00 to 59
Name	To type an alphanumeric name ranging from 1 to 15 characters. This field is optional.
Number	To view the table number. This field is a display-only field.
Start	To determine the parameters on which vector processing for the holiday begins.
	Fill in the following details:
	• Month: 1 to 12
	• Day: 1 to 31
	Hour (Optional): 00 to 23
	Minute (Optional): 00 to 59

Hunt Group screen

Use the Hunt Group screen to administer splits or skills than can receive calls from more than one business function, such as Sales, Service, or Billing.

Hunt Group administration commands

Use the following commands to administer the Hunt Group screen.

Command name	Parameter	Qualifier
add	hunt-group	1-system limit
		next

Command name	Parameter	Qualifier
change	hunt-group	1-system limit
display	hunt-group	1-system limit
		next
		['number' x] ['to-number' x] ['count' n] ['schedule']
remove	hunt-group	1-system limit
list	hunt-group	['number' x] ['to-number' x] ['name' x] ['type' x] ['ext' x] ['to-ext' x] ['count' n] ['schedule']
duplicate	hunt-group	master_grp [start nnnn] [count xx]
[] indicates the q	ualifier is optional.	

duplicate Use the command to create up to 16 duplicate hunt-group screens. hunt-group Is the assigned hunt group number, which is up to four digits, of the hunt master grp group you want to duplicate. [start nnnn] Is an optional parameter that you can use to specify the number from which to start the number sequence for the duplicate hunt group. If you do not specify a start number, the first available hunt group after the master hunt

group number is selected automatically. Only one hunt group is selected. [count xx] Is an optional parameter that you can use to specify the number of

> duplicates. You can create a maximum of 16 duplicates. If you do not specify a count number, the first available hunt group after the master hunt group

number is selected automatically. Only one hunt group is selected.

Hunt Group field descriptions

Field title	Field description
AAS	To use the hunt group as an Auto-Available Split (AAS).
ACD	To use ACD for the hunt group.
	The field is applicable if the ACD field on the System-Parameters Customer-Options screen is active for the system.
Administered Members (min/max)	To administer the minimum and maximum number of members for a hunt group. The field is available for all member pages.

Field title	Field description
AUDIX Extension	To assign a 4-digit to 5-digit Uniform Dial Plan (UDP) extension that identifies the AUDIX [™] hunt group on the host switch used as the Message Center for this hunt group. The field supports AUDIX [™] in a Distributed Communications Services (DCS) arrangement.
	The field is applicable when you administer the Message Center field as rem-audix .
AUDIX name	The name of the AUDIX [™] machine, which must match the IP node name.
	Administer this field after you configure an IP node.
Calling Party Number to INTUITY AUDIX	To determine whether to send the Calling Party Number (CPN) to INTUITY AUDIX [™] . Valid entries are y and n .
Calls Warning Threshold	To determine the number of calls that the system can place in a queue.
	When the number of calls exceeds this threshold, the system flashes the queue status and the optional Auxiliary Queue Call Warning Threshold lamp that you can assign to the split or skill. The lamps glow steadily when a minimum of one call is in a queue.
	You can assign a number from 1 to 999, but ensure that the value is less than or equal to the queue limit.
	Do not leave the field blank if you administer the associated Calls Warning Port field.

Field title	Field description
Controlling Adjunct	To determine the type of adjunct processor that controls the members of the split or skill or the hunt group.
	Valid entries are:
	none: Members of the split/skill or hunt group are not controlled by an adjunct processor.
	asai: All agent login IDs are controlled by an associated adjunct and logged-in agents can use only their data terminal keyboards, for example, to change the agent work state.
	You can use the field option if the controlling adjunct is a CONVERSANT IVR.
	adjlk: Computer Telephony Adjunct Links.
	asai-ip: ASAI links administered without hardware.
	adj-ip: Adjunct links administered without hardware.
	You must select a field option other than none for the ASAI Link Core Capabilities and Computer Telephony Adjunct Links fields on the System-Parameters Customer-Options screen.
(Calls Warning) Port	To assign a port number to connect the optional external Auxiliary Queue Call Warning Threshold lamp that flashes when the number of calls in a queue exceeds the queue warning threshold, as assigned on the Calls Warning Threshold field.
	The field is applicable if you allow calls to queue to a hunt group.
	Valid entries are:
	• 1 to 64: The first and second characters are the cabinet number.
	A to E: The third character is the carrier.
	0 to 20: The fourth and fifth characters are the slot number.
	• 01 to 04 (Analog TIE trunks) 01 to 31: The six and seventh characters are the circuit number.
	The port is assigned to an analog line circuit pack or given an <i>x</i> designation if an extension is used.
	Table continues

Field title	Field description
COR	To assign a Class of Restriction (COR) number that reflects the desired restriction for the hunt group.
	If the hunt group supports voice messaging in DCS, the CORs on the screen must be the same for each communication server.
Coverage Path	To assign a coverage path for the hunt group. The field is applicable if the hunt group is not vector controlled.
	Valid entries are:
	• 1 to 999: To assign a coverage path number .
	• t1 to t999: To assign a time of day table.
Dynamic Percentage Adjustment	To determine whether to automatically adjust the agent work allocations to meet the service-level targets. Valid entries are y and n .
	The field is applicable if:
	ACD is active.
	Business Advocate is active.
	The hunt group is a Percent Allocation Distribution (PAD) hunt group.
Dynamic Queue Position	To determine whether to dynamically change the position of calls in a queue based on the service objective of the originating VDN. Valid entries are y and n .
	This field is applicable if the following fields are active for the system:
	• ACD
	Business Advocate
	Expert Agent Selection (EAS)
	• Skill
Dynamic Threshold Adjustment	To determine whether to automatically adjust the overload thresholds to meet the administered service levels. Valid entries are y and n .
	This field is applicable if the following fields are active for the system:
	• ACD
	Business Advocate
	Service Level Supervisor

Field title	Field description
Expected Call Handling Time (sec)	To determine the expected call handling time.
	Valid entries are 1 to 9999 with increments of 1 second.
	This field is applicable if the following fields are active for the system:
	• ACD
	Business Advocate
	Vectoring (Advanced Routing)
First Announcement Extension	To assign an extension number for a recorded announcement. Leave the field blank to indicate no announcement.
	If this is the forced first announcement, the caller hears ringback after the announcement.
	This field is applicable if you administer the Queue field as y and the Vector field as n .
(First Announcement) Delay (sec)	To determine the number of seconds that a call remains in a queue before the system plays the associated first announcement. The call retains its position in the queue while the caller listens to the recorded announcement.
	If the call is not answered after the announcement, the caller hears:
	 Music if music-on-hold is provided. Music is only for the first announcement.
	Ringing till the call remains in the queue
	Valid entries are from 0 to 99. Leave the field blank if there is no first announcement.
	If you enter 0, the first announcement is provided immediately to the caller. This field option is automatically administered to 0 if there is no queue.
	This field is applicable if you administer the Queue field as y and the Vector field as n .
Forced Entry of Stroke Counts or Call Work Codes	To determine whether an agent must enter a stroke/ event count or a Call Work Code (CWC) for each call that the agent receives while in the manual-in work mode.
	The field is applicable if you use ACD and if the hunt group does not have a controlling adjunct.

Field title	Field description
Group Extension	To assign an unused extension number to the hunt group. Do not leave the field blank.
(Group Member Assignments) Ext	To assign a station or attendant console extension. The extension cannot be a Vector Directory Number (VDN). The data module cannot be a member of an ACD split or skill.
	You can administer the assigned station or attendant console extension only if the Controlling Adjunct field is administered as none .
	ℜ Note:
	Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, do not display correctly on a BRI station.
(Group Member Assignments) Name	To assign a name to the extension number.
(Group Member Assignments) At End of Member List	To view the current page and the last page.
Group Name	To type a character string that uniquely identifies the hunt group. You can type up to 27 characters.
	ℜ Note:
	This field is supported by Unicode language display for the 4610SW, 4620SW, 4621SW, and 4622SW phones.
Group Number	To assign a hunt group number.

Field title	Field description
Group Type	To determine an agent or extension selection method when more than two extensions or agents are available to receive a call.
	Valid entries are:
	circ: The order of administration of extensions determines the order of call routing.
	This field option is applicable if you do not use ACD.
	ddc: Also known as hot seat distribution. Communication Manager routes a call to the first available agent based on the sequence of administration.
	This field option is applicable if you do not use Expert Agent Selection (EAS).
	ead-loa: Communication Manager routes a call to the agent with the highest skill level and the lowest percentage of time on ACD calls since login.
	This field option is applicable if you administer the Expert Agent Selection field to y and the Least Occupied Agent field or the Business Advocate field to y.
	ead-mia: Communication Manager routes a call to the agent with the highest skill level and the longest idle time.
	This field option is applicable if you use EAS.
	pad: Communication Manager routes a call to the agent with the lowest ratio of adjusted work time to the target allocation for the skill.
	This field is applicable if you use Dynamic Advocate
	slm: Communication Manager routes a call to the agent whose other assigned skills meet the administered service level targets.
	This field option is applicable if you administer the Service Level Maximizer field to y and the Business Advocate field to n .
	ucd-loa: Communication Manager routes a call to the agent with the lowest percentage of time on ACD calls since agent login.
	This field option is applicable if you administer the Expert Agent Selection field to y and the Least

Field title	Field description
	Occupied Agent field or the Business Advocate field to y.
	ucd-mia: Communication Manager routes a call to the agent with longest idle time.
	This field option is applicable if you use ACD.
Inflow Threshold (sec)	To determine the number of seconds that a call can remain in a queue before the queue stops accepting calls. Valid entries are from 0 to 999.
	If you administer the field as 0, Communication Manager redirects calls to a split or skill only if there is an available agent.
	This field is applicable if:
	ACD is active for the system.
	The system has a queue for calls.
	The hunt group is not vector controlled.
Interruptible Aux Threshold	To specify which threshold triggers an event to interrupt agents that are interruptible for a skill and then to specify the threshold value in the corresponding field.
	Valid entries are:
	calls-warning-threshold
	service-level-target
	time-warning-threshold
	• none
Interruptible Aux Deactivation Threshold	Based on the Interruptible Aux Threshold policy and the associated threshold value, administer a deactivation threshold to turn off agent notification.
	If you select calls-warning-threshold, the threshold is less than X calls in the hunt group queue. You can assign a value from 0 to 998.
	• If you select service-level-target , you can assign a value from 0 to 100.
	If you select time-warning-threshold, the threshold is the oldest call that has been in a queue for less than Y seconds. You can assign a value from 0 to 998.

Field title	Field description
ISDN/SIP Caller Display	To display the hunt group name to the originating user. This field is required for ISDN-PRI, ISDN-BRI, and SIP trunks.
	Valid entries are:
	grp-name: To display the hunt group name to the originating user.
	mbr-name: To display the member name to the originating user.
	blank: To display the VDN name to the originating user.
	* Note:
	Avaya BRI stations support only ASCII characters. Non-ASCII characters, such as Eurofont or Kanafont, do not display correctly on a BRI station.
Level 1 Threshold (sec) Level 2 Threshold (sec)	To determine the number of seconds for the first EWT threshold. Valid entries are 0 to 99.
	This field is applicable if the ACD and Service Level Supervisor fields are active for the system.
	To determine the number of seconds for the second EWT threshold. Valid entries are 0 to 99.
	This field is applicable if the ACD and Service Level Supervisor fields are active for the system.

Field title	Field description
LOA Increased Agts in Skill	To support having a maximum of 7,000 agents logged in the same skill.
	The LOA Increased Agts in Skill field appears on the Hunt Group screen only if the following conditions are fulfilled:
	You must enable Expert Agent Selection (EAS).
	You must set the group type to ead-loa or ucd-loa.
	Communication Manager is not installed on a S8300D system.
	Valid entries are:
	n: Distributes calls to the least-occupied available agent in the skill. However, this limits the number of logged-in agents per skill to 1,500 agents.
	• y: Approximates the least-occupied available agent within 5%. Distributes call to an agent within the bucket of the least-occupied agents where the least-occupied bucket contains all agents with less than 70% occupancy. The next bucket contains all agents with 70 – 74% occupancy, and so forth up to the most-occupied agents at 95-100% occupancy. This method of distribution is less precise, but allows up to 7,000 agents logged in per skill. This approximation is required due to performance issues with the more-precise LOA algorithm when large numbers of agents are available.
Local Agent Preference	To handle agent and call surplus conditions. Valid entries are y or n .
	Use the field to administer agent surplus conditions. To set up an algorithm for call surplus conditions, administer the Local Call Preference field on the Agent ID screen.
	This field is applicable if the Multiple Locations field is active for the system.

Field title	Field description
LWC Reception	To select a storage point for Leave Word Calling (LWC) messages.
	Valid entries are:
	audix: LWC messages are stored in the voice messaging system.
	none: LWC messages are not stored.
	spe: LWC messages are stored in the system or on the Switch Processor Element (SPE).
Measured	To send measurement data for the ACD split or skill to VuStats or BCMS.
	This field is applicable if you activate the ACD field for the hunt group and the VuStats field or the BCMS field for the system.
	Valid entries are:
	both: To send measurement data collected both internally and externally.
	external: To send measurement data tracked by Call Management System that are external to the server running Communication Manager.
	internal: To send measurement data tracked by Call Management System that are internal to the server running Communication Manager.
	none: To not send measurement data for the hunt group.
Member Range Allowed	To determine the number of allowed members. The values vary based on the system or the configuration.
Message Center AUDIX Name	To type the name of the Message Center AUDIX [™] .
	This field is applicable if the messaging type is audix or rem-vm.
Message Center MSA Name	To type the name of the Message Center Messaging Server Adjunct (MSA).
	This field is applicable if the messaging type is msa .

Field title	Field description
Message Center	To select the type of messaging adjunct for the hunt group.
	Valid entries are:
	• audix: To select AUDIX [™] located on the server running Communication Manager.
	• fp-mwi : To select a public network allowing AUDIX [™] to be located on another Communication Manager. This field option is applicable if the ISDN Feature Plus field is administered as y .
	• msa: To select an MSA.
	msa-vm: To select a voice mail system integrated using mode codes or digital station emulation.
	rem-vm: To select DCS allowing voice mail to be located on another server.
	qsig-mwi: To select a QSIG network allowing voice mail to be located on another server.
	sip-adjunct: To select a SIP message center server.
	none: The hunt group does not serve as a message hunt group.
	You can administer only one hunt group in the system with type audix , fp-mwi , and rem-audix . You can administer up to six hunt groups with type qsig-mwi .
Messaging Server Name	To type the name of the server as displayed on the User-Defined Adjunct Names screen.
MM Early Answer	To allow multimedia early answer.
	The system answers an H.320 call and establishes an audio channel before offering the conversion call to a hunt group. This action starts billing for the call when the call is first placed in a queue.
	This field is applicable to systems using Multimedia Call Handling.

Field title	Field description
Night Service Destination	To type an extension number as the destination where calls to this split redirect when the split is in the Night Service mode. This extension can be a VDN extension but must be a local extension for all features to work correctly.
	You can select attd as an attendant group code.
	This field is inapplicable to a vector-controlled hunt group.
Primary	To administer the specified AUDIX [™] as the primary adjunct. This field is applicable if the messaging type is audix or rem-audix .
Priority on Intraflow	To assign priority for calls routed from a split to a covering split over calls waiting in the covering split queue.
	This field is applicable if the ACD field is administered as y and the hunt group is not vector-controlled.
Provide Ringback	To provide ringback to the caller till the system receives a connect indication. Valid entries are y and n .
	If you select y , a call covering to the message center provides ringback to the caller during the coverage interval. The system discontinues ringback upon receipt of a connect indication.
	Administer this field if you use a Separation of Bearer and Signaling (SBS) trunk for the QSIG MWI hunt group.
	This field is applicable if you administer the Message Center field on the Hunt Group screen as fp-mwi or qsig-mwi .
Queue Limit	To dynamically allocate the queue limit.
	Valid entries are:
	unlimited: To allocate the queue dynamically. All calls to this hunt group are put in a queue when an agent or a station is unavailable. This field option is the default.
	1-999 : To assign a limit to the number of calls that queue to this hunt group.
	This field is applicable if you administer the Queue field as y .
Queue	To assign a queue for the hunt group.

Field title	Field description
Redirect on IP/OPTIM Failure to VDN	To specify the VDN for call redirection due to Redirection on IP Failure (ROIF) or Redirection on OPTIM Failure (ROOF).
	If you leave this field blank, Communication Manager places calls back in the queue.
(Redirect on IP/OPTIM Failure to VDN) Retain Active VDN Context	To retain and use the VDN context from the previous active VDN after Communication Manager redirects a call due to ROIF and ROOF.
Redirect on No Answer (rings)	To determine the number of rings after which Communication Manager must redirect calls.
	Valid entries are 1 to 20 rings. You can leave this field blank so that Communication Manager redirects calls without waiting for a ring.
Redirect on No Answer to VDN	To specify the VDN for call redirection due to Redirection on No Answer (RONA).
	If you leave this field blank, Communication Manager places calls back in the queue.
(Redirect on No Answer to VDN) Retain Active VDN Context	To retain and use the VDN context from the previous active VDN after Communication Manager redirects a call due to RONA.
Routing Digits (e.g. AAR/ARS Access Code)	To assign a 1-digit to 4-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 AUDIX [™] .
	The field accepts the star (*) and pound (#) characters.
	This field is applicable if the messaging type is qsig-mwi or fp-mwi .
Second Announcement Extension	To assign an extension number to a recorded announcement. Leaving the field blank indicates that there is no second announcement.
	This field is applicable if:
	ACD is active for the system.
	The Queue field is administered as y.
	The Vector field is administered as n.

Field title	Field description
(Second Announcement) Delay (sec)	To determine the time before a call in a queue receives a second recorded announcement or if the second announcement is repeated.
	Valid entries are from 1 to 99. If this hunt group is a coverage point for another split or skill, the delay must not be more than 15 seconds.
	This field is applicable if:
	ACD is active for the system.
	The Queue field is administered as y.
	The Vector field is administered as n .
Security Code	To enter a 4-digit security code for the Demand Print feature. You can leave this field blank.
Send Reroute Request	To determine whether to send rerouting invocation when a call covers to a qsig-mwi hunt group.
	This field is applicable if you administer the supplementary services with rerouting and the messaging type as qsig-mwi .
Service Level Interval	To determine 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. You can administer the interval to the same value as the target objectives for the application.
	Valid entries are:
	hourly: To set the ASL algorithm calculations for accepted call and total call components to 0 (zero) at hourly intervals.
	daily: To set the ASL algorithm calculations for accepted call and total call components to 0 (zero) at daily intervals. This field option is the default.
	weekly: To set the ASL algorithm calculations for accepted call and total call components to 0 (zero) at weekly intervals. The weekly interval starts as 00:00 hours on Sunday.
	This field is applicable if you administer the SLM algorithm as actual and the hunt group is of an SLM type.

Field title	Field description
Service Level Supervisor	To reduce the need to move agents from skill to skill during emergencies or unanticipated peaks in call volume.
	The field is applicable if you use ACD and if the hunt group is of the EAS skill.
Service Level Target (% in sec)	To determine the service level targets.
	This field is applicable if you administer the ACD field as y , the Measured field is not blank, and when more than one of the following features is set to y.
	BCMS or VuStats Service Level, BCMS/VuStats Service Level customer option license is active and the hunt group measured field is set to internal or both.
	The seconds component of the service level target is used as the acceptable level for reporting the percentage of calls answered within the specified time. Leave the default percentage of 80 unchanged.
	The service level target in seconds is used for the Business Advocate Service Level Supervisor Objective.
	You can use the service level target for the dynamic percentage adjustment when:
	- The Dynamic Threshold Adjustment field on the Hunt Group screen is y .
	- The Group Type field on the Hunt Group screen is pad .
	- The Dynamic Percent Adjustment field on the Hunt Group screen is y .
	SLM service level target. Applicable if the Group Type field on the Hunt Group screen is slm , the SLM customer option license is active, and the Business Advocate customer option license is not active.
	Interruptible Aux Work service level target. Applicable if the Interruptible Aux Threshold field on the Hunt Group screen is service-level-target.
	The Interruptible Aux feature is triggered if the service level is less than the administered percentage of calls in the specified seconds.

Field title	Field description
Service Objective	To assign a service objective to a skill as the number of elapsed seconds before a call is answered.
	Valid entries are from 1 to 9999 seconds. The default value is 20 seconds.
	This field is applicable if you administer:
	The ACD field as y.
	The Business Advocate field as y.
	The hunt group as an EAS skill.
Skill	To administer a hunt group as an EAS skill. This field is applicable if ACD and EAS are active for the system.
SLM Count Abandoned Calls	To determine whether to include abandoned calls in the ASL algorithm calculations for SLM. Valid entries are $\bf y$ and $\bf n$.
	If you select y , abandoned calls are included in the ASL algorithm calculations for SLM. If you select n , abandoned calls are not included in the ASL algorithm calculations for SLM.
	The field is best used when reporting for this application does not account for calls that are abandoned while in the skill queue.
	This field is applicable if you administer the SLM algorithm as actual and the hunt group is of an SLM type.
SLM Max Auto Reserve Agents	To determine the maximum number of auto reserve agents for the hunt group.
	Valid entries are 0 to 9.
Supervisor Extension	To assign an extension number to the ACD split or skill that agents reach when using the Supervisor Assist feature. The extension number cannot be a VDN.

Field title	Field description
Time Warning Threshold	To administer a time warning threshold that activates Interruptible Aux if the oldest call has been in a queue for longer than the specified number of seconds.
	If you administer the threshold at 60 seconds, interruptible agents start getting interrupted when the duration of the oldest call in a queue for a hunt group exceeds 60 seconds. An entry of 0 provides a warning whenever a call is queued.
	Valid entries are from 0 to 999.
	This field is applicable if you assign a queue to the hunt group and if you do not administer a port number to the call warning and time warning ports.
Timed ACW Interval (sec)	To determine the number of seconds an agent in the auto-in work mode remains in the ACW mode after a call drops. After this time interval expires, the agent automatically becomes available.
	You cannot administer Timed ACW if the hunt group is adjunct-controlled or is an AUDIX [™] Message Center. Valid entries are from 1 to 9999.
	You can override the field by adjusting the settings for a vector.
	This field is applicable if you administer the ACD field for the hunt group as y and the Timed ACW field for the system as y .
(Timed ACW Interval (sec)) After Xfer or Held Call Drops	To place an auto-in agent in the Timed ACW mode for incoming ACD or direct agent calls instead of immediately making the agent available. Use the field for instances when the caller drops a held call or the agent transfers a call.
	You can administer this field for the agents in a hunt group or for calls delivered from a VDN when the Timed ACW Interval field is administered as a nonzero value.
TN	To assign a Tenant Partition number. Valid entries are from 1 to 250.
Total Administered Members	To determine the total number of members for the hunt group.
Vector	To administer the hunt group as vector-controlled. This field is applicable if you administer the Basic Vectoring field as y .

Field title	Field description
Voice Mail Handle	To assign a SIP Enablement Services (SES) handle that can receive voice mails. Leave the field blank if you have assigned a voice mail number.
Voice Mail Number	To assign a 1-digit 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.
	This field is applicable if you administer the Basic Call Setup and Basic Supplementary Services fields and the messaging type is qsig-mwi or fp-mwi.
VuStats Objective	To assign a numeric objective for calls. Valid entries are from 0 to 99999.
	An objective is a goal of the split or skill. This value can be an agent objective, such as a specific number of calls handled or the average talk time. The objective can also be a percentage within the service level.
	The objective is available on the VuStats display and agents and supervisors can compare the current performance with the objective for the split or skill. This value applies to customized VuStats display formats.
	This field is applicable if you administer the ACD field for the hunt group and the VuStats field for the system as y .
	You must administer the hunt group to collect internal or external measurement data for VuStats.

Off-PBX Feature-Name-Extensions screen

Use the Off-PBX Feature-Name-Extensions screen to administer an Off-PBX dialed extension to a feature. The extension is called a Feature-Name-Extension (FNE).

Off-PBX Feature-Name-Extensions administration commands

Use the following commands to administer Off-PBX Feature-Name-Extensions (FNEs).

Command name	Parameter	Qualifier
change	off-pbx feature-name-extensions	set x
		You edit set 1 if you omit adding set x as a qualifier.
list	fne	No qualifier.
		The system displays all the administered FNEs.

Off-PBX Feature-Name-Extensions field descriptions

Field title	Field description
After Call Work Access Code	Mobile agents dial the After Call Work Access Code FNE to change their work state to ACW.
Agent Availability Query Access Code	Mobile agents dial the Agent Availability Query Access Code FNE to understand their current work state. Depending on the agent state, Communication Manager plays a tone, for information on Agent Availability Query tones, see "Agent Availability Query dial tones" in <i>Avaya Aura</i> Call Center Elite Feature Reference.
Auto-In Access Code	Mobile agents dial the Auto-In Access Code FNE to change their work state to Auto-In.
Aux Work Access Code	Mobile agents dial the Aux Work Access Code FNE to change their work state to AUX Work.
Login Access Code	Mobile agents dial the Login Access Code FNE to login.
Logout Access Code	Mobile agents dial the Logout Access Code FNE to logout.
Manual-In Access Code	Mobile agents dial the Manual-In Access Code FNE to change their work state to Manual-In.

! Important:

You must administer corresponding FACs for all FNEs except for **Agent Availability Query Access Code**.

You can also define more FNEs that agents might need to use, such as **Idle Appearance Select**, **Conference Complete**, **Conference on Answer**, **Transfer Complete**, and **Transfer on Hang-Up**.

Policy Routing Table screen

Use the Policy Routing Table screen to administer and monitor the percentage allocation routing by assigning destination routes and target percentages.

Policy Routing Table administration commands

Use the following commands to administer the Policy Routing Table screen.

Command name	Parameter	Qualifier
add	policy-routing-table	1–8000 next
change	policy-routing-table	1–8000
display	policy-routing-table	1–8000
		['schedule']
remove	policy-routing-table	1–8000
list	policy-routing-table	No qualifier since the list command displays all the Policy Routing Tables

Policy Routing Table field descriptions

Field title	Field description
Number	To view the table number.
Name	To type a string of up to 15 characters as the name of the policy routing table. Any alphanumeric character is valid. You can leave this field blank.
Туре	To specify the type of algorithm that the policy routing table supports. The valid entry in this field is percentage.

Field title	Field description
Period	To specify the time period for resetting the call counts and actual percentages.
	Valid entries are:
	100_count (default): To reset the call count and the displayed percentage when the total calls for the policy routing table reach 100, which is when the total calls match the target routing pattern percentages. Using this field option ensures that the routing points have equal distribution of calls all the time.
	max_count: To maintain the call count until calls delivered to one of the VDNs exceed 65,400. At that point, calls continue to be distributed over the VDNs, but the system resets the call count when the actual percentages equal the targets for all of the VDNs at the same time.
	half-hour: To reset the call count at the top of the hour and at the 30–minute point.
	• hour: To reset the call count at the top of the hour.
	daily: To reset the call count every night at midnight.
	weekly: To reset the call count every Saturday at midnight.
Index	To display the sequential number of the rows. You can enter up to 15 route-to VDN entries in a policy routing table.
Route-to VDN	To assign a valid extension that is up to 13-digits long to which calls are to be routed. You can leave this field blank.
VDN Name	To view the assigned name of the VDN specified in the Route-to VDN field or a <i>name not assigned</i> message if the VDN name is not assigned.
	You must assign the name or change the name on the Vector Directory Number screen.
Target %	To specify the target percentage of total calls to be routed to a VDN. Valid entries are from 0 to 100. Use only whole numbers. Do not use fractions.
Actual %	To view the actual percentage of total calls routed to a VDN. The actual percentage is calculated to 6 decimal places, but the system displays only the first decimal place.

Field title	Field description
Call Counts	To view the current number of calls routed to a VDN.
Totals	To view the values in the Target % and Call Counts fields for all the assigned VDNs in the policy routing table. The total for the Target % field must add up to 100%.

Reason Code Names screen

Use the Reason Code Names screen to assign names to reason codes.

Reason Code Names administration commands

Use the following administration commands to access the Reason Code Names screen.

Command name	Parameter
display	reason-code-names
change	reason-code-names

Reason Code Names field descriptions

Field title	Field description
Aux Work	To type a name to be associated with the Auxiliary (AUX) work reason code. The name can be up to 16 characters long.
	You can also leave this field blank.
Logout	To type a name to be associated with the logout reason code. The name can be up to 16 characters long.
	You can also leave this field blank.
Interruptible	To determine whether the reason code can be interruptible. Valid entries are y and n .
	You cannot make the following types of reason codes interruptible:
	IP Failure Aux Work
	Maximum Agent Occupancy Aux Work
	Redirection on No Answer Aux Work
	Redirection on OPTIM Failure Aux Work

Service Hours Table screen

Use the Service Hours Table screen to specify the office hours. You can administer up to 999 different tables.

Service Hours Table administration commands

Command name	Parameter	Qualifier
add	service-hours-table	1 through 999
change	service-hours-table	1 through 999
display	service-hours-table	1 through 999
remove	service-hours-table	1 through 999
list	service-hours-table	none
list usage	service-hours-table	none

Service Hours Table field descriptions

Field title	Field description
Description	To include a description for the table. You can type an alphanumeric table name of up to 27 characters.
	You can leave this field blank.
Number	To view the table number.
Start and End	To define the range of service hours for each day of the week. Ensure that the start time is earlier than the end time.
	• hour - 0-23
	• minute - 0-59
	The hour range must be within the specified day, from 00:00 (midnight) till 23:59. If a time range goes past midnight, for example, Friday 19:00 to Saturday 02:00, type the time in two ranges. Set up the first range as Friday from 19:00 to 23:59 and the second range as Saturday from 00:00 to 01:59.
	A time is in the table from the first second of the start time, for example, 08:00:00. The time is in the table until the last second of the end time, for example, 17:00:59.
Use time adjustments from location	To indicate the location number on the Locations screen that specifies how to adjust the time zone offset and daylight saving time.

SIT Treatment for Call Classification screen

Use the SIT Treatment for Call Classification screen to specify the treatment of Special Information Tones (SITs) used for Outbound Call Management (OCM) type calls with USA tone characteristics. Use the port network TN744 Call Classifier circuit pack ports or H.248 Media Gateway internal tone detector resources in the classified mode to detect SITs.

SIT Treatment for Call Classification administration commands

Use the following administration commands to administer the SIT Treatment For Call Classification screen. In some cases, the screen displays only the most commonly used commands.

Command name	Parameter	Qualifier
change	sit-treatment	_
display	sit-treatment	['print' or 'schedule']

SIT Treatment for Call Classification field descriptions

In the field following each type of SIT, select **answered** to indicate that the call is classified as answered and is sent to a staffed agent or **dropped** to indicate that the call is classified as not answered and is not sent to a staffed agent.

Field title	Field description
Answering Machine Detected (AMD)	To determine the AMD treatment if the system detects an answering machine.
	AMD treatment has two separately administrable sub fields. Talk duration is for full seconds and Pause duration is for fractions of a second, separated by a display-only decimal point.
	The default for talk duration is 2 seconds from a range of 0.1 seconds to 5.0 seconds with increments of 0.1 seconds. The default for pause duration is 0.5 seconds from a range of 0.1 seconds to 2.0 seconds with increments of 0.1 seconds.
SIT Ineffective Other	To play an announcement following the SIT. For example, You are not required to dial a 1 when calling this number.
SIT Intercept	To play an announcement following the SIT. For example, XXX-XXXX has been changed to YYY-YYYY.
SIT No Circuit	To play an announcement following the SIT. For example, All circuits are busy, try again later.

Field title	Field description
SIT Reorder	To play an announcement following the SIT. For example, Your call did not go through, hang up and dial again.
SIT Unknown	To indicate that the network encountered an unknown situation or condition.
SIT Vacant Code	To play an announcement following the SIT. For example, Your call cannot be completed as dialed, check the number and dial again.

Station screen

Use the Station screen to administer hardphones and softphones.

Station administration commands

Use the following administration commands to administer the Station screen for individual phones.

Command name	Parameter	Qualifier
add	station	xxx or next, where xxx is the extension number.
change	station	xxx
display	station	xxx
list	station	No qualifier as this command lists all the administered stations.
remove	station	xxx

Station field descriptions

Field title	Field description
AUDIX Name	To type the name of the voice messaging system that is associated with the station.

Field title	Field description
Auto Answer	To determine whether Communication Manager must send all ACD and non-ACD calls to the station.
	Valid entries are:
	acd: To allow Communication Manager to send only ACD and direct agent calls to the station.
	all: To allow Communication Manager to send all ACD and non-ACD calls to the station.
	icom: To allow the station user to respond to an intercom call.
	none: To provide an audible ringing treatment to all calls that Communication Manager delivers to the station.
BCC	To determine compatibility when non-ISDN facilities are connected to ISDN facilities. If you assign 0 to this field, the Bearer Capability Class (BCC) indicates voice and voice-grade data.
	The field is applicable if the field option in the ISDN-BRI Trunks and ISDN-PRI fields on the System-Parameters Customer-Options screen is y.
Button Assignments	To assign feature buttons on the station.
Button Modules	To determine the number of button modules for the station.
COR	To assign a Class of Restriction (COR) number to the station.
cos	To assign a Class of Service (COS) number to the station.
Coverage Path 1	To assign a coverage path number or Time of Day (TOD) table to the station so that Communication Manager can route calls to the coverage point if no station user answers calls to the station.
Coverage Path 2	To assign a second coverage path or TOD table if the first coverage point is unavailable.
Coverage Msg Retrieval	To determine whether station users at the coverage point can retrieve Leave Word Calling (LWC) messages for the station from where Communication Manager sent the call to the coverage point. Valid entries are y and n .
	This field is applicable if you administer the LWC Reception field.

Field title	Field description
Enable Reachability for Station Domain Control	To enable Communication Manager to poll domain- controlled SIP stations and send the station reachability information to CTI applications that require to track the status of the station, on an individual station basis.
	Valid entries are s , y , or n . The default option is s .
	• y : Enables polling on the station.
	• n : Does not enable polling on the station.
	• s: The system setting determines the polling for this station.
Extension	To assign an extension number for the station.
Hunt-to Station	The extension number that Communication Manager must search when the station is busy.
Idle/Active Ringing (Callmaster)	To determine how calls must ring at a Callmaster [™] station.
	Valid entries are:
	continuous: For calls to the station to ring continuously.
	if-busy-single: For calls to the station to ring one time if the station user is busy.
	silent-if-busy: For calls to ring silently if the station user is busy.
	single: For calls to ring one time.
Location	To assign a number that identifies where the station is located in the system.
	This field is applicable if:
	• The field option in the Multiple Locations field on the System-Parameters Customer-Options screen is y .
	• The Type field on the Station screen is an H.323 or SIP phone type.
Lock messages	To determine whether other station users can gain access to voice messages.
LWC Activation	To determine whether to use LWC.

Field title	Field description
LWC Reception	To determine the storage point for LWC messages.
	Valid entries are:
	 audix: To store LWC messages in an AUDIX[™] system.
	none: To not store LWC messages.
	msa: To store LWC messages on Messaging Server Adjunct (MSA).
	spe. To store LWC messages in the system or on Switch Processor Element (SPE). The default is spe.
Message Lamp Ext	To assign the extension number of the station that is associated with the message waiting lamp.
Name	To assign a name to the station.
Per Station CPN - Send Calling Number	To determine whether Communication Manager must send Calling Party Number (CPN) information.
	Valid entries are:
	• n: To not send CPN information.
	• r: To restrict the level of CPN information.
	• y: To send CPN information.
	You can leave this field blank.
Port	To assign an auxiliary or analog port to the station.
Service Link mode	Valid entries are as-needed and permanent . Default value is as-needed . For EC500–enabled stations, this field set this field as as-needed .
SIP Trunk	To select a trunk that corresponds to the field entry on the Off-PBX-Telephone Station-Mapping screen.
	Valid entries are aar , ars , or a SIP trunk value from 1 to 2000.
Time of Day Lock Table	To assign a TOD Lock/Unlock table. Valid entries are from 1 to 5.
	You can leave this field blank.
TN	To assign a tenant partition number. Valid entries are from 1 to 250.
Type of 3PCC Enabled	To determine whether an Avaya Third Party Call Control (3PCC) or a Computer Telephony Integration (CTI) adjunct can control the station.
	Valid entries are Avaya and none .

Vector Directory Number screen

Use the Vector Directory Number screen to define VDNs for Call Vectoring. Each VDN is mapped to a call vector.

VDN administration commands

Use the following administration commands to administer the Vector Directory Number screen:



Note:

Square brackets [] indicate that the qualifier is optional. Single quotes ('') indicate that you must enter the text inside the quote exactly as shown or enter an abbreviated screen of the word. MAX is the maximum number available in your system configuration.

Command name	Parameter	Qualifier
add	vdn	xxxxx or next, where xxxx is the extension number of the VDN.
change	vdn	xxxxx
display	vdn	xxxxx ['print' or 'schedule']
list	vdn	xxxx 'count' 1-MAX ['print' or 'schedule']
	vdn	bsr xxx, which is the number of a BSR application plan.
remove	vdn	xxxxx
duplicate	vdn	master_ext [start nnnn] [count xx]

duplicate vdn

The command to create up to 16 duplicate VDNs.

master ext

The assigned extension number, which can be up to 16 digits, of the VDN

you want to duplicate.

[start nnnn]

An optional parameter that you can use to specify the number from which to start the number sequence for the duplicate VDN. If you do not specify a start number, the first available VDN after the master VDN gets selected

automatically. Only one VDN is selected.

[count xx]

An optional parameter that you can use to specify the number of duplicates. You can create a maximum of 16 duplicates. If you do not specify a start number, the first available VDN after the master VDN is selected

automatically. Only one VDN is selected.

Vector Directory Number field descriptions

Field title	Field description
1st Skill	To assign a skill number. Valid entries are from 1 to 8000. You can leave
2nd Skill	this field blank.

Field title	Field description
3rd Skill	This field is applicable if:
	 The Expert Agent Selection (EAS) field on the System-Parameter Customer-Options screen is active for the system. The field option in the Meet-me Conferencing field on the Vector Directory Number screen is n.
Acceptable Service Level (sec)	To determine the number of seconds within which agents must answer all calls to the Vector Directory Number (VDN). Basic Call Management System (BCMS) tracks the percentage of calls answered within the administered time.
	Valid entries are from 0 to 9999 seconds. You can leave this field blank.
	This field is applicable if:
	The BCMS/VuStats Service Level field on the System-Parameter Customer-Options screen is active for the system.
	The field option in the Measured field on the Vector Directory Number screen is internal or both .
Allow VDN Override	To determine whether to change the active VDN for a call. Valid entries are y and n .
	If you select n , the routed-to VDN does not replace the active VDN. This field option is the default.
	If you select y , the routed-to VDN replaces the active VDN and the VDN parameters associated with the call use the routed-to VDN.
	This field is applicable if the field option in the Meet-me Conferencing field on the Vector Directory Number screen is n .
Apply Ringback for Auto Answer Calls	To prevent ringback to the caller for a call that Communication Manager delivers to an agent in the auto-answer work mode.
	Valid entries are ${\bf y}$ and ${\bf n}$. The default is ${\bf y}$ to provide ringback to the caller.
	This field is applicable if the field option in the Auto Answer field on one of the following screens is y :
	• Station
	Agent LoginID
	Note:
	The field follows VDN Override Rules.

Field title	Field description	
Attendant Vectoring	To use the VDN as an Attendant Vectoring VDN. Valid entries are y and n .	
	If you select y , Communication Manager deletes all call center-related fields, such as skills and Best Service Routing (BSR).	
	Important:	
	Before you assign a VDN as an Attendant Vectoring VDN, ensure that the VDN is not administered on the following screens:	
	Console Parameters	
	Tenant Partitioning	
AUDIX Name	To type the name of the AUDIX [™] messaging system if the VDN is associated with an AUDIX [™] vector.	
	You must type the name of the messaging system as is available on the Adjunct Names screen.	
BSR Application	To use multisite BSR with a VDN. You can assign a 1-digit to 3-digit number to specify an application plan for the VDN.	
	This field is applicable if the Lookahead Interflow (LAI) and Vectoring (Best Service Routing) fields on the System-Parameter Customer-Options screen are active for the system.	
	Note:	
	The field follows VDN Override Rules.	

Field title	Field description
BSR Available Agent Strategy	To determine the BSR agent selection strategy for identifying the best split or skill for a call.
	Valid entries are:
	1st-found: BSR stops agent selection when a consider series finds an available, that is, idle agent.
	EAD-LOA: BSR selects the least occupied agent with the highest level for the skill.
	EAD-LOA is Expert Agent Distribution-Least Occupied Agent.
	EAD-MIA: BSR selects the most idle agent with the highest level for the skill.
	EAD-MIA is Expert Agent Distribution-Most Idle Agent.
	UCD-LOA: BSR selects the least occupied agent.
	UCD-LOA is Uniform Call Distribution-Least Occupied Agent.
	UCD-MIA: BSR selects the most idle agent.
	UCD-MIA is Uniform Call Distribution-Most Idle Agent.
	The EAD-LOA and EAD-MIA field options are available with EAS.
	The UCD-LOA field option is applicable if the Least Occupied Agent field or the Business Advocate field is active for the system.
	This field is applicable if:
	The Vectoring (Best Service Routing) field on the System-Parameter Customer-Options screen is active for the system.
	The field option in the BSR Tie Strategy field on the Vector Directory Number screen is system or alternate.
	* Note:
	The field follows VDN Override Rules.
BSR Local Treatment	To provide local audio feedback to interflow calls in a queue. You can administer this field on local and remote VDNs.
	Note:
	The field follows VDN Override Rules.

Field title	Field description
BSR Tie Strategy	To determine the BSR selection strategy when a tie occurs for calls from the same VDN.
	Valid entries are:
	1st-found: BSR uses the previously selected best choice as the best skill or best location.
	alternate: BSR alternates the selection algorithm during a tie between Expected Wait Time (EWT) calculations and Available Agent Strategy.
	For every tie, BSR selects the consider series with the tie to send the call instead of the first selected split/skill or location. This field option balances call routing when the cost of routing to remote locations is not a concern.
	system: BSR uses the BSR Tie Strategy field option on the Feature- Related System Parameters screen.
	This field option is the default.
	This field is applicable if the Vectoring (Best Service Routing) field on the System-Parameter Customer-Options screen is active for the system.
	* Note:
	The field follows VDN Override Rules.
COR	To assign a Class of Restriction (COR) to the VDN. Valid entries are from 0 to 995.
	You cannot leave this field blank.

Field title	Field description
Daylight Saving Rule	To define the Daylight Saving Time (DST) rule.
	You can apply the DST rule and the time zone offset to the goto time-of-day commands in the vector that you assign to the VDN. The time-of-day (TOD) calculations depend on the local time of the receiving call VDN.
	The assigned rule number applies start rules and stop rules that you administer in the Daylight Saving Rule field for that rule number.
	① Tip:
	Type list usage vdn-time-zone-offset to view all VDNs containing a DST rule.
	Valid entries are:
	• 0: The DST rule is inapplicable. If the system time has a daylight saving rule, the system deletes this rule before evaluation of the goto if time-of-day conditional.
	• 1–15: To use the rule as defined in this field.
	When you use a number other than 0 (zero), the system does not use the rule associated with the main server clock display time and the main server offset. 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 TOD step.
	system: To use the same DST rule as the system clock shown in the display/set time field.
	Note:
	The field follows VDN Override Rules.
Destination	To indicate whether calls are routed using a vector number or a policy routing table.
	Valid entries are Vector Number and Policy Routing Table .
Display VDN for Route-To DAC	To display the VDN for routed-to direct agent calls. Valid entries are y and n .
	You can use this field when the following vector commands with the Coverage field option as y route EAS direct agent calls:
	• adjunct routing
	• route-to number or route-to digits
	Note:
	The field follows VDN Override Rules.

Field title	Field description
Extension	To view the extension associated with the VDN. The extension is a number that begins with a valid first digit and conforms to the length defined by the system dial plan.
Measured Associated field: Report	To collect measurement data for the VDN. BCMS, Call Management System (CMS), or Avaya IQ collect data for reporting purpose.
Adjunct Calls as ACD	Valid entries are:
	• both
	• external
	• internal
	• none
	The field options both and internal are applicable if the BCMS field on the System-Parameter Customer-Options screen is active for the system.
	You can administer a CMS release on the Feature-Related System Parameters screen to change the field options to both or external .
	The system displays the Report Adjunct Calls as ACD field if the Measured field is present on the Vector Directory Number screen.
	To determine whether adjunct routed calls are identified in SPI messaging as ACD calls to the indicated skill when the call is adjunct routed. This feature is administered per VDN basis and the default value is no.
	Valid entries are and .
	• y: Adjunct routed calls are identified in SPI messaging as ACD calls. Therefore, an adjunct routed call is also counted as an ACD call for a skill in reports.
	• n: Adjunct routed calls are not counted as ACD calls. This is the default value
	If you set the Report Adjunct Calls as ACD field to yes then you must set the value of the the Measured field to either external or both .
	Note:
	The field follows VDN Override Rules.
Meet-me Conferencing	To determine whether to use the Meet-me Conferencing feature. Valid entries are y and n .
	This field is applicable if the Meet-me Conference field is active for the system.

Field title	Field description
Name	To assign an alphanumeric name of up to 27 characters that identifies the VDN. You can leave this field blank.
	When users press the new VDN-INFO button on H.323 or DCP phones, Communication Manager displays the complete VDN name of the active call.
Observe on Agent Answer	To determine when Communication Manager must connect an observer to a call.
	Valid entries are:
	• y: Communication Manager connects the observer to a call after delivering the call to an agent.
	 n: Communication Manager connects a non-Observe by Location observer to a call at the beginning of vector processing. This field option is the default.
Reporting for PC or POM Calls	To assign the VDN for Proactive Outreach Manager (POM) or Proactive Contact (PC) outbound calling.
Associated field: PC	Valid entries are y and n .
Predictive Reports Skill	If you have switch-classified outbound calling with PC, use this field and the associated skill field to activate the improved integration with the PC feature.
	If you have a POM system, use this field to activate the VDN for indication to reporting of a POM call to the VDN.
	This field is applicable if the field option in the Measured field on the Vector Directory Number screen is external or both .
	This associated field is applicable if you administer the following fields as \mathbf{y} :
	 The ASAI Link Plus Capabilities field or the Computer Telephony Adjunct Links field on the System-Parameter Customer-Options screen.
	The Reporting for PC or POM Calls field on the Vector Directory Number screen.
	If you assign a skill number, the VDN option applies to PC reporting. If you leave this field blank, the VDN option applies to POM calls.
Return Destination Associated field: Call Origin	To assign the VDN extension number to which an incoming trunk call is routed if the call returns to vector processing after an agent drops the call.
	You can leave this field blank.

Field title	Field description
	This associated field is applicable if you assign a VDN extension in the Return Destination field.
	Use this field to administer the types of calls that Communication Manager must redirect to the extension in the Return Destination field.
	Valid entries are:
	both: To apply VDN Return Destination (VRD) to external or internal calls.
	external: To apply VRD only to incoming external trunk calls directly to the VDN. This field option is the default.
	• internal: To apply VRD 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 Adjunct Switch Application Interface (ASAI) routed.
Send VDN as Called Ringing Name Over QSIG	To determine whether to display the VDN name to the receiver when the phone is ringing. Valid entries are y and n .
Service Objective (sec)	To determine the service level for the VDN. Valid entries are 1 to 9999. The default value is 20 seconds.
	This field is applicable if the Business Advocate field on the System-Parameter Customer-Options screen is active for the system.
	This field activates the Dynamic Queue Position (DQP) feature, which is also referred to as Service Objective by VDN. With DQP, you can place calls from multiple VDNs in a single skill queue, while maintaining different service objectives for each VDN.
TN	To assign a Tenant Partition number.
	Valid entries are 1 to 250.
	* Note:
	The field follows VDN Override Rules.
	Table continues

Field title	Field description
Used for BSR Polling	To prevent Communication Manager from sending polling-related messages to CMS or Avaya IQ for unmeasured calls.
	Valid entries are y and n .
	Use this field for unmeasured VDNs that route to vectors with a BSR reply-best command.
	The field is applicable for unmeasured VDNs assigned to a vector with no queue-to command, but with consider steps terminated by the reply-best command. The trunk groups that are used to route calls to the VDNs are also unmeasured.
	You can use this field if the field option in the Measured field on the Vector Directory Number screen is none .
	The field is applicable when you use Best Service Routing (BSR) and Look Ahead Interflow (LAI).
Use VDN Time Zone For Holiday Vectoring	To determine whether to use VDN Time Zone with Holiday Vectoring. Valid entries are y and n .
	If you select y , Communication Manager uses VDN Time Zone with Holiday Vectoring. If you select n , Communication Manager uses the system time with Holiday Vectoring.
	Note:
	The field follows VDN Override Rules.
VDN of Origin Annc. Extension	To type the extension of the VDN of Origin Announcement (VOA). You can leave this field.
	This field is applicable if the VDN of Origin Announcements field on the System-Parameter Customer-Options screen is active for the system.
	Note:
	The field follows VDN Override Rules.

Field title	Field description
VDN Override for ASAI Messages	To determine whether to send the called number as the active VDN for ISDN Trunk ASAI messages.
	If you select n in the Meet-me Conferencing field, this field follows VDN Override Rules when the system changes the <i>active</i> VDN for a call.
	Valid entries are:
	all: To allow VDN Override Rules for the ASAI messages so that Communication Manager can use the active VDN for the called number for all types of calls to the VDN, including local or internal calls and external incoming ISDN trunk calls.
	ISDN Trunk: When an incoming ISDN trunk call is routed to the VDN, the called number information sent in the ASAI event and adjunct route request ASAI messages is the active VDN extension that becomes associated with the call based on the VDN Override Rules. This field option is not applicable to local or internal calls.
	no: The called number information sent for the Call Offered, Alerting, Queued, and Connect ASAI event notification messages and the adjunct-request message is always the called VDN extension in the called number Information Element (IE) sent in the incoming ISDN SETUP message or the called number of the local call and does not change after routing to the called VDN and subsequent routed-to VDNs.
	This field is applicable if:
	The ASAI Link Core Capabilities field on the System-Parameter Customer-Options screen is active for the system.
	The G3 Version field on the System-Parameter Customer-Options screen is V10 or later.
VDN Timed ACW Interval Associated field: After Xfer or	To assign an After Call Work (ACW) interval for auto-in agents who receive calls from this VDN.
Held Call Drops	If you administer this field, Communication Manager places an auto-in agent who receives a call from this VDN in the ACW mode after the agent releases the call.
	When the administered interval is complete, Communication Manager makes the agent available to receive calls.
	The field has priority over the Timed ACW Interval field on the Hunt Group screen.
	Note:
	The field follows VDN Override Rules.

Field title	Field description
	For incoming ACD or direct agent calls, Communication Manager places an auto-in agent in the Timed After Call Work (TACW) mode, instead of making the agent available, if the held caller drops or the agent transfers the call.
	Use this associated field for the agents in a hunt group or for calls delivered from a VDN when the Timed ACW Interval field is administered to a non-zero value.
VDN Time-Zone Offset	To apply VDN Time Zone Offset to the Communication Manager clock when a TOD vector command is executed. The Communication Manager clock handles the DST changes by using the existing operation.
	The syntax is +HH:MM, that is, [+ or -] [0 - 23] : [0 - 59]
	Valid entries are:
	• + or -: Use the minus (-) sign if the VDN local time is earlier than the server local time.
	Use the plus (+) sign if the VDN local time is later than the server local time.
	• 0 - 23: Time in hours
	• 0 - 59: Time in minutes
	The default is +00:00. When the default is set, the Communication Manager time is used without modification.
	Note:
	The field follows VDN Override Rules.

VDN variables



Note:

The field follows VDN Override Rules.

Var	To assign up to nine VDN variables from V1 to V9.
Description	To type up to 15 alphanumeric characters to describe each VDN variable.
Assignment	To assign unvalidated decimal numbers of up to 16 digits to each VDN variable.
	You can leave this field blank to assign no decimal numbers to the VDN variables.
VDN Occasida Dulas assess that the assessment as that are assessinted with each VDN and also as a second as	

VDN Override Rules means that the parameters that are associated with each VDN can change based on the field settings of the next VDN that the call routes to.

Vector Routing Table screen

Use the Vector Routing Table screen to store Automatic Number Identification (ANI) or digits that you refer to in the goto vector steps.

The fields on this screen are applicable if the **Vectoring (G3V4 Enhanced)** field on the System-Parameters Customer-Options screen is active for the system.

Vector Routing Table administration commands

Use the following commands to administer Vector Routing Tables (VRT). You can use the List Usage command to view the vectors and digit fields.

Command name	Parameter	Qualifier
add	vrt	1–999
		next
change	vrt	1– 999
display	vrt	1–999
		['schedule']
remove	vrt	1–999
list	vrt	No qualifier since the list command displays all the Vector Routing Tables

Vector Routing Table field descriptions

Field title	Field description
Name	To assign an alphanumeric name of up to 15 characters. You can leave this field blank.
Number	To enter a number. The default is blank.
	You can use wildcards, such as a plus (+) sign and a question (?) mark. The plus (+) sign represents a group of digits. The question (?) mark represents a single digit.
	The field is limited to 16 characters with the following restrictions:
	• You can use a plus sign (+), a question mark (?), or numbers from 0 to 9. No other entries are valid.
	 You can use a plus sign (+) as the first or the last character in the number field. You cannot use this character as the sixteenth character of the number field.
	You can use many question (?) marks.
	You cannot use blank spaces in the number field.
	If you leave the field blank, the system stores a null value.

Field title	Field description
Sort	To determine whether to sort the digit fields. Valid entries are y and n .
You can sort the number fields as follows:	
	Plus (+) signs sort first.
	Question (?) marks sort second.
	All numbers from 0 to 9 sort last.
	Leading zeros are significant that means that 02 sorts ahead of a 2 followed by a space.

VuStats Display Format screen

Use the VuStats Display Format screen to describe the content and layout of the information that appears on the phone display when an agent presses **vu-display**.

The system has 50 different display formats. You can modify the first display, which is a predefined example format. Each display format can contain up to 10 data items. However, the amount of data to be displayed is limited to 40 characters due to the physical limitations of the phones.

VuStats Display Format administration commands

Use the following administration commands to administer the VuStats Display Format screen.

Command name	Parameter	Qualifier
change	vustats-display-format	1-50
display	vustats-display-format	1-50
list	vustats-display-format	(optional) count 1-50

You can use the list vustats-display-format command to view the following information about the administered VuStats displays:

- Data items, such as data type, format, threshold, and skill reference
- · Format description
- · Format number
- Next format number to indicate links with the current VuStats display
- Number of intervals
- · Object type
- Update interval

VuStats Display Format field descriptions

Field title	Field description	
Data Field Character	To assign a character that is used in the Format Description field to identify the position and length of each data field. The default field option is \$.	
	You can assign another character if you use the default character for fixed text in the Format Description field. Any character is valid except a space.	
Display Interval	To determine the interval, in seconds, for which the phone displays data if you do not update the Update Interval field.	
	Valid entries are:	
	• 5: To display data for 5 seconds.	
	• 10: To display data for 10 seconds.	
	• 15: To display data for 15 seconds.	
	• 30: To display data for 30 seconds.	
	not cleared: To display data until another operation displaces the data or the agent using the phone presses normal.	

Field title	Field description
Format Description	To define the layout for the 40-character display.
	You can assign the starting position and the length of the data items by typing an optional label for each field followed by the appropriate number of data field characters, such as \$. Each \$ represents one character in the display. For example, for data of five characters, type \$\$\$\$.
	Some data types have preset field length limits based on the Communication Manager administration. For example, the system takes the acceptable-service-level data type from the Acceptable Service Level (sec) field on the Vector Directory Number screen. As you can type a maximum of four characters in this field, do not create a VuStats Display field that consists of more than four characters for the acceptable-service-level data type. Other data types have similar limits.
	Field lengths for data items that appear as time must match the field option in the Format field. Account for possible colons when the display is in a time format.
	Format descriptions can be all text such as a message for the day. The format description can also be all data fields, but the users must memorize the labels or use customer-provided overlays.
	If the numeric data for a field is too large for the number of data field characters, the VuStats display shows asterisks instead of data. If the name database items are too large for the number of data field characters, the VuStats display truncates data to fit the field limit. The split or skill objective, as entered on the Hunt Group screen, appear as asterisks if the information exceeds the data field limit.
Format Number	The system generates a format number when the system creates a VuStats display. You cannot change this number. You can create 50 different display formats.
	You can modify Format Number 1, which is a predefined sample format.
Int	To view the number of measurement intervals.
	This is a display-only field and is available when you use the list vustats-display-format command to compare VuStats displays.

Field title	Field description	
Next	To view the number of the next display if the current display is followed by another display format. If the current display format is not followed by another display format, the system does not display any numbers.	
	This is a display-only field and is available when you use the list vustats-display-format command to compare VuStats displays.	
Next Format Number	To link this display to another display of the same object type.	
	Valid entries are from 1 to 50 or none .	
No	To view the identifying number of each VuStats display format.	
	This is a display-only field and is available when you use the list vustats-display-format command to compare VuStats displays.	
Number of Intervals	To determine the number of BCMS intervals to collect data when you specify the interval as the period for a historical data type.	
	You can assign a number from 1 to 24 or leave the field blank for the current interval.	
	If you assign 24 intervals and field option in the BCMS/VuStats Measurement Interval field on the Feature-Related System Parameters screen is hour, the system collects hourly information for the last 24 hours. If the field option in the BCMS/VuStats Measurement Interval field is half-hour, the system collects half-hourly information for the last 12 hours.	

Field title	Field description
Object Type	To determine the type of object for which the system displays data.
	Valid entries are:
	agent: To provide agents with their performance statistics or statistics about the splits or skills that agents log in to.
	agent-extension: To provide supervisors with statistics about agents or splits/skills that agents log in to. VuStats can automatically display statistics for an agent if you administer agent login IDs or BCMS/Vustats login IDs. Supervisors can also type the login ID of an agent to view the performance statistics for the agent.
	split/skill: To display the statistics about a split or skill. You must administer the Measured field on the Hunt Group screen to display the split or skill statistics to the phone user.
	trunk-group: To display the statistics about a trunk group. You must administer the Measured field on the Trunk Group screen to display the trunk statistics to the phone user.
	vdn: To display the statistics about a VDN. You must administer the Measured field on the Vector Directory Number screen to display the VDN statistics to the phone user.
Update Interval	To determine the intervals when the system updates data.
	Valid entries are:
	• 10: To update data after 10 seconds.
	• 20: To update data after 20 seconds.
	• 30: To update data after 30 seconds.
	60: To update data after 60 seconds.
	• 120: To update data after 120 seconds.
	no-update: To not update data.
	The system displays data only for the interval specified in the Display Interval field.
	polled: To update data based on the field option in the BCMS/VuStats Measurement Interval field on the Feature-Related System Parameter screen.

VuStats Display Format data types

Agent and agent-extension data types

VuStats data type	Description	BCMS report	Field or column heading in the report
acd-calls	Split or skill calls and direct agent calls answered by a staffed agent.	Split Status/VDN Status/Agent	ACD CALLS
agent-extension	The extension for an agent. If BCMS/ VuStats login IDs or EAS is active, the data type shows the login ID or the agent.	Split Status	Login ID or EXT
agent-name	The administered name for an agent.	Split Status/Agent/ Agent Summary	Agent
agent-state	The current work mode of an agent.	Split Status	STATE
average-acd- call-time	The average of hold time plus talk time.	_	_
average-acd- talk-time	The average time an agent spent talking on completed ACD calls during a specified time period for all internally measured splits or skills that the agent was logged in to. This time does not include the time a call was ringing or was on hold at the agent terminal.	Agent/Agent Summary	AVG TALK TIME
average- extension-time	The average time that an agent spends on non-ACD calls while logged in to a minimum of one split or skill during the reported interval. This average does not include the time when the agent was holding EXTN calls.	Agent/Agent Summary	AVG EXTN TIME
call-rate	The current rate of ACD calls handled by each agent every hour for all split or skills.	_	_
current-reason- code	The number of the reason code associated with the current Auxiliary (AUX) work mode or with agent logout.	_	_
current-reason- code- name	The name of the reason code associated with the current AUX work mode or with agent logout.		
elapsed-time-in- state	The duration that a staffed agent was in the current state.	_	_

VuStats data type	Description	BCMS report	Field or column heading in the report
extension-calls	The number of incoming and outgoing non-ACD calls that an agent completed while logged in to a minimum of one split or skill.	Agent	EXTN CALLS
extension- incoming- calls	The number of non-ACD calls that an agent received while logged in to a minimum of one split or skill.	Split	EXT IN CALLS
extension- outgoing-calls	The number of non-ACD calls that an agent placed while logged in to a minimum of one split or skill.	Split	EXT OUT CALLS
percent-acd-call- time	The current calculated occupancy for the staffed agent.	_	_
	The data type indicates the percentage of time that the agent talks and holds ACD calls and is calculated as follows:		
	ACD time + hold time (ACD calls only) / (staffed time in interval + 100)		
shift-acd-calls	The number of ACD calls answered by an agent during the administered period.	_	_
shift-aux-time-1	The time that an agent spent in AUX work for reason code 1 during the administered period.	_	_
shift-aux-time-2	The time that an agent spent in AUX work for reason code 2 during the administered period.	_	_
shift-aux-time-3	The time that an agent spent in AUX work for reason code 3 during the administered period.	_	_
shift-aux-time-4	The time that an agent spent in AUX work for reason code 4 during the administered period.	_	_
shift-aux-time-5	The time that an agent spent in AUX work mode for reason code 5 during the administered period.	_	_
shift-aux-time-6	The time that an agent spent in AUX work for reason code 6 during the administered period.	_	_

VuStats data type	Description	BCMS report	Field or column heading in the report
shift-aux-time-7	The time that an agent spent in AUX work for reason code 7 during the administered period.	_	_
shift-aux-time-8	The time that an agent spent in AUX work for reason code 8 during the administered period.	_	
shift-aux-time-9	The time that an agent spent in AUX work for reason code 9 during the administered period.	_	
shift-aux-time-all	The time that an agent spent in AUX work for all reason codes during the administered period.	_	_
shift-aux-time- default	The time that an agent spent in AUX work for the default reason code 0 during the administered period.	_	_
shift-aux-time- non-default	The time that an agent spent in AUX work for reason codes 1 through 99 during the administered period.	_	
shift-aux-time- reason-code	The time that an agent spent in AUX work for the current reason code during the administered period.	_	_
shift-average- acd-talk-time	The average talk time for ACD calls for an agent during the administered period.	_	_
skill-level	The skill level at which the skill was assigned to an agent.	_	
split-acceptable- service-level	The number of seconds within which agents must answer calls so that the system can treat calls as accepted.	Split Status	Acceptable Service Level
	The system identifies calls for each hunt group and begins tracking calls when calls enter a hunt group queue.		
split-acd-calls	Split or skill calls and direct agent calls answered by an agent.	System Status/Split/ Split Summary	ACD CALLS
split-after-call- sessions	The number of times that all agents entered After Call Work (ACW) for a split or skill.	_	_

VuStats data type	Description	BCMS report	Field or column heading in the report
split-agents- available	The number of staffed agents currently available to receive ACD calls to a split or skill. This number includes agents in the auto-in or manual-in work mode.	Split Status	Avail
split-agents-in- after- call	The number of agents currently in ACW for a split or skill.	Split Status	ACW
split-agents-in- aux-1	The number of agents of a split or skill who are currently in AUX work with reason code 1.	_	_
split-agents-in- aux-2	The number of agents of a split or skill who are currently in AUX work with reason code 2.	_	_
split-agents-in- aux-3	The number of agents of a split or skill who are currently in AUX work with reason code 3.	_	_
split-agents-in- aux-4	The number of agents of a split or skill who are currently in AUX work with reason code 4.	_	_
split-agents-in- aux-5	The number of agents of a split or skill who are currently in AUX work with reason code 5.	_	_
split-agents-in- aux-6	The number of agents of a split or skill who are currently in AUX work with reason code 6.	_	_
split-agents-in- aux-7	The number of agents of a split or skill who are currently in AUX work with reason code 7.	_	_
split-agents-in- aux-8	The number of agents of a split or skill who are currently in AUX work with reason code 8.	_	_
split-agents-in- aux-9	The number of agents of a split or skill who are currently in AUX work with reason code 9.	_	_
split-agents-in- aux-all	The total number of agents of a split or skill who are currently in AUX work for all reason codes.	Split Status	AUX
split-agents-in- aux- default	The number of agents of a split or skill who are currently in AUX work with the default reason code.	_	_

VuStats data type	Description	BCMS report	Field or column heading in the report
split-agents-in- aux- non-default	The number of agents of a split or skill who are currently in AUX work with reason codes 1 through 99.	_	_
split-agents-in- other	The number of agents who are on calls for another split or skill, are in ACW for another split or skill, have calls on hold but are not in another work mode, have calls ringing at their stations, or are dialing numbers while in the auto-in or manual-in work mode.	Split Status	Other
split-agents-on- acd-calls	The number of agents who are currently on split or skill or direct agent calls for a split or skill.	Split Status	ACD
split-agents-on- extension-calls	The number of agents in a split or skill who are currently on non-ACD calls.	Split Status	Extn
split-agents- staffed	The number of agents currently logged in to a split or skill.	Split Status	Staffed
split-average- acd- talk-time	The average talk time for ACD calls during a period or day for a specified split or skill.	System Status/Split/ Split Summary	AVG TALK TIME
split-average- after- call-time	The average time for call-related ACW completed by agents of a split or skill. This data type is similar to the average after-call-time data type, but is available only for the agent and agent-extension object types.	System Status	AVG AFTER CALL
	The system records the call-related ACW time when an agent leaves ACW. If an agent is in call-related ACW when an interval ends, the system records the ACW time for the interval in which the agent leaves ACW.		
split-average- speed- of- answer	The average speed for answering completed split or skill and direct agent calls to a split or skill.	System Status/Split/ Split Summary	AVG SPEED ANS
split-average- time-to- abandon	The average time that calls waited in a queue and were ringing before the callers abandoned the calls.	System Status/Split/ Split Summary	AVG ABAND TIME
split-call-rate	The current hourly rate of ACD calls handled by each agent for a split or skill.	_	_

VuStats data type	Description	BCMS report	Field or column heading in the report
split-calls- abandoned	The number of calls that left the queue or abandoned ringing, provided the queue is the first split or skill that the call queued to.	System Status/Split/ Split Summary	ABAND CALLS
split-calls- flowed-in	The total number of calls for a split or skill that were received as a coverage point (intraflowed) from another internally measured split or skill or were call-forwarded (interflowed) to the split or skill.	Split/Split Summary	FLOW IN
split-calls- flowed-out	The total number of calls for a split or skill that successfully extended to the split or skill coverage point, call-forwarded, or answered using call pick up.	Split/Split Summary	FLOW OUT
split-calls-waiting	The number of calls that reached a split or skill but were not answered, abandoned, or outflowed.	System Status	CALLS WAIT
split-extension	The administered extension for a split or skill.	_	_
split-name	The administered name for a split or skill.	Split/Split Status/Split Name/System Status	SPLIT
split-number	The administered number for a split or skill.	Split/Split Status/Split Name/System Status	SPLIT
split-objective	The administered objective for a split or skill.	_	_
split-oldest-call- waiting	The time that the oldest call has been waiting for a split or skill.	System Status	OLDEST CALL
split-percent-in- service-level	The percentage of calls that an agent answered within the administered service level for a split or skill.	System Status/Split/ Split Summary	% WITHIN SERVICE LEVEL
split-total-acd- talk-time	The total time that an agent spent on a split or skill calls and on direct agent calls to the split or skill.	_	_
split-total-after- call-time	The total time that an agent spent in call-related ACW for a split or skill and non-call-related ACW for any split or skill during a specific time period, excluding the time spent on incoming or outgoing extension calls while in ACW.	Split/Split Summary	TOTAL AFTER CALL

VuStats data type	Description	BCMS report	Field or column heading in the report
split-total-aux- time	The total time that an agent spent in AUX work for a split or skill.	Split/Split Summary	TOTAL AUX/OTHER
total-acd-call- time	The total talk time plus the total hold time for split or skill and direct agent calls.	_	_
total-acd-talk- time	The total time that agents spent talking on split or skill calls and direct agent calls.	_	_
total-after-call- time	The total time that an agent spent in call-related or non-call-related ACW for any split during a specific time period, excluding the time spent on incoming or outgoing extension calls while in ACW.	Agent/Agent Summary	TOTAL AFTER CALL
	With EAS, all non-call-related ACW time is associated with the first skill that the agent logged in to.		
total-aux-time	The total time that an agent spent in AUX work for all splits or skills that the agent was logged in to.	Agent/Agent Summary	TOTAL AUX/OTHER
	If an agent entered AUX work in one interval but ended AUX work in another, the system tracks the time spent in each interval.		
	Agent reports also include OTHER time		
total-available- time	The total time that an agent is available in a split or skill.	Agent	TOTAL AVAIL TIME
total-hold-time	The total time ACD calls are on hold at an agent phone. This time is the caller hold time and is independent of the agent work mode. The system does not include the hold time for non-ACD calls.	Agent	TOTAL HOLD TIME
total-staffed-time	The total time that an agent is logged in to more than one split or skill during a specific period or day. An agent is clocked for staff time as long as the agent is logged in to a split or skill.	Agent	TOTAL TIME STAFFED

Required and allowed fields for agent and agent-extension data types

VuStats data type	Format	Period	Threshold	Reference
acd-calls	_	required	allowed	required
agent-extension	_	_	_	_
agent-name	_	_	_	_
agent-state	_	_	_	required
average-acd-call-time	required	required	allowed	_
average-acd-talk-time	required	required	allowed	required
average-extension-time	required	required	allowed	_
call-rate	_	required	allowed	_
current-reason-code	_	_	allowed	_
current-reason-code-name	_	_	allowed	_
elapsed-time-in-state	_	_	_	_
extension-calls	_	required	allowed	_
extension-incoming-calls	_	_	allowed	_
extension-outgoing-calls	_	_	allowed	_
percent-acd-call-time	_	required	allowed	_
shift-acd-calls	_	_	allowed	required
shift-aux-time-1	required	_	allowed	_
shift-aux-time-2	required	_	allowed	_
shift-aux-time-3	required	_	allowed	_
shift-aux-time-4	required	_	allowed	_
shift-aux-time-5	required	_	allowed	_
shift-aux-time-6	required	_	allowed	_
shift-aux-time-7	required	_	allowed	_
shift-aux-time-8	required	_	allowed	_
shift-aux-time-9	required	_	allowed	_
shift-aux-time-all	required	_	allowed	_
shift-aux-time-default	required	_	allowed	_
shift-aux-time-non-default	required	_	allowed	_
shift-aux-time-reason-code	required	_	allowed	_
shift-average-acd-talk-time	required	_	allowed	required
skill-level	_	_	_	required
split-acceptable-service-level	required		_	required
split-acd-calls	_	required	allowed	required
split-after-call-sessions	_	-	allowed	required

VuStats data type	Format	Period	Threshold	Reference
split-agents-available	_	_	allowed	required
split-agents-in-after-call	_	_	allowed	required
split-agents-in-aux-1	_	_	allowed	required
split-agents-in-aux-2	_	_	allowed	required
split-agents-in-aux-3	_	_	allowed	required
split-agents-in-aux-4	_	_	allowed	required
split-agents-in-aux-5	_	_	allowed	required
split-agents-in-aux-6	_	_	allowed	required
split-agents-in-aux-7	_	_	allowed	required
split-agents-in-aux-8	_	_	allowed	required
split-agents-in-aux-9	_	_	allowed	required
split-agents-in-aux-all	_	_	allowed	required
split-agents-in-aux-default	_	_	allowed	required
split-agents-in-aux-non-default	_	_	allowed	required
split-agents-in-other	_	_	allowed	required
split-agents-on-acd-calls	_	_	allowed	required
split-agents-on-extension-calls	_	_	allowed	required
split-agents-staffed	-	_	allowed	required
split-average-acd-talk-time	required	required	allowed	required
split-average-after-call-time	required	_	allowed	required
split-average-speed-of-answer	required	required	allowed	required
split-average-time-to-abandon	required	required	allowed	required
split-call-rate	_	_	allowed	required
split-calls-abandoned	_	required	allowed	required
split-calls-flowed-in	_	required	allowed	required
split-calls-flowed-out	_	required	allowed	required
split-calls-waiting	_	_	allowed	required
split-extension	_	_	_	required
split-name	_	_	_	required
split-number	_	_	-	required
split-objective	_	_	_	required
split-oldest-calling-waiting	required	_	allowed	required
split-percent-in-service-level	_	required	allowed	required
split-total-acd-talk-time	required	required	allowed	required
split-total-after-call-time	required	required	allowed	required

VuStats data type	Format	Period	Threshold	Reference
split-total-aux-time	required	required	allowed	required
time-agent-entered-state	_	_	_	required
total-acd-call-time	required	required	allowed	_
total-acd-talk-time	required	required	allowed	_
total-after-call-time	required	required	allowed	_
total-aux-time	required	required	allowed	_
total-available-time	required	required	allowed	_
total-hold-time	required	required	allowed	_
total-staffed-time	required	required	allowed	_

Split or Skill data types

VuStats data type	Description	BCMS report	Field or column heading in the report
acceptable-service- level	The number of seconds within which agents must answer calls.	Split Status/Split	Acceptable Service Level
	The system identifies this data type for each hunt group and begins tracking the data type when a call enters a vector.		
acd-calls	Split or skill calls and direct agent calls answered by an agent.	Split Status/VDN Status/Agent	ACD CALLS
after-call sessions	The number of times that all agents have entered the After Call Work (ACW) mode.	_	_
agents-available	The number of agents who are currently available to receive ACD calls. The system includes agents in the auto-in or manual-in work mode.	Split Status	Avail
agents-in-after-call	The number of agents who are currently in the ACW mode.	Split Status	ACW
agents-in-aux-1	The number of agents currently in the Auxiliary (AUX) work mode for reason code 1 for the referenced skill.	_	_
agents-in-aux-2	The number of agents currently in AUX work for reason code 2 for the referenced skill.	_	_

VuStats data type	Description	BCMS report	Field or column heading in the report
agents-in-aux-3	The number of agents currently in AUX work for reason code 3 for the referenced skill.	_	_
agents-in-aux-4	The number of agents currently in AUX work for reason code 4 for the referenced skill.	_	_
agents-in-aux-5	The number of agents currently in AUX work for reason code 5 for the referenced skill.	_	_
agents-in-aux-6	The number of agents currently in AUX work for reason code 6 for the referenced skill.	_	_
agents-in-aux-7	The number of agents currently in AUX work for reason code 7 for the referenced skill.	_	_
agents-in-aux-8	The number of agents currently in AUX work mode for reason code 8 for the referenced skill.	_	_
agents-in-aux-9	The number of agents currently in AUX work for reason code 9 for the referenced skill.	_	_
agents-in-aux-all	The number of agents currently in AUX work for all reason codes for the referenced split or skill.	Split Status	AUX
agents-in-aux-default	The number of agents currently in AUX work for the default reason code 0 for the referenced split or skill.	_	_
agents-in-aux-non- default	The number of agents currently in AUX work for reason codes 1 through 99 for the referenced skill.	_	_
agents-in-other	The number of agents who are currently on a call for another split, in the ACW work mode for another split, have a call on hold but are not in another state, or have a call ringing at their terminal, or are dialing a number when in the auto-in or manual-in work mode.	Split Status	Other
agents-on-acd-calls	The number of agents who are currently on split or skill or direct agent ACD calls for a specific split.	Split Status	ACD

VuStats data type	Description	BCMS report	Field or column heading in the report
agents-on-extension- calls	The number of agents in a specific split who are currently on non-ACD calls.	Split Status	Extn
agents-staffed	The number of agents who are currently logged in to the specified split.	Split Status	Staffed
average-acd-talk- time	The average talk time for ACD calls during a specific period or day for a specified split.	System Status/Split	AVG TALK TIME
average-after-call- time	The average time for call-related ACW completed by agents in this split. Call-related ACW time is recorded when an agent leaves ACW. If an agent is in call-related ACW when an interval completes, the system records the ACW time for the interval in which the agent leaves ACW.	System Status	AVG AFTER CALL
average-speed-of- answer	The average speed for answering for a split or skill and direct agent calls that have completed for a specified split or skill during a specified time. This includes queue time and ringing time for this split.	System Status/Split	AVG SPEED ANS
average-time-to- abandon	The average time that calls were in a queue before the callers abandoned the calls.	System Status/Split	AVG ABAND TIME
call-rate	The current rate of ACD calls handled by each agent every hour for all splits or skills.	_	_
calls-abandoned	The number of abandoned calls.	System Status/Split	ABAND CALLS
calls-flowed-in	The total number of calls for a specific split that were received as a coverage point (intraflowed) from another internally-measured split, or were call-forwarded (interflowed) to the split.	Split Report/Split Summary	FLOW IN
	The system does not include calls that were interflowed from a remote server by the Look Ahead Interflow (LAI) feature.		

VuStats data type	Description	BCMS report	Field or column heading in the report
calls-flowed-out	The number of calls the split extended to its coverage point, calls that call-forward out or are answered by call pickup, calls that queued to this split as a primary split and were answered or abandoned from ringing in another split.	Split Report/Split Summary	FLOW OUT
calls-waiting	The number of calls that reach a split or skill but have not been answered, abandoned, or outflowed.	System Status	CALLS WAIT
oldest-call-waiting	The time that the oldest call has been waiting in the split or skill.	System Status	OLDEST CALL
	The system begins tracking when a call enters a split or skill.		
percent-in-service- level	The percentage of calls offered to the split that agents answered within the service level administered on the hunt group screen.	System Status/ Split Report/Split Summary	% IN SERV LEVL
split-extension	The administered extension for a split.	_	_
split-name	The administered name for a split.	Split/Split Status/Split Name/System Status	SPLIT
split-number	The administered number for a split.	Split/Split Status/Split Name/System Status	SPLIT
split-objective	The administered objective for a split.	_	_
total-acd-talk-time	The total time agents spent talking on split or skill calls and direct agent calls for this split.	_	_
total-after-call-time	The total time agents spent in call- related or non-call-related ACW for any split during a specific time period.	Split Report/Split Summary	TOTAL AFTER CALL
total-aux-time	The total time agents spent in AUX work for all reason codes for the referenced split or skill during the administered period.	Split Report/Split Summary	TOTAL AUX/ OTHER

Required and allowed fields for split data types

VuStats data type	Format	Period	Threshold
acceptable-service-level	required	_	-
acd-calls	_	required	allowed
after-call sessions	_	_	allowed
agents-available	_	_	allowed
agents-in-after-call	_	_	allowed
agents-in-aux-1	_	_	allowed
agents-in-aux-2	_	_	allowed
agents-in-aux-3	_	_	allowed
agents-in-aux-4	_	_	allowed
agents-in-aux-5	_	_	allowed
agents-in-aux-6	_	_	allowed
agents-in-aux-7	_	_	allowed
agents-in-aux-8	_	_	allowed
agents-in-aux-9	_	_	allowed
agents-in-aux-all	_	_	allowed
agents-in-aux-default	_	_	allowed
agents-in-aux-non-default	_	_	allowed
agents-in-other	_	_	allowed
agents-on-acd-calls	_	_	allowed
agents-on-extension-calls	_	_	allowed
agents-staffed	_	_	allowed
average-acd-talk-time	required	required	allowed
average-after-call-time	required	_	allowed
average-speed-of-answer	required	required	allowed
average-time-to-abandon	required	required	allowed
call-rate	_	_	allowed
calls-abandoned	_	required	allowed
calls-flowed-in	_	required	allowed
calls-flowed-out	_	required	allowed
calls-waiting	_	_	allowed
oldest-calling-waiting	required	_	allowed
percent-in-service-level	_	required	allowed
split-extension	_	_	_
split-name	_	_	_

VuStats data type	Format	Period	Threshold
split-number	_	_	-
split-objective	_	_	-
total-acd-talk-time	required	required	allowed
total-after-call-time	required	required	allowed
total-aux-time	required	required	allowed

Trunk Group data types

VuStats data type	Description	BCMS report	Field or column heading in the report
average-incoming- call-time	Average holding time for incoming trunk calls.	Trunk Group	INCOMING TIME
average-outgoing- call-time	Average holding time for outgoing trunk calls.	Trunk Group	OUTGOING TIME
incoming-abandoned-calls	Incoming calls abandoned during a specified time period for a specified trunk group.	Trunk Group	INCOMING ABAND
incoming-calls	Incoming calls carried by a specified trunk group.	Trunk Group	INCOMING CALLS
incoming-usage	The total trunk holding time for incoming calls in hundred call seconds.	Trunk Group	INCOMING CCS
number-of-trunks	The number of trunks in a specified trunk group.	Trunk Group	Number of Trunks
outgoing-calls	The number of outgoing calls carried by a specified trunk group.	Trunk Group	OUTGOING CALLS
outgoing-completed- calls	The number of outgoing calls that received answer supervision or answer timeout.	Trunk Group	OUTGOING COMP
outgoing-usage	The total trunk holding time for outgoing calls in hundred call seconds.	Trunk Group	OUTGOING CCS
percent-all-trunks- busy	The percent of time all the trunks in a specified trunk group were busy during a specified period or day.	Trunk Group	% ALL BUSY
	The system begins tracking when the last trunk is seized.		
percent-trunks-maint- busy	The percent of time trunks were busied out for maintenance during a specified period or day.	Trunk Group	% TIME MAINT

VuStats data type	Description	BCMS report	Field or column heading in the report
trunk-group-name	The name administered for a specific trunk group.	Trunk Group	Trunk Group Name
trunk-group-number	The number administered for a specific trunk group.	Trunk Group	Trunk Group Number
trunks-in-use	The number of trunks currently in use.	_	_
trunks-maint-busy	The number of trunks currently busied out for maintenance.	_	_

Required and allowed fields for trunk group data types

VuStats data type	Format	Period	Threshold
average-incoming-call-time	required	required	allowed
average-outgoing-call-time	required	required	allowed
incoming-abandoned-calls	_	required	allowed
incoming-calls	_	required	allowed
incoming-usage	required	required	allowed
number-of-trunks	_	_	-
outgoing-calls	_	required	allowed
outgoing-completed-calls	_	required	allowed
outgoing-usage	required	required	allowed
percent-all-trunks-busy	_	required	allowed
percent-trunks-maint-busy	_	required	allowed
trunk-group-name	_	_	_
trunk-group-number	_	_	-
trunks-in-use	_	_	allowed
trunks-maint-busy	_	required	allowed

VDN data types

VuStats data type	Description	BCMS report	Field or column heading in the report
acceptable-service- level	The number of seconds within which agents must answer calls.	VDN Status/VDN	Acceptable Service Level
	The system identifies this data type for each VDN and begins tracking the data type when a call enters a vector.		

VuStats data type	Description	BCMS report	Field or column heading in the report
acd-calls	The split or skill calls and direct agent calls answered by an agent.	VDN Status	ACD CALLS
average-acd-talk- time	The average talk time for ACD calls during a specified period or day for a specified VDN.	VDN Status/Split	AVG TALK HOLD
average-speed-of- answer	The average speed for answering ACD and CONNect calls that were completed for a specified VDN during a specified time. The system includes the time in vector processing.	VDN Status/VDN/VDN Summary	AVG SPEED ANS
average-time-to- abandon	The average time that calls were in a queue before the callers abandoned the calls.	VDN Status/VDN	AVG ABAND TIME
calls-abandoned	The number of abandoned calls.	VDN Status/VDN/VDN Summary	ABAND CALLS
calls-flowed-out	The total number of calls for a specific VDN that successfully routed to another VDN or off the communication server.	VDN Status/VDN/VDN Summary	FLOW OUT
calls-forced-busy- or-disc	The number of calls that received a forced busy or a forced disconnect tone.	VDN Status/VDN/VDN Summary	CALLS BUSY/DISC
calls-offered	All calls offered to a VDN, including ACD calls, connected calls, abandoned calls, busy calls, disconnected calls, and outflow calls.	VDN Status/VDN/VDN Summary	CALLS OFFERED
calls-waiting	The number of calls that reach a VDN but have not been answered, abandoned, or outflowed.	VDN Status	CALLS WAIT
non-acd-calls- connected	The number of non-ACD calls routed from a specific VDN that were connected to an extension.	VDN Status/VDN/VDN Summary	CONN CALLS
oldest-calling- waiting	The time that the oldest call has been waiting in the VDN. The system begins tracking when a call enters a vector.	VDN Status	OLDEST CALL
percent-in-service- level	The percentage of calls offered to the VDN that agents answer within the service level that you administer for the VDN.	VDN Status/VDN/VDN Summary	% IN SERV LEVL

VuStats data type	Description	BCMS report	Field or column heading in the report
total-acd-talk-time	The total time agents spent talking on split or skill calls and direct agent calls.	_	_
vdn-extension	The extension of a VDN.	VDN Status/VDN	VDN EXT
vdn-name	The name of a VDN.	VDN Status/VDN Summary	VDN NAME

Required and allowed fields for VDN data types

VuStats data type	Format	Period	Threshold
acceptable-service-level	required	_	-
acd-calls	_	required	allowed
average-acd-talk-time	required	required	allowed
average-speed-of-answer	required	required	allowed
average-time-to-abandon	required	required	allowed
calls-abandoned	_	required	allowed
calls-flowed-out	_	required	allowed
calls-forced-busy-or-disc	_	required	allowed
calls-offered	_	required	allowed
calls-waiting	_	_	allowed
non-acd-calls-connected	_	required	allowed
oldest-calling-waiting	required	_	allowed
percent-in-service-level	_	required	allowed
total-acd-talk-time	required	required	allowed
vdn-extension	_	_	-
vdn-name	_	_	-

Chapter 3: Administering features

AAS

Before you begin

Ensure that Automatic Call Distribution (ACD) is active for the system.

Procedure

- 1. At the command prompt, type change hunt-group xxx, where xxx is the number of the hunt group. Press **Enter**.
- 2. In the **AAS** field, select **y** to use the hunt group as Auto-Available Split/Skill (AAS).
- 3. Press **Enter** to save the changes.
- 4. At the command prompt, type change agent-loginid xxx, where xxx is the login ID of the agent.
- 5. In the **AAS** field, select **y** to use the login ID as an AAS port.
- 6. Press **Enter** to save the changes.

Abandoned Call Search

Procedure

- 1. At the command prompt, type change trunk-group xxx, where xxx is the number of the trunk group. Press **Enter**.
- 2. In the **Abandoned Call Search** field, select **y** for each Central Office (CO), Foreign eXchange (FX), and Wide Area Telecommunications Service (WATS) trunk group.
- 3. Press **Enter** to save the changes.

Add/Remove Skills

Administer Add/Remove Skills on the following screens.

Screen title	Field title
Class of Restriction (COR)	Add/Remove Agent Skills on page 1 of the screen.
Class of Service	Console Permissions on page 1 of the screen.
	You must administer this field so that supervisors can change third-party skill.
Feature Access Code (FAC)	Add Agent Skill Access Code and Remove Agent Skill Access Code on page 6 of the screen under the Miscellaneous header.
	The fields do not apply to the 96X1 SIP agent deskphones.
Hunt Group	COR on page 1 of the screen and Skill on page 2 of the screen.
Language Translations	On page 5 of the screen under Miscellaneous features. Provide translations for the following fields:
	Add Skill: Enter number, then # sign
	Remove Skill: Enter number, then # sign
	• Enter skill level, then # sign
	• Enter Agent LoginID

Adding skills

About this task

The system plays an intercept tone if a station user:

- Does not log in to the system.
- Does not have a Class of Restriction (COR) number that permits the user to change skills.
- Has the maximum number of assigned skills and attempts to add another skill.
- Types a skill that is not a valid skill hunt group.

A supervisor can change agent skills if the supervisor has console permissions in addition to the COR number.

The system plays a reorder tone if a move is pending from Call Management System (CMS) or Communication Manager.



Note:

96X1 SIP agent deskphones do not support this feature.

Procedure

- 1. Go off-hook on an idle call appearance.
 - The system plays a dial tone.
- 2. Type the Add Skill Feature Access Code (FAC). You can administer this FAC on the Abbreviated Dial (AD) button.

The system plays the dial tone again and displays the following message on a deskphone with a display screen: Add Skill: Enter number, then the pound (#) sign.

- 3. Type the skill number with or without the leading zeros, and press the pound (#) sign on the deskphone.
- 4. Assign a skill level from 1 and 16, and press the pound (#) sign on the deskphone.

Result

If the skill assignment is valid, the system plays a confirmation tone and displays the new skill on the deskphone for five seconds.

Removing skills

About this task

The system plays an intercept tone if a station user:

- Does not log in to the system.
- Does not have a Class of Restriction (COR) number that permits the user to change skills.
- · Attempts to remove an unassigned skill.
- Attempts to remove a skill from a login ID that has only one assigned skill.

A supervisor can change agent skills if the supervisor has console permissions in addition to the COR number.

The system plays a reorder tone if:

- An agent is on an ACD call or in the After Call Work (ACW) mode for the skill.
- · A direct agent call is in a queue for the skill.
- A move is pending from Call Management System (CMS) or Communication Manager.

Procedure

1. Go off-hook on an idle call appearance.

The system plays a dial tone.

2. Type the Remove Skill Feature Access Code (FAC). You can administer this FAC on the Abbreviated Dial (AD) button.

The system plays the dial tone again and displays the following message on a deskphone with a display screen: Remove Skill: Enter skill, then the pound (#) sign.

3. Type the skill number with or without the leading zeros, and press the pound (#) sign on the deskphone.

Result

The system plays a confirmation tone if the system removes the skill. You can view the other skill assignments on the deskphone for five seconds.

Changing skills from a voice terminal

The procedure for changing skills from a voice terminal with console permission is the same as for agents changing their skills except that after entering the Add Skill or Remove Skill FAC, the voice terminal user must enter the agent login ID. The user must have the Add or Remove Skills Class of Restriction (COR) and a Console Permissions Class of Service (COS).

Agent Call Handling

Before you begin

Ensure that Automatic Call Distribution (ACD) and Expert Agent Selection (EAS) are active for the system.

Procedure

- 1. Administer the following fields on the Vector Directory Number screen:
 - VDN Timed ACW Interval
 - After Xfer or Held Call Drops
- 2. Administer the following fields on the Hunt Group screen for the skill number that is associated with the VDN:
 - Timed ACW Interval (sec)
 - After Xfer or Held Call Drops
 - Forced Entry of Stroke Counts or Call Work Codes

You can view the associated skill numbers in the following fields on the Vector Directory Number screen:

- 1st Skill
- 2nd Skill
- 3rd Skill
- 3. Administer the **Auto Answer** field on the Station screen of the associated station so that the station automatically receives calls.

If the field entry in the **Group Extension** field on page 1 of the Hunt Group screen matches the field entry in the **Hunt-to Station** field on page 1 of the Station screen, the station is associated with the hunt group.

4. Administer features buttons on the Station screen.

You can administer the **Active Station Ringing** field for Digital Communication Protocol (DCP) and hybrid phones and the **Idle/Active Ringing** field for Callmaster[™] phones.

- 5. Administer the following fields on the Feature Access Code (FAC) screen:
 - 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
- 6. Administer all the fields on the Agent LoginID screen.

Caller information

Procedure

- 1. Administer **callr-info** on the Station screen as a feature button.
- 2. For one-line display phones, select y in the Enhanced Callr-Info display for 1-line phones field on the Station screen.

Administer the Enhanced Callr-Info display for 1-line phones field to determine when Communication Manager must clear the caller information (callr-info).



■ Note:

The **Clear callr-info** field applies for one-line display phones if you:

- Select a one-line phone type in the **Type** field on the Station screen.
- On the Station screen, select y in the Enhanced Callr-Info display for 1-line display phones field.
- 3. On page 13 of the Feature-Related System Parameters screen, select one of the following options in the Clear Callr-info field:
 - leave-ACW
 - next-call: This field option is the default.
 - · on-call-release
- 4. Administer the Callr-info Display Timer (sec) field on the Feature-Related System Parameters screen for softphones, H.323 phones, and one-line display phones that have the Enhanced Callr-Info display for 1-line phones field set to n.

The Enhanced Callr-Info display for 1-line phones field does not apply to two-line display phones, 96X1 SIP phones, 96X1 H.323 phones, and softphones.

Agent/Caller Disconnect Tones

Procedure

- 1. At the command prompt, type change system-parameters features and press
- 2. In the Agent/Caller Disconnect Tones field, select y to play one of the two distinct disconnect tones that indicate whether the caller or the agent disconnected the call first.

Agent Mobility

Before you begin

- See the "Extension to Cellular" topic in Avaya Aura® Communication Manager Feature Description and Implementation guide and the Avaya Extension to Cellular User Guide.
- Ensure sufficient licensing of Off-PBX Telephones type EC500 is available.
- Administer the Expert Agent Selection (EAS) Enabled field on the Feature-Related System Parameters screen as y. For more information, see "Expert Agent Selection".
- · Add the agent ID and configure the required hunt group or VDN Contact center administration. Agent LoginID and other Call Center Elite administration is the same for both mobile and local agents.
- Configure Timed-ACW on skills that route to mobile agents. This gives agents the time to change work-modes in-between calls.

Procedure

1. Ensure that the agent's station extension is of type DCP or H.323 using a unique station extension for each mobile agent.



Note:

Avaya one-X[®] Agent does not support Agent Mobility.

- 2. Ensure that the station is EC500-enabled and specify the Service Link mode as asneeded.
- 3. (Optional) Add station buttons, ec500 and extnd-call.
- 4. (Optional) If mobile agents require to perform transfer or conference, configure at least three line appearances on the station where the agent is logged in.



Note:

When mobile agents perform any ACD operations, such as answer, hold, release, and transfer using applications that use ASAI/AES, the ACD operations are performed on the deskphone instead of the mobile phone.

If agents are going to perform "Conference On Answer", configure the no-hld-conf button on the station where the agent is logged in.

5. On the Off-PBX-Telephone Station-Mapping screen, add EC500 entry for station extension to a PSTN mobile phone.

Important:

You must use the EC500 application and can use only one EC500 application per station extension.

6. Configure Agent work mode related Feature Name Extensions (FNEs) on the EXTENSIONS TO CALL WHICH ACTIVATE CALL CENTER FEATURES BY NAME page on the feature-name-extensions screen.

Note:

You must also administer Feature Access Codes for each agent FNE except the Agent Availability Query FNE.

You can configure the following agent FNEs:

- · After Call Work Access Code
- Agent Availability Query Access Code
- Auto-In Access Code
- · Aux Work Access Code
- Login Access Code
- · Logout Access Code
- Manual-in Access Code

For more information, see "Off-PBX Feature-Name-Extensions screen".

Automatic Call Distribution

Administer Automatic Call Distribution (ACD) on the following screens:

- Abbreviated Dialing
- Agent Login ID
- · Announcement/Audio Sources
- Call Vectors
- · Class of Service
- · Date and Time
- Dial Plan Parameters
- Dial Plan Analysis Table
- Feature Access Code (FAC)

- Feature-Related System-Parameters
- Hunt Group
- Media Gateway
- Station
- System-Parameters Customer-Options
- Trunk Group
- Vector Directory Number (VDN)

Avaya IQ measurements

Before you begin

EAS and Universal Call Identification (UCID) must be active for the system.

Procedure

- 1. Add a row to the Avaya IQ and Call Management System (CMS) table to administer the following screens:
 - Feature-Related System Parameters
 - · Processor Channel Assignment
- 2. In the **Measured** field on the following screens, select **y**:
 - Hunt Group
 - Trunk Group
 - Vector Directory Number (VDN)

Basic Call Management System

Before you begin

Ensure that all agents have logged out of the system and the field option in the following fields on the System-Parameter Customer-Options screen is \mathbf{y} :

- ACD
- · BCMS (Basic)
- BCMS/VuStats Service Level
- VuStats

Procedure

- 1. In the **Measured** field on the Trunk Group screen, select **both** or **internal**.
- 2. Administer the following fields on the Call Center System Parameter pages of the Feature-Related System Parameters screen:
 - BCMS/VuStats LoginIDs
 - BCMS/VuStats Measurement Interval
 - BCMS/VuStats Abandon Call Timer (seconds)
 - · Clear VuStats Shift Data
 - Validate BCMS/VuStats Login IDs
- 3. Administer the **Measured** and **Acceptable Service Level (sec)** fields on the Vector Directory Number screen.
- 4. Administer the **Measured** and **Service Level Target (% in sec)** fields on the Hunt Group screen for the skill number that is associated with the VDN.

You can view the associated skill numbers in the following fields on the Vector Directory Number screen:

- 1st Skill
- 2nd Skill
- 3rd Skill
- 5. Administer all the fields on the BCMS/VuStats Login ID and Agent Login ID screens when you use BCMS with Expert Agent Selection (EAS).

Administering Business Advocate

Administer Business Advocate through one of the following:

- Communication Manager: To create new login IDs.
- CMS Supervisor: To administer existing agent login IDs.

Note:

Do not administer Business Advocate and Service Level Maximizer (SLM) on the same system as these two features are mutually exclusive.

The following table lists the tasks that you can perform through Communication Manager and CMS Supervisor.

Administration tasks	Communication Manager	CMS Supervisor
Activating Service Objective (SO) by agent.	Yes	Yes
Adding or deleting skills per agent.	Yes	Yes
Administering call handling preference.	Yes	Yes
Administering call selection measurement.	Yes	No
Administering Dynamic Queue Position (DQP).	Yes	No
Administering hunt group types.	Yes	No
Administering Service Level Supervisor (SLS).	Yes	No
Administering SO by skill.	Yes	No
Assigning reserve agents.	Yes	Yes
Creating agent login IDs.	Yes	No
Creating hunt groups.	Yes	No
Including After Call Work (ACW) in Least Occupied Agent (LOA) calculations.	Yes	No
Viewing or changing agent skills.	Yes	Yes

Business Advocate screen reference

The following table lists the screens and the fields for agent, skill, system, and VDN levels of administration.

Screen title	Field title
Agent-level administration	
	Call Handling Preference
	• greatest-need
	percent-allocation
Agent LoginID	• skill-level
, tgont Loginib	Direct Agent Calls First
	Percent Allocation (PA)
	Reserve Level (RL)
	Service Objective
Skill-level administration	
	Activate on Oldest Call Waiting
Hunt Group	Dynamic Percentage Adjustment
	Dynamic Queue Position
	Dynamic Threshold Adjustment

Screen title	Field title
	Expected Call Handling Time (sec)
	Group Type
	• ead-loa
	• ead-mia
	• pad
	• ucd-loa
	• ucd-mia
	Level 1 Threshold (sec)
	Level 2 Threshold (sec)
	Service Level Supervisor
	Service Level Target (%)
	Service Objective
System-level administration	
	ACW Agents Considered Idle
	AUX Agent Remains in LOA Queue
	AUX Agents Considered Idle (MIA)
Feature-Related System Parameters	Auto Reserve Agents
	Call Selection Measurement
	MIA Across Splits/Skills
	Service Level Supervisor Call Selection Override
VDN-level administration	
Vector Directory Number (VDN)	Service Objective

Best Service Routing

Singlesite BSR

Before you begin

Ensure that the following fields are administered for the system:

- Adjunct CMS Release on the Feature-Related System Parameters screen.
- G3 Version, Vectoring (G3V4 Advanced Vector Routing), and Vectoring (Best Service Routing) on the System-Parameters Customer-Options screen.
- CPN Prefix, Ext Len, and Ext Code on the ISDN Numbering Public/Unknown screen.

Ensure that the Call Vector screen is administered for each vector that uses Best Service Routing (BSR).

Procedure

- 1. At the command prompt, type add vdn xxx or change vdn xxx, where xxx is a valid Vector Directory Number (VDN) extension as defined in the system dial plan. Press **Enter**.
- 2. In the **Allow VDN Override** field, select **y** to allow the settings of the subsequent VDN to replace the settings of the current VDN.
- 3. In the BSR Available Agent Strategy field, select one of the following field options:
 - 1st-found
 - · ead-loa
 - · ead-mia
 - ucd-loa
 - · ucd-mia

When this VDN is the active VDN for a vector that uses BSR, Available Agent Strategy determines how calls are directed when more than one of the specified resources have available, that is, idle agents. If there is only one split or skill with available agents, Communication Manager delivers calls to that resource.

4. Press **Enter** to save your changes.

Multisite BSR

Administer multisite BSR on the following screens.

Screen title	Field title
System-Parameters Customer-Options	• G3 Version
	Vectoring (Best Service Routing)
	Vectoring (G3V4 Advanced Routing)
	Look-Ahead Interflow (LAI)
Feature-Related System Parameters	Adjunct CMS Release
Trunk Group (ISDN-BRI)	Outgoing Display
Settings in the fields Codeset to Send TCM,	Supplementary Service Protocol
Lookahead and Send Codeset 6/7 LAI IE on the ISDN trunk screen do not affect BSR.	• UUI Treatment
Trunk Group (ISDN-PRI)	Outgoing Display
	Supplementary Service Protocol
	UUI Treatment
Trunk Group (SIP)	UUI Treatment
Best Service Routing Application Plan	Complete one screen for each BSR application.

Screen title	Field title
Vector Directory Number	BSR Application
	BSR Available Agent Strategy
Call Vector	Complete a screen for each vector, that is, primary, status poll, and interflow vectors, in a BSR application.

Related links

<u>Look-Ahead Interflow</u> on page 154 <u>UUI Treatment for ISDN trunks</u> on page 172 <u>UUI Treatment for SIP trunks</u> on page 173

BSR Local Treatment

Before you begin

Ensure that the following fields on the System-Parameters Customer-Options screen are administered as \mathbf{y} :

- Look-Ahead Interflow (LAI)
- BSR Local Treatment for IP & ISDN
- Vectoring (Best Service Routing)

! Important:

You must administer the **BSR Local Treatment** field as **y** on both the local and remote VDNs. If you administer the field as **n** for the local VDN and **y** for the remote VDN, Communication Manager displays an ISDN_PROGRESS message with a progress indicator of in-band information. The local Communication Manager treats this type of progress message as invalid unless the local treatment flag is set and all interflow attempts result in dropped calls.

Procedure

- 1. At the command prompt, type change vdn xxx, where xxx is the number of the VDN. Press **Enter**.
- 2. In the BSR Local Treatment field, select y.
- 3. Press Enter to save the changes.

BSR Tie Strategy

Before you begin

Ensure that the following fields are administered for the system.

- **Vectoring (Prompting)** or **Vectoring (Basic)** on the System-Parameters Customer-Options screen is administered as **y**.
- Attendant Vectoring and Meet-me Conferencing on the Vector Directory Number screen are administered as **n**.

Procedure

- At the command prompt, type change system-parameters features and press Enter.
- 2. In the **BSR Tie Strategy** field on the Feature-Related System Parameters screen, select one of the following options:
 - 1st-found
 - alternate
- 3. Press Enter to save the changes.
- 4. At the command prompt, type change vdn xxx, where xxx is the number of the VDN. Press Enter.
- 5. In the **BSR Tie Strategy** field, select one of the following fields:
 - 1st-found
 - alternate
 - system
- 6. Press **Enter** to save the changes.

Administering BSR polling over IP without the B-channel

Before you begin

- You must administer multisite BSR. If the Communication Manager version is lower than R11, the CISC SETUP does not start vector processing and the poll operation ends in a time out, logging a vector event. Vector processing continues with the next step.
- You must set the following fields to y:
 - IP Trunk
 - QSIG Basic Call Setup
 - QSIG Basic Supplementary Services
- You must have the TN799 C-LAN circuit pack to support D-channel signaling.

About this task

You can configure BSR polling to prevent using a Bearer (B) channel when a polling signal is sent over an H.323 IP trunk. This feature offers the following advantages:

- Improved trunk efficiency: BSR polling uses QSIG Call Independent Signaling Connections/ Temporary Signaling Connections (CISC/TSCs) that send BSR polls over the D-channel without associated seizure of a B-channel. With this polling strategy, more trunk bandwidth is available for other forms of voice or data traffic.
- Reduced hardware requirements: If you do not require the Voice over IP (VoIP) functionality for the trunk, an IP Media Processor (MedPro) circuit pack (TN2302) is not required.

₩ Note:

BSR polling over IP without the B-channel uses non-call associated TSCs.

Procedure

- 1. Set up the signaling group for H.323, QSIG, and NCA-TSCs.
- 2. Set up the designated trunk group for ISDN and IP with a minimum of one assigned trunk.

Important:

If an IP MedPro circuit pack is not installed, the Trunk Group Status screen indicates the trunk as out-of-service. The circuit pack installation does not affect the status poll signaling over the D-channel.

The Pattern screen

Procedure

- 1. Enter the display route-pattern xx command to view the Pattern screen. Where xx is the route pattern used by the trunk group that supports TSCs.
- 2. In the TSC column, set the value to y.

Result

A sample of the Pattern screen configured for BSR polling over IP without the B-channel is as follows.

display route-pattern 32	Page	1 of 3
Pattern	Number: 32	
Grp FRL NPA Pfx Hop Toll No. I	nserted	DCS/ IXC
No Mrk Lmt List Del D	igits	QSIG
Dgts		Intw
1: 32 0 3		n user
2:		n user
3:		n user
4:		n user
5:		n user
6:		n user
BCC VALUE TSC CA-TSC ITC BCIE Service/Feature	BAND No. Numbering LAR	
0 1 2 3 4 W Request	Dgts Form	mat
·	Subaddress	
1: y y y y y n		
as-needed rest	none	
2: y y y y y n n rest		none
3: y y y y y n n rest		none
4: y y y y y n n rest		none
5: y y y y y n n rest		none
6: y y y y y n n rest		none

The BSR screen

Procedure

- 1. Enter the display best-service-routing xx command to view the Best Service Routing screen. Where xx is a BSR application plan number.
- 2. The **Status Poll VDN** field must specify an AAR or AAS pattern that routes over an IP trunk.



Do not specify a Trunk Access Code (TAC) in the **Status Poll VDN** field. If you specify a TAC, the poll routes through a B-channel, if a B-channel is available.

The Signaling Group screen

Procedure

- 1. Enter the display signaling-group xx command to view the Signaling Group screen. Where xx is the signaling group number.
- 2. Specify the TSC related fields in the upper-right corner of the field.

The following are relevant fields:

- Max number of NCA TSC
- Trunk group for NCA TSC
- 3. Set the Supplementary Services Protocol field to b.

Result

A sample of the Signaling Group screen configured for BSR polling over IP without the B-channel is as follows.

display signaling-group 32	
SIGN	ALING GROUP
Group Number: 32 Group	Type: h.323
Remote Of Max number of NCA TSC: 10	fice? n
	Max number of CA TSC: 10
	Trunk Group for NCA TSC: 32
Trunk Group for Channel Selec	tion: 32
Supplementary Service Protoc Network Call Transfer? n	ol: b
Near-end Node Name: clan-01D	12 Far-end Node Name: cland12-loop
Near-end Listen Port: 1720	Far-end Listen Port: 1720
	Far-end Network Region:
LRQ Required? n	Calls Share IP Signaling Connection?y
RRQ Required? n	
	Bypass If IP Threshold Exceeded?n
	Direct IP-IP Audio Connections?y
	IP Audio Hairpinning?y
	Interworking Message: PROGress

The Trunk Group screen

Procedure

- 1. Enter the display trunk-group xx command to view the Trunk Group screen. Where xx is the signaling group number.
- 2. Set the Group Type field to isdn.

- 3. Set the Carrier Medium field to IP.
- 4. Set the Supplementary Service Protocol field to b.



- 5. Specify a trunk group member in the NCA-TSC Trunk Member field.
- 6. You must associate the group member used to make the BSR status polls with an appropriate signaling group. Specify The signaling group in the "Sig Grp" column. The group member is also specified in the **NCA-TSC Trunk Member** field on page 2 of the Trunk Group screen.

The Feature-Related System Parameters screen (ISDN) Procedure

- 1. Enter the **change system-parameters feature** command to view the Feature-Related System Parameters screen.
- 2. Specify an unassigned extension number for the dial plan in the **QSIG TSC Extension** field.

Result

A sample of the Feature-Related System Parameters screen configured for BSR polling over IP without the B-channel is as follows.

change system-parameters features	Page 7 of 12
FEATURE-RELATED SYSTEM PARAMETERS	
ISDN PARAMETERS	
Send Non-ISDN Trunk Group Name as Connected Name? n	
Display Connected Name/Number for ISDN DCS Calls? n	
Send ISDN Trunk Group Name on Tandem Calls? n	
QSIG TSC Extension: 3999	
MWI - Number of Digits Per Voice Mail Subscriber: 5	
National CPN Prefix:	
International CPN Prefix:	
Pass Prefixed CPN to ASAI? n	
Unknown Numbers Considered Internal	for AUDIX? n
USNI Calling Name for Outgoing Calls	s? n
Path Replacement with Measurements?	У
QSIG Path Replacement Extension: 2999	9
Path Replace While in Queue/Vectoring	1; À

Call Prompting

Administer Call Prompting on the following screens.

Screen title	Field title
System-Parameters Customer-Options	Vectoring (Prompting)
	Vectoring (CINFO)
	ISDN-PRI - for CINFO only
Feature-Related System Parameters	Prompting Timeout
Vector Directory Number	Complete all fields.
Announcements/Audio Sources	Complete all fields for each extension that provides a Call Prompting announcement.
Hunt Group	Vector
Call Vector	Complete the fields for each Call Prompting vector.
Station (multi-appearance)	Button/Feature Button Assignments: callr-info
Attendant Console	Button/Feature Button Assignments: callr-info

If the Vectoring (Basic) field on the System-Parameters Customer-Options screen is administered as n. Call Prompting cannot gueue calls or make conditional checks based on the queue or agent status, time of day, or day of week.

Ensure that AT&T Intelligent Call Processing (ICP), ISDN-PRI and Vectoring (Prompting) are active for CINFO.

You can administer callr-info for any phone equipped with a display or attendant console. When an agent presses callr-info, the phone displays the digits collected for the last collect digits command.

You must administer Call Vectoring to use Call Prompting with Call Management System (CMS).

Call Vectoring

Administer Call Vectoring on the following screens.



Note:

Do not change a vector while the vector is processing calls because calls already in the vector might experience problems. Instead, add a new vector and change the VDN to point to the new vector.

Screen title	Field title
System-Parameter Customer-	Vectoring (Basic)
Options	Vectoring (G3V4 Enhanced)
	Vectoring (G3V4 Advanced Routing)
	Vectoring (ANI/II-Digits Routing)
	Vectoring (Attendant Vectoring)
	Vectoring (Holiday Vectoring)
	Vectoring (Variables)
	Vectoring (3.0 Enhanced)
Feature-Related System	Vector Disconnect Timer
Parameters	Music/Tone on Hold
	• Port
	Music (or Silence) on Transferred Trunk Calls
Vector Directory Number	Complete all fields.
Announcements/Audio	Complete all fields for each extension that provides a vectoring
Sources	announcement.
Hunt Group	• ACD
	• Vector

Screen title	Field title
Call Vector	Complete the screen for each vector.
Holiday Table	Complete all fields.
Vector Routing Table	Complete all fields.

Fields that do not allow VDN extensions

You cannot enter a VDN extension in the fields listed in the following table.

Screen title	Field title
Announcements/Audio Sources	Extension Number
Call Coverage Answer Group	Group Member Assignments
Call Coverage Paths	Coverage Point Assignments, other than the last coverage point in a coverage path
Console Parameters	Centralize Attendant Service (CAS) Back-up Extension
Feature-Related System Parameters	Automatic Circuit Assurance (ACA) Long Holding Time Originating Extension
	ACA Short Holding Time Originating Extension
	Extensions With System wide Retrieval Permission
	Controlled Outward Restriction Intercept Treatment
	Controlled Termination Restriction (Do Not Disturb)
	Controlled Station-to-Station Restriction
Hospitality	Extension of Property Management System (PMS) Log Printer
	Extension of Journal/Schedule Printer
	Extension of PMS
	Extension to Receive Failed Wakeup LWC Messages
Hunt Group and Agent	Supervisor Extension
LoginID with EAS	Member Extensions
Intercom Group	Member Extensions
Listed Directory Numbers	LDN Extensions
Loudspeaker Paging and Code Calling Access	Extension Numbers Assigned to Codes
Pickup Groups	Member Extensions
Remote Access	Remote Access Extension
Station Forms	Hunt to Station
Terminating Extension Group	Member Extensions

Fields that allow VDN extensions

Screen title	Field title
Call Coverage Paths	Allow it as the last coverage point only in coverage path
Hunt Group	Night Destination
Listed Directory Numbers	Night Destination
Trunk Groups	Night Destination
	Incoming Destination

You cannot enter a VDN extension as auxiliary data for the following buttons:

- Bridged Appearance (brdg-app)
- Data Call Setup (data-ext)

You can enter a VDN extension as auxiliary data for the following buttons:

- Remote Message Waiting Indicator (aut-msg-wt)
- Facility Busy Indication (busy-ind)
- Manual Message Waiting (man-msg-wt)
- Manual Signaling (signal)

Expert Agent Selection

Before you begin

Ensure that the following fields on the System-Parameters Customer-Options screen are administered as **y**:

- ACD
- Expert Agent Selection (EAS)
- EAS-PHD
- Vectoring (Basic)

Ensure that the **Direct Agent Calling** field on the Class of Restriction screen is administered as y.

Procedure

- 1. At the command prompt, type change system-parameters features and press Enter.
- 2. Administer the following fields on the Feature-Related System Parameters screen:
 - Direct Agent Announcement Extension/Delay
 - Expert Agent Selection (EAS) Enabled
 - Message Waiting Lamp Indicates Status For

- Minimum Agent-loginID Password Length
- Work Mode on Login
- 3. Press **Enter** to save the changes.

Result

Skill hunt groups replace splits. Help messages, error messages, and field titles change from split to skill.

Physical aspects of the phone, such as the set type and button layout, do not change as these aspects are associated with the phone and not the login ID.

EAS-related fields

Administer the following fields related to EAS.

Screen title	Field title
Agent Login ID	Administer all fields.
Call Vector	Administer the fields and edit vectors.
CDR System Parameters	Record Called Vector Directory Number Instead of Group or Member
Dial Plan Analysis Table and Dial Plan Parameters	All fields related to EAS for assignment of digits to the agent login IDs.
	Agent login IDs can be from 3 to 16 digits and must be in the dial plan, but the IDs must be different from the assigned phone extensions.
Hunt Group	• ACD
	Group Type
	• Skill
	• Vector
	If the Message Center field is administered as AUDIX , you must administer the ACD and Skill fields as y . You can administer the Vector field as y or n .
Station	EAS works with a single set of work mode button. Use this screen to remove additional sets of buttons.
Vector Directory Number	• 1st Skill
	• 2nd Skill
	• 3rd Skill
	Administering these fields is optional.

Other parameters that support EAS Agent LoginID

Enhanced Ac Group Ac Personal Ac System Ac Agent-LoginID Port extension Re Announcements Re Buttons abrdg_app Re	ccepts ccepts ccepts ccepts ccepts ccepts cepts ejects ejects
Enhanced Ac Group Ac Personal Ac System Ac Agent-LoginID Port extension Re Announcements Re Buttons abrdg_app Re	ccepts ccepts ccepts ccepts ccepts ejects ejects ejects
Group Ad Personal Ad System Ad Agent-LoginID Re Port extension Re Announcements Re Buttons Re abrdg_app Re	ccepts ccepts ccepts ejects ejects ejects
Personal Ad System Ad Agent-LoginID Re Port extension Re Announcements Re Buttons Re abrdg_app Re	ccepts ccepts ejects ejects ejects
System Ad Agent-LoginID Re Port extension Re Announcements Re Buttons Re abrdg_app Re	ejects ejects
Agent-LoginID Port extension Re Announcements Re Buttons abrdg_app Re	ejects ejects
Port extension Re Announcements Re Buttons abrdg_app Re	ejects ejects
Announcements Re Buttons abrdg_app Re	ejects ejects
Buttons abrdg_app Re	ejects
abrdg_app Re	-
	-
	1-
aut-msg-wt Ac	ccepts
brdg_app Re	ejects
busy-ind Ye	es
data_ext Re	ejects
man_msg_wt Re	ejects
q-calls Re	ejects
q-time Re	ejects
signal Re	ejects
Call processing	
Auto callback Re	ejects
Call forward from agent login ID	ejects
Call forward to agent login ID Ac	ccepts
Call park Ac	ccepts
Hundreds group Re	ejects
LWC retriever gets last msgs Ac	ccepts
Service observe agent login ID Ac	ccepts
Call Detail Recording (CDR)	
Primary extension Re	ejects
Secondary extension Re	ejects
Code calling Ac	ccepts
Communication Link	
Communication link digits Re	

Parameters	Agent loginID	
Console parameters		
CAS-backup ext	Rejects	
IAS Att access code	Rejects	
Coverage groups		
Answer group member	Rejects	
Path	Accepts	
Measured principals		
Coverage measurement	Rejects	
Feature-related system parameters		
ACA-referral dest.	Rejects	
ACA - long holding	Rejects	
ACA - short holding	Rejects	
Controlled out restriction	Rejects	
Controlled terminal	Rejects	
Controlled Stn-to-Stn	Rejects	
DAA Extension	Rejects	
DID/Tie/ISDN announcement	Rejects	
Emergency access redirection	Rejects	
CDR output extension	Rejects	
SVN referral destination (announcement)	Accepts	
System LWC retriever	Rejects	
System printer	Rejects	
Hospitality		
Journal printer	Rejects	
LWC wakeup	Rejects	
PMS ext	Rejects	
PMS log	Rejects	
Routing on voice synthesis	Rejects	
Hunt Group		
Announcement extension	Rejects	
ASAI link	Rejects	
AUDIX extension	Rejects	
Calls warning extension	Rejects	
Member	Rejects	
Night service	Rejects	

Table continues...

Parameters	Agent loginID	
Supervisor	Accepts	
Time warning extension	Rejects	
Intercom group member	Rejects	
Intraswitch CDR	Accepts	
Listed Directory Number (LDN)		
Member	Rejects	
Night destination	Accepts	
Malicious Call Trace (MCT)		
MCT member	Rejects	
Permanent switched calls	Rejects	
Personal CO line	Rejects	
Pickup group member	Rejects	
Remote access extension	Rejects	
Term Extension Group (TEG) member	Rejects	
Trunk group		
Night service	Accepts	
Incoming destination	Accepts	
Member night service	Accepts	
Vector administration		
Adjunct extension	Rejects	
Announcement	Rejects	
Messaging	Accepts	
Route-to	Accepts	

Direct Agent Calling

Procedure

- 1. In the **Direct Agent Skill** field on the Agent LoginID screen, assign a skill number for handling direct agent calls.
- 2. Use the Hunt Group screen to administer a skill for all direct agent calls.

Setting a skill:

- The system communicates how to handle calls to the skill to Communication Manager.
- The system indicates to the report users how much time each agent has spent on DA calls.

Note:

For an agent who receives direct agent calls, ensure that you assign a minimum of one non-reserve skill to the agent login ID.

3. Add the skill to the list of skills administered for an agent on the Hunt Group screen.

When an outside caller dials the agent extension, Communication Manager determines the skill for tracking call data by viewing the entry in the field.

- 4. On page 8 of the Feature-Related System Parameters screen, specify:
 - A Direct Agent Announcement (DAA) extension that plays an announcement to direct agent callers waiting in queue.
 - The period of delay, in seconds, before the announcement.
- 5. Administer a Class of Restriction (COR) for direct agent calls.
- 6. Use the Trunk Group screen to administer Direct Inward Dialing (DID).
- 7. On page 2 of the Hunt Group screen, administer Multiple Call Handling (MCH) On-Request for this hunt group.
 - With this feature, agents can see if the incoming call is a direct agent call and put the current call on hold to answer the direct agent call.
- 8. If there is no answer after an administered number of rings, use Redirection on No Answer (RONA) to redirect the caller to a VDN that points to a vector.
 - You can set up the vector to provide appropriate routing and treatment for the redirected call.
- 9. On page 3 of the Hunt Group screen, administer messaging for the direct agent hunt group.
- 10. Assign this hunt group to agents who must answer direct agent calls.

Direct Agent Announcement

Before you begin

Ensure that the following fields on the System-Parameters Customer-Options screen are administered for the system:

- ACD
- Expert Agent Selection (EAS) or ASAI Adjunct Routing
- Vectoring (Basic)

- 1. At the command prompt, type change system-parameters features and press **Enter**.
- 2. In the **Direct Agent Announcement Extension** field on the Feature-Related System Parameters screen, type the extension of the direct agent announcement.

- 3. In the **Delay** field, assign a delay period from 0 to 99 seconds to determine how long must a caller hear ringback before listening to a direct agent announcement. You can leave this field blank for no delay.
- 4. Press **Enter** to save the changes.
- 5. On the command prompt, type change announcement xxx, where xxx is an extension number. Press Enter.

■ Note:

You refer to the Avaya Aura® Media Server-based announcement source by the letter M followed by its media-server number. For example, you must enter M5 when you are entering media-server 5 into the announcement form or the audio-group form.

- 6. Administer all the fields on the Announcements/Audio Sources screen.
- 7. Press **Enter** to save the changes.

Display VDN for Route-To DAC

Before you begin

To use the Display VDN for Route-to DAC feature for incoming trunk calls, ensure that the following fields are administered as y:

- The EAS field on the System-Parameters Customer-Options and the Features-Related System Parameters screens.
- The Direct Agent Calling field on the Class of Restriction screen of the Class of Restriction (COR) number that is associated with the VDN and the Expert Agent Selection (EAS) login ID to which the VDN routes the direct agent call.

About this task

The active VDN name display treatment applies to the initial EAS agent who receives the vectorinitiated direct agent call. The treatment also applies to an EAS agent in the coverage path of the initial EAS agent who receives the call.

- 1. At the command prompt, type change vdn xxx, where xxx is the number of the VDN. Press **Enter**.
- 2. On page 2 of the Vector Directory Number screen, select y in the Display VDN for Route-To DAC field.
- 3. Press **Enter** to save the changes.

Forced Agent Logout by Clock Time

Before you begin

Ensure that the following fields are administered for the system:

- The EAS field on the System-Parameters Customer-Options screen is administered as y.
- The **AAS** field on the Agent LoginID screen is administered as **n**.
- The Call Center Release field on the System Parameter Customer-Options screen is 4.0 or later.

Procedure

- 1. At the command prompt, type change system-parameters features and press Enter.
- On page 14 of the Feature-Related System Parameters screen, administer the Clock Time Forced Logout Reason Code field to specify the logout reason.
- 3. Press **Enter** to save the changes.
- 4. At the command prompt, type change agent-loginid xxx, where xxx is the login ID of an agent. Press Enter.
- 5. Administer the **Forced Agent Logout Time** field on the Agent Login ID screen to automatically log agents out of the system based on a timer.
- 6. Press **Enter** to save the changes.
- 7. At the command prompt, type change station xxx, where xxx is the extension number. Press **Enter**.
- 8. Administer a **logout-ovr** button on the Station screen to override forced logout.
- 9. Press Enter to save the changes.

Forced Agent Logout/Aux Work by Location/Skill

- 1. On the Feature Access Code screen, administer the following fields:
 - 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
- 2. On the Feature-Related System Parameters screen, administer the following fields:
 - Forced Agent Logout by Location Reason Code

- Forced Agent Logout by Skill Reason Code
- Forced Agent Aux Work by Location Reason Code
- Forced Agent Aux Work by Skill Reason Code

When an agent is forcibly logged out of the system or moved to Auxiliary (AUX) work, Communication Manager sends the reason code with the logout event messages to the reporting adjuncts and the ASAI-connected adjunct.

- 3. On the Class of Restriction screen, administer the Class of Restriction (COR) for the relevant station and agent:
 - a. In the **Can Force a Work State Change** field on the Class of Restriction screen relevant to the station to be used to force the change, select **y**.
 - b. In the **Work State Change Can Be Forced** field on Class of Restriction screen relevant to the agent or VDN/trunk for which to force the change, select **y**.

The Forced Logout/Aux Work by Location/Skill feature is applicable only to the server on which you administer the feature. If the system has multiple servers, you must administer the feature on each server.

The two fields are applicable only with this feature and not with other similar features, such as Forced Logout by Clock Time and Forced Logout from ACW.

Forced Agent Logout from ACW

Before you begin

Ensure that the following fields are administered as \mathbf{y} in the System-Parameters Customer-Options screen:

- Expert Agent Selection (EAS)
- Reason Codes

If **Reason Codes** is administered as **n**, you can administer the maximum time that an agent can be in the After Call Work (ACW) mode on a systemwide basis or for each agent, but you cannot administer a reason code for agent logout.

- 1. At the command prompt, type change system-parameters features and press Enter.
- 2. On page 15 of the Feature-Related System Parameters screen, administer the **Maximum Time Agent in ACW before Logout (sec)** field to assign a systemwide timer.
- 3. Press **Enter** to save the changes.
- 4. At the command prompt, type change agent-loginid xxx, where xxx is the login ID of the agent.

- 5. Administer the **Maximum time agent in ACW before logout (sec)** field to assign a timer for each agent.
- 6. Press **Enter** to save the changes.

Forced Agent Logout from ACW

You can use Forced Agent Logout from ACW only if all of the following conditions are true:

- **EAS** is set to y.
- **Reason Code** is set to y. If the Reason Code feature is not active, you can still set the maximum time that an agent can be in ACW on a system-wide basis or for each agent, but you cannot administer a reason for the logout.

Administer Forced Agent Logout from ACW on the following screens.

Screen name	Field name
Page 13 of Feature-Related System Parameters	Maximum time agent in ACW before logout (sec) : Set the maximum time that an agent can be in the ACW mode.
	This is a system-wide setting.
Agent Login ID	Maximum time agent in ACW before logout (sec): Set the maximum time that an agent can be in the ACW mode. Set the reason code that describes why Communication Manager logged the agent out.
	This setting is for each agent.

Tips for administering Forced Agent Logout from ACW

You can use the following suggestions when administering Forced Agent Logout from ACW.

Requirement	Tip
To force agent logout from ACW and use the same time out period for all agents.	 Administer the Maximum Time Agent in ACW before Logout (sec) field on the Feature-Related System Parameters screen to a time out value in seconds. The field value can be from 30 to 9999 seconds. Leave the default system setting on the Agent Login ID screen.
To maintain different time out periods assigned to specific agents.	Administer the Maximum time agent in ACW before logout (sec) field on the Agent Login ID screen to a time out value for each agent.
To use the time out feature only to certain agents.	Administer the Maximum time agent in ACW before logout (sec) field on the Agent Login ID screen to none for those agents.

Inbound Call Management

Before you begin

Ensure that the following fields on the System-Parameters Customer-Options screen are administered as \mathbf{y} :

- ACD
- ASAI Link Core Capabilities
- ASAI Link Plus Capabilities
- Vectoring (Basic) or Vectoring (Prompting)

Procedure

- 1. At the command prompt, type change trunk-group xxx, where xxx is the number of the trunk group. Press **Enter**.
- 2. Administer the Per Call CPN/BN field for the relevant ISDN-PRI trunk group.
- 3. Press **Enter** to save the changes.
- 4. At the command prompt, type change hunt-group xxx, where xxx is the number of the hunt group. Press Enter.
- 5. Administer the fields on the **Hunt Group** screen for each split or skill that the Inbound Call Management (ICM) adjunct monitors.
- 6. Press Enter to save the changes.
- 7. At the command prompt, type change vector xxx, where xxx is the vector number.
- 8. Add an Adjunct Switch Application Interface (ASAI) link extension number for adjunct routing vector commands. The number must match the extension number on the Station screen.
- 9. Press Enter to save the changes.

Interruptible Aux

Administer Interruptible Aux on the following screens.

Screen title	Field title
Agent LoginID	Reserve Level (RL)
Feature-Related System	Interruptible Aux Deactivation Threshold (%)
Parameters	Interruptible Aux Notification Display
	Interruptible Aux Notification Timer (sec)

Table continues...

Screen title	Field title
Hunt Group	Interruptible Aux Threshold and Interruptible Aux Deactivation Threshold
Reason Code Names	Interruptible

Intraflow and Interflow

Administer Intraflow and Interflow on the following screens.

Screen title	Field title	
Coverage Path	Coverage criteria	
	• Busy	
	Don't Answer	
	Number of Rings	
Feature Access Code (FAC)	Call Forwarding Activation All	
	Call Forwarding Busy/DA	
	Call Forwarding Deactivation	
Feature-Related System Parameters	Coverage - Don't Answer Interval for Subsequent Redirection	
Hunt Group	Calls Warning Threshold	
	Inflow Threshold (sec)	
	Priority on Intraflow	
	Service Level Target (% in sec)	
	Time Warning Threshold	

Location Preference Distribution

Before you begin

Ensure that the following fields on the System-Parameters Customer-Options screen are administered as \mathbf{y} :

- Expert Agent Selection (EAS)
- Multiple Locations

Procedure

1. At the command prompt, type change hunt-group xxx, where xxx is the number of the hunt group. Press **Enter**.

- 2. Administer the Local Agent Preference field for agent surplus conditions.
- 3. Press **Enter** to save the changes.
- 4. At the command prompt, type change agent-loginid xxx, where xxx is the login ID of the agent.
- 5. Administer the **Local Call Preference** field for call surplus conditions.
- 6. Press **Enter** to save the changes.

Look-Ahead Interflow

Administer Look-Ahead Interflow (LAI) on the following screens.

Screen title	Field title
System-Parameters Customer-	• ISDN-PRI
Options	Lookahead Interflow (LAI)
	Vectoring (Basic)
Trunk Group (ISDN)	Codeset to Traveling Class Mark, Look Ahead
	Outgoing Display
	Supplementary Service Protocol
	• UUI IE Treatment
Feature-Related System Parameters	Interflow-qpos EWT Threshold
ISDN Public-Unknown-	CPN Prefix
Numbering	• Ext Code
	• Ext Len
Call Vector	Complete a screen for each LAI vector.

- If Lookahead Interflow (LAI) is administered as n on the receiving communication server, interflow still results on a look-ahead basis. However, the forwarded DNIS, sending communication server VDN name, and information is ignored and tandem LAI is not provided.
- Administer the **Outgoing Display** field to **n** to not update the call originator display on each LAI call attempt.
- Administer the Interflow-qpos EWT Threshold field when working with enhanced LAI.
 Any calls answered before this threshold do not interflow, saving CPU resources on the communication server.
- Administer a CPN prefix for each VDN that maps to a vector used to place LAI calls. If you do
 not administer the CPN Prefix field, the phone displays an LAI DNIS with all blanks.

For private network non-QSIG connectivity with direct facilities between the communication servers, administer LAI DS1/E1 circuit packs with **Country Protocol Option 1** independent of the country where the system is located.

Maximum Agent Occupancy

Procedure

- 1. At the command prompt, type change system-parameters features and press Enter.
- 2. Administer the following fields on the Feature-Related System Parameters screen:
 - Maximum Agent Occupancy AUX Reason Code
 - Maximum Agent Occupancy Percentage
- 3. Press **Enter** to save the changes.

Multinational CPN Prefix

Before you begin

Examine the field settings on the following screens:

- Call Vector
- Numbering Public/Unknown Format
- Route Pattern
- Tandem Calling Party Number Conversion
- Trunk Group
- Uniform Dial Plan Table
- Vector Routing Table

- 1. At the command prompt, type change system-parameters features and press Enter.
- 2. Administer the following fields on page 8 of the Feature-Related System Parameters screen.
 - International CPN Prefix
 - National CPN Prefix
 - Pass Prefixed CPN: VDN/Vector

- 3. Press **Enter** to save the changes.
- 4. At the command prompt, type change vdn xxx, where xxx is the number of the VDN. Press Enter.
- 5. Administer the **Pass Prefixed CPN to VDN/Vector** on the Vector Directory Number screen.
- 6. Press **Enter** to save the changes.

Multiple Call Handling

Before you begin

Ensure that the field option in the **Multiple Call Handling (On Request)** and **Multiple Call Handling (Forced)** fields on the System-Parameters Customer-Options screen is **y**.

Procedure

- 1. At the command prompt, type change hunt-group xxx, where xxx is the number of the hunt group. Press **Enter**.
- 2. Administer the **ACD** and **Multiple Call Handling** fields on the Hunt Group screen.
- 3. Press **Enter** to save the changes.

Network Call Redirection

NCR for ISDN trunks



The administration requirements described in this section do not apply if you start Network Call Redirection (NCR) with the AT&T In-Band Transfer Connect[™] service.

To administer NCR, ensure that the field option in the **ISDN/SIP Network Call Redirection** field on the System-Parameters Customer-Options screen is **y**.

Administer the following fields to use NCR for Integrated Services Digital Network (ISDN) trunks.

Administration command	Field title	Field option
Best Service Routing Application screen		
change best-service- routing xxx	Net Redir	У
DS1 Circuit Pack		

Table continues...

Administration command	Field title	Field option
change ds1 [board location]	Country Protocol	1a or any field option for MCI Network Call Transfer (NCT).
Board location parameter values are: [cabinet(1-1)];carrier(A-		• 1b/1d for Two Bearer Channel Transfer (TBCT)
E);slot(0-20) OR [gateway(1-10)];module(V1-V9)		etsi for European Telecommunications Standards Institute (ETSI) Explicit Call Transfer (ECT) and Network Call Deflection (NCD)
Signaling Group screen		
change signaling group xxx	Group Type	ISDN for ETSI ECT, MCI NCT, NCD, and TBCT
	Network Call Transfer	• y for ETSI ECT, MCI NCT, and TBCT
		• n for NCD
Trunk Group screen		
change trunk-group xxx	Direction	two-way
	Disconnect Supervision - In and Out	У
	Group Type	ISDN for ETSI ECT, MCI NCT, NCD, and TBCT
	Network Call Redirection	У
	Service Type	cbc
	Supplementary Service Protocol	Settings specific to the PSTN redirection options
		• a (national) for TBCT
		• c (ETSI) for ETSI ECT
		• c (ETSI) for NCD
		• g (ANSI) for MCI NCT
	Usage Alloc	у
	This field option is applicable if you administer the Service Type field as a Call-by-Call service, that is, the cbc field option.	

NCR for SIP trunks

To administer NCR, ensure that the field option in the ISDN/SIP Network Call Redirection field on the System-Parameters Customer-Options screen is y.

Administer the following fields to use Network Call Redirection (NCR) for Session Initiation Protocol (SIP) trunks.

Administration command	Field title	Field option
Best Service Routing Application so	creen	
change best-service- routing xxx	Net Redir	У
Trunk Group screen		
change trunk-group xxx	Direction	two-way
	Group Type	SIP
	Network Call Redirection	у

NCR with the AT&T In-Band Transfer Connect service

The following table includes the administration requirements and administration of Dual-Tone Multi-Frequency (DTMF) announcement and Best Service Routing (BSR) vectoring methods that are associated with use of the AT&T In-Band Transfer Connect[™] service.

To administer NCR, ensure that the field option in the **ISDN/SIP Network Call Redirection** field on the System-Parameters Customer-Options screen is **y**.

Administer the following fields to use Network Call Redirection (NCR) with the AT&T In-Band Transfer ConnectSM service.

Administration command	Field title	Field option
Best Service Routing Application se	creen	
change best-service- routing xxx	Net Redir You must administer this field if	n
	BSR is active for the system.	
DS1 Circuit Pack		
change ds1 [board location]	Country Protocol	1b/1d
Board location parameter values are: [cabinet(1-1)]; carrier(A-E); slot(0-20) OR [gateway(1-10)]; module(V1-V9)		
Signaling Group screen		
change signaling group xxx	Network Call Transfer	У
Trunk Group screen		
change trunk-group xxx	Group Type	ISDN
	Network Call Redirection	none
	Supplementary Service Protocol	a (national)
	UU IE Treatment	shared

Administering DTMF announcements for AT&T In-Band Transfer Connect

You can use the following methods to create an announcement that provides the Dual-Tone Multi-Frequency (DTMF) digits required to start the AT&T In-Band Transfer Connect[™] service operations:

- Use a Communication Manager analog DTMF station to activate the recording session for the announcement. When the session begins, use the keypad to enter the touchtone digits that correspond to the *T + PSTN endpoint number that is used to invoke the AT&T In-Band Transfer Connect service. For example, you must perform the following steps to redirect an incoming ISDN call to an number, such as 3035552104:
 - 1. Add a silence period of 100-200 milliseconds (ms).
 - 2. Type *83035552104.
 - 3. Add a silence period of 200-500 ms.

Important:

For out-of-band DTMF methods that are used to relay the DTMF digits in IP-connected center stage and IP-trunked environments, you must record periods of silence before and after you enter the DTMF digits. The silence period accommodates the connection time, that is, the time for tone detection by MEDPRO and VOIP resources, and prevents the announcement from ending before the resources complete tone regeneration. To increase the DTMF transmission accuracy, record the DTMF digits with at least 80 ms of digit duration with an inter-digit silence of at least 80 ms.

You cannot use a digital phone, such as Callmaster[®], BRI, ISDN, or IP, to record the announcement as the station keypads on these sets do not generate audible DTMF tones during an announcement recording session. However, if you choose to record DTMF digits with these phones, you must make arrangements to connect an external keypad.

 Use a computer with VAL boards with an internal or external keypad or a commercially available computer software tool. Include a silence period for recording with an analog phone.

For more information about how to use DTMF announcements in vectors, see *Avaya Aura*[®] *Call Center Elite Feature Reference*.

Administering station or ASAI transfer, conference, or release

Before you begin

Ensure that the field option in the **ISDN/SIP Network Call Redirection** field on the System-Parameters Customer-Options screen is **y**.

- 1. For MCI Network Call Transfer (NCT) or Two Bearer Channel Transfer (TBCT) protocols, use the trunk group with a Bearer (B) channel that has the same Delta (D) channel as the incoming ISDN call to create a second call leg of the call transfer.
- 2. For the ETSI Explicit Call Transfer (ECT) protocol, the second leg of the call can be over a different trunk group with a different signaling group than the incoming call leg.

- 3. You must also perform the following tasks to administer station or ASAI transfer:
 - a. On the ARS Digit Analysis screen, add the PSTN number that a station or ASAI user dials to transfer an incoming call to another PSTN endpoint.
 - b. Administer the entry with an Alternate Route Selection (ARS) routing pattern that routes the second call leg to the same trunk group being used for the incoming call.
 - c. When you use the MCI NCT feature for the Route Pattern screen associated with the ARS Digit Analysis screen entry, administer the **Service/Feature (sdn)** and **Number Format (lev0-pvt)** fields to be consistent with the service-type and dialingplan configuration of the PSTN trunk.
 - d. For the other NCT-type NCR protocols, no administration is required for the Route Pattern screen associated with the ARS Digit Analysis screen entry. NCR call processing automatically causes the **Service/Feature** and **Number Format** for the NCR second leg call to be **unknown/unknown**.
 - e. Contact your PSTN service provider to verify the configuration of the PSTN switch used for the NCT operation. Configure the PSTN switch to accept the outgoing digits used by the station or the ASAI application to set up the second leg of the call transfer or conference.

Percentage Allocation Routing

Procedure

- 1. At the command prompt, type change vdn xxx, where xxx is the number of the VDN. Press Enter.
- 2. Administer the **Destination** and **Number** fields on the Vector Directory Number screen.
- 3. Press **Enter** to save the changes.
- 4. At the command prompt, type change policy-routing-table xxx, where xxx is the number of the policy routing table. Press **Enter**.
- 5. Administer all the fields on the Policy Routing Table screen.
- 6. Press **Enter** to save the changes.

Queue Status Indications

Administer Queue Status Indications on the following screens.

Screen title	Field title
Attendant Console	Feature Button Assignments

Table continues...

Screen title	Field title	
Hunt Group	Calls Warning Threshold and Port	
	Queue Limit	
	Time Warning Threshold and Port	
Station (multi-appearance)	Feature buttons	
	• q-calls	
	• q-time	

Reason Codes

You can administer Reason Codes on the following screens if the field option in the **ACD**, **Expert Agent Selection (EAS)**, and **Reason Codes** fields on the System-Parameters Customer-Options screen is **y**.

Screen title	Field title
Attendant Console	RC
Agent LoginID	Aux Work Reason Code Type
	Logout Reason Code Type

Table continues...

Screen title	Field title
Feature-Related System Parameters	Expert Agent Selection (EAS) Enabled
	Aux Work Reason Code Type
	Logout Reason Code Type
	Two-Digit Aux Work Reason Codes
	Redirection on No Answer Aux Work Reason Code
	Redirection on OPTIM Failure and Unreachable Aux Work Reason Code
	IP Failure Aux Work Reason Code
	Maximum Agent Occupancy Aux Work Reason Code
	ACW Forced Logout Reason Code
	Clock Time Forced Logout Reason Code
	Forced Agent Logout by Location Reason Code
	Forced Agent Logout by Skill Reason Code
	Forced Agent Aux Work by Location Reason Code
	Forced Agent Aux Work by Skill Reason Code
	Forced Agent Logout for Unreachable Reason Code
Language Translations	English: ENTER REASON CODE
Reason Code Names	Aux Work
	• Logout
	Interruptible
Station	RC for the aux-work feature button

Redirection on IP Failure

Administer Redirection on IP Failure (ROIF) on the following screens.

Screen title	Field title
Feature-Related System Parameters	IP Failure AUX Reason Code
	Switch Hook Query Response Timeout
	Caution:
	If you administer a low value for the switch hook query time out, the system takes agents out of service even when agents are available to receive calls.
Hunt Group	Redirect on IP/OPTIM Failure to VDN
	Retain Active VDN Context

Setting the switch hook query time out value

When you set the switch hook query time out value, the degree of network congestion or delay determines the speed of the switch hook query response.

The range value for the switch hook timer is from 500 to 5000 milliseconds (ms). IP hard phones located geographically close to Communication Manager respond to the timer within the 500-750 ms range. However, IP Agent or one-X® Agent endpoints on a personal computer take longer to respond, so you must set the timer to more than 2000 ms. In all cases, set the time out long enough to prevent false triggering. A longer time out period does not delay the delivery of calls when connectivity is intact as the response to the query returns quickly. The caller hears a portion of ringback while waiting for a response to the switch hook query with an auto-answer delivery.

Redirection on No Answer

Administer Redirection on No Answer (RONA) on the following screens.

Screen title	Field title
Feature-Related System Parameters	Redirection on No Answer Aux Work Reason Code

Table continues...

Screen title	Field title	
Hunt Group	• ACD	
	· AAS	
	Controlling Adjunct	
	Message Center	
	• Redirect On No Answer (rings)	
	Redirect On No Answer to VDN	
	Retain Active VDN Context	
	• Vector	
	Note:	
	The field option in the Controlling Adjunct field must be none .	
Station	Feature buttons	
	• noans-airt	

Redirection on OPTIM Failure

Administer Redirection on Off-PBX Telephony Integration and Mobility (OPTIM) Failure (ROOF) on the following screens.

Screen title	Field title	
Feature-Related System Parameters	Redirection on OPTIM Failure and Unreachable Aux Work Reason Code	
Hunt Group	Redirect on IP/OPTIM Failure to VDN	
	Retain Active VDN Context	
Trunk Group	Redirect On OPTIM Failure	
	☆ Note:	
	The range for this field is from 250 to 32000 milliseconds (ms). The default value for this field is 5000 ms. You must increase the ROOF field value if the system places agents in the Auxiliary (AUX) work mode when agents are actually available to receive calls.	

Remote Logout of Agent

Before you begin

Ensure that the field option in the following fields is **y** for users of this feature:

- Remote Logout of Agent on the Class of Restriction screen.
- Console Permissions on the Class of Service screen.

You must also ensure that feature user and the agent are in the same tenant partition.

Procedure

- 1. At the command prompt, type change feature-access-codes. Press Enter.
- 2. In the **Remote Logout of Agent Access Code** field on the Feature Access Code (FAC) screen, assign an FAC of up to four digits that the user must type to activate this feature. You can use the asterisk (*) or pound (#) sign as the first digit.
- 3. Press **Enter** to save the changes.

Reporting adjuncts on Communication Manager

In Call Center, there are two fields dedicated to each of the reporting adjuncts, Avaya Call Management System and Avaya IQ. You can set each field to CMS or Avaya IQ or both. High availability configurations are also supported.

Measured trunks versus unmeasured facilities

Unmeasured facilities are not actual trunks, but are tracking records on the Avaya Call Management System (CMS).

Avaya IQ does not require data allocation for the unmeasured facilities. The number of measured trunks is based on the communication server capacity that Avaya IQ is monitoring. Avaya IQ does not require data storage allocation for unmeasured facilities.

Adding reporting adjunct nodes

- 1. At the command prompt, type change node-names ip. Press Enter.
- 2. In the **Name** field on the IP Node Names screen, type node names.
- 3. In the **IP Address** field, type the IP address of each reporting adjunct.
 - For Avaya IQ, type the host name and IP address for the Data Collection Host that monitors the communication server as a *ccr* adjunct.
- 4. Press **Enter** to save the changes.

Communication Manager to reporting adjunct interface

For information about administering the interface between Communication Manager and Avaya IQ, see *Administering Avaya IQ*.

For information about administering the interface between Communication Manager and Call Management System (CMS), see *Avaya Call Management System Switch Connections, Administration, and Troubleshooting.*

Administering reporting adjuncts with Communication Manager

Before you begin

On the following screens, set the **Measured** field to both or external:

- Hunt Group
- Trunk Group
- Vector Directory Number

Procedure

- 1. At the command prompt, type change system-parameters features and press Enter.
- 2. On page 12 of the FEATURE-RELATED SYSTEM PARAMETERS screen, set the reporting adjunct release in the following fields:
 - · CMS (appl mis)
 - AAPC/IQ (appl ccr)

The releases must be compatible with the Switch Protocol Interpreter (SPI) language. For example:

- If you set the CMS (appl mis) field to R16.1/R16.x/R17.0, set the IQ (appl ccr) field to 5.1/5.2 because both releases support the SPI language 23.
- If you set the CMS (appl mis) field to R18 or R18.1, set the IQ (appl ccr) field to 5.2.6+ because both releases support the SPI language 24.
- 3. Save the changes.

Next steps

Administer the fields on the Processor Channel Assignment screen.

Processor Channel Assignment

- At the command prompt, type change communication-interface processorchannels and press Enter.
- 2. In **Enable** field on the Processor Channel Assignment screen, select **y**.

- 3. In the Appl field, select ccr for Avaya IQ or mis for CMS.
- 4. In the Mode field, select s.
- 5. In the **Interface Link** field, assign the data link that is administered to communicate with the adjunct.
- 6. In the **Interface Chan** field, assign an unused value in the range allotted to the link type being used for this channel.
- 7. In the **Destination Node** field, select the node name that you assigned on the IP Node Names screen.
- 8. In the **Destination Port** field, type 0.
- 9. In the **Session Local/Remote** field, type 1.
- 10. Press **Enter** to save the changes.

Proactive Contact predictive calls

Before you begin

Ensure that the field option in the **Measured** field on the Vector Directory Number screen is **both** or **external**.

Procedure

- 1. At the command prompt, type change vdn xxx, where xxx is the number of the VDN. Press Enter.
- 2. In the **Reporting for PC or POM Calls** field on the Vector Directory Number screen, select **y**.
- 3. In the associated **PC Predictive Reports Skill** field, assign a skill number from 1 to 8000 or leave the field blank to indicate use by Proactive Outreach Manager (POM).
- 4. Press **Enter** to save the changes.

Proactive Contact non-predictive calls

- At the command prompt, type change system-parameters features and press Enter.
- 2. In the **Reporting for PC Non-Predictive Calls** field on the Feature-Related System Parameters screen, select **v**.

- 3. In the associated **PC Non-Predictive Reports Skill** field, assign a skill number from 1 to 8000.
- 4. Press **Enter** to save the changes.

Service Hours Table

Procedure

- 1. At the command prompt, type change service-hours-table xxx, where xxx is number of a service hours table. Press **Enter**.
- 2. Administer the following fields on the Service Hours Table screen:
 - Description
 - Number
 - Start and End
 - Use time adjustments from location
- 3. Press **Enter** to save the changes.

Service Level Maximizer

Before you begin

Ensure that the following fields are administered on the System-Parameters Customer-Options screen:

- Business Advocate to n.
- Expert Agent Selection (EAS) to n.
- Service Level Maximizer to y.

- 1. At the command prompt, type change hunt-group xxx, where xxx is the number of the hunt group. Press **Enter**.
- 2. In the **Group Type** field on the Hunt Group screen, select **slm**.
- 3. In the Service Level Target (% in sec) field, administer the target level.
- 4. In the SLM Count Abandoned Calls field, select y.
- 5. In the **SLM Max Auto Reserve Agents** field, administer the maximum number of agents for the hunt group.
- 6. Press **Enter** to save the changes.

Administering the ASL algorithm

Before you begin

Ensure that the field option in the **Service Level Maximizer** field on the System Parameters Customer-Options screen is **y**.

Procedure

- 1. Administer the **Service Level Maximizer Algorithm** field on the Feature-Related System Parameters screen as **actual**.
- 2. Administer the **SLM Count Abandoned Calls** field on the Hunt Group screen to determine whether to include abandoned calls in the ASL algorithm calculations for SLM.
- 3. Administer the time interval in the **Service Level Interval** field on the Hunt Group screen.

Service Observing

Administer Service Observing on the following screens.

Screen title	Field title	
Class of Restriction	Can Be Service Observed: For agents	
	Can Be A Service Observer: For supervisors	
	Service Observing by Recording Device: For recording devices	
	Service Observing Permissions: For granting permission	
Feature Access Code (FAC)	Service Observing by Location Listen Only Access Code	
	Service Observing by Location Listen/Talk Access Code	
	Service Observing Listen Only Access Code	
	Service Observing Listen/Talk Access Code	
	Service Observing Next Call Listen Only Access Code	
	Service Observing No Talk Access Code	
	Note:	
	SIP phones do not support Service Observing using FAC.	
Station	Button Assignment	
	serv-obsrv - Use this button type for non-SIP devices.	
	sip-sobsrv - Use this button type for SIP devices.	

Table continues...

Screen title	Field title
Feature-Related System	Expert Agent Selection (EAS) Enabled
Parameters	Service Observing: Warning Tone
	Allow Two Observers in Same Call
	Allow with Exclusion: Service Observing
	Allow with Exclusion: SSC
System-Parameters Customer- Options	Service Observing (Basic): For basic or Logical Agent ID observing
	Service Observing (Remote/By FAC): For remote observing or observing by FAC
	Service Observing (VDNs): For VDN observing
	Vectoring (Prompting): For vector-initiated observing
Vector Directory Number	Observe on Agent Answer

Service Observing with Multiple Observers

Before you begin

Ensure that the field options in the following fields is y:

- Service Observing (Basic) on the System Parameter Customer-Options screen.
- Can Be A Service Observer or Can Be Service Observed on the Class of Restriction screen for supervisors or agents.
- Service Observing by Recording Device on the Class of Restriction screen to use a recording device.

Procedure

- 1. At the command prompt, type change system-parameters features and press **Enter**.
- 2. In the **Allow Two Observers in Same Call** field on the Features-Related System Parameters screen, select **v**.
- 3. Press **Enter** to save the changes.

Universal Call ID

Before you begin

• Ensure that the field option in the **ASAI Link Core Capabilities** and **ASAI Link Plus Capabilities** fields on page 3 of the System-Parameters Customer-Options screen is **y**. If

the field option is **n**, the system displays the following error message: ASAI Interface feature not assigned.

• Busy out the *mis* links if you have Call Management System (CMS) systems in the network. Release the *mis* links after you administer all the UCID-related fields on the Feature-Related Customer-Options screen.

Procedure

- 1. At the command prompt, type change system-parameters features and press Enter.
- 2. In the **Create Universal Call ID (UCID)** field on the Feature-Related System Parameters screen, select **y**.
- 3. In the **UCID Network Node ID** field, assign a unique number from 1 to 32767 for each communication server and Interactive Voice Response (IVR) system.
- 4. In the **Send UCID to ASAI** field, select **y** if the network includes ASAI adjuncts.
- 5. In the CMS (appl mis) field, select the release of the CMS system that must track UCIDs.
- 6. Press Enter to save the changes.

Next steps

You must administer the Trunk Group screen to relay UCIDs over ISDN or SIP trunks.

Sending UCIDs over ISDN or SIP trunks

Before you begin

To send UCIDs over ISDN trunks, ensure that the field option in the **ISDN-PRI** or **ISDN-BRI Trunks** field on the System-Parameters Customer-Options screen is **y**.

- 1. At the command prompt, type change trunk-group xxx, where xxx is the number of the trunk group. Press **Enter**.
- 2. In the **Group Type** field on the Trunk Group screen, select **ISDN** or **SIP**.
- 3. In the **Supplementary Service Protocol** field, select **b** for QSIG or other field options for User-to-User (UUI). This field is applicable with ISDN trunks.
- 4. In the **UUI IE Treatment** field for ISDN trunks or the **UUI Treatment** field for SIP trunks, select **shared**.
- 5. In the Send UCID field, select y.
- 6. Press **Enter** to save the changes.

User-to-User Information

UUI Treatment for ISDN trunks

About this task

You must select shared in the UUI IE Treatment field on the Trunk Group screen for outgoing and incoming trunks at the remote end if you use:

- Shared User-to-User Information (UUI) Information Element (IE).
- Supplementary service protocol other than **b**.

If you select service-provider in the UUI IE Treatment field, Communication Manager forwards the ASAI user data in a non-shared codeset 0 UUI IE when forwarding the other data as MSI.

In non-QSIG networks, use protocol options other than **b** to relay information. In QSIG networks and supplementary service protocol of type b, you must use shared UUI IE treatment to include the Adjunct Switch Application Interface (ASAI) user information with MSI transport.

Note:

For non-QSIG networks, if you use shared UUI IE and select y in the Send Codeset 6/7 LAI IE field on page 3 of the Trunk Group screen, you can send the Lookahead Information (LAI) data twice, unless you leave the priorities of the LAI Name and Other LAI data items blank. For QSIG networks, Communication Manager sends LAI data that exceeds the maximum ISDN message size.

For more information, see Avaya Aura® Communication Manager Screen Reference.

Procedure

- 1. At the command prompt, type change trunk-group xxx, where xxx is the number of the trunk group. Press Enter.
- 2. In the **UUI IE Treatment** field on page 3 of the Trunk Group screen, select **shared** if the trunk group is not connected to an early version of Communication Manager or you do not require the service provider functionality.



Caution:

For service provider functionality, you must select service-provider in the UUI IE Treatment field.

- 3. (Optional) In the Maximum Size of UUI Contents field, type a number from 32 to 128 as the maximum User-to-User Information (UUI) size. The default field option is 128 bytes.
 - You must administer the trunk groups to send the appropriate UUI size over connected networks. For example, if the public network supports only 32 bytes of UUI and you administer a number greater than 32, the network rejects the UUI.
- 4. On the Shared UUI Feature Priorities page, which is applicable when you select **shared** in the **UUI Treatment** field, assign numbers from 1 to 6, where 1 is the highest priority. You can use the default field options.

If you leave a field blank, the system does not send any information related to the field in UUI Information Element (IE). If the public network supports less than 128 bytes, you must determine which feature information to send in UUI IE and assign a high priority to the related field.

5. Press **Enter** to save the changes.

UUI Treatment for SIP trunks

Procedure

- 1. At the command prompt, type change trunk-group xxx, where xxx is the number of the trunk group. Press **Enter**.
- 2. In the **UUI Treatment** field on page 3 of the Trunk Group screen, select **shared**.
- 3. (Optional) In the **Maximum Size of UUI Contents** field, type a number from 32 to 128 as the maximum User-to-User Information (UUI) size. The default field option is 128 bytes.
 - You must administer the trunk groups to send the appropriate UUI size over connected networks. For example, if the public network supports only 32 bytes of UUI and you administer a number greater than 32, the network rejects the UUI.
- 4. On the Shared UUI Feature Priorities page, which is applicable when you select **shared** in the **UUI Treatment** field, assign numbers from 1 to 6, where 1 is the highest priority. You can use the default field options.
 - If you leave a field blank, the system does not send any information related to the field in UUI Information Element (IE). If the public network supports less than 128 bytes, you must determine which feature information to send in UUI IE and assign a high priority to the related field.
- 5. Press **Enter** to save the changes.

UUI-info feature button

Before you begin

Ensure that the **Station-Button Display of UUI IE Data** field on page 2 of the Class of Restriction screen is administered as **y**.

About this task

Agents can press **uui-info** to view call-related information, such as account numbers of customers. The phones can display up to 32 characters of Adjunct Switch Application Interface (ASAI) user data or data inserted by an ASAI or Computer Telephony Integration (CTI) application.

Note:

The UUI button does not display the ASAI UUI transported over QSIG trunks.

Procedure

1. At the command prompt, type change station xxx, where xxx is the number of the station. Press **Enter**.

- 2. On the Station screen, administer the **uui-info** feature button.
- 3. Press **Enter** to save the changes.

VDN in a Coverage Path

Administer all the fields on the following screens to use VDN in a Coverage Path (VICP):

- · Call Vector
- Coverage Path
- Vector Directory Number

For each coverage path, you can assign up to six coverage points from Point 1 to Point 6.

Note:

Calls in a vector cover twice if you select **y** in the **Cvg Enabled for VDN Route-to Party** field on the Coverage Path screen.

VDN of Origin Announcement

Administer VDN of Origin Announcement (VOA) on the following screens.

Screen title	Field title
Announcements/Audio Sources	Administer all fields, but do not administer analog and integrated repeating announcement types as VOAs.
Attendant Console	Feature Button Assignments
Class of Restriction	Hear VDN of Origin Annc.
Feature-Related System Parameters	Hear Zip Tone Following VOA
Vector Directory Number	VDN of Origin Annc. Extension

VDN Return Destination

- 1. At the command prompt, type change vdn xxx, where xxx is the number of the VDN. Press Enter.
- 2. In the **Return Destination** field on the Vector Directory Number screen, assign a VDN extension to which Communication Manager must route an incoming trunk call if the call returns to vector processing after an agent release the call.

- 3. In the associated **Call Origin** field, select one of the following field options to determine the type of calls that Communication Manager must redirect to the VDN extension that you assign in the **Return Destination** field:
 - both
 - external
 - internal
- 4. Press **Enter** to save the changes.

VDN Time-Zone Offset

Procedure

- 1. At the command prompt, type change vdn xxx, where xxx is the number of the VDN. Press **Enter**.
- 2. In the **VDN Time-Zone Offset** field on page 3 of the Vector Directory Number screen, assign the time zone offset to change the system local time that is used for the Time-of-Day (TOD) conditionals in the vectors processed for the VDN to the VDN local time.
- 3. Press Enter to save the changes.

Voice Response Integration

Administer the following fields to allow passing of data between Voice Response Units (VRUs) and Communication Manager.

Screen title	Field title
Call Vector	Administer all fields.
Feature Access Code (FAC)	Converse Data Return Code
Feature-Related System Parameters	Converse First Data Delay and Second Data Delay
	Converse Signaling Tone (msec) and Pause (msec)
System-Parameters Customer-Options	Call Prompting

VuStats

Administer VuStats on the following screens.

Screen title	Field title
Attendant Console	Feature Button Assignments
BCMS/VuStats Login ID	Login ID
	• Name
Feature-Related System Parameters	ACD Login Identification Length
	BCMS/VuStats Abandoned Call Timer (seconds)
	BCMS/VuStats Measurement Interval
	Clear VuStats Shift Data
	Validate BCMS/VuStats LoginIDs
Hunt Group	• ACD
	Acceptable Service Level
	Measured
	Objective
Station	Feature Buttons
System-Parameters Customer-Options	• ACD
	BCMS/VuStats Login ID
	BCMS/VuStats Service Level
	VuStats or VuStats (G3V4 Enhanced)
Trunk Group	Measured
Vector Directory Number	Acceptable Service Level (sec)
	Measured
VuStats Display Format	Administer all fields.

Vu-display feature button

Administer the vu-display feature button and the following fields on the Station screen.

Field title	Field description
Fmt	To associate a vu-display feature button with a VuStats display format. Valid entries are from 1 to 50, where 1 is the default field option.
ID number	To assign an identification number for each vu-display button. The ID can be an agent login ID or extension number, a split or skill or trunk group number, or a VDN extension. For example, you can administer a vu-display button with split or skill ID 6 to view statistics for split or skill 6.
	Do not administer IDs for VuStats displays with the agent object type. Agent object type displays are limited to statistics for logged-in agents.
	With ID numbers, supervisors and agents do not have to type the agent extension, split or skill, or VDN number to view statistics. You can use ID numbers to limit access of specific statistics to designated phones.

Zip Tone Burst for Callmaster® Endpoints

Procedure

- 1. At the command prompt, type change system-parameters features and press Enter
- 2. In the **Zip Tone Burst for Callmaster Endpoints** field on the Feature-Related System Parameters screen, select **single** to eliminate the second burst of zip tone or Incoming Call Identification (ICI) tone.

The single field option reduces:

- The time for an agent to begin a conversation with the caller.
- The possibility of an agent and a caller hearing an open mike background noise between the first tone and second tone.
- 3. Press Enter to save the changes.

Chapter 4: Administering the 96X1 SIP agent deskphones

Before you begin

To select the **Station Type** field value as a 96X1SIPCC type phone:

- Ensure that the Call Center release is 6.0 or later.
- Administer the Expert Agent Selection (EAS) Enabled field on the Feature-Related System Parameters screen as v.

About this task

Administer the Logged-In SIP EAS Agent field on the System-Parameters Customer-Options screen to restrict the maximum number of logged-in SIP Expert Agent Selection (EAS) agents. The license file usually determines the maximum field value. To view the limit set in the license file, use the display capacity command.

The 96X1 SIP deskphones use the Avaya Outboard Proxy SIP (OPS) features on the trunk side of Communication Manager.



You can assign only one **ops** type on the Off-PBX-Telephone Station-Mapping screen.

For more information, see Using Avaya 96X1 SIP Agent Deskphones with Avaya Aura® Call Center Elite.

Procedure

- 1. On the Station screen, administer the 96X1 SIP agent deskphones as one of the following 96XISIPCC types:
 - 9608SIPCC
 - 9611SIPCC
 - 9621SIPCC
 - 9641SIPCC

Note:

After you administer the **Station Type** field to one of the 96X1SIPCC type deskphone, you cannot administer the release button as the 96X1 SIP agent deskphones provide the Release and Drop functionalities during a call and conference respectively.

2. In the **Measured** field on the Trunk Group screen, select **none** to ensure that the trunk groups for SIP signaling to SIP stations are unmeasured. Select none also to prevent Call Management System (CMS) or Avaya IQ reporting issues, including loss of reporting. Use dedicated trunk groups for SIP signaling to stations that are not shared with call routing SIP trunk groups. For more information, see Dedicating station and call routing SIP trunk groups on page 179.

- 3. Assign the following feature buttons:
 - after-call. This button is optional.
 - agnt-login. This button toggles to agnt-logout after an agent logs in to the system. You must administer the login/logout button because the login/logout Feature Access Codes (FACs) do not work with the 96X1 SIP agent deskphones.
 - assist
 - · aut-msg-wt
 - auto-in or manual-in
 - · aux-work
 - · logout-ovr
 - · q-calls
 - sip-sobsrv and the associated Listen-Only? and Coach fields.
 - Note:

If you administer the **Listen-Only?** field as **y**, Communication Manager denies change to the Listen/Talk observing mode.

- stroke-cnt
- · uui-info
- vu-display
- · work-code

Agent Greetings is unavailable with the 96X1 SIP agent deskphones. The feature is supported only with phones that use the Avaya Deskphone H.323 application. Call Center features, such as login and logout, function differently in the 96X1 SIP agent deskphones as these features use the capabilities of the Avaya SIP architecture.

Dedicating station and call routing SIP trunk groups

SIP agent deskphones must have dedicated SIP trunk groups for:

- 1. Add SIP trunks on the Private Numbering and the Public Unknown Numbering screens.
- 2. Do not add extra digits to station extensions and agent extensions on the Private Numbering or the Public Unknown Numbering screens. The extra digits might cause

- Session Manager to send calls to Communication Manager for further processing without terminating the calls to the 96X1 SIP agent deskphones. If private unknown numbering modifies agent extensions, an agent can log out and log in to a station but cannot use a feature button to change the work state.
- 3. Ensure that Session Manager has two entities for Communication Manager: One for inbound traffic and the other for agent deskphone traffic. All SIP trunks must be on the Processor Ethernet (PE) interface.
- 4. Ensure that the signaling groups for the deskphone SIP OPS signaling trunk groups have a listen port number that is different from the listen port number of the inbound SIP trunking signal groups and that the trunk groups are dedicated for each function.
- 5. The Session Manager Profile configuration for the SIP agent deskphones must point to the Communication Manager entity that has been defined for the OPS trunk (station) signaling in the Origination Application Sequence field and the Termination Application Sequence field. Otherwise, the appropriate station signaling trunks are not used to deliver calls to the SIP agent deskphones. This is important with an existing configuration that uses the same Communication Manager entity for all traffic and now you want to separate it into dedicated inbound trunking and OPS signaling trunking.
- 6. Administer Sequenced Applications for users based on whether the user is on the originating or on the terminating side of a call. When designing sequences for applications that act on the deskphones that are controlled by Communication Manager Evolution Server, Communication Manager must be the last application defined on the origination side of a call and the first application defined on the termination side of the call. An Evolution Server must be the last application in the origination sequence and the first application in the termination sequence because the Evolution Server uses the full call model for call processing and all origination and termination feature processing occurs on the origination side of the call. For more information, see *Implementing End-to-End SIP*.
- 7. For dedicated trunk groups that are used for agent deskphone signaling with Session Manager, you must administer the **Measured** field on the Trunk Group screen as **none** to prevent Call Management System (CMS) or Avaya IQ reporting issues, including loss of reporting.

Administering the Signaling Group screen

- 1. In the Group Type field, select SIP.
- 2. In the **IMS Enabled** field, select **n**.
- 3. In the **Transport Method** field, select **tls**.
- 4. Administer Near-end Listen Port and Far-end Listen Port. You must administer separate SIP entities on Session Manager that map to different signaling groups in Communication Manager to configure dedicated trunking for inbound trunks and for OPS station signaling trunks. Session Manager must have two entities for Communication Manager: One for inbound traffic and the other for agent phone traffic. For example, use the following settings for the Transport Layer Security (TLS) protocol.

Communication Manager side settings	Session Manager side settings	
Station signaling	SIP entities	
Signaling-group 1: Set Near-end Listen Port to 5062 and Far-end Listen Port to 5062	CM1OPS, CMTRNK1, and SM1. Both the CM1xx entities must point to the same Communication Manager Fully Qualified Domain Name (FQDN)	
Inbound trunk signaling	Entity links	
Signaling-group 2: Set Near-end Listen Port to 5061 and Far-end Listen Port to 5061	Entity Link 1: SM1 port 5062 to CM1OPS port 5062	
	Entity Link 2: SM1 port 5061 to CM1TRNK port 5061	

For TCP, use port 5060 instead of 5061.



Note:

With the sample configuration, signaling-group 1 and entity link 1 are logically paired-up to handle SIP OPS station signaling, and signaling-group 2 and entity link 2 are logically paired-up to handle inbound call signaling. However, you must administer appropriate routing in Session Manager, as well as in Communication Manager, to ensure that the appropriate traffic is routed over the appropriate signaling facilities. You can add additional signaling groups on Communication Manager by using the same entity links defined in Session Manager, as indicated in the table, as more OPS signaling trunks and more inbound signaling trunks might be required in Communication Manager to handle the required call traffic load and the required number of SIP agent deskphones.

- 5. In the **DTMF over IP** field, select **rtp-payload**.
- 6. In the Session Establishment Timer field, enter 3, which is the recommended time period for call center use.
- 7. In the Far-end Network Region field, assign a number from 1 to 250 that represents the region of Session Manager.
- 8. In the **Far-end Domain** field, assign the IP address of the SIP domain.
- 9. In the Initial IP-IP Direct Media field, select n. This field option is necessary to prevent unexpected interactions with the Call Center Elite features as some features depend on the media that Communication Manager handles during the first few seconds.
- 10. In the Direct IP-IP Audio Connections field, select y.

Administering the Trunk Group screen

The following procedure and settings refer to the trunk groups that are used only for SIP signaling to the SIP agent deskphones.

Procedure

1. In the **Group Type** field, select **SIP**.

- 2. In the **Signaling Group** field, assign the number of the signaling group defined on the Signaling Group screen.
- 3. In the **Number of Members** field, assign the number of 96X1 SIP phones that the trunk must support. Communication Manager automatically assigns the IP port numbers.
- 4. In the Redirect on OPTIM Failure (ROOF) field, enter 5000 (5 seconds).
- 5. In the **Preferred Minimum Session Refresh Interval (sec)** field, assign a value from 90 to 64800. The default setting is 600. To prevent spikes in the processor occupancy, assign a value that is greater than the sum of the average call handling time and the average call queuing time. For example, if the average call queuing time is 3 minutes and the average call handling time is 10 minutes, assign a field value that is greater than 780 seconds.

*

Note:

Although Communication Manager uses the configured value for *outbound* calls when the traffic is relatively low, during peak traffic conditions Communication Manager increases the session refresh interval value dynamically for protection against the processor occupancy spikes.

If the service provider network for SIP uses a session refresh interval value that is higher than the configured value for **Preferred Minimum Session Refresh Interval** but relatively low compared to the amount of SIP traffic that Communication Manager currently handles, Communication Manager rejects the calls with a *4xx* response and includes the suggested session refresh interval value for call attempt. Therefore, large call centers must ensure that the service provider network uses a session refresh interval value that is greater than 1800 seconds within the INVITE message to ensure that Communication Manager accepts calls in the first attempt, regardless of the SIP call traffic load that Communication Manager might be handling.

- 6. In the **Measured** field, select **none** for the signaling trunks to Session Manager.
- 7. In the **Numbering Format** field, select **private**, which is the default for SIP.
- 8. In the Show ANSWERED BY on Display field, select y.
- 9. In the **Support Request History** field on the Protocol Variations page, select **y**. If the field setting is **n**, the SIP agent deskphone displays do not function correctly.

Chapter 5: Administering C and D tones for a VRU port

Administering C and D tones for a VRU port

About this task

Connect (C) and Disconnect (D) Tones provides DTMF (touch tone) to a VRU/IVR port when the incoming caller is on soft hold to indicate the following transfer operation events:

- The VRU-placed call is being connected to the transferred-to agent.
- The caller disconnected while on hold.

Procedure

- 1. On the System-Parameters Customer-Options screen, set the **DTMF Feedback Signals** For VRU field to y.
- 2. On page 6 of the Feature-Related System Parameters screen, select values for the **Connection** and **Disconnect** fields. Values, such as 0-9, asterisk (*), pound (#), and A-D, represent the buttons that are included in the 16-button DTMF array.

The default connection and disconnection tones are C and D respectively. The tones are applied for 350 millisecond with a 100 millisecond pause period.

- 3. On page 1 of the Station screen of the VRU port, set the type to one of the following:
 - VRU (analog line)
 - VRUFD (Line Side DS1-FD)
 - VRUSA (Line Side DS1-SA)

IVR or VRU ports

When using Interactive Voice Response (IVRs) systems or Voice Response Units (VRUs) as station ports in a hunt group, you must administer the station ports on the Station screen with the type required by the IVR or VRU ports.

The following table lists the types for VRU or IVR ports supported by Communication Manager.

VRU or IVR port type	Forward disconnect	C&D tones support	Station type	Description	
Analog T&R	_	No	2500	Standard station set interface. Uses an analog line circuit pack, such as TN2135.	
Analog T&R	_	Yes	VRU	Provides standard station set interface with C&D tones support using the DTMF Feedback Signals feature.	
Lineside DS1/DS0 or lineside T1/E1	No	No	ds1–ops	OPS is a DS1 type that provides a TIA/EIA off-premise station type DS1 interface used where the device does not require or support forward	
DS1 circuit packs (TN767E or later or TN464F or later) must be equipped.				disconnect.	
Lineside DS1/DS0 or lineside T1/E1	Yes	No	ds1–fd	ds1–fd provides a TIA/EIA Foreign eXchange (FX) type DS1 interface. The forward disconnect signal is a toggle of the A bit from 0 to 1 and then back to 0 after 600 milliseconds. This type is used for line side T1/E1 ports on the IVR system when used as an analog-like VRU device and is the recommended method for interfacing.	
Lineside DS1/DS0 or lineside T1/E1	Yes	Yes	VRUFD	VRUFD is the same as ds1–fd but with C&D tone support. An IVR system does not use this type of administration.	
Lineside DS1/DS0 or lineside T1/E1	Yes	No	ds1–sa	ds1–sa provides a TIA/EIA special- access type DS1 interface. The forward disconnect signal is a toggle of the A bit from 1 to 0 and then back to 1 after 600 milliseconds.	
Lineside DS1/DS0 or lineside T1/E1	Yes	Yes	VRUSA	VRUSA is the same as ds–1sa but with C&D tone support.	

Station field descriptions

Field title	Field description	
AUDIX Name	To type the name of the voice messaging system that is associated with the station.	
Auto Answer	To determine whether Communication Manager must send all ACD and non-ACD calls to the station.	
	Valid entries are:	
	acd: To allow Communication Manager to send only ACD and direct agent calls to the station.	
	all: To allow Communication Manager to send all ACD and non-ACD calls to the station.	
	icom: To allow the station user to respond to an intercom call.	
	none: To provide an audible ringing treatment to all calls that Communication Manager delivers to the station.	
BCC	To determine compatibility when non-ISDN facilities are connected to ISDN facilities. If you assign 0 to this field, the Bearer Capability Class (BCC) indicates voice and voice-grade data.	
	The field is applicable if the field option in the ISDN-BRI Trunks and ISDN-PRI fields on the System-Parameters Customer-Options screen is y.	
Button Assignments	To assign feature buttons on the station.	
Button Modules	To determine the number of button modules for the station.	
COR	To assign a Class of Restriction (COR) number to the station.	
cos	To assign a Class of Service (COS) number to the station.	
Coverage Path 1	To assign a coverage path number or Time of Day (TOD) table to the station so that Communication Manager can route calls to the coverage point if no station user answers calls to the station.	
Coverage Path 2	To assign a second coverage path or TOD table if the first coverage point is unavailable.	

Field title	Field description	
Coverage Msg Retrieval	To determine whether station users at the coverage point can retrieve Leave Word Calling (LWC) messages for the station from where Communication Manager sent the call to the coverage point. Valid entries are y and n .	
	This field is applicable if you administer the LWC Reception field.	
Enable Reachability for Station Domain Control	To enable Communication Manager to poll domain- controlled SIP stations and send the station reachability information to CTI applications that require to track the status of the station, on an individual station basis.	
	Valid entries are s , y , or n . The default option is s .	
	• y : Enables polling on the station.	
	• n : Does not enable polling on the station.	
	s: The system setting determines the polling for this station.	
Extension	To assign an extension number for the station.	
Hunt-to Station	The extension number that Communication Manager must search when the station is busy.	
Idle/Active Ringing (Callmaster)	To determine how calls must ring at a Callmaster [™] station.	
	Valid entries are:	
	continuous: For calls to the station to ring continuously.	
	if-busy-single: For calls to the station to ring one time if the station user is busy.	
	silent-if-busy: For calls to ring silently if the station user is busy.	
	single: For calls to ring one time.	
Location	To assign a number that identifies where the station is located in the system.	
	This field is applicable if:	
	The field option in the Multiple Locations field on the System-Parameters Customer-Options screen is y .	
	The Type field on the Station screen is an H.323 or SIP phone type.	

Field title	Field description	
Lock messages	To determine whether other station users can gain access to voice messages.	
LWC Activation	To determine whether to use LWC.	
LWC Reception	To determine the storage point for LWC messages.	
	Valid entries are:	
	• audix: To store LWC messages in an AUDIX [™] system.	
	none: To not store LWC messages.	
	msa: To store LWC messages on Messaging Server Adjunct (MSA).	
	spe. To store LWC messages in the system or on Switch Processor Element (SPE). The default is spe.	
Message Lamp Ext	To assign the extension number of the station that is associated with the message waiting lamp.	
Name	To assign a name to the station.	
Per Station CPN - Send Calling Number	To determine whether Communication Manager must send Calling Party Number (CPN) information	
	Valid entries are:	
	• n: To not send CPN information.	
	• r: To restrict the level of CPN information.	
	• y: To send CPN information.	
	You can leave this field blank.	
Port	To assign an auxiliary or analog port to the station.	
Service Link mode	Valid entries are as-needed and permanent . Default value is as-needed . For EC500–enabled stations, this field set this field as as-needed .	
SIP Trunk	To select a trunk that corresponds to the field entry on the Off-PBX-Telephone Station-Mapping screen.	
	Valid entries are aar , ars , or a SIP trunk value from 1 to 2000.	
Time of Day Lock Table	To assign a TOD Lock/Unlock table. Valid entries are from 1 to 5.	
	You can leave this field blank.	
TN	To assign a tenant partition number. Valid entries are from 1 to 250.	

Field title	Field description
Type of 3PCC Enabled	To determine whether an Avaya Third Party Call Control (3PCC) or a Computer Telephony Integration (CTI) adjunct can control the station.
	Valid entries are Avaya and none .

Chapter 6: Recorded announcements

Screen title	Field title	
Announcements/Audio Sources (includes Integrated Announcement Translations)	Administer all the fields.	
Feature Access Code (FAC)	Announcement Access Code	
Station	cos	
Data Modules (for Save/Restore/Copy)	Administer all the fields.	
Netcon Data Module		
System Port Data Module (SAP)		
Announcement Data Module		
Circuit Pack	Administer the fields on this screen if you:	
	Administer the Board Location field on the Announcements/Audio Sources or Data Modules screen.	
	Do not have the circuit pack plugged in.	
Feature-Related System Parameters	Administer the following fields on this screen to use recorded announcements with the screen's associated features. For example, to use announcements with the Hospitality features, you must complete the Hospitality screen.	
	Controlled Outward Restriction Intercept Treatment	
	Controlled Station-to-Station Restriction	
	Controlled Termination Restriction (Do Not Disturb)	
	DID/Tie/ISDN Intercept Treatment	
Hospitality	Announcement Type	
	Length of Time to Remain Connected to Announcement	
Trunk Groups (All)	Incoming Destination	
Coverage Path	Coverage Points	

Screen title	Field title	
Hunt Group	First Announcement Extension	
	Second Announcement Extension	
Call Vector	Administer all the fields that require announcements.	

For more information, see Avaya Aura® Communication Manager Feature Description and Implementation.

Recorded announcement types

Analog line types

You can use one of the analog line types to interface with the external announcement machines for recorded announcements. You can then connect the external announcement machine by an analog line port.

Analog

The analog announcement type provides an analog phone interface using an analog line port for use with an announcement or audio source device that emulates analog phones. The communication server starts playback by applying ringing. The device indicates that the playback has stopped by going on-hook. The communication server does not indicate to the device to stop playback.

Use the analog type for announcements that play for a specific period and then go on-hook at the end. When the device goes on-hook to indicate that the playback ended, the caller listening to the announcement hears the sound of a click.

Analog-fd

Like the analog type, analog-fd provides an analog line interface and starts the playback with a ringing sound. However, when there are no callers, the system sends a forward disconnect signal, that is, an open loop for about one-half second, to the device to stop playback.

Analog-m

Analog-m provides an analog line interface. However, playback does not start with a ringing sound. Use this announcement type for continuous playing music or audio sources. The device stays in an off-hook state when active and goes on-hook when turned off or disconnected. You can use the announcement type when you set the **Q** field to b to provide barge-in repeating or continuous-play announcements.

DS1 types

The DS1 announcement types provide analog-like interfaces with DS1 line ports, which are called Line Side DS1 or Line Side T1. Each of these types indicate to the announcement, music, or audio-source device to start playback using the Line Side T1 equivalent of a ringing sound. The DS1 types also stay off-hook from the device to indicate that the playback is active and on-hook to indicate that the playback is inactive.

The ds1-id and ds1-sa types provide a forward disconnect using transitions of the A signaling bit to the device, which indicates when to stop playback. Callers listening to announcements do not hear clicks when the device disconnects or goes on-hook.

ds1-fd

The ds1-fd announcement type provides a TIA/EIA Foreign eXchange (FX) type DS1 interface. The forward disconnect signal is a toggle of the A bit from 0 to 1 and then back to 0 after 600 milliseconds. IVR systems use this announcement type for Line Side T1 ports when the systems are used as an analog-like announcement device.

ds1-sa

The ds1-sa announcement type provides a TIA/EIA special-access type DS1 interface. The forward disconnect signal is a toggle of the A bit from 1 to 0 and then back to 1 after 600 msecs.

ds1-ops

The ds1-ops announcement type provides a TIA/EIA off-premises-station type DS1 interface that is used when the device does not support forward disconnect.

Auxiliary trunk types

The Auxiliary Trunk announcement type supports an external announcement machine connected using a 4-wire auxiliary trunk interface, such as a 15A announcement system. The communication server indicates to the device to start or stop the playback on the S lead. The device indicates that the playback is active on the S1 lead.

aux-trunk

Use the Auxiliary Trunk (aux-trunk) announcement type with a 4-wire interface external device when you want to stop and start the playback using the S1 lead. The external device uses the S1 lead to indicate the start of the playback.

aux-trk-m

Use the auxiliary trunk music (aux-trk-m) with a 4-wire interface device for continuous music or audio sources that do not indicate that playback is active on the S1 lead. This announcement type is used when the $\bf Q$ field is set to $\bf b$ to provide barge-in repeating or continuous-play announcements.

Integrated types

The Integrated announcement type stores announcements internally on an Integrated Announcement circuit pack or embedded gateway processor equivalent. Integrated announcement or VAL announcement sources can include the following:

- TN2501AP Voice Announcements with LAN (VAL)
- H.248 Media Gateway VAL source
- Avaya Aura[®] Media Server hosted announcement

The following announcement boards or sources are obsolete and not used in the Communication Manager configurations:

- TN750, TN750B, or TN750C announcement boards
- Co-resident SSP sources (DEFINITY One or S8100)

integrated

You can use this announcement type for general, ACD, and vectoring announcements and for VDN of Origin Announcements.

integ-rep

The integrated-repeating (integ-rep) announcement type provides integrated and repeated automatic wake-up announcements which is implemented along with the multi-integ hospitality announcement type. Call center applications in vectoring require the continuous repeating announcement type.

integ-mus

The integ-mus announcement is similar to the integ-rep announcement except that the Q field is always set to b to provide a continuous, repeating barge-in operation. Use the integ-mus announcement type to provide music on delay or music on hold.



Note:

Use integ-rep and integ-mus, which are repeating announcements, only in the wait-time vector steps. The use of integ-mus or integ-rep in an announcement vector step or a collect digits step can halt processing of the subsequent vector steps.

When to use recorded announcements

The most common applications for recorded announcement include when:

- Direct Inward Dialed (DID) calls cannot be completed as dialed.
- Incoming private-network access calls cannot be completed as dialed.
- Calls enter a split or skill (first announcement).

- Direct Department Calling, Uniform Call Distribution, or Direct-Agent calls have been in queue for an assigned interval.
- ACD and Call Vectoring calls have been in queue for an assigned interval.
- A call destination is a recorded-announcement extension.
- A call routes to a vector that contains an announcement step.
- An announcement extension is specified as a coverage point.
- An announcement is the incoming destination of a trunk group.
- A VDN of Origin Announcement (VOA) occurs.
- · A Security violation notification occurs.
- The Hospitality Automatic Wakeup feature is in use.

Barge-in

The system connects multiple callers to the beginning of an announcement regardless of the announcement type. However, you can administer auxiliary trunk announcements, DS1 announcements, and integrated announcement so callers can listen to an announcement when the system plays the message. This capability is called barge-in.

Barge-in operational details

When you administer barge-in by setting the **Q** field to b, only one port plays the announcement at any one time. When the system routes a call to that announcement, the call immediately connects to the port and the caller hears the announcement playing. Most administrators administer barge-in announcements to repeat continually while callers are connected to the port. In this way, the caller listens until the system plays the entire announcement.

Nonbarge-in operational details

If an announcement port is available when a call arrives, the system connects the call to the announcement. If an announcement port is not available and the announcement is administered with *no* as the queue option, the call does not enter the queue for the announcement and the caller hears a busy tone or another feedback.

If an announcement port is not available and the announcement is administered with yes as the queue option, the call enters the announcement queue. When a port becomes available, the communication server connects the calls waiting in the queue to the beginning of the announcement. The system first connects the call that has been waiting in queue the longest and then connects as many calls as possible.

Integrated and externally recorded announcements

With recorded announcements, you can administer integrated announcements and those recorded on external devices. The external devices connect to the communication server using analog line circuit packs or auxiliary trunk interfaces, such as TN2183 or TN763.

The system stores an integrated announcement on a VAL announcement source. The system can store multiple announcements on each circuit pack depending on the system capacity. The integrated announcement files can be transferred to the announcement sources using the System Manager announcement manager interface.

Announcements are wave files that are recorded as CCITT u-law/a-law, 8kHz, 8-bit mono files using Microsoft Sound Recorder on a computer or using an Avaya switch phone.

Announcements that are stored on a circuit pack can play through any port on the circuit pack. Announcements that are not administered for barge-in can be played through multiple ports.

Storage type	Storage time	Playback ports
TN2501AP	60 minutes	31
G700 MG	20 minutes	15
G350 MG	10 minutes	6
G450 MG	45 minutes with internal flash memory or 240 minutes with external compact flash card	63
G430 MG	45 minutes with internal flash memory or 240 minutes with external compact flash card	15
Avaya Aura [®] Media Server	240 minutes	Limited only by the number of media channels

On the Announcements/Audio Sources screen, set the $\bf Q$ field to y to queue each extension for Integrated Announcements.

Calls hearing integrated announcements at extensions that have a queue are assigned a queue only when all the ports on the source that contains the announcement are busy. When a port is available, all callers queued to hear a specific announcement up to the maximum supported by the server platform are simultaneously connected to that port to hear the announcement from the beginning. The same queuing pool is used over all integrated sources.

The communication server controls the announcement queue length for integrated announcements, but you must set the queue length for analog or aux-trunk announcements.

How to record announcements

You can transfer announcement files to and from a computer or delete over the LAN for the VAL announcement sources using System Manager or an FTP client in conjunction with System Administration Terminal (SAT) commands. To transfer Avaya Aura[®] Media Server-based announcement files, use the System Manager announcement manager interface.

Announcements for the Voice Announcement over Local Area Network (VAL) announcement sources can also be recorded with a phone using the procedures discussed in this section.

Announcement recording

With VAL announcement sources, recording by phone always uses port 1, which is dedicated for phone access with sources. VAL announcement sources also support recording announcements as .wav files on a local computer or made by a professional recording studio. Move the files to the VAL source using File Transfer Protocol (FTP). To move Avaya Aura® Media Server-based announcement files, use the System Manager announcement manager interface.

You cannot use a phone to record an announcement with an audio group assignment. For VAL announcements use FTP or System Manager and for Avaya Aura® Media Server-based announcements use System Manager to move each prerecorded file to each of the sources defined for the audio group.

For more information, see Administering Avaya Aura® Communication Manager.

Announcement sessions

You can record, play back, or delete integrated announcements by initiating an announcement session. You must have console permissions assigned to your Class of Service (COS) for the internal station or remote access barrier code to initiate an announcement session.

You can start an announcement session by dialing the administered FAC followed by the announcement extension. If an announcement session is in progress, you will hear a reorder tone and the system will drop the call.

If the phone session port to an integrated board is in use, you will hear a reorder tone followed by silence. This indicates that the port is reserved for an announcement session. You can redial the FAC and extension every 45 seconds to access the port.

Note:

You can use multiple phone sessions with one session associated with each active integrated announcement board.

Once you access an announcement session, you can dial 1 to record an announcement, 2 to play an announcement, or 3 to delete an announcement. If the circuit pack memory is more than 90 percent full, the communication server gives a stutter dial tone when the you try to access an announcement session. Even if you hear a stutter tone, you can record the announcement.

Recording the announcement

About this task

If you dial 1, the communication server attempts to start a recording session with one of the following outcomes:

Procedure

- 1. If an announcement already exists and is protected, you will hear an intercept tone. Protection status is indicated by y. Hang up and determine the correct announcement extension to use.
- 2. If the announcement is currently being played to callers, you will hear the reorder tone.
- 3. If the communication server has started the recording session, you will hear a record tone to record the announcement.

Stop recording the announcement

Procedure

Choose one of the following methods to stop the recording after the announcement is complete:

- Hybrid, digital, or IP phone: Dial the pound (#) key to end the recording. When you end the recording with a pound key, you will hear the dial tone, and you can request a playback, delete, or record over operation. The "#" tones or a click sound produced when you hang up are not recorded. If the circuit pack memory becomes full during recording, you will hear a reorder tone, the system will drop you, and the announcement will not be retained.
- Analog phone: Hang up. Otherwise, ending with a pound key puts a tone in the message. If you are using an analog phone that is not connected with lineside T1 (DS1 type), Communication Manager records a click when you hang up. After you hang up, you must redial the FAC plus the announcement extension to start a new recording session. If the circuit pack memory becomes full during recording, you hear a reorder tone, Communication Manager drops the call, and does not retain the announcement.

Playing back the announcement

About this task

After you complete a recording and hang up, do not immediately dial the extension. The new announcement remains busy for approximately 15 seconds. To play back the new announcement, dial the FAC, the announcement extension, and the number two before the 15–second timer expires.

Upon completion of the recording session, Communication Manager sets a timer for 15 seconds. During this interval, the system restricts you to one of two tasks:

Procedure

- 1. Listen to the announcement just recorded.
- 2. Record another announcement.

Result

To listen to the announcement before the announcement is available to others, dial the FAC, the extension, and the number two. The announcement plays the dial tone, you can perform another operation, such as record a message.

Note:

If a caller attempts to dial an announcement that does not exist, the caller hears silence. Therefore, ensure that you have copied the media file to the announcement source.

Deleting the announcement

Procedure

Dial the feature access code, the extension, and the number three (3) to delete the announcement.

If an announcement is protected or is currently being played, Communication Manager does not delete the announcement and you hear a reorder tone. Delete a recorded announcement before rerecording another announcement with the dial 1 function.

Recorded announcements for ACD and other call center features

You can use recorded announcements for ACD, Call Vectoring, Call Prompting, Expert Agent Selection, VDN of Origin Announcement, Direct Department Calling, and UCD.

Recorded announcements and automatic wakeup

Recorded announcements allow Automatic Wakeup to use the built-in integrated announcement circuit pack or sources in place of the Audichron adjunct.

If you use an integrated, multiple integrated, or external type of announcement for Automatic Wakeup, you can also administer the announcement to repeat with the integ-rep announcement type and to allow barge-in as a queue type. The benefit of repeating announcements and barge-in queues is that you do not need to use a separate port for each wakeup announcement.

Locally sourced music and announcements

Definitions

Term	Definition
VAL announcement	A voice announcement with a TN2501AP board.
source	

Term	Definition
vVAL announcement source	A virtual VAL (vVAL) source integrated in a Media Gateway (G700, G250, G350, G450, and so on). The vVAL source is referred to as a virtual VAL source or an embedded VAL source.
announcement file	The recorded announcement file that is played for the specific announcement extension assigned to the audio group. This file must reside on every announcement source defined in the audio group. The file name on each source must match exactly what is administered for the announcement extension.
Avaya Aura® Media Server announcement source	A voice announcement using an Avaya Aura [®] Media Server source.

Locally sourced music and announcements

Call centers can use any or all of their VAL announcement sources in the gateways as sources for the same announcement.



Note:

You refer to the Avaya Aura® Media Server-based announcement source by the letter M followed by its media-server number. For example, you must enter M5 when you are entering media-server 5 into the announcement form or the audio-group form.

Locally sourced music offers the following benefits:

- Improves the quality of the audio.
- Reduces resource usage, such as VoIP resources, by selecting the nearest available source when playing the announcement.
- Provides backup for announcements because a working announcement source with the same announcement file is selected from the sources if the primary announcement source is not available.

Audio groups

The VAL announcement sources that contain a particular announcement file are assigned to an audio group, for example, G1. The audio group is then assigned to the announcement or audio extension port location as a group-sourced location instead of a single-sourced location. The caller usually hears an announcement from a source that is local to the incoming call trunk facility.

Selecting the most local source of an audio group

The algorithm skips the nonworking sources to find a working source in the group. The audio group algorithm selects one of the following sources:

- A source that is local to the trunk or the user in the same media gateway or in the same group of TDM-connected or ATM-connected Port Network Gateway.
- A source in a gateway in the same network region.

- One of the following sources in a gateway in the interconnected network region:
 - Adjacent with the most available bandwidth and highest quality codec.
 - Non adjacent based on the shortest number of hops.
- An Inter-Gateway Alternate Routing (IGAR)-connected source on a gateway interconnected through a PSTN trunk. This selection is based on IGAR percentage usage.

If you administer an announcement file extension for queuing and the source that is selected to play the file has no available playback ports, the request to play the announcement is held in a queue until a port on the source becomes available. With Call Vectoring, callers can listen to a previously-started feedback, such as ringback, until the system connects the call to an announcement. If you do not administer queuing for the announcement file extension, the system continues to search for a local source based on the listed criteria.

Capabilities of locally sourced music and announcements

Single or group sourced recorded announcement extensions as Music On Hold (MOH) sources

You can use integrated announcement to:

- Assign the announcement to a system MOH source instead of a port location as the system MOH source.
- Play MOH and music in vectors. As with audio groups, the callers hear music from a local source. The integrated announcement provides a repeating barge-in operation by combining the integrated repeating type with a forced assignment of barge-in. To assign MOH sources, use the Music Sources or the Feature-Related System Parameters screen if you are not using tenant partitioning.

Separate MOH groups with multiple analog or auxiliary trunk music source port locations

You can create separate MOH groups that can assign multiple analog or auxiliary trunk music source port locations. You can assign the MOH groups, for example, group 1, as system MOH sources instead of a single port location on the Music Sources screen. The music played is from a local music source of the assigned group.

You can use an announcement or audio source extension with an assigned audio group anywhere a single-sourced announcement is used. For example, you can use a group-sourced announcement extension of type integrated in any of the following Call Vectoring commands:

- announcement xxxxxx
- collect ... after announcement xxxxxx
- disconnect after announcement xxxxxx
- wait-time < time > [secs, mins, hrs] hearing [audio source ext] then [music, ringback, silence, continue]

Apply a partition-defined system music source as the system music

You can apply a partition-defined system music source with an MOH group or music audio group-sourced extension of type integrated as the system music using the following commands:

- wait-time < time > [secs, mins or hrs] hearing music
- wait-time < time > [secs, mins or hrs] hearing [audio source ext] then music

Chapter 7: Time of Day Clock Synchronization

With the Time of Day (TOD) Clock Synchronization feature, you can maintain synchronous clock times across a multisite call center network. Maintenance of accurate TOD settings is important for many functions such as the following:

- Creation of time stamps for items such as error logs, Malicious Call Trace (MCT) records, and CMS data.
- Scheduling of a large number of diverse task activities on Communication Manager and the adjuncts.

Related links

Using NTP/SNTP to synchronize Communication Manager to the UTC time on page 202

TOD synchronization methods

The TOD clock synchronization capabilities are available in Communication Manager releases with Multiple Locations and Daylight Saving rules. You can use the following methods to implement TOD clock synchronization:

- NTP/SNTP to administer direct switch synchronization.
- Avaya Site Administration to schedule synchronization tasks.

How to use NTP/SNTP to administer direct switch synchronization

In this method, individual communication servers use either Network Time Protocol (NTP) or Simple Network Time Protocol (SNTP) to synchronize their Operating System (OS) clocks with highly accurate Coordinated Universal Time (UTC) from an Internet Time Server. The Communication Manager clock time also uses the OS clock time.

How to use Avaya Site Administration to schedule synchronization tasks

In this method, you can use Avaya Site Administration to set up a regularly scheduled synchronization task for all servers in the call center network. You can use the time from the system clock of the client computer to synchronize the communication servers. A best practice is

to run the NTP or SNTP software on the client computer to synchronize the client computer time with the UTC time from an Internet Time Server.

You can use this method of time synchronization for the following communication servers:

- DEFINITY G3csi
- Avaya DEFINITY Server CSI
- DEFINITY G3si
- Avaya DEFINITY Server SI
- DEFINITY G3r
- Avaya DEFINITY Server R

Using NTP/SNTP to synchronize Communication Manager to the UTC time

The method applies to the Communication Manager server on which the platform OS, Linux or Windows 2000, uses NTP or SNTP to obtain highly accurate UTC data from an Internet Time Server. The time on the OS clock, which is continuously adjusted to match the polled UTC time also provides the basis for the Communication Manager clock time. This synchronization method is accurate to the fraction of a second.

When a multisite network includes Communication Manager that uses this synchronization method, each Communication Manager server maintains its own clock time. However, as all Communication Manager servers that use this method maintain settings based on the UTC time and are essentially identical, clock synchronization is still achieved.

For requirements and procedures associated with this screen of clock synchronization, see *Administering Avaya Aura*[®] *Communication Manager*.

Related links

Time of Day Clock Synchronization on page 201

How to use Avaya Site Administration to set up a TOD synchronization schedule

This method for TOD clock synchronization, which applies to certain communication servers, uses the Avaya Site Administration tool installed on a client computer to set up a synchronization task schedule.

You must install Avaya Site Administration on the client computer along with the NTP or SNTP software. Connect the client computer to an Internet Time Server so that the computer can continuously poll the Internet Time Server for the UTC time. Synchronize the client clock time

with the clock time of the switches on the network using the Avaya Site Administration Time Synchronization feature.

On communication servers with software releases prior to R11, the synchronization command is ignored if the minute time specified for the incoming time is the same as that currently being counted at the switch. Consequently, this synchronization method is only accurate to within 59 seconds or less on communication servers that are installed with releases prior to R11.

On communication servers that are installed with a software release of R11 or later, if the minute time specified for the incoming synchronization command is the same as that being counted at the switch, the minute count on the switch is set back to the 0-second mark for the minute. When potential network delays are factored in, this method is accurate to within 5 seconds or less.

Prerequisites for using Avaya Site Administration to set up a TOD synchronization schedule

- You must install and run Avaya Site Administration on the client computer when synchronization runs are scheduled to occur.
- You must ensure that the client computer can establish a Local Area Network (LAN) or dialup connection with the target communication servers when synchronization runs are scheduled to occur.

Important:

Before you set up TOD synchronization tasks in Avaya Site Administration, administer dedicated synchronization connections from Avaya Site Administration to each Communication Manager.

- You must install an SNTP/NTP software on the client computer and connect the client computer to an NTP Time Server through the internet.
- You must configure the Daylight Saving Time (DST) rules on the client computer since the same rule applies to the computer clock.

Things to know before you set up a synchronization schedule

Read the following details before you use Avaya Site Administration to set up a TOD clock synchronization schedule.

Specify offset values in Standard Time equivalents

When you set up a regular schedule for a time synchronization task in Avaya Site Administration, specify an offset value that reflects the difference in local time between the client computer and a target Communication Manager location.

Avaya Site Administration uses the set time command to synchronize the Communication Manager clock time with the clock time of the Avaya Site Administration client. Avaya Site Administration always sends the set time command in Standard Time. Communication Manager checks the DST rule specified in the set time command, which is always Standard Time, and compares the Daylight Saving Rule with the existing rule. If a DST rule other than Standard Time is in effect, Communication Manager sends an error message back to Avaya Site

Administration. When Avaya Site Administration receives the error message from Communication Manager, Avaya Site Administration automatically corrects the synchronization time to comply with the DST rule on the local computer and again sends the adjusted time back to Communication Manager.

When you calculate offset values to use as input in the Avaya Site Administration Time Synchronization feature, you must do the following:

- If either the client or the target Communication Manager is located in a time zone where the DST rule is in effect, convert the local time to reflect the Standard Time rule. A best practice is to always normalize Communication Manager and client times to Standard Time before you calculate offset values.
- After you normalize the local times to their Standard Time equivalents, calculate the offset time as the difference between the local Standard Time at the client end and the local Standard Time at the Communication Manager end. This value is the offset between the client computer and Communication Manager that you specify when you use the Avaya Site Administration Time Synchronization feature.

For more information on how to specify offset values in Standard Time equivalents, see the Administering Avaya Aura® Communication Manager document.

Possible lag times

You can set up multiple TOD clock synchronization tasks to run simultaneously on the client computer for possible lag between the synchronization start time on the client and actual run time at the Communication Manager server. Synchronization tasks run in a sequence and the actual synchronization run times in the Communication Manager server varies from the run time specified on the client by several minutes. However, the accuracy of the synchronization setting is not affected.

EPN locations do not require synchronization

Expansion Port Network (EPN) cabinets that are located in a different time zone from their connecting Communication Manager do not require separate TOD clock synchronization. EPNs receives the synchronized clock time from Communication Manager, which is adjusted according to any settings that are specified in the Locations screen. You can use the change multiple locations command to access the Locations screen.



Note:

CMS data synchronizes with the Communication Manager time for the ACD from which CMS data is generated. CMS data uses the same DST rules that apply to Communication Manager.

Run synchronization tasks during low traffic periods

Avaya Site Administration uses the set time command to adjust the Communication Manager clock time. Since the Communication Manager traffic can delay the completion of the set time command, schedule the clock synchronization task to run during a low traffic period on each Communication Manager.

Run synchronization tasks in the middle of CMS archive intervals

About this task

Avaya Call Management System is designed to run archives at regular intervals of 15 minutes, 30 minutes, or 1 hour. Schedule the synchronization tasks to run near the middle of an archive interval. The approach minimizes potential redundancies in archive interval records for an ACD.

For example, an archive interval begins at 09:00. If Communication Manager receives a synchronization command and changes the Communication Manager clock time to 08:59, a second archive interval begins when the Communication Manager clock changes to 09:00. In this case, two archive intervals that have the same 09:00 time stamp are recorded. One interval extends from 09:00 to 09:01. The other interval also begins at 09:00 and extends for the normal duration specified for archive intervals on the ACD, which is either 15, 30, or 60 minutes.

To prevent such as situation, schedule synchronization tasks to run near the middle of the archive interval specified for each ACD.

To determine the Call Management System archive interval length specified for an ACD:

Procedure

- 1. Start Avaya Call Management System Supervisor.
- 2. From the **main** menu, select:

Tools > System Setup

The CMS System Setup dialog appears.

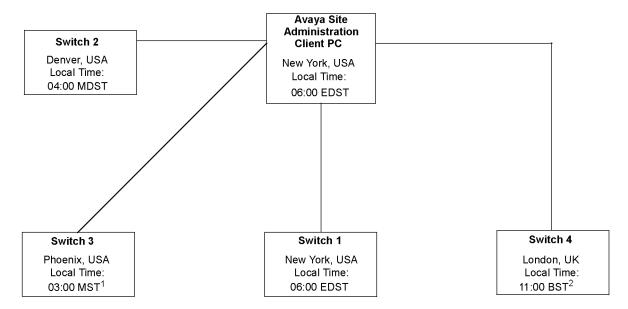
- 3. Perform the following:
 - a. In the Operations list, select Storage Intervals.
 - b. To retrieve archive interval information on an ACD, select the ACD from the **ACD** field box.
- 4. Select OK.

CMS Supervisor displays the Storage Intervals window, which includes the archive interval for the ACD.

Designing a TOD clock synchronization schedule

Example of a multisite call center network

The following example scenario shows how to design an Avaya Site Administration Time Synchronization schedule for a multisite call network that spans several time zones. The example sites and the respective time zones are shown in the following figure:



- 1. Site is not on Daylight Saving Time (DST)
- 2. British Summer Time (BST) is UK Daylight Saving Time (GMT + 1)

The call center network example in this figure includes four Communication Manager locations that are located in different time zones. In this example, all Communication Manager and the client are on Daylight Saving Time, with the exception of Communication Manager 2 located in Phoenix.

Using Avaya Site Administration to create a TOD synchronization schedule requires careful planning and consideration. The steps described in the following page represent the most error-free method you can use to design your synchronization schedule.

How to determine the location offset values

The Avaya Site Administration Time Synchronization feature sends synchronization messages specified in Standard Time to Communication Manager using the set time command. To calculate the offset values that represent the time difference between the Avaya Site Administration client computer and the Communication Manager locations, normalize all location times to their Standard Time equivalents.

Important:

Always calculate the offset values based on comparisons between Standard Time equivalents. If DST rules are not the same for the Avaya Site Administration client location and a target Communication Manager location, significant synchronization errors can result.

The following table uses the Communication Manager locations described in the example scenario to derive correct offset values for the client computer and the Communication Manager locations.

Note:

The local time listed in the table are random and only to illustrate the time differences between locations. You can use any set of relative location time.

Local time or normalized standard time		Calculated offset value
Avaya Site Administration	Communication Manager in New York	
Client computer in New York		
Local Time: 06:00 EDST	Local Time: 06:00 EDST	
Adjusted Standard Time: 05:00 EST	Adjusted Standard Time: 05:00 EST	0
Client computer in New York	Communication Manager in Denver	
Local Time: 06:00 EDST	Local Time: 04:00 MDST	
Adjusted Standard Time: 05:00 EST	Adjusted Standard Time: 03:00 EST	-2
Client computer in New York	Communication Manager in Phoenix	
Local Time: 06:00 EDST	Local Time: 03:00 MST	
Adjusted Standard Time: 05:00 EST (New York)	Adjusted Standard Time: 03:00 MST	-2
Client computer in New York	Communication Manager in London	
Local Time: 06:00 EDST	Local Time: 11:00 BST	
Adjusted Standard Time: 05:00 EST	Adjusted Standard Time: 10:00 BST	+5

The table demonstrates the importance of normalizing all local times to Standard Time when calculating time offsets. The unadjusted time difference between Denver and New York is 2 hours and the calculated offset is minus 2 hours.

However, despite the fact that the unadjusted time difference between New York and Phoenix is 3 hours, the calculated offset is also minus 2 hours, which is the same offset value that is calculated for New York and Denver.

How to determine synchronization run time

After calculating the offset values, you can schedule the synchronization task to run on the Avaya Site Administration client computer. A best practice is to normalize all local time to Standard Time equivalents.

The following table shows how to obtain the synchronization task run time from the Communication Manager locations. The task run time represents the values that are entered in the Scheduler dialog in Avaya Site Administration.

Note:

Always establish clock synchronization run time on the basis of low traffic time intervals for Communication Manager. In the following example, a single local Communication Manager time is used as the synchronization run time for all Communication Manager. In actual practice, low traffic periods for Communication Manager in a multisite network may not always be the same for each Communication Manager location.

Local time or normalized standard time for synchronization at a Communication Manager location	Client computer or Communication Manager offset	Synchronization run time set on client computer
Communication Manager in New York		
Local Synchronization Run Time: 03:07 EDST		
Adjusted Standard Time: 02:07 EST	0	2:07 AM
Communication Manager in Denver		
Local Synchronization Run Time: 03:00 MDST		
Adjusted Standard Time: 02:07 EST	-2	0:07 AM
Communication Manager in Phoenix		
Local Time: 03:07 MST		
Adjusted Standard Time:		
03:07 MST	-2	1:07 AM
Communication Manager in London		
Local Time: 03:07 BST		
Adjusted Standard Time: 03:07 BST	+5	8:07 AM

Synchronization run times must occur during low traffic periods at Communication Manager. In this example, the simplifying assumption is made that a single low traffic time is common to all Communication Manager locations. This assumption may not be true for all call center operations.

Listed offset values are those that were derived in determining synchronization run times.

If the client computer clock is currently set to DST rules, add 1 hour to the synchronization start time that you specify in the Avaya Site Administration Schedule dialog.

Specify the run time entered in the Schedule dialog of the Avaya Site Administration Time Synchronization feature in a 12-hour AM/PM time format.

Synchronization run time considerations

If the client computer and the Communication Manager locations do not use the same DST rules, actual synchronization run time vary by an hour over the course of the year. Depending on how the DST rules between Communication Manager and client computer vary, the actual synchronization run time at Communication Manager occurs either one hour earlier or later than the time specified in Avaya Site Administration. The following basic rules apply:

Note:

The following exceptions relate only to the synchronization start times. If the offsets are calculated correctly, the following start time exceptions have no effect on the accuracy of the synchronization.

• When DST is in effect at Communication Manager but not on the client computer, the actual synchronization run time at Communication Manager occurs 1 hour later than the run time that is specified on the client computer.

• When DST is in effect at the client computer but not at Communication Manager, the actual synchronization run time at Communication Manager occurs 1 hour earlier than the synchronization run time that is specified on the client computer.

Setting up a TOD synchronization task schedule in Avaya Site Administration

Before you begin

Before you set up a TOD synchronization task, create dedicated connections from Avaya Site Administration to each target Communication Manager.

Procedure

- 1. Start Avaya Site Administration on the client computer and select a Communication Manager from the drop-down list on the main tool bar.
- 2. In the browser pane located on the right side of the main application window, select the Fault & Performance tab.
- 3. Select the **Time Synchronization** option.
- 4. In the Time Synchronization Properties dialog box that the system displays:
 - If the target Communication Manager is located in a different time zone, check the
 Offset field and specify the time offset between the client computer and Communication
 Manager.

Important:

When different Daylight Saving Time rules are in effect at the Avaya Site Administration client location and a target Communication Manager location, synchronization errors can result if you do not calculate offset values based on Standard Time equivalents.

- · Click Next.
- 5. In the Time Synchronization Schedule dialog box that the system displays:
 - · Check the Schedule this task to run field.
 - · Click Schedule.
- 6. Perform the following tasks in the Scheduler dialog box that the system displays:

Important:

Schedule to run the synchronization task during a low traffic period at Communication Manager. Heavy Communication Manager traffic can delay the execution of the synchronization command.

- a. In the **Date** field, click the arrow to view the calendar and select a day to start the synchronization task.
- b. In the **Time** field, enter the time to run the synchronization task.

The time you specify in this field is the time on the client computer and not the time at the target Communication Manager. To determine the correct time to enter in the Time field, perform the following tasks:

- Determine the local time at Communication Manager when the synchronization runs. If necessary, adjust the local time to the Standard Time equivalent.
- Subtract the offset factor that you used in step 4 from Communication Manager run time that you derived in the preceding sub step. The time you calculate is the run time on the client expressed in Standard Time. If DST is in effect at the client computer, increase the time by 1 hour to account for DST.
- Enter the calculated time in the **Time** field.
- c. Select one of the following options for the **Recurrence Pattern** field: frequent, weekly, or monthly. Enter the time parameters specified with the option.

Note:

If you select the Frequent setting, set the task to run at 24-hour intervals.

- d. Click Finish.
- e. Verify the synchronization schedule information that you provided on the Time Synchronization - Schedule dialog that appears, and click **Finish**.
- f. Click Finish on the Time Synchronization Summary window that the system displays.
- 7. Repeat steps 1 through 6 to synchronize any other Communication Manager.

NTP or SNTP and Internet Time Servers

The NTP synchronizes the system time on a computer with that of an Internet Time Server that has been synchronized to a reference source such as radio, Global Positioning Service (GPS) receiver that provides Coordinated Universal Time (UTC). NTP communicates with the Internet Time Server by using a dialup modem or direct LAN connection.

The SNTP is a basic version of NTP that allows for a greater degree of error, but can still deliver time to an accuracy on the order of fractions of a second.

SNTP on switch platforms that support direct synchronization

The following Avaya switch platforms can use the SNTP software on the platform OS to directly synchronize the switch clock to the UTC time from an Internet Time Server:

- Avaya IP600
- DEFINITY One
- Avaya S8000 Media Server

- Avaya S8300 Media Server
- Avaya S8700 Media Server

The platforms listed include either Red Hat Linux or Windows 2000 as the platform OS. The following conditions are in effect for the SNTP configuration on the systems:

- For Linux platforms, configure the IP addresses for a minimum of three different Internet Time Servers.
- For Linux platforms, go to the following Web sites for information on how to obtain NTP/ SNTP software:

http://www.ubr.com/clocks/timesw/timesw.html http://www.ntp.org/software/index.html

- Linux platforms support the authentication/encryption mode provided in NTP/SNTP version 3 or later support. This capability is not enabled by default.
- Windows 2000 platforms can use only one SNTP client (W32Time), which limits UTC polling to one Internet Time Server IP address at a time. For more information, see the "Microsoft Windows 2000" documentation. Search for the keywords Window Time Service.
- Windows 2000 platforms use the W32Time service for SNTP functions. This service does not support an authentication/encryption mode for the SNTP protocol.
- W32Time service allows optional polling of a Microsoft network domain controller as the primary time server. Avaya does not support this configuration.

Platforms that synchronize through an Avaya Site Administration client computer

The following Avaya switch platforms must use the Avaya Site Administration Time Synchronization feature to maintain switch synchronization:

- DEFINITY G3csi
- DEFINITY G3si
- DEFINITY G3r
- Avaya S8100 Media Server
- Avaya S8200 Media Server
- Avaya S8500 Media Server

The following Web sites provide information on how to obtain the NTP/SNTP software for the Avaya Site Administration client computer:

http://www.ubr.com/clocks/timesw/timesw.html

http://www.ntp.org/software/index.html

Setting up ACD offset times for CMS reporting

The time stamp for CMS data is obtained from the local switch on which the data is generated. When a CMS system includes ACDs that are located in different time zones, CMS reports reflect the time zone differences based on unadjusted data. However, you can use Avaya CMS Supervisor to adjust CMS data derived from remote ACDs so that you to view data from different time zones in a common time format. The common time format is a more convenient way to view and assess simultaneous call center activity across time zones.

To adjust CMS data to reflect a common time format, you must:

- Designate a master ACD.
- Determine the appropriate offsets for each remote ACD.
- Set the switch time zone offset values for each ACD in the CMS Supervisor Storage Windows dialog box.

Setting switch time zone offset values for CMS report times **Procedure**

- 1. In the main Supervisor Controller window of CMS Supervisor, select **Tools > System**
- 2. In the CMS System Setup window:
 - a. Select the **Operations** tab.
 - b. From the displayed list, select Storage Intervals.
 - c. In the ACD field, select an ACD.
 - d. Click OK.

The system displays the Storage Intervals window.

3. In the Switch time zone offset (-23 to +23) field, assign an offset value that reflects the time difference between the target ACD and the designated master ACD.



▼ Note:

For more information, see Avaya Call Management System Administration.

- 4. On the main menu, select **Actions** > **Modify**.
- 5. Repeat the procedure for any other ACDs that require a time zone offset.

Chapter 8: Resources

Documentation

Title	Use the document to:	Audience
Supporting		
Avaya Aura® Communication Manager System Capacities Table	Read about the system capacity and system scalability.	Implementation engineers, sales engineers, and solution architects
Programming Call Vectoring Features in Avaya Aura® Call Center Elite	Write and edit call vectors.	Implementation engineers and system administrators
Avaya Aura [®] Call Center Elite Call Vectoring Feature Description	Learn how the Call Vectoring feature work and get to now details about feature characteristics, capabilities, capacities, and interactions.	Implementation engineers and system administrators
Understanding		
Avaya Aura [®] Call Center Elite Feature Reference	Read about Automatic Call Distribution (ACD) and Call Vectoring features.	All users of Avaya Aura [®] Call Center Elite
Avaya Aura® Communication Manager Feature Description and Implementation	Read about Avaya Aura [®] Communication Manager features.	All users of Communication Manager
Using		
Using Avaya Business Advocate	Learn how to use Business Advocate for agent selection and call selection.	Contact center managers, system administrators, and supervisors
Using Avaya 96X1 SIP Agent Deskphones with Avaya Aura [®] Call Center Elite	Know the prerequisites for using Avaya 96X1 SIP agent deskphones with Call Center Elite.	Implementation engineers and system administrators
Administering		
Administering Avaya Aura® Call Center Elite	Administer all the Call Center Elite features.	System administrators
Troubleshooting		

Title	Use the document to:	Audience
Troubleshooting Avaya Aura® Call Center Elite	Know how to troubleshoot common problems and denial events related to Call Center Elite.	All users who perform troubleshooting tasks in Call Center Elite
Implementation		
Planning for an Avaya Aura® Call Center Elite Implementation	Know how to transition from a basic call center environment to an Expert Agent Selection (EAS) and a Call Vectoring environment.	All users who perform Call Center Elite site preparation and planning tasks, including implementation engineers and sales engineers.

Finding documents on the Avaya Support website

Procedure

- 1. Go to https://support.avaya.com.
- 2. At the top of the screen, type your username and password and click Login.
- 3. Click Support by Product > Documents.
- 4. In **Enter your Product Here**, type the product name and then select the product from the list.
- 5. In **Choose Release**, select the appropriate release number.
 - The Choose Release field is not available if there is only one release for the product.
- 6. In the **Content Type** filter, click a document type, or click **Select All** to see a list of all available documents.
 - For example, for user guides, click **User Guides** in the **Content Type** filter. The list only displays the documents for the selected category.
- 7. Click Enter.

Accessing the port matrix document

Procedure

- 1. Go to https://support.avaya.com.
- 2. Log on to the Avaya website with a valid Avaya user ID and password.
- 3. On the Avaya Support page, click **Support by Product > Documents**.
- 4. In **Enter Your Product Here**, type the product name, and then select the product from the list of suggested product names.

- 5. In **Choose Release**, select the required release number.
- 6. In the **Content Type** filter, select one or both the following categories:
 - Application & Technical Notes
 - Design, Development & System Mgt

The list displays the product-specific Port Matrix document.

7. Click Enter.

Avaya Documentation Center navigation

For some programs, the latest customer documentation is now available on the Avaya Documentation Center website at https://documentation.avaya.com.

! Important:

For documents that are not available on Avaya Documentation Center, click **More Sites** > **Support** on the top menu to open https://support.avaya.com.

Using the Avaya Documentation Center, you can:

Search for keywords.

To filter by product, click Filters and select a product.

· Search for documents.

From **Products & Solutions**, select a solution category and product, and then select the appropriate document from the list.

- Sort documents on the search results page.
- Click **Languages** (((a)) to change the display language and view localized documents.
- Publish a PDF of the current section in a document, the section and its subsections, or the entire document.
- Add content to your collection using My Docs (☆).

Navigate to the **Manage Content > My Docs** menu, and do any of the following:

- Create, rename, and delete a collection.
- Add topics from various documents to a 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 collection that others have shared with you.
- Add yourself as a watcher using the Watch icon (

Navigate to the Manage Content > Watchlist menu, and do the following:

- Enable **Include in email notification** to receive email alerts.
- Unwatch selected content, all content in a document, or all content on the Watch list page.

As a watcher, you are notified when content is updated or deleted from a document, or the document is removed from the website.

- Share a section on social media platforms, such as Facebook, LinkedIn, and Twitter.
- Send feedback on a section and rate the content.

Note:

Some functionality is only available when you log in to the website. The available functionality depends on your role.

Training

The following courses are available on www.avaya-learning.com. Enter the course code in the Search field, and click Go to search for the course.

Course code	Course title
ACIS-7391	
73600V	Implementing Avaya Aura [®] Call Center Elite 40 hours
7391X	Avaya Aura [®] Call Center Elite and Avaya Aura [®] Call Center Elite Multichannel Implementation Exam 1.50 hours
ACSS-7491	
74600V	Supporting Avaya Aura [®] Call Center Elite 16 hours
7491X	Avaya Aura [®] Call Center Elite and Avaya Aura [®] Call Center Elite Multichannel Support Exam 1.50 hours
2416W	Avaya Aura [®] Call Center Elite Fundamentals 0.5 hour for all audiences
2412W	Using Avaya Workspaces for Call Center Elite – Agents 0.5 hour for end-users
2414W	Using Avaya Workspaces for Call Center Elite – Supervisors 0.5 hour for endusers

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 Content Type.
 - In Search, type the product name. On the Search Results page, click Clear All and select Video in the 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 key word or key words in the Search Channel to search for a specific product or
 - Scroll down Playlists, and click a topic name to see the list of videos available for the topic. 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 guestions, or request an agent to connect you to a support team if an issue requires additional expertise.

Related links

Using the Avaya InSite Knowledge Base on page 217

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 http://www.avaya.com/support.
- 2. Log on to the Avaya website with a valid Avaya user ID and password. The system displays the Avaya Support page.
- 3. Click Support by Product > Product-specific Support.
- 4. In Enter Product Name, enter the product, and press Enter.
- 5. Select the product from the list, and select a release.
- 6. Click the **Technical Solutions** tab to see articles.
- 7. Select relevant articles.

Related links

Support on page 217

Glossary

AAR When resources are unavailable, Communication Manager uses the

Automatic Alternate Routing (AAR) feature to route calls to a different

route than the first-choice route.

abandoned call An inbound call in which the caller disconnects the call before an agent

can answer the call.

access code A 1-digit, 2-digit, or 3-digit dial code that activates or cancels a feature, or

accesses an outgoing trunk.

access trunk A trunk that connects a main communications system with a tandem

communications system in an <u>Electronic Tandem Network (ETN)</u> on page 225. You can use an access trunk to connect a system or tandem to

a serving office or service node. Also called an access tie trunk.

ACD Automatic Call Distribution (ACD) is a telephony feature for processing

and distributing inbound, outbound, and internal calls to groups of

extensions.

ACW An agent enters the After Call Work (ACW) mode to complete ACD call-

related activities, such as filling forms or taking notes. An agent in the

ACW mode is unavailable to receive ACD calls.

AD Abbreviated Dialing (AD) makes agent login easier as agents can press

the AD button to dial an access code, split number, or login ID.

adjunct A processor that does tasks for another processor and is optional in the

configuration of the other processor. See also application on page 220.

adjunct routing A means of evaluating calls before the calls are processed by

requesting information from an adjunct. Communication Manager requests instructions from an associated adjunct and makes a routing

decision based on agent availability or caller information.

adjunct-controlled

split

An ACD split that is administered to be controlled by another application. Agents logged in to such splits must do all telephony work, ACD login, ACD logout, and work mode changes through the adjunct, except for auto-available adjunct-controlled splits, wherein agents cannot log in, log

out, or change the work modes.

adjunct-monitored

call

An adjunct-controlled call, active-notification call, or call that provides

event reporting over a domain-control association.

adjusted EWT

A Best Service Routing (BSR) term for Expected Wait Time (EWT) plus a

user adjustment set by a consider command.

administration

terminal

A terminal that is used to administer and maintain a system.

AES Application Enablement Services (AES) is an Avaya product that

provides a platform for the development of CTI-based applications for

Communication Manager.

agent A member of an ACD hunt group, ACD split, or skill. Depending on the

ACD software, an agent can be a member of multiple splits/skills.

agent report A report that provides historical traffic information for internally measured

agents.

ANI Automatic Number Identification (ANI) is a display of the calling number

for agents to gain access to information about the caller.

appearance A software process that is associated with an extension and whose

purpose is to supervise a call. An extension can have multiple appearances. Also called call appearance, line appearance, and

occurrence.

application An adjunct that requests and receives ASAI services or capabilities.

Applications can reside on an adjunct. However, Communication Manager cannot distinguish among several applications residing on the same adjunct. Hence, Communication Manager treats the adjunct and all resident applications as a single application. The terms application and

adjunct are used interchangeably throughout the document.

application plan A plan used in multisite Best Service Routing (BSR) applications. The

application plan identifies remote Communication Manager servers for comparison in a consider series. The plan specifies information that is required to contact each Communication Manager server and to interflow

calls to the selected Communication Manager server.

applications

processor

A micro-computer based, program controlled computer providing application services for the switch. The processor is used with several user-controlled applications such as traffic analysis and electronic

documentation.

ARS Automatic Route Selection (ARS) is a feature that Communication

Manager uses to automatically select the least cost route to send a toll

call.

ASAI Adjunct-Switch Application Interface (ASAI) is an Avaya protocol that

applications use to gain access to the call-processing capabilities of

Communication Manager.

association A communication channel between adjunct and switch for messaging

purposes. An active association is one that applies to an existing call on

the switch or to an extension on the call.

attendant A person at a console who provides personalized service for incoming

callers and voice-services users by performing switching and signaling

operations.

attendant console The workstation used by an attendant. The attendant console allows the

> attendant to originate a call, answer an incoming call, transfer a call to another extension or trunk, put a call on hold, and remove a call from hold. Attendants using the console can also manage and monitor some

system operations. Also called console.

AUDIX™ Audio Information Exchange (AUDIX[™]) is an Avaya messaging system.

auto-in A call-answering mode in which an agent automatically receives ACD

calls without pressing any button to receive calls.

Automatic Callback A feature that 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.

Automatic Circuit Assurance (ACA)

A feature that tracks calls of unusual duration to facilitate troubleshooting.

A high number of very short calls or a low number of very long calls

signify a faulty trunk.

automatic trunk A trunk that does not require addressing information because the

> destination is predetermined. A request for service on the trunk, called a seizure, is sufficient to route the call. The normal destination of an automatic trunk is the communications-system attendant group. Also

called automatic incoming trunk and automatic tie trunk.

AUX work Agents enter the Auxiliary (AUX) work mode for non-ACD activities, such

as taking a break, going for lunch, or making an outgoing call. Agents in

the AUX work mode are unavailable to receive ACD calls.

auxiliary trunk A trunk used to connect auxiliary equipment, such as radio-paging

equipment, to a communications system.

available agent

strategy

A strategy that determines how the Best Service Routing (BSR) commands in a vector identify the best skill when multiple skills have

available, that is, idle agents.

Avaya Aura® A converged communications platform unifying media, modes, network,

devices, applications. Avaya Aura® is based on the SIP architecture with

Session Manager at the core.

Avaya Aura[®] Media

Server

Avaya Aura® Media Server (Avaya Aura® MS) is a software-based media platform. Communication Manager uses Avaya Aura® MS to provide IP audio, tone generation and detection, and announcement capabilities similar to legacy H.248 media gateways or port networks with media

processors.

AWOH Administration Without Hardware (AWOH) is a feature that allows

administration of ports without associated terminals or other hardware.

barrier code A security code used with remote access to prevent unauthorized access

to the system.

BCMS A software package residing on Communication Manager that monitors

the operations of ACD systems. Basic Call Management System (BCMS)

collects data related to the calls on Communication Manager and

organizes data into reports that help supervisors manage ACD facilities

and personnel.

best The split/skill or location that can provide the best service to a caller as

determined by BSR.

BHCC Busy-Hour Call Completion (BHCC) is a measure of the number of calls

that Communication Manager successfully completes during the peak

hour of a network.

BRI Basic Rate Interface (BRI) is an ISDN configuration that offers two bearer

(B) channels for voice and data and one data channel for signaling.

bridge (bridging) The appearance of a phone extension at other phones.

bridged appearance A call appearance on a telephone that matches a call appearance on

another telephone for the duration of a call.

BSR A feature that provides singlesite and multisite load balancing and

maximizes staffing resources. Communication Manager uses Best Service Routing (BSR) to compare skills and to route calls to the best

skill.

Business Advocate A Call Center Elite feature that establishes different service levels for

different types of calls. For example, a company decides that a premium

customer must receive service before the other types of customers.

call appearance For an attendant console, the six buttons labeled **a** to **f** for making calls,

receiving calls, or holding calls. For a deskphone, a button labeled with an

extension for making calls, receiving calls, or holding calls.

cause value

A value that is returned in response to requests or in event reports when a denial or unexpected condition occurs.

CCS or hundred call seconds

A unit of call traffic. Call traffic for a facility is scanned every 100 seconds. There are 3600 seconds per hour. The Roman numeral for 100 is the capital letter C. The abbreviation for call seconds is CS. Therefore, 100 call seconds is abbreviated CCS. If a facility is busy for an entire hour, the facility is said to have been busy for 36 CCS.

channel

- 1. A circuit-switched call.
- 2. A communications path for transmitting voice and data.
- 3. In wideband, all of the time slots (contiguous or noncontiguous) necessary to support a call. Example: an H0-channel uses six 64-kbps time slots.
- 4. A DS0 on a T1 or E1 facility not specifically associated with a logical circuit-switched call; analogous to a single trunk.

circuit

- 1. An arrangement of electrical elements through which electric current flows.
- 2. A channel or transmission path between points.

circuit pack

A card with microprocessors, transistors, and other electrical circuits. A circuit pack is installed in a switch carrier or bay. Also called a circuit board or circuit card.

CMS

A software program for reporting and managing agents, splits, trunks, trunk groups, vectors, and VDNs. With Call Management System (CMS), you can also administer some ACD features.

CO

Central Office (CO) is a switch that a local phone company owns to provide local phone service (dial-tone) and access to toll facilities for long-distance calling.

communications server

A software-controlled processor complex that interprets dialing pulses, tones, and keyboard characters and makes the proper connections both within the system and external to the system. The communications system itself consists of a digital computer, software, storage device, and carriers with special hardware to perform the connections. A communications system provides voice and data communications services, including access to public and private networks, for telephones and data terminals on a customer's premises. Previously called a switch or a Private Branch eXchange (PBX).

confirmation tone

A telephone tone confirming that feature activation, deactivation, or cancellation has been accepted.

connectivity A connection of disparate devices within a single system.

consider sequence A consider series plus a queue-to best, check-best, or reply-

best step is called a consider sequence.

consider series A series of consider commands typically written in sets. A set of

consider commands is called a consider series.

COR Class of Restriction (COR) is a feature that allows classes of call-

origination and call-termination restrictions for phones, phone groups,

data modules, and trunk groups.

Class of Service (COS) is a feature that uses a number to specify if phone

users can activate the Automatic Callback, Call Forwarding All Calls, Data

Privacy, or Priority Calling features.

coverage answer

group

A group of up to eight telephones that ring simultaneously when a call is redirected by Call Coverage. Any one of the group can answer the call.

coverage call A call that is automatically redirected from the called party's extension to

an alternate answering position when certain coverage criteria are met.

coverage path An order in which calls are redirected to alternate answering positions.

coverage point An extension or attendant group, VDN, or ACD split designated as an

alternate answering position in a coverage path.

covering user A person at a coverage point who answers a redirected call.

CWC Call Work Codes (CWCs) are up to 16–digit sequences that agents type

to record the occurrence of customer-defined events, such as account

codes or social security numbers.

data link A configuration of physical facilities enabling end terminals to

communicate directly with each other.

data terminal An input/output (I/O) device that has either switched or direct access to a

host computer or to a processor interface.

dial-repeating tie

trunk

A tie trunk that transmits called-party addressing information between two

communications systems.

dial-repeating trunks A PBX tie trunk that is capable of handling PBX station-signaling

information without attendant assistance.

direct agent A feature, accessed only through ASAI, that allows a call to be placed

in a split queue but routed only to a specific agent in that split. The call receives normal ACD call treatment (for example, announcements) and is measured as an ACD call while ensuring that a particular agent answers.

Direct Inward Dialing (DID) trunk

An incoming trunk used for dialing directly from the public network into a communications system without help from the attendant.

DMCC

Device, Media, and Call Control (DMCC) is the new name for Communication Manager Application Programming Interface (API), that is, CMAPI.

domain

A group of VDNs, ACD splits, and stations.

Dynamic Host Configuration Protocol (DHCP)

A protocol that dynamically assigns IP addresses to devices when the devices connect to the network.

Dynamic Percentage Adjustment

A Business Advocate feature that automatically adjusts the agent target allocations to meet the administered service level targets.

Dynamic Queue Position

A Business Advocate feature that queues calls from multiple VDNs to a single skill, while maintaining the service objectives of each originating VDN. For instance, DQP positions a premium customer call with an assigned service objective of 10 seconds before a regular customer call with an assigned service objective of 25 seconds. Dynamic Queue Position (DQP) is also known Service Objective by VDN.

Dynamic Threshold Adjustment

A Business Advocate Service Level Supervisor (SLS) feature that meets the administered service levels by automatically adjusting the overload thresholds to engage reserve agents.

EAD-LOA

Expert Agent Distribution - Least Occupied Agent (EAD-LOA) is an agent selection method for call delivery. With EAD-LOA, calls are delivered to the available agent with the highest skill level and the lowest percentage of work time since login, when compared to other available agents with the same skill level.

EAD-MIA

Expert Agent Distribution - Most Idle Agent (EAD_MIA) is an agent selection method for call delivery. With EAD-MIA, calls are delivered to the available agent with the highest skill level who has been idle the longest since the last ACD call, when compared to other available agents with the same skill level.

ECC

External Call Controller (ECC) is an external Media Gateway Controller (MGC) that communicates with the G250 or G350 media gateways in a network.

Electronic Tandem Network (ETN)

A large private network that has automatic call-routing capabilities based on the number dialed and the most preferred route available. Each switch in the network is assigned a unique private network office code (RNX), and each telephone is assigned a unique extension.

Exclusion A feature that allows multi-appearance telephone users to keep other

users with the same extension from bridging onto an existing call.

Expansion Port Network (EPN)

A port network that is connected to the Time Division Multiplex (TDM) bus and packet bus of a processor port network. Control is achieved by indirect connection of the EPN to the processor port network using a

port-network link.

Expected Wait Time (EWT)

A prediction of how long a call waits in queue before the call is answered.

extension-in (EXT-IN)

A work state agents go into when answering a non-ACD call. If the agent is in manual-in or auto-in and receives an EXT-IN call, the call is recorded by the reporting adjunct as an AUX-IN call.

extension-out (EXT-OUT)

A work state that agents go into when placing non-ACD calls.

external call

A connection between a communications system user and a party on the public network, or on another communications system in a private network.

facility

A telecommunications transmission pathway and the associated equipment.

glare

A simultaneous seizure of a 2-way trunk by two communications systems resulting in a standoff.

ground-start trunk

A trunk on which, for outgoing calls, the system transmits a request for services to a distant switching system by grounding the trunk ring lead. To receive the digits of the called number, that system grounds the trunk tip lead. When the system detects this ground, the digits are sent.

holding time

A total length of time in minutes and seconds that a facility is used during a call.

ICC

Internal Call Controller (ICC) is an internal MGC that communicates with the G250 or G350 media gateways in a network.

IMS

IP Multimedia Subsystem (IMS) is an architectural framework for delivering IP multimedia services.

in-use lamp

A red light on a multiappearance telephone that lights to show which call appearance will be selected when the handset is lifted or which call appearance is active when a user is off-hook.

intelligent polling

An automatic feature of Best Service Routing (BSR) that significantly reduces the number of status polls executed. When a remote location

cannot be the best resource at a given moment in time, the intelligent

polling feature temporarily suppresses polls to that location.

intercept tone An tone that indicates a dialing error or denial of the service requested.

interflow An ACD term that refers to the ability to establish a connection to a

second ACD and overflow a call from one ACD to the other.

internal call A connection between two users within a system.

internal A BCMS measurement that is made by the system.

intraflow An ACD term that refers to the ability for calls to redirect to other splits on

the same Communication Manager to backup the primary split.

ISDN Integrated Services Digital Network (ISDN) is a communication standard

for digital transmission of voice and data in a public switched telephone

network.

ISDN Gateway (IG) A feature allowing integration of the switch and a host-based

> telemarketing application using a link to a gateway adjunct. The gateway adjunct is a 3B-based product that notifies the host-based telemarketing

application of call events.

ISDN trunk A trunk administered for use with ISDN-PRI. Also called ISDN facility.

line A transmission path between a communications system or Central Office

(CO) switching system and a telephone.

line port A piece of hardware that provides the access point to a communications

system for each circuit associated with a telephone or data terminal.

link A transmitter-receiver channel that connects two systems.

maintenance Activities involved in keeping a telecommunications system in proper

working condition: the detection and isolation of software and hardware

faults, and automatic and manual recovery from these faults.

major alarm An indication of a failure that has caused critical degradation of

service and requires immediate attention. Major alarms are automatically

displayed on LEDs on the attendant console and maintenance or alarming circuit pack, logged to the alarm log, and reported to a remote

maintenance facility, if applicable.

management A terminal that the system administrator uses to administer the switch. terminal

The administrator can also use the management terminal to gain access

to the BCMS feature.

measurement

manual-in A call-answering mode in which an agent must press manual-in to

receive an ACD call.

MAO Maximum Agent Occupancy (MAO) is a feature that Communication

Manager uses to set thresholds on the amount of time that an agent spends on a call. The MAO threshold is a system-administered value that places an agent in the AUX work mode when the agent exceeds the MAO

threshold for calls.

message center An answering service that supplies agents and stores messages for later

retrieval.

message-center

agent

A member of a message-center hunt group who takes and retrieves

messages for telephone users.

MGC Media Gateway Controller (MGC) controls the phone services on a media

gateway.

minor alarm An indication of a failure that affects customer service. Minor alarms

are automatically displayed on LEDs on the attendant console and maintenance or alarming circuit pack, sent to the alarm log, and reported

to a remote maintenance facility.

multiappearance

telephone

A telephone equipped with several call-appearance buttons for the same extension, allowing the user to handle more than one call on that same

extension at the same time.

network region A group of IP endpoints and Communication Manager IP interfaces that

are interconnected by an IP network.

Network Specific

Facility (NSF)

An information element in an ISDN-PRI message that specifies which public-network service is used. NSF applies only when Call-by-Call

Service Selection is used to access a public-network service.

node A network element that connects more than one link and routes voice

or data from one link to another. Nodes are either tandem or terminal. Tandem nodes receive and pass signals. Terminal nodes originate a transmission path or terminate a transmission path. A node is also known

as a switching system.

non switch-classified

outbound calls

Proactive Contact outbound calls that are automatically launched by

Communication Manager.

Non-Facility
Associated Signaling

(NFAS)

A method that allows multiple T1 or E1 facilities to share a single D-channel to form an ISDN-PRI. If D-channel backup is not used, one facility is configured with a D-channel, and the other facilities that share the D-channel are configured without D-channels. If D-channel backup is used, two facilities are configured to have D-channels (one D-channel

on each facility), and the other facilities that share the D-channels are configured without D-channels.

pickup group A group of individuals authorized to answer any call directed to an

extension within the group.

poll suppression An automatic feature of BSR that significantly reduces the number of

status polls executed. When a remote location cannot be the best resource at a given moment in time, the intelligent polling feature

temporarily suppresses polls to that location. See status poll on page 230.

primary extension A main extension associated with the physical telephone or data terminal.

principal A terminal that has the primary extension bridged on other terminals.

principal (user) A person to whom a telephone is assigned and who has message-center

coverage.

private network A network used exclusively for the telecommunications needs of a

particular customer.

Processor Port A port network (PN) controlled by a switch-processing element that is

directly connected to that PN's TDM bus and LAN bus.

public network A network that can be openly accessed by all customers for local and

long-distance calling.

queue An ordered sequence of calls waiting to be processed.

redirection criteria Information administered for each telephone's coverage path that

determines when an incoming call is redirected to coverage.

Redirection on No

Network (PPN)

Answer

An optional feature that redirects an unanswered ringing ACD call after an administered number of rings. The call is then redirected back to the

agent.

reorder tone A tone to signal that one of the facilities such as a trunk or a digit

transmitter, was not available.

Service Level

Maximizer (SLM)

An agent selection strategy that ensures that a defined service level of X% of calls are answered in Y seconds. When SLM is active, the software verifies that inbound calls are matched with agents in a way that makes sure that the administered service level is met. SLM is an optional Call Vectoring feature that is used with Expert Agent Selection (EAS), and

without Business Advocate.

simulated bridged

appearance

A feature with which a terminal user can bridge onto a call answered by another user. A simulated bridged appearance is a temporary bridged

appearance.

SIP Session Initiation Protocol (SIP) is an application-layer control signaling

protocol for creating, modifying, and terminating sessions with more than

one participant using http like text messages.

split (agent) status

report

A report that provides real-time status and measurement data for internally-measured agents and the split to which agents are assigned.

split condition A condition whereby a caller is temporarily separated from a connection

with an attendant. A split condition automatically occurs when the

attendant, active on a call, presses the start button.

split number An identification of the split to Communication Manager and to BCMS.

split report A report that provides historical traffic information for internally measured

splits.

Single Step Conference (SSC) is a client-side IP call recording method

that uses:

The AES DMCC service to provide the required media control by

registering standalone recording devices.

• The AES TSAPI service to provide call information, call monitoring,

and third-party call control functionality.

staffed An indication that an agent position is logged in. A staffed agent functions

in one of four work modes: auto-in, manual-in, ACW, or Aux.

status lamp A green light that shows the status of a call appearance or a feature

button by the state of the light (lit, flashing, fluttering, broken flutter, or

unlit).

status poll A call that Communication Manager makes to gain status data from a

remote place in a multisite BSR application plan.

stroke counts A method used by ACD agents to record up to nine customer-defined

events per call when a reporting adjunct is active.

switch-classified

outbound calls

Outbound calls placed by the Proactive Contact dialer and connected to

the agents.

system printer An optional printer that used to print scheduled reports using the report

scheduler.

system report A report that provides historical traffic information for internally-measured

splits.

system-status report A report that provides real-time status information for internally-measured

splits.

trunk A dedicated telecommunications channel between two communications

systems or Central Offices (COs).

trunk allocation The manner in which trunks are selected to form wideband channels.

trunk group An arrangement of communication channels that carry multiple calls for

the same phone number.

UCD-LOA Uniform Call Distribution-Least Occupied Agent (UCD-LOA) is an agent

selection method for delivery of calls under agent surplus conditions. With UCD-LOA implemented, calls are delivered to the available, that is, idle

agent with the lowest percentage of work time since login.

UCD-MIA Uniform Call Distribution-Most Idle Agent (UCD-MIA) is an agent selection

method for delivery of calls under agent surplus conditions. With UCD-MIA implemented, calls are delivered to the available agent who has been

idle the longest since the last ACD call that the agent received.

UDP Uniform Dial Plan (UDP) is a feature that allows a unique number

assignment for each terminal in multi-switch configurations, such as a Distributed Communications System (DCS) or main-satellite-tributary

system.

VDN Vector Directory Number (VDN) is an extension number that directs calls

to a vector. VDNs can represent a call type or a service category, such as

Billing or Customer Service.

vector-controlled

split

A hunt group that you can gain access to only by dialing a VDN

extension.

work mode A function that an agent performs during the work shift. ACD work modes

include AUX work, auto-in, manual-in, and ACW.

Index

A		VAL (continued)	
		externally recorded	
AAS ports		file	
AAS, administering		integrated	<u>194</u>
abandoned call search		integrated types	
acceptable service level (ASL)		integ-mus	
accessing port matrix		integ-rep	<u>192</u>
ACD screen reference		playback	
ACW agents in MIA queue		ports	
ACW timeout	<u>12</u>	recorded	<u>189</u>
add/remove skills		recording	
adding skills		stop	
screen reference		sessions	<u>195</u>
adding login IDs	<u>11</u>	source	<u>197</u>
administering		storage time	<u>194</u>
reporting adjuncts	<u>166</u>	types	
administration commands		analog	<u>190</u>
Agent LoginID	<u>11</u>	VAL	
Policy Routing Table	<u>76</u>	audio groups	<u>198</u>
Reason Code Names	<u>78</u>	vVAL	<u>198</u>
Service Hours Table	<u>79</u>	application sequencing	<u>179</u>
system parameters	<u>34</u>	ASAI	
VuStats Display Format	<u>98</u>	administering	
after call work (ACW)	<u>129</u>	conference	<u>159</u>
agent data types	<u>103</u>	release	<u>159</u>
agent disconnect tones	<u>126</u>	transfer	<u>159</u>
agent login IDs, creating	<u>129</u>	ASAI links	152
agent mobility		ASAI routing	<mark>.27</mark>
agent skills		ASL algorithm	
adding	129	administering	169
changing		assigning a BCMS login ID	
deleting	129	assigning AUDIX name	
viewing		AT&T In-Band Transfer Connect	
agent TN and skill TN, matching		attendant vectoring	
agent-level administration		audio group algorithm	
allowed with exclusion		AUDIX extension	
service observing	49	AUDIX name	81, 1 <mark>85</mark>
single step conference (SSC)		AUDIX ports	
analog announcements		auto available split/skill (AAS)	
ANI vector routing		auto-answer	
announcement	_	auto-answer agents	
analog line types		auto-answer field preference	
analog-fd	190	AUX work reason code	
analog-m		Avaya support website	
auxiliary trunk types	<u></u>	,, a sappert necessary	
aux-trk-m	191	_	
aux-trunk		В	
circuit packs		hanna in	400
deleting		barge-in	
DS1 types	<u>101</u>	BCMS service level	
ds1–fd	101	BCMS-measured split or skill	
ds1–ops		BCMS, administering	
ds1–opsds1–ops		bearer capability class (BCC)	<u>81,</u> <u>185</u>
us I—sa	<u>131</u>	best service routing	

polling (continued)	collection (continued)	
polling	generating PDF	<u>215</u>
administering <u>134</u>	sharing content	<u>215</u>
without B-channel 134	conference tone on service observing	<u>49</u>
singlesite <u>131</u>	configuring agent mobility	
BSR agent selection <u>85</u>	configuring EC500-enabled mobile agents	<u>126</u>
BSR local treatment <u>133</u>	connecting calls to announcements	<u>198</u>
BSR tie strategy <u>133</u>	console permission	<u>124</u>
Business Advocate screens <u>130</u>	console permissions	<u>121</u> , <u>165</u>
	content	
C	publishing PDF output	<u>215</u>
U	searching	<u>215</u>
call handling <u>124</u>	sharing	<u>215</u>
call handling preference	sort by last updated	<u>215</u>
call handling time	watching for updates	<u>215</u>
call origin85, 174	controlling adjunct	<u>56</u>
call prompting27	converse data delay	<u>175</u>
screen reference	converse data return	<u>175</u>
call queuing time	converse signaling	<u>175</u>
call routing	COR commands	<u>30</u>
dedicated179	COR field descriptions	30
call selection measurement 129, 130	COR screen	30
call selection method	coverage path	<u>56,</u> 174
greatest need	coverage path for login ID	
percent allocation 130	CPN prefixes	
skill level	CPN, sending	
call vectoring		
administering	D	
call warning threshold	D	
call work code (CWC)	data passing	175
caller disconnect tones	dedicated trunk groups	
caller information	destination node	
calling part number, sending	destination port	
calling party number (CPN)	direct agent announcements	
callmaster phones	direct agent call preference	
callr-info, administering	direct agent calling	
calls warning threshold <u>56</u>	administering	
cer link	disconnect tones	
cor links41	displaying VDN name	
central office (CO)	distributed communications services (DCS)	
changing agent skills	document changes	
console permissions	documentation center	
COR number	finding content	
changing login IDs11	navigation	
changing skills	documentation portal	
class of restriction (COR)30, 56	finding content	
clear caller information	navigation	
leave-ACW39	duplicate VDNs	
next-call	duplicate vectors, creating	
on-call-release39	dynamic percentage adjustment	
CM interfaces	dynamic queue position (DQP)	
CM screens	dynamic threshold adjustment	
Signaling Group	Ly.Lao an control adjustment	<u>oc</u>
coach on single step conference (SSC)	_	
collection	E	
delete215	TAC administration	4.40
edit name	EAS, administering	
<u>210</u>	enhanced callr-info	<u>125</u>

Index

enhanced vectoring	incoming call identification (ICI)	<u>39</u>
event counts	InSite Knowledge Base	<u>217</u>
expected call handling time <u>56</u>	integrated announcements	<u>194</u>
extension number manipulation, preventing 179	intercept tone	<u>122, 123</u>
	intercom calls	
F	interface channel	166
F	interface link	166
field descriptions	interflow and intraflow	
field descriptions	screen reference	153
Call Management System	interflow VDN at remote locations	
Hunt Group56	interruptible aux	
fields related to EAS	deactivation	152
finding content on documentation center	notification	
finding port matrix	screen reference	
forced agent aux work	threshold	
location	INVITE message	
administering, prerequisites	S .	
skill	ISDN trunks, NCR	
administering, prerequisites <u>149</u>	IVR ports	<u>183</u>
forced agent logout <u>150</u>		
ACW	L	
administering, prerequisites		
clock time	least occupied agent (LOA)	
administering, prerequisites <u>149</u>	leave word calling (LWC)	<u>12</u>
location	listen port numbers	<u>179</u>
administering, prerequisites	local agent preference	153
skill	local audio source	
administering, prerequisites	local call preference	The second secon
forced logout	local session	
foreign eXchange (FX)	local VDN administration	
	locally sourced announcements	
	location preference	
G	location preference distribution	
	logging agents out	
group extension	login ID characters	
group type	logout reason code	
ead-loa <u>130</u>	look-ahead interflow	<u>12</u>
ead-mia <u>130</u>	screen reference	15/
pad <u>130</u>	LWC messages, storing	
ucd-loa <u>130</u>	LVVC messages, storing	<u>12</u>
ucd-mia <u>130</u>		
group types, administering <u>129</u>	M	
	MAO, administering	155
H	master vector	
	matching tenant numbers	
holiday table	maximum agent occupancy (MAO)	
administration commands <u>54</u>	maximum logged-in SIP agents	
Holiday Vectoring administration <u>54</u>		
Hunt Group administration <u>55</u>	maximum time in ACW	
hunt groups <u>129</u>	MCH, farrad	
hunt-group	MCH, forced	
commands <u>55</u>	MCH, on request	<u>156</u>
duplicate <u>55</u>	message	. = -
	INVITE	
1	messaging server adjunct (MSA)	
I	MIA Across Skills	
ICI tones	mis link	
inbound call management (ICM) 152	mis links	the state of the s
inbound can management (row) <u>102</u>	multimedia call handling	<u>27</u>

multiple locations		R	
multiple observers	<u>170</u>		
multisite BSR		reason code options	<u>12</u>
screen reference		reason codes	
multisite call center	<u>205</u>	screen reference	
music		reason codes, assigning names	<u>78</u>
locally sourced	<u>198</u>	recorded announcements	<u>189</u> , <u>192</u>
music on hold (MOH)	<u>199</u>	recording announcements	<u>194</u>
MWI Served User Type	<u>12</u>	recording device	<u>170</u>
My Docs	<u>215</u>	redirect on OPTIM failure (ROOF)	<u>181</u>
		redirection	
N		IP failure	<u>162</u>
IN .		no answer	<u>163</u>
NCR for ISDN	156	OPTIM failure	<u>164</u>
NCR for SIP		relaying UCIDs	<u>171</u>
NCR, administering		remote location VDNs, accessing	<u>25</u>
network call redirection (NCR)		remote logout	<u>165</u>
network redirection		remote logout FAC	<u>165</u>
non predictive calls		remote session	<u>166</u>
non-predictive calls		remote VDN administration	<u>133</u>
nonbarge-in		removing skills	121, 123
NTP or SNTP	<u>190</u>	reorder tone	
internet time servers	210	reporting adjunct	
litterilet tille servers	<u>210</u>	adding nodes	165
		administering	
0		reporting adjunct compatibilities	
		reporting adjunct interfaces	
off-pbx FNE administration commands		reporting adjuncts	
off-pbx FNE screen		administering	
office hours, administering	<u>79</u>	sending data	
offset times for CMS reporting		reporting issues, preventing	
setting up		reserve agents, assigning	
outboard proxy SIP (OPS)	<u>178</u>	reserve levels	
		return destination	
P		ringback duration	
•		ringer-off	
passing prefixed CPN	<u>155</u>	ROIF	
percent allocation	<u>12</u>	screen reference	162
policy routing table (PRT)	<u>160</u>	RONA	
poll suppression	<u>25</u>	screen reference	163
port matrix	<u>214</u>	ROOF	
oredictive calls	<u>167</u>	screen reference	164
Proactive Contact calls, reporting	<u>167</u>		
Proactive Contact, reporting	<u>167</u>	•	
proc channel	<u>166</u>	S	
PRT administration		ashaduling aurahranization tasks	207
		scheduling synchronization tasks	<u>207</u>
^		screen reference	44
Q		Agent LoginID	
q-calls button	160	BCMS	
q-time button		Business Advocate	
queue limit		VuStats	
queue status indications	<u>100</u>	searching for content	
screen reference	160	security codes	
301551115151105	<u>100</u>	sending calling party number	
		sending direct agent calls	
		sending name on ISDN trunk group	
		service hours table	

Index

service hours table (continued)		storing LWC messages	<u>12</u>
administering	<u>168</u>	support	<u>217</u>
service level maximizer (SLM)	<u>168</u>	suppress polling	<u>25</u>
service level supervisor (SLS)	<u>129</u>	switch node, assigning IDs	<u>25</u>
service level targets	<u>39</u> , <u>168</u>	switch processor element (SPE)	<u>12</u>
service objective (SO)	<u>129</u>	system-level administration	<u>130</u>
service observer	<u>170</u>	systemwide feature administration	<u>3</u> 4
service observing	<u>49</u>		
multiple observers	<u>170</u>	Т	
screen reference	<u>169</u>	•	
service observing permissions	<u>30</u>	tenant number (TN)	56. 85
Session Manager		tenant partitions	
entities		time adjustments	
profile configuration		time of day	
session refresh interval	<u>181</u>	location offset values	206
sharing content	<u>215</u>	set up	
singlesite BSR		synchronization	
administering	<u>131</u>	considerations	208
SIP		methods	
adding trunks	<u>179</u>	prerequisites	
dedicated trunk groups	<u>179</u>	time of day (TOD)	
SIP phone buttons		time warning threshold	
after-call	<u>178</u>	time zone offset	
agnt-login		timed ACW, administering	
aut-msg-wt	<u>178</u>	tracking outbound calls	
auto-in		transmitting UCIDs	
aux-work		Trunk Group data types	
coach		two observers	
manual-in			
q-calls	<u>178</u>	III	
stroke-cnt	<u>178</u>	U	
uui-info		UCID, administering	170
vu-display		UCID, sending	
work-code		uniform dial plan (UDP)	
SIP phones, administering		universal call identification (UCID)	170
SIP traffic		measured trunks	
SIP trunks, NCR		unmeasured trunk groups	
skill access codes		user-to-user information (UUI)	
skill levels, assigning	<u>12</u>	using extensions as AAS ports	
skill priority		UUI	<u>12</u>
skill-level administration		transport	
skill, assigning	<u>12</u>	ISDN trunks	172
skills, changing	<u>124</u>	SIP trunks	
skills, removing		UUI treatment	
SLM, administering		uui-info	
sort documents by last updated		uui-info feature button	
SPI links		dul-inio leature button	<u>173</u>
split or skill data types	<u>112</u>		
station		V	
administering		\(\mathrea{\tau}\)	
conference		VDN	
release		coverage path	
transfer		field descriptions	
station administration	<u>81</u>	holiday vectoring	
station ports	<u>183</u>	origin announcement	
Station screen, administering		override	
status poll VDN	<u>25</u>	return destination	<u>85</u> , <u>17</u> 4

VDN (continuea)
time zone <u>85</u>
time zone offset
variables <u>85</u>
VDN administration85
VDN commands85
VDN data types <u>118</u>
VDN in a coverage path (VICP)
VDN of origin announcement (VOA)
vector names, assigning33
vector variables <u>27</u>
videos
viewing login IDs12
VOA
voice response integration (VRI)
VRD
VRI
screen reference
vrt administration
VRU port
tones
administering <u>183</u>
VRU ports
VuStats
data type split <u>116</u>
trunk group
VDN
screen reference
VuStats button
VuStats data types <u>103</u> , <u>112</u> , <u>117</u> , <u>118</u>
W
warning tone on service observing
watch list
wide area telecommunications service (WATS)121
Z
L
zip tones
Zip torios <u>59, 177</u>