Administering Avaya Aura® Call Center Elite

Release 7.1.x
Issue 3
June 2018

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Contents

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.

New in this release

New features for Avaya Aura® Call Center Elite 7.1:

- Agent Mobility integrates with Avaya Extension to Cellular (EC500) enabling Expert Agent Selection (EAS) agents to function while outside the corporate network.
- Administrators can ensure that when agents log in to Call Center Elite, they are automatically logged into the available work mode instead of the aux work mode.
- Support for treating AUX work mode as idle for controlling the LOA queue by adding the **AUX Agent Remains in LOA Queue** field to the Agent LoginID and Feature-Related System Parameters screens.
- Supported length of vector names increased from 15 characters to 27 characters.
- New station button, **VDN-INFO**, added on Communication Manager for H.323 and DCP phones. When users press the **VDN-INFO** button, Communication Manager displays the complete VDN name of the active call.

New features for Avaya Aura® Call Center Elite 7.0.1:

- Support for treating AUX work mode as idle for controlling the MIA queue by adding the **AUX Agent Considered Idle (MIA)** field to the Agent LoginID and Feature-Related System Parameters screens.
- Support to allow ASAI Single Step Conference calls to ignore **Exclusion by adding the Allowed with Exclusion: Service Observing** field to the Feature-Related System Parameters screen.
- Support for treating adjunct routed calls as ACD calls by adding an the **Report Adjunct Calls as ACD** associated field to the **Measured** field to the Feature-Related System Parameters screen.

- Addition of the **sip-sobsrv** feature button with the associated **Listen-Only?** and **Coach** fields in case of 96X1 SIP agent deskphones.

Document changes since last issue

Issue	Date	Summary of changes
1	May 2017	Initial issue for Release 7.1 of Administering Avaya Aura® Call Center Elite.
2	October 2017	Second issue for Release 7.1 of Administering Avaya Aura® Call Center Elite. Updated the BSR Local Treatment on page 125 topic to remove an incorrect Call Center release information.
3	June 2018	Third issue for Release 7.1.x of Administering Avaya Aura® Call Center Elite. Updated the Call Management System field descriptions on page 39 topic to remove incorrect Avaya IQ release information in the UI.

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.

<table>
<tr><th>Command name</th><th>Parameter</th><th>Qualifier</th></tr>
<tr><td><code>add</code></td><td>agent-loginID</td><td>xxxxx or next</td></tr>
<tr><td><code>change</code></td><td>agent-loginID</td><td>xxxx [auto]</td></tr>
<tr><td><code>duplicate</code></td><td>agent-loginID</td><td>xxxxx [start nnn] [count x]</td></tr>
<tr><td><code>remove</code></td><td>agent-loginID</td><td>xxxxx</td></tr>
<tr><td><code>list</code></td><td>agent-loginID</td><td>staffed [name] [AAS y or n]
unstaffed [name] [AAS y or n]</td></tr>
<tr><td colspan="3">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: <code>Agents must log in again before non-skill changes take effect.</code>
• If you do not use auto, you will see the following warning message: <code>Agents must log in again for the changes to take effect.</code></td></tr>
<tr><td colspan="3">xxxx is the extension.</td></tr>
<tr><td colspan="3">nnn is the extension number of the first duplicate agent login ID in the number sequence.</td></tr>
<tr><td colspan="3">[ ] indicates that the qualifier is optional.</td></tr>
</table>

Agent LoginID field descriptions

Field title	Field 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.
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.

Table continues…

Field title	Field description
	• **aux**: Agents log in to Call Center Elite in the aux work mode and are unavailable for an ACD call. **Note:** Call Center Elite release must be 7.0 or later and Communication Manager release must be 7.1 or later.
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. **Note:** Call Center Elite release must be 7.0 or later and Communication Manager release must be 7.1 or later.
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.

Table continues…

Field title	Field description
	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. **Note:** Call Center Elite release must be 7.0 or later.
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.
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.

Table continues…

Field title	Field description
	• **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.
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**.

Table continues…

Field title	Field description
	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`.
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 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**.

Table continues…

Field title	Field description
	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.
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 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.

Table continues…

Field title	Field 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.
Name	To assign a name of up to a 27 characters. Any alphanumeric character is valid. You can also leave this field blank.
Percent Allocation	To assign a number from `1` to `100` for each skill. All entries must add up to 100 percent.
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.
Port Extension	To type the assigned extension for an AAS or AUDIX port. This extension cannot be a VDN or an agent login ID.

Table continues…

Field title	Field description
	The field is applicable if you enter **y** in the **AAS** field or the **AUDIX** field.
Reserve Level (RL)	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.
Skill Number (SN)	To assign skills to the agent login ID. Do not enter the same skill twice.

Table continues…

Field title	Field description
	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.
Skill Level (SL)	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.
TN	To assign a Tenant Partition number. Valid entries are from `1` to `250`.

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

Field title	Field 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.
Skl/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.

Table continues…

Field title	Field description
	• 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
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.
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**.

Table continues…

Field title	Field description
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 Poll 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.
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.

Table continues…

Field title	Field description
	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.
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.

Table continues…

Field title	Field description
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.
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.

Table continues…

Field title	Field description
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**.
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**.

Table continues…

Field title	Field description
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 **y** and **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 number of the vector to be duplicated or to be used as a template.

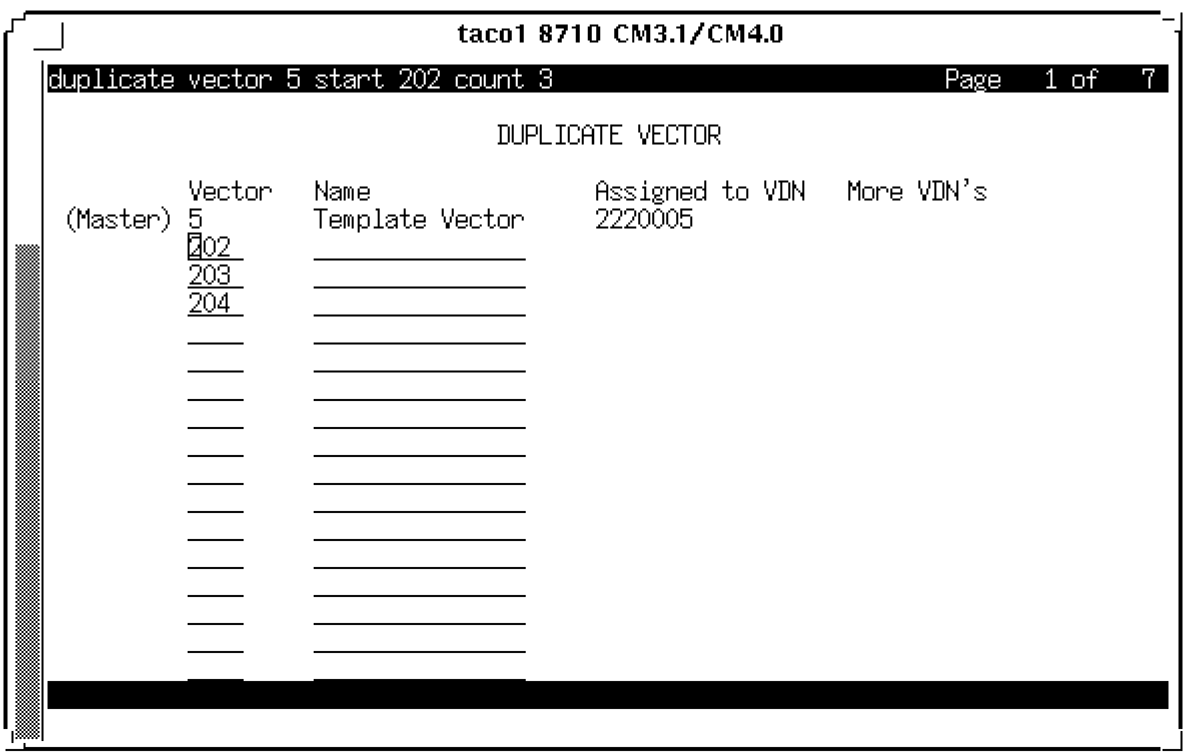
[start nnnn] — Specifies the first vector number to be used as a duplicate. The parameter is optional. If you do not specify a start number, the first available vector after the master vector number is selected. Only one vector is selected.

[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.
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.

Table continues…

Field title	Field description
	• **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).
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.
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-

Table continues…

Field title	Field description
	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.

Table continues…

Field title	Field description
	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

Field title	Field 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.

Table continues…

Field title	Field description
	• **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.
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

Table continues…

Field title	Field description
	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 Contactsoft dialer places through Adjunct Switch Application Interface (ASAI). Valid entries are **y** and **n**.
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**. Valid entries 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 *open mike* 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 (determines protocol used by 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, that is, 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.
CMS (appl mis)	The option to select the release of Call Management System (CMS) to which you are connecting. The options are: • **R15/R16**: These value apply to CMS R15 and R16 CMS R16 and later with Communication Manager 5.2.1 and later support full Expanded Dial Plan (EDP) without Special Application (SA). • **R16.1/R16.x/R17.0**: These value apply to CMS R16.1, R16.2, R16.3, or R17.0 • **R18**: This value applies to CMS R18.0.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 field option is the default.
IQ (appl ccr)	The option to select a release of Avaya IQ. The options are: • 5.0

Table continues…

Name	Description
	• 5.1/5.2: These values 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+: This value applies to 5.2.6 and 5.3.x 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 field 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
R15/R16	**5.0**	22
R16.1/R16.x/R17.0	**5.1/5.2**	23
R18	**5.2.6+** **Note:** Selecting R5.2.6+ applies to R5.2.6 or R5.3.x.	24

To use other combinations of CMS and Avaya IQ releases, select the lowest release of one reporting adjunct. For example, 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.

Note:

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.
	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.
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.

Table continues…

Field title	Field description
	• **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.

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.

Table continues…

Field title	Field description
	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

Table continues…

Field title	Field description
	the System-Parameters Customer-Options screen are active for the system.
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**.

Table continues…

Field title	Field description
	• `0-9`: If you administer the **Two-Digit Aux Work Reason Code** field as **n**.
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

Warning:

The use of Service Observing features is subject to federal, state, or local laws, rules or regulations, or requires the consent of one or both the parties in a conversation. You must comply with all the applicable laws, rules, and regulations before using Service Observing features.

Field title	Field description
Allow Two Observers in Same Call	To determine whether two observers can monitor the same call. Valid entries are **y** and **n**. 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	To play a warning tone that phone users and calling parties can hear when a supervisor monitors calls. Valid entries are **y** and **n**. The field is applicable if the **Service Observing (Basic)** field on the System-Parameters Customer-Options screen is active for the system.
Service Observing: Conference Tone	To play a conference tone that the phone users and calling parties can hear when a supervisor monitors calls. Valid entries are **y** and **n**. The field is applicable if the **Service Observing (Basic)** field on the System-Parameters Customer-Options screen is active for the system. This field is inapplicable if you administer a warning tone for Service Observing.

Table continues…

Field title	Field description
Allowed with Exclusion: Service Observing	To determine whether multi-appearance phone users can prevent other service observers from bridging on to an existing call. Valid entries are **y** and **n**. 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. This field option is the default.
Allowed with Exclusion: SSC	To determine whether multi-appearance phone users can prevent ASAI Single Step Conference parties from bridging on to an existing call. Valid entries are **i**, **b**, and **n**. If you select **i**, Communication Manager allows invisible SSC parties a connection towards a station with Exclusion active, either by Class of Service (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 Class of Service (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. This field option is the default.

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 `y` and `n`. The default value is n.

Table continues…

Field title	Field description
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.

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.

Table continues…

Field title	Field description
	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.
Converse Signaling Pause	To determine the delay period in milliseconds (msec) between the digits being passed. The

Table continues…

Field title	Field description
	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.

Table continues…

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
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 qualifier is optional.		

duplicate hunt-group
Use the command to create up to 16 duplicate hunt-group screens.

master_grp
Is the assigned hunt group number, which is up to four digits, of the hunt 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.

Table continues…

Field title	Field description
	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.
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.
Controlling Adjunct	To determine the type of adjunct processor that controls the members of the split or skill or the hunt group. • **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.

Table continues…

Field title	Field description
	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. 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 *x* designation if an extension is used.
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.

Table continues…

Field title	Field description
	• `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**
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)**

Table continues…

Field title	Field description
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.
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**.

Table continues…

Field title	Field description
	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. **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.
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.

Table continues…

Field title	Field description
	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 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 Managerredirects 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**

Table continues…

Field title	Field description
	• **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`.
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)	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.
Level 2 Threshold (sec)	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.
LOA Increased Agts in Skill	To support having a maximum of 7,000 agents logged in the same skill.

Table continues…

Field title	Field description
	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.
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).

Table continues…

Field title	Field description
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**.
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.

Table continues…

Field title	Field description
	• **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.
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

Table continues…

Field title	Field description
	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.
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

Table continues…

Field title	Field description
	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**.
(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.

Table continues…

Field title	Field description
	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.
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**.

Table continues…

Field title	Field description
	- 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.
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 **y** and **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.

Table continues…

Field title	Field description
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.
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.

Table continues…

Field title	Field description
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**.
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 *Avaya Aura® 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.
Type	To specify the type of algorithm that the policy routing table supports. The valid entry in this field is percentage.

Table continues…

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 *name not assigned* 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.

Table continues…

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

Table continues…

Field title	Field description
	• 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

Table continues…

Field title	Field description
	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.

Table continues…

Field title	Field description
	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.`
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.
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

Table continues…

Field title	Field description
	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.
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.

Table continues…

Field title	Field description
	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.

Command name	Parameter	Qualifier
`add`	vdn	xxxxx or next, where xxxx is the extension number of the VDN to be added.
`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]
Square brackets [ ] indicate that the qualifier is optional. Single quotes (‘ ’) indicate that the text inside the quote must be entered exactly as shown or an abbreviated screen of the word can be entered. MAX is the maximum number available in your system configuration.		

`duplicate vdn`
Use the command to create up to 16 duplicate VDNs.

`master_ext`
Is the assigned extension number, which can be up to 13 digits, of the VDN 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 VDN. If you do not specify a start number, the first available VDN after the master VDN is selected automatically. Only one VDN 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 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 **2nd Skill** **3rd Skill**	To assign a skill number. Valid entries are from `1` to `8000`. You can leave this field blank. 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 **y** and **n**. The default is **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

Table continues…

Field title	Field description
	Note: The field follows VDN Override Rules.
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.
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.

Table continues…

Field title	Field description
	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.
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.

Table continues…

Field title	Field description
COR	To assign a Class of Restriction (COR) to the VDN. Valid entries are from `0` to `995`. You cannot leave this field blank.
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. **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`**

Table continues…

Field title	Field description
	• `route-to number` or `route-to digits` **Note:** The field follows VDN Override Rules.
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 Adjunct Calls as ACD**	To collect measurement data for the VDN. BCMS, Call Management System (CMS), or Avaya IQ collect data for reporting purpose. 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.

Table continues…

Field title	Field description
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.
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 Associated field: **PC Predictive Reports Skill**	To assign the VDN for Proactive Outreach Manager (POM) or Proactive Contact (PC) outbound calling. Valid entries are **y** and **n**. 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 **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.

Table continues…

Field title	Field description
	You can leave this field blank.
	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.
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

Table continues…

Field title	Field description
	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.
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 *active* 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

Table continues…

Field title	Field description
	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 Held Call Drops**	To assign an After Call Work (ACW) interval for auto-in agents who receive calls from this VDN. 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.
	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.

Table continues…

Field title	Field description
	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 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.
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.

Table continues…

Field title	Field description
	• **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**.
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.

Table continues…

Field title	Field description
	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.
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.
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.

Table continues…

Field title	Field description
	• **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.	—	—

Table continues…

VuStats data type	Description	BCMS report	Field or column heading in the report
elapsed-time-in-state	The duration that a staffed agent was in the current state.	—	—
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.	—	—

Table continues…

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. The system identifies calls for each hunt group and begins tracking calls when calls enter a hunt group queue.	Split Status	Acceptable Service Level
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.	—	—
split-agents-available	The number of staffed agents currently available to receive ACD calls to a split or skill. This number	Split Status	Avail

Table continues…

VuStats data type	Description	BCMS report	Field or column heading in the report
	includes agents in the auto-in or manual-in work mode.		
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.	—	—
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.	—	—

Table continues…

VuStats data type	Description	BCMS report	Field or column heading in the report
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. 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.	System Status	AVG AFTER CALL
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.	—	—
split-calls-abandoned	The number of calls that left the queue or abandoned ringing, provided the	System Status/Split/ Split Summary	ABAND CALLS

Table continues…

VuStats data type	Description	BCMS report	Field or column heading in the report
	queue is the first split or skill that the call queued to.		
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
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

Table continues…

VuStats data type	Description	BCMS report	Field or column heading in the report
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. With EAS, all non-call-related ACW time is associated with the first skill that the agent logged in to.	Agent/Agent Summary	TOTAL AFTER CALL
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. 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	Agent/Agent Summary	TOTAL AUX/OTHER
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

Table continues…

VuStats data type	Format	Period	Threshold	Reference
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
split-agents-available	–	–	allowed	required
split-agents-in-after-call	–	–	allowed	required

Table continues…

VuStats data type	Format	Period	Threshold	Reference
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
split-total-aux-time	required	required	allowed	required
time-agent-entered-state	–	–	–	required

Table continues…

VuStats data type	Format	Period	Threshold	Reference
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. The system identifies this data type for each hunt group and begins tracking the data type when a call enters a vector.	Split Status/Split	Acceptable Service Level
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.	—	—
agents-in-aux-3	The number of agents currently in AUX work for reason code 3 for the referenced skill.	—	—

Table continues…

VuStats data type	Description	BCMS report	Field or column heading in the report
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
agents-on-extension-calls	The number of agents in a specific split who are currently on non-ACD calls.	Split Status	Extn

Table continues…

VuStats data type	Description	BCMS report	Field or column heading in the report
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. The system does not include calls that were interflowed from a remote server by the Look Ahead Interflow (LAI) feature.	Split Report/Split Summary	FLOW IN
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	Split Report/Split Summary	FLOW OUT

Table continues…

VuStats data type	Description	BCMS report	Field or column heading in the report
	queued to this split as a primary split and were answered or abandoned from ringing in another split.		
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. The system begins tracking when a call enters a split or skill.	System Status	OLDEST CALL
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

Table continues…

VuStats data type	Format	Period	Threshold
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	–	–	–
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. The system begins tracking when the last trunk is seized.	Trunk Group	% ALL BUSY
percent-trunks-maint-busy	The percent of time trunks were busied out for maintenance during a specified period or day.	Trunk Group	% TIME MAINT
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. The system identifies this data type for each VDN and begins tracking the data type when a call enters a vector.	VDN Status/VDN	Acceptable Service Level
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

Table continues…

VuStats data type	Description	BCMS report	Field or column heading in the report
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
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	–	–

Table continues…

VuStats data type	Format	Period	Threshold
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 **Enter**.
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 **as-needed**.
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 **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	
Agent LoginID	**Call Handling Preference** • **greatest-need** • **percent-allocation** • **skill-level**
	Direct Agent Calls First
	Percent Allocation (PA)
	Reserve Level (RL)
	Service Objective
Skill-level administration	
Hunt Group	**Activate on Oldest Call Waiting**
	Dynamic Percentage Adjustment

Table continues…

Screen title	Field title
	Dynamic Queue Position
	Dynamic Threshold Adjustment
	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	
Feature-Related System Parameters	**ACW Agents Considered Idle**
	AUX Agent Remains in LOA Queue
	AUX Agents Considered Idle (MIA)
	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) Settings in the fields Codeset to Send TCM, Lookahead and Send Codeset 6/7 LAI IE on the ISDN trunk screen do not affect BSR.	• **Outgoing Display** • **Supplementary Service Protocol** • **UUI Treatment**

Table continues…

Screen title	Field title
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.
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

Look-Ahead Interflow on page 146
UUI Treatment for ISDN trunks on page 163
UUI Treatment for SIP trunks on page 164

BSR Local Treatment

Before you begin

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

1. 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.  Inserted                             DCS/ IXC
    No          Mrk Lmt List Del  Digits                               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 Format
                                                  Subaddress
 1: y y y y y n
y
  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.

   Important:

   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

                                SIGNALING GROUP

 Group Number: 32             Group Type: h.323

                           Remote Office? n
Max number of NCA TSC: 10

                                                      Max number of CA TSC: 10

                                                   Trunk Group for NCA TSC: 32

       Trunk Group for Channel Selection: 32

        Supplementary Service Protocol: b
        Network Call Transfer? n

        Near-end Node Name: clan-01D12       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`.

```
display trunk-group 32                                            Page   2 of  22
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                 Internal Alert? n        Maintenance Tests? y
                               Data Restriction? n
NCA-TSC Trunk Member: 1

                                      Send Name: y      Send Calling Number: y
            Used for DCS? n             Hop Dgt? n
   Suppress # Outpulsing? n    Numbering Format: public
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider
Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                                      Send Connected Number: n
               Send UCID? y
 Send Codeset 6/7 LAI IE? n
 Path Replacement with Retention? n
 Path Replacement Method: better-route
                            Network (Japan) Needs Connect Before Disconnect? n
```

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? n
                        Path Replacement with Measurements? y
                       QSIG Path Replacement Extension: 2999
                       Path Replace While in Queue/Vectoring? y
```

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 queue 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-Options	• **Vectoring (Basic)** • **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 Parameters	• **Vector Disconnect Timer** • **Music/Tone on Hold** • **Port** • **Music (or Silence) on Transferred Trunk Calls**
Vector Directory Number	Complete all fields.
Announcements/Audio Sources	Complete all fields for each extension that provides a vectoring announcement.
Hunt Group	• **ACD** • **Vector**

Table continues…

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 LoginID with EAS	• **Supervisor Extension** • **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 13 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

Parameters	Agent loginID
Abbreviated Dialing (AD) buttons	
7103A	Accepts
Enhanced	Accepts
Group	Accepts
Personal	Accepts
System	Accepts
Agent-LoginID	
Port extension	Rejects
Announcements	Rejects
Buttons	
abrdg_app	Rejects
aut-msg-wt	Accepts
brdg_app	Rejects
busy-ind	Yes
data_ext	Rejects
man_msg_wt	Rejects
q-calls	Rejects
q-time	Rejects
signal	Rejects
Call processing	
Auto callback	Rejects
Call forward from agent login ID	Rejects
Call forward to agent login ID	Accepts
Call park	Accepts
Hundreds group	Rejects
LWC retriever gets last msgs	Accepts
Service observe agent login ID	Accepts
Call Detail Recording (CDR)	
Primary extension	Rejects
Secondary extension	Rejects
Code calling	Accepts
Communication Link	

Table continues…

Parameters	Agent loginID
Communication link digits	Rejects
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

Table continues…

Parameters	Agent loginID
Night service	Rejects
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)**

Procedure

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 vector-initiated direct agent call. The treatment also applies to an EAS agent in the coverage path of the initial EAS agent who receives the call.

Procedure

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**.
2. 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

Procedure

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 **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.

Procedure

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.	1. 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. 2. 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 **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 Parameters	• **Interruptible Aux Deactivation Threshold (%)** • **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 **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-Options	• **ISDN-PRI** • **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-Numbering	• **CPN Prefix** • **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

Procedure

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

Important:

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**	**y**

Table continues…

Administration command	Field title	Field option
DS1 Circuit Pack		
`change ds1 [board location]` Board location parameter values are: [cabinet(1-1)];carrier(A-E);slot(0-20) OR [gateway(1-10)];module(V1-V9)	**Country Protocol**	• **1a** or any field option for MCI Network Call Transfer (NCT). • **1b/1d** for Two Bearer Channel Transfer (TBCT) • **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**	**y**
	Group Type	**ISDN** for ETSI ECT, MCI NCT, NCD, and TBCT
	Network Call Redirection	**y**
	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 This field option is applicable if you administer the **Service Type** field as a Call-by-Call service, that is, the **cbc** field option.	**y**

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 screen		
`change best-service-routing xxx`	**Net Redir**	**y**
Trunk Group screen		
`change trunk-group xxx`	**Direction**	**two-way**
	Group Type	**SIP**
	Network Call Redirection	**y**

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 Connect℠ service.

Administration command	Field title	Field option
Best Service Routing Application screen		
`change best-service-routing xxx`	**Net Redir** You must administer this field if BSR is active for the system.	**n**
DS1 Circuit Pack		
`change ds1 [board location]` Board location parameter values are: [cabinet(1-1)]; carrier(A-E); slot(0-20) OR [gateway(1-10)]; module(V1-V9)	**Country Protocol**	**1b/1d**
Signaling Group screen		

Table continues…

Administration command	Field title	Field option
`change signaling group xxx`	**Network Call Transfer**	**y**
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**.

Procedure

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 dialing-plan 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**
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**
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**

Table continues…

Screen title	Field title
	• **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**
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-alrt**

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**

Table continues…

Screen title	Field title
	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

If you are using Call Center Release 4.0 or later, 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. Configurations with both reporting adjuncts are supported for all compatible combinations with a particular Communication Manager release starting with Communication Manager R4.0, Call Management System R13.1, and R14 Call Management System and **Avaya IQ R4.0.** 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

Procedure

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

Ensure that the field option in the **Measured** field on the following screens is **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. In the **CMS (appl mis)** and **IQ (appl ccr)** fields on page 12 of the Feature-Related System Parameters screen, select the reporting adjunct release.

CMS (appl mis) field options	IQ (appl ccr) field options	Switch Protocol Interpreter (SPI) language supported
R15/R16	**5.0**	22
R16.1/R16.x/R17.0	**5.1/5.2**	23
R18	**5.2.7+**	24

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

Next steps

Administer the fields on the Processor Channel Assignment screen.

Processor Channel Assignment

Procedure

1. At the command prompt, type `change communication-interface processor-channels` 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

Procedure

1. 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 **y**.
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**.

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 **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.
Feature-Related System Parameters	• **Expert Agent Selection (EAS) Enabled** • **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 **y**.
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**.

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 **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

Procedure

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**

Table continues…

Screen title	Field title
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 **y**.

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.

Note:

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 170.

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:

Procedure

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

Procedure

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

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 **Note:** DS1 circuit packs (TN767E or later or TN464F or later) must be equipped.	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 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.
Coverage Msg Retrieval	To determine whether station users at the coverage point can retrieve Leave Word Calling (LWC) messages for the station from where

Table continues…

Field title	Field description
	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.
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.

Table continues…

Field title	Field description
	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**.

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) Netcon Data Module System Port Data Module (SAP) Announcement Data Module	Administer all the fields.
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**
Hunt Group	• **First Announcement Extension**

Table continues…

Screen title	Field title
	• **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 **Q** field is set to `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 **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 source	A voice announcement with a TN2501AP board.
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 192

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 191

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. Depending on the Communication Manager version, one of the following results occur:

- If the receiving Communication Manager is installed with R11 or later, Communication Manager checks the DST rule specified in the **`set time`** command, which is always Standard Time, and compares the DST rule with the existing rule. Communication Manager adjusts the incoming synchronization time if a DST rule other than Standard Time is in effect.
- If the receiving Communication Manager is installed with a release prior to R11, 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 resends 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 perform 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.

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 Communication Manager. However, synchronization tasks actually run in sequence. Actual synchronization run times at Communication Manager may vary 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.

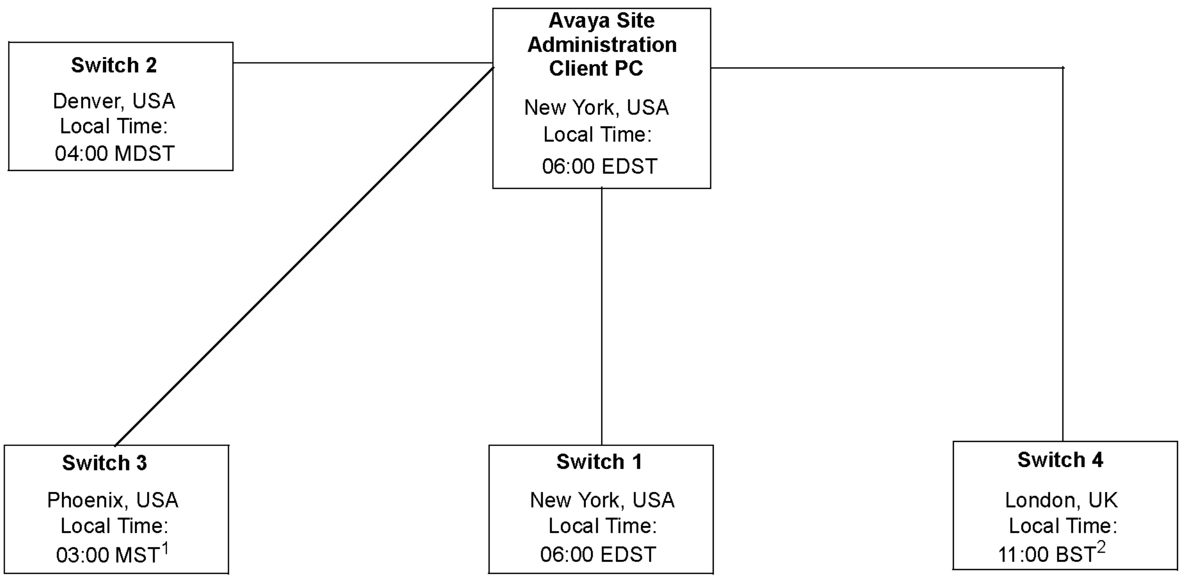
4. Select **OK**.

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 their 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 (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 Client computer in New York	Communication Manager 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:

      a. Determine the local time at Communication Manager when the synchronization runs. If necessary, adjust the local time to the Standard Time equivalent.
      b. 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.
      c. 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 Setup**.

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

See the following related documents.

Title	Use this document to:	Audience
Planning		
Planning for an Avaya Aura® Call Center Elite Implementation	Transition a basic call center environment to a Call Vectoring and Expert Agent Select (EAS) environment.	Implementation engineers and sales engineers
Supporting		
Avaya Aura® Call Center Elite Feature Reference	Learn about Automatic Call Distribution (ACD) and Call Vectoring features available in Call Center Elite.	All users of Call Center Elite
Programming Call Vectoring Features in Avaya Aura® Call Center Elite	Write and edit call vectors.	Implementation engineers and system administrators
Troubleshooting Avaya Aura® Call Center Elite	Perform basic feature-related troubleshooting tasks.	All users of Call Center Elite
Understanding		
Avaya Aura® Call Center Elite Overview and Specification	Learn about Call Center Elite features, interoperability with other products, performance specifications, security, and licensing information.	Implementation engineers, sales engineers, and solution architects
Using		
Using Avaya 96X1 SIP Agent Deskphones with Avaya Aura® Call Center Elite	Learn how to set up the 96X1 SIP agent deskphones with Call Center Elite.	Implementation engineers

Finding documents on the Avaya Support website

Procedure

1. Navigate to http://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 an appropriate release number.
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 displays the documents only from the selected category.
7. Click **Enter**.

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
AVA00741WEN	Introduction to Call Center Operations
AVA00742WEN	Avaya Call Center - Analyze, Design, and Plan Implementation
5C00091E	Avaya Aura® Call Center Elite Virtual Campus Offering
5C00091I	Avaya Aura® Call Center Elite Implementation and Configuration
5C00091V	Avaya Aura® Call Center Elite Implementation and Configuration

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.

Procedure

- To find videos on the Avaya Support website, go to http://support.avaya.com and perform one of the following actions:
  - In **Search**, type `Avaya Mentor Videos` to see a list of the available videos.
  - In **Search**, type the product name. On the Search Results page, select **Video** in the **Content Type** column on the left.

- To find the Avaya Mentor videos on YouTube, go to www.youtube.com/AvayaMentor and perform one of the following actions:
  - Enter a key word or key words in the **Search Channel** to search for a specific product or topic.
  - Scroll down Playlists, and click the name of a topic to see the available list of videos posted on the website.

  Note:

  Videos are not available for all products.

Support

Go to the Avaya Support website at http://support.avaya.com for the most up-to-date documentation, product notices, and knowledge articles. You can also search for release notes, downloads, and resolutions to issues. Use the online service request system to create a service request. Chat with live agents to get answers to questions, or request an agent to connect you to a support team if an issue requires additional expertise.

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 Electronic Tandem Network (ETN) on page 213. 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 208.
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 3.0 or later.
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.
COS	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 measurement	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 terminal	A terminal that the system administrator uses to administer the switch. The administrator can also use the management terminal to gain access to the BCMS feature.

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 218.

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 Network (PPN) 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 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.
SSC	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.

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