



Avaya Aura[®] Contact Center Configuration — Avaya Communication Server 1000 Integration

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Contents

Chapter 1: New in this release	7
Features	7
Avaya Aura® System Manager support	7
Avaya Aura® Session Manager support	8
Other changes	8
SIP-enabled contact center commissioning	8
Chapter 2: Introduction	9
Related resources	10
Support	10
Avaya Mentor videos	10
Part I: Avaya Communication Server 1000 — AML	11
Chapter 3: AML Configuration Fundamentals	13
Configuration overview	13
CCMA Global settings	14
Voice-processing and PABX subsystems	16
Typical uses of voice-processing commands	17
Operational modes for voice-processing commands	18
Resource acquisition	21
Voice port partitioning rules	21
Voice segment variables	22
Scripts	23
LD overlays	25
Agent phones	25
IVR ACD-DNs	29
Meridian Link Services	29
Chapter 4: Avaya Communication Server 1000 deployment	31
Prerequisites for Avaya Communication Server 1000 configuration	31
Configuring general parameters	32
Configuring Give IVR (Avaya CallPilot®)	33
Configuring a third-party voice-processing engine	36
Configuring Give Controlled Broadcast Announcement and Open Voice Session (Avaya CallPilot®)	37
Configuring NSBR networking	40
Chapter 5: Subsystem connections configuration	43
Prerequisites	43
Subsystem connections configuration procedures	43
Defining the ELAN subnet with LD 17	45
Configuring the System Parameters (LD 17)	46
Configuring the Home Location Code (LD 15)	47
Enabling the ELAN subnet link	48
Checking the ELAN subnet link	49
Defining a default gateway address	50
Establishing a physical connection to Contact Center Manager Server from Avaya CallPilot®	50
Configuring the ACCESS link to Contact Center Manager Server from Avaya CallPilot®	51
Chapter 6: Avaya Communication Server 1000 Configuration	53

Prerequisites to Avaya Communication Server 1000 configuration.....	53
Avaya Communication Server 1000 configuration procedures.....	54
Configuring ACD queues.....	56
Configuring CDNs on the PABX.....	57
Changing CDNs on the PABX.....	59
Configuring a new NACD-DN with LD 23.....	60
Configuring an existing ACD-DN as an NACD-DN with LD 23.....	61
Configuring the Night Routing table with LD 23.....	62
Creating an Avaya CallPilot® ACD-DN in LD 23.....	63
Creating an Avaya CallPilot® voice port with LD 11.....	64
Defining a digital ACD phone.....	66
Associating an ACD queue with an agent.....	68
Defining single-line ACD phones.....	68
Verifying the TN configuration after it is acquired.....	70
Configuring a TN for the phantom loop with LD 10.....	72
Configuring a RAN route with LD 16.....	73
Configuring a MUS route with LD 16.....	74
Associating a MUS route with TNs in LD 14.....	75
Chapter 7: Meridian Link Services configuration.....	77
Meridian Link Services configuration procedures.....	77
Enabling Meridian Link Services on the PABX.....	79
Configuring Meridian Link Services.....	79
Defining multiline ACD phones as associated phones.....	80
Defining single-line ACD phones as associated phones.....	81
Defining a non-ACD multiline phone as an associated phone.....	82
Defining a non-ACD single-line phone as an associated phone.....	83
Defining a controlled DN.....	84
Connecting the host application.....	86
Chapter 8: Avaya CallPilot® configuration.....	87
Avaya CallPilot® configuration procedures.....	87
Updating Avaya CallPilot® configuration.....	89
Defining IVR and ACCESS ACD-DNs.....	90
Defining treatment DNs.....	90
Creating voice segments.....	91
Part II: Avaya Communication Server 1000 — SIP.....	93
Chapter 9: Avaya Communication Server 1000 configuration for SIP-enabled Contact Centers.....	95
Prerequisites to Avaya Communication Server 1000 configuration for SIP-enabled Contact Centers.....	95
Avaya Communication Server 1000 configuration procedures for SIP-enabled Contact Centers... ..	96
Adding SIP endpoints for Contact Center Manager Server.....	98
Adding a SIP Routing Entry for Contact Center Manager Server.....	99
Configuring the Signaling Server for SIP CTI (Remote Call Control).....	100
Configuring the Call Server for SIP CTI (Remote Call Control).....	101
Configuring Agent DN with SIP CTI (RCC) on the Call Server.....	102
Chapter 10: Avaya Communication Server 1000 Hotdesking configuration.....	105
Logging on to Virtual Office remotely.....	105
Logging off of Virtual Office.....	106

Chapter 11: Avaya Aura® System Manager configuration.....	107
Prerequisites.....	108
Logging on to the System Manager Web interface.....	108
Adding Avaya Unified Communications Management.....	109
Chapter 12: Avaya Aura® Session Manager configuration.....	111
Prerequisites.....	113
Avaya Aura® Session Manager configuration procedures.....	113
Creating a routing domain.....	115
Creating a routing location.....	116
Creating a SIP Entity for the Avaya Communication Server 1000.....	118
Creating a SIP Entity for the Session Manager.....	121
Creating a SIP Entity for the Avaya Aura® Contact Center.....	124
Creating a SIP Entity Link to the Avaya Communication Server 1000.....	128
Creating a SIP Entity Link to the Avaya Aura® Contact Center.....	130
Creating a routing policy from the Session Manager to Contact Center.....	132
Creating a routing policy from the Session Manager to Avaya Communication Server 1000.....	133
Creating a dial pattern to route calls to the Contact Center.....	135
Index.....	139

Chapter 1: New in this release

The following sections detail what is new in the *Avaya Aura® Contact Center Configuration – Avaya CS 1000 Integration* (NN44400-512) for Release 6.2.

Navigation

- [Features](#) on page 7
- [Other Changes](#) on page 8

Features

See the following sections for information about feature changes:

- [Avaya Aura® System Manager support](#) on page 7
- [Avaya Aura® Session Manager support](#) on page 8

Avaya Aura® System Manager support

Avaya Aura® System Manager delivers a set of shared, secure management services and a common console across multiple products. You use Avaya Aura® System Manager to manage and configure the routing policies for all Session Manager instances in your solution. Session Manager communicates with Session Manager using secure links.

You can use System Manager to access the Avaya Unified Communications Management (UCM) Web interface. This grants System Manager Administrators access to the Avaya Communication Server 1000 Elements manager.

For more information about Avaya Aura® System Manager, see [Avaya Aura® System Manager configuration](#) on page 107.

Avaya Aura® Session Manager support

Avaya Communication Server 1000 Release 7.5 supports integration with the Session Manager.

Session Manager is a SIP routing and integration tool. It integrates all the SIP entities across the entire enterprise network within a company. Session Manager offers a core communication service that builds on existing equipment but adds a SIP-based architecture. Session Manager connects to and acts as a system-wide dial plan for call processing applications such as:

- Avaya Communication Server 1000 Release 7.5 SIP-enabled PBX.
- Communication Manager using direct SIP connections.

You use Avaya Aura® System Manager to configure Session Manager.

For more information about configuring Session Manager to support Avaya Communication Server 1000 see, [Avaya Aura® Session Manager configuration](#) on page 111.

Other changes

See the following sections for information about other changes in this document:

- [SIP-enabled contact center commissioning](#) on page 8

SIP-enabled contact center commissioning

This document now contains additional content about commissioning your Contact Center in a SIP-enabled environment. For more information see, [Avaya Communication Server 1000 — SIP](#) on page 93.

Chapter 2: Introduction

This document provides conceptual and procedural information to configure the following components for use with the Contact Center Manager Server:

- Avaya Communication Server 1000 (CS 1000) in AML-based contact center solutions
- Avaya Communication Server 1000 (CS 1000) in SIP-enabled contact center solutions
- Meridian Link Services (MLS)
- Avaya CallPilot® or a third-party voice-processing system
- Avaya NES Meridian Integrated Recorded Announcement (MIRAN)

Application Module Link (AML) is an internal protocol used by Contact Center Manager Server to communicate directly with Avaya Communication Server 1000.

For more information about configuring Avaya Communication Server 1000 to support Contact Center Manager Server in AML-based solutions, see [Avaya Communication Server 1000 — AML](#) on page 11.

For more information about configuring Avaya Communication Server 1000 to support Contact Center Manager Server in SIP-enabled solutions, see [Avaya Communication Server 1000 — SIP](#) on page 93.

Prerequisites

- Contact Center Manager Server is installed and operational. For more information, see *Avaya Aura® Contact Center Server Administration* (NN44400-610).
- The Avaya Communication Server 1000 PABX is installed and operational. All current Dependency Lists (DL) are applied. For information about which DL to install, see www.avaya.com.
- The minimum supported release is Avaya Communication Server 1000 Release 5.0.

Navigation

- [Avaya Communication Server 1000 — AML](#) on page 11
- [Avaya Communication Server 1000 — SIP](#) on page 93

Related resources

Support

Visit the Avaya Support website at <http://support.avaya.com> for the most up-to-date documentation, product notices, and knowledge articles. On the Avaya Support website at <http://support.avaya.com>, search for notices, release notes, downloads, user guides, and resolutions to issues. Use the Web service request system to create a service request. Chat with live agents to help answer questions. If an issue requires additional expertise, agents can quickly connect you to a support team.

Avaya Mentor videos

Avaya Mentor is an Avaya-run channel on YouTube that includes technical content on how to install, configure, and troubleshoot Avaya products.

Visit <http://www.youtube.com/AvayaMentor> and do one of the following:

- Enter a key word or key words in the Search channel to search for a specific product or topic.
- Click the name of a playlist to scroll through the posted videos.

Part I: Avaya Communication Server 1000 — AML

- [AML Configuration fundamentals](#) on page 13
- [Avaya Communication Server 1000 deployment](#) on page 31
- [Subsystem Connections configuration](#) on page 43
- [Avaya Communication Server 1000 configuration](#) on page 53
- [Meridian Link Services configuration](#) on page 77
- [Avaya CallPilot® configuration](#) on page 87

Chapter 3: AML Configuration Fundamentals

This chapter provides the conceptual information that you need to configure the Avaya Communication Server 1000 in AML-based contact centers.

Navigation

- [Configuration overview](#) on page 13
- [Voice-processing and PABX subsystems](#) on page 16
- [Typical uses of voice-processing commands](#) on page 17
- [Operational modes for voice-processing commands](#) on page 18
- [Resource acquisition](#) on page 21
- [Voice port partitioning rules](#) on page 21
- [LD overlays](#) on page 25
- [Agent phones](#) on page 25
- [IVR ACD-DNs](#) on page 29
- [Meridian Link Services](#) on page 29

Configuration overview

This document describes how to configure the following components for use with Contact Center Manager Server (CCMS):

- Avaya Communication Server 1000
- Meridian Link Services (MLS)
- Avaya CallPilot® or a third-party voice-processing system
- Avaya NES Meridian Integrated Recorded Announcement (MIRAN)

Guide Assumptions

This guide is based on the following assumptions:

- You installed CCMS.
If necessary, install CCMS. For more information, see *Avaya Aura® Contact Center Installation* (NN44400-311).
- You installed the Avaya Communication Server 1000 PABX and applied all current Dependency Lists (DL). For information about which DL to install on the PABX, see www.avaya.com.
- You installed Avaya Communication Server 1000 Release 5.0 or later.

CCMA Global settings

To support voice-processing in Contact Center, you must configure the following items in the Contact Center Manager Administration (CCMA) Global Settings window.

- The number of voice ports to use for broadcast announcements.
- The wait time for a start and stop broadcast announcement (the amount of time between the arrival of the first call for the start and stop broadcast announcement and when the announcement actually starts).
- The default ACCESS treatment DN for the Give Controlled Broadcast Announcement and Voice Session script commands (this is the ACCESS ACD-DN).
- The DN and password for the mailbox containing the voice files and segments used by Give Controlled Broadcast Announcement and Voice Session script commands (if you use Contact Center Voice Services).

If you use Avaya CallPilot®, you must enter a value (even though you do not use the fields).

Maximum number of broadcast ports

Configure the number of voice ports available for use for Give Controlled Broadcast Announcement at any time. Up to 50 calls can be attached simultaneously to a single voice port on a broadcast announcement. After 50 calls, subsequent calls for an announcement are connected to a new voice port. If the maximum number of broadcast voice ports is reached, the system does not play a broadcast announcement to new callers. Instead, the system runs the next command in the Voice Session script.

If the voice ports are partitioned so that broadcast calls are directed to a dedicated IVR ACD-DN (that is, they do not share voice ports with Open Voice Session), then configuring this

parameter is not important as long if it is configured to greater than the number of voice ports in this IVR ACD-DN. The system can then queue new calls to the broadcast voice ports.

If voice ports are shared between broadcast announcements and voice sessions, you must limit the number of voice ports used by broadcasts. This ensures that Voice Session calls can obtain a voice port.

You can estimate the number of broadcast voice ports needed to use the following items:

- the call arrival rate
- the length of the announcement
- the Broadcast Wait Timer (if start and stop operation is used)

This limit is to minimize the number of voice ports used by broadcast and to maximize the number of voice ports used by Voice Session. Because Voice Session requires a one-call-to-one-port arrangement, the Voice Session voice port used for the same call traffic generally has a greater limit.

Broadcast voice port wait timer

The value of the Broadcast voice wait port timer determines the number of calls that can connect to a single voice port. You can configure broadcast voice-processing to use either of two modes: continuous, or start/stop. The Broadcast voice port wait timer value affect call handling only if you configure the system to use start and stop mode. Continuous mode connects calls immediately upon arrival.

Use a high timer value to permit the maximum number of calls to connect to the same voice port. A low timer value reduces the amount of time callers wait before the announcement plays. The lower the timer value, the smaller the number of calls that can use a single voice port (which means less efficient use of voice ports). The default timer value is 10 seconds.

Default ACCESS treatment DN

Do not explicitly specify a treatment DN in the Open Voice Session or Give Controlled Broadcast Announcement command within a script. Use the default ACCESS treatment DN.

The default ACCESS treatment DN must be the same as the ACCESS IVR DN.

Voice-processing and PABX subsystems

The Voice-processing feature interacts automatically with callers. You can classify interactions in the following ways:

- Passive—playing prerecorded messages to a caller
- Interactive—collecting input from a caller, usually with Dual Tone Multi-Frequency (DTMF)

Various Voice-processing methods

Contact Center Manager Server supports voice-processing with the following methods:

- Script commands—Calls terminate on CDNs and enter the Contact Center Manager Server script. Script statements can direct a call to connect to a voice port or RAN trunk so that voice-processing interaction can occur.
- Front-end IVR—Calls terminate directly on an Avaya CallPilot® voice menu or an IVR system and are not controlled by Contact Center Manager Server until the voice-processing system transfers the call to a CDN. Contact Center Manager Server must not acquire the voice ports used for front-end IVR.

Manual caller interaction

You can manually interact with callers in the following ways:

- Play a message to a caller—You can use any script command listed in [Script commands](#) on page 17 to play a message to a caller. You can use the Give Controlled Broadcast Announcement command to play a recorded announcement to a caller in either start/stop mode (the system plays the entire message from start to finish), or continuous mode (the system plays the message in a continuous loop, and callers can enter and exit at any point in the message). Other announcements do not use these specific modes.
- Broadcast announcements to multiple callers—You can use Avaya CallPilot® in stop/start mode to provide the same announcement to multiple callers.
- Interact with an external voice system—The CCMS interacts with an external voice system to control communication with the caller through commands and treatments in the scripts. You can use the Open Voice Session and End Voice Session commands to interact with a caller directly. To use these commands, you must use Contact Center Voice Services on Avaya CallPilot®.
- Interact with a caller indirectly—the voice-processing system controls communication with the caller. You can interact with a caller indirectly by using the Give IVR command to

connect the caller to a voice port controlled by Avaya CallPilot® or an external IVR system.

Script commands

Contact Center Manager Server supports the following voice-processing commands:

- **Give RAN**—Provide a recorded announcement (RAN) to a call through the specified RAN trunk. The RAN is interrupted if an agent becomes available to take the call. Otherwise, the RAN finishes and the next command in the script runs. If the GIVE RAN command is the first treatment in the script, and a delay occurs before the RAN is available, then the call hears ringback until the announcement plays.
- **Give IVR**—Play an announcement or IVR session using an Avaya CallPilot® voice menu or an external IVR system. The voice-processing system determines call handling. You can base the treatment on the IVR ACD-DN or the treatment DN. You can collect digits from the caller, but you cannot access the information from the script unless you use Host Data Exchange (HDX).
- **Give Controlled Broadcast Announcement**—Play a message to multiple callers by using a single voice port. This option requires Contact Center Voice Services on Avaya CallPilot®.

The GIVE CONTROLLED BROADCAST feature is not available on an Avaya Communication Server 1000E (CS 1000E) platform. You can use Open Voices sessions with one caller for each access channel for small Contact Centers on Avaya Communication Server 1000E instead of GIVE CONTROLLED BROADCAST. For more information, contact Avaya technical support.

- **Open/End Voice Session**—provide an interactive voice session in which you can play prompts and collect digits. The command requires Contact Center Voice Services on Avaya CallPilot®.

During script execution, all voice-processing commands and the Give RAN command pause until the command is completed. After the command finishes, the next statement in the script runs.

Typical uses of voice-processing commands

The following sections describe typical uses of voice-processing commands:

- **Use Give RAN**
 - Messages must be spoken to callers.
 - Legacy RAN equipment exists.

- RAN equipment is installed instead of a voice-processing system as a cost-reduction measure, and other voice-processing functions are not required.
- Use Give IVR
 - If you use Avaya CallPilot® to play announcements or give voice menus to callers, and you do not want to use an ACCESS link. You usually use this command if you migrate from Avaya NES Meridian CCR and you do not want to re-record your announcements or voice menus.
 - If you use Avaya CallPilot® and you want to give the caller the option to leave a message in a mailbox.
 - If you use a third-party voice-processing system for announcements or voice menus while the call is controlled by a Contact Center Manager Server script. Usually, if an external IVR system is used, the call is directed to the IVR system before the call enters the Contact Center Manager Server script (front-end IVR). Give IVR is not used unless you use Avaya integrated IVR CTI applications.

Give Controlled Broadcast Announcement

Use Give Controlled Broadcast Announcement if you use Contact Center Voice Services on Avaya CallPilot®, and you want to simultaneously play the same message to multiple callers (for example, the traditional RAN scenario “All agents are busy...”). Give Controlled Broadcast Announcement uses ports more efficiently than Give IVR.

Open/End Voice Session

Use Open/End Voice Session commands if

- you use Contact Center Voice Services on Avaya CallPilot®, and you must give custom messages (for example, a caller's expected wait time)
- you use Contact Center Voice Services on Avaya CallPilot®, and caller interaction (collect digits) is required

For more information about using these script commands in scripts, see *Avaya Aura® Contact Center Configuration – Orchestration Designer Application Development* (NN44400-510).

Operational modes for voice-processing commands

The following sections describe the operation modes for voice-processing commands:

Listen only or interactive

If callers listen only to the recorded message, use the following voice-processing commands:

- Give RAN
- Give IVR
- Give Controlled Broadcast Announcement
- Open/End Voice Session, Play Prompt

If callers can interact, use the following voice-processing commands:

- Open/End Voice Session, Collect Digits (Contact Center Voice Services on Avaya CallPilot®)
- Give IVR (Avaya CallPilot® voice menus or external IVR)

Single connection or broadcast

You can use any listen-only command to play a single message for more than one caller. You can optimize the use of voice ports by using broadcast commands. A broadcast connection can sustain much higher call rates than a single connection and uses fewer voice ports.

Use the one-call-per-port commands if you must give custom messages to callers (for example, Expected Wait Time), if caller input is collected, or if you have a third-party voice-processing system.

The following commands connect multiple calls for each voice port:

- Give Controlled Broadcast Announcement
- Give RAN (if RAN broadcast is used)

The following commands connect one call for each voice port:

- Give IVR
- Open/End Voice Session

Start or stop and continuous

The system can play messages using two modes: Start or Stop, and Continuous.

Start or stop mode

In Start or Stop mode, the system plays the entire message from the beginning. You can use the following voice-processing commands in start or stop mode:

- Give RAN
- Give IVR
- Give Controlled Broadcast Announcement
- Open/End Voice Session

Continuous mode

In continuous mode, the system plays the message in a repeating loop, and users can enter at any point during the message. You can use the following commands with continuous mode:

- Give RAN
- Give Controlled Broadcast Announcement

Give Controlled Broadcast Announcement in continuous mode connects you immediately and continues the script only after playing one full cycle of the message.

RAN operation remains the same with the introduction of Contact Center Manager Server.

Interruptible or non-interruptible

In Contact Center Manager Server, only the Give IVR command supports both interruptible (option) and non-interruptible (default) operation. Both the Give Controlled Broadcast Announcement and Open Voice Session commands support interruptible (default) operation only.

For more information about using these script commands in scripts, see *Avaya Aura® Contact Center Configuration – Orchestration Designer Application Development* (NN44400-510).

Interruptible operation

The voice-processing stops if an agent becomes available, and the agent receives the call.

Use this mode when the system plays a message for the caller's information (for example, "All agents are busy...") or for amusement (for example, advertising) and when you need to transfer a call to an agent as quickly as possible.

Non-interruptible operation

If the call is queued before the voice-processing command starts, the call does not qualify to be answered until the voice-processing session ends. The call, however, remains in place in queue during the voice-processing session.

If an agent becomes available during the voice-processing session, the next call that can be answered is presented instead. After the call in the voice-processing session ends, the system moves the call to the next available agent.

This mode is useful if callers must listen to a full cycle of a message before speaking to an agent, or if interactive menus are presented to the caller, and the input must be complete before the caller can speak to an agent.

If a call does not queue before a voice-processing session, the interruptible versus non-interruptible operation has no effect; the call always operates in a non-interruptible fashion.

Resource acquisition

The following table summarizes the resources that Contact Center Manager Server (CCMS) must acquire for the various voice-processing commands.

! Important:

Do not configure extra voice ports on the Avaya Communication Server 1000 or on the CCMS that are not accessed or acquired by Contact Center Manager Server.

Table 1: Resources required by CCMS for voice-processing

Command	IVR ACD-DN	Voice Port TN	Voice Port channel
Give IVR	Yes	Yes	No
Give Controlled Broadcast Announcement	Yes	Yes	Yes
Open Voice Session	Yes	Yes	Yes
Front-end IVR	No	No	No

Voice port partitioning rules

Your voice-processing system can provide a variety of services:

- Auto-attendant
- Voice menus
- Fax
- Voice mail
- Application services to Meridian Link Services (MLS), Avaya NES Meridian (Integrated) IVR, and Contact Center Manager Server

While some services and applications can share voice ports, Contact Center Manager Server requires a dedicated group of voice ports behind a dedicated ACD-DN to operate correctly. If

you require other voice-processing services, such as Call Answering or Voice Menus, configure a separate queue.

Voice port partitioning

Contact Center Manager Server uses two types of voice ports: IVR and ACCESS ports. For proper operation of Contact Center Manager Server and the voice-processing system, each ACD-DN can contain voice ports of only one type (Give IVR or ACCESS).

Contact Center Manager Server Give IVR voice ports

The Give IVR script command uses Give IVR voice ports. Make sure that only Contact Center Manager Server Give IVR calls arrive at an ACD-DN containing Give IVR voice ports.

Configure Give IVR voice ports as standard IVR voice ports. Contact Center Manager Server must acquire the TNs for the voice ports.

Contact Center Manager Server ACCESS voice ports

The Give Controlled Broadcast Announcement and Open Voice Session commands use the ACCESS voice ports. ACCESS voice ports are available for integrated voice-processing systems only, such as Avaya CallPilot®. Configure the ports as ACCESS voice ports, giving each voice port a unique ACCESS class (channel number). Contact Center Manager Server must acquire the TNs and Class IDs for these voice ports.

Configure the maximum number of broadcast ports parameter [CCMA Global settings](#) on page 14 to limit the number of voice ports used by broadcast announcements. After reaching this limit, Contact Center Manager Server skips this command and runs the next command in the script.

Voice segment variables

Contact Center Manager Server scripts reference voice segments on Avaya CallPilot® by using voice segment variables. Voice segment variables can contain one or more voice segments. Voice segments contain specific words or phrases recorded in the Voice Prompt Editor or Application Builder. Each voice segment variable has a name, number, and value that indicates the language used to record the segment.

Two types of voice segments are available: User-defined, and System-defined.

User-defined voice segments

Record user-defined voice segments for Avaya CallPilot® using Application Builder.

Define the variables for a user-defined voice segment using the Script Variables dialog box. A voice segment variable has the type VOICE SEGMENT and is a global variable. You can define any number of variables.

You can define the variables for a user-defined voice segment on Contact Center Manager Administration at any time. Contact Center Manager Administration and Avaya CallPilot® looks for segments on the other platform when the script runs, but does not check at any other time.

Voice segment file names are case-sensitive. Enter voice segment names exactly as they appear in Application Builder or the Voice Prompt Editor.

For more information about creating voice segments, see *Avaya Aura® Contact Center Manager Administration – Client Administration* (NN44400-611).

System-predefined phrases

To generate spoken numbers, Contact Center Manager Server provides a number of predefined voice segments representing spoken numbers. Contact Center Manager Server strings the segments together to speak numbers 0 to 999 999 999 999 999.

Predefined voice segment variables have file and segment number placeholders when you install the Contact Center Manager Server system.

Script example

For example, in a voice session, a message plays that prompts the caller to enter an identification number by pressing the phone keys. The seven digits are collected in a variable named vardigit_cv. The system plays a second message in which the numbers entered are spoken back:

```
OPEN VOICE SESSION 2299
  PLAY PROMPT VOICE SEGMENT enter_ID_number_vs
  COLLECT 7 DIGITS INTO vardigit_cv
  PLAY PROMPT NUMBERBYDIGIT vardigit_cv
END VOICE SESSION
```

Scripts

Use script commands to determine call handling. The services that a caller hears depend on the path the call follows through the master script and any secondary scripts. Information about the voice-processing treatment that a call receives by Contact Center Manager Server is pegged in the database. You can run reports that show details about voice-processing and the effects in your Contact Center.

For more information about scripts, see *Avaya Aura® Contact Center Configuration – Orchestration Designer Application Development* (NN44400-510).

CDNs

Ensure that the script references a CDN that is configured in the Contact Center Manager Administration application.

NACD ACD-DNs

To route calls to a remote ACD-DN, the Contact Center Manager Server script must contain the following command:

```
QUEUE TO NACD acd-dn [WITH PRIORITY priority]
```

The script can contain other commands to control the wait time or to change the priority.

IVR ACD-DNs and treatment DNs

All voice-processing script commands need and use both an IVR ACD-DN and a treatment DN. Both parameters are optional in the script statement, and the defaults are drawn from various places.

The IVR ACD-DN on the voice-processing script statement specifies the PABX ACD-DN to which the voice port TNs belong in the PABX configuration. Contact Center Manager Server directs voice-processing calls to the IVR ACD-DN, and the PABX ACD software distributes the calls over the voice port (the PABX selects the actual voice port, not the Contact Center Manager Server software).

If the PABX IVR ACD-DN does not have TRDN configured, you must include with treatment in the Give IVR script element.

Contact Center Manager Server must acquire the IVR ACD-DN for the voice-processing to function correctly.

Routes

To generate all trunks busy (ATB) reports, configure trunk routes on Contact Center Manager Administration, and acquire the routes.

To use Give Music or Give RAN commands, in the Contact Center Manager Server script, you must reference a RAN or MUS route. You do not need to acquire music or RAN routes.

LD overlays

LD overlays provide a command line interface (CLI) to configure the Avaya Communication Server 1000. You can connect a hyperterminal to an Avaya Communication Server 1000 and connect to a CLI. You enter the overlay LD number, for example LD 48, and a number of prompts appear. Each overlay has a different purpose and accepts specific commands. For example, overlays are available to add phones, CDNs, new cards, and to configure trunks.

Agent phones

This section describes the phones supported by Contact Center Manager Server. For more information, see www.avaya.com.

! Important:

Agents assigned multimedia capabilities must log on to a phone through the Communication Control Toolkit. These agents cannot manually log on to a phone.

Contact Center phones

Avaya recommends that you use the following phones specifically for Contact Centers.

- Avaya 3905 Digital Deskphone

Avaya Communication Server 1000 ACD phones

Contact Center Manager Server also supports the following phones that you can configure for use with Avaya Communication Server 1000 ACD.

- Avaya 39xx Digital Deskphone
 - Avaya 3904 Digital Deskphone
 - Avaya 3905 Digital Deskphone
- IP Phones and Softphones
 - Avaya 2004 IP Deskphone
 - Avaya 2002 IP Deskphone
 - Avaya 1120E IP Deskphone

- Avaya 1140E IP Deskphone
- Avaya 1150E IP Deskphone
- Avaya 2050 IP Softphone

These phones are not designed for a Contact Center environment and some Contact Center features are not available.

Support of specific types of phones can change with each software release of the call server (Avaya Communication Server 1000). Consult the Avaya Communication Server 1000 documentation for an up-to-date list of supported phone types for your software release.

Display Waiting Calls key and lamp

Contact Center Manager Server supports the Display Waiting Calls (DWC) key. This feature lists skillset information when an agent presses the DWC key.

Contact Center Manager Server does not support the Directory Log feature.

Note:

The information is different from the DWC feature used in the Avaya Communication Server 1000 ACD environments.

Supervisor phone display

The DWC key and associated lamp configured on a supervisor's phone do not support the display of Contact Center Manager Server skillset information. If you press the DWC key on a supervisor's phone, it shows ACD queue information for that supervisor. The lamp also responds to ACD queue loading and activity for that supervisor, as determined by the PABX configuration. Calls do not normally queue to ACD queues for Contact Center Manager Server; therefore, this feature for Contact Center Manager Server supervisors ensures the Contact Center can handle Network ACD calls or operate in default mode, and to ensure that the PABX ACD features can route calls.

Skillset information

Skillset information display is available only on phones that have numeric display capabilities.

Display format

The information on the phone appears with spaces between the fields. For the phone display of type 1 x 12, the data appears in three digits. For phone displays larger than 1 x 12, the data appears in four digits. The maximum displayable number of calls in queue is 9999, and the maximum number of agents that Contact Center Manager Server currently supports is 40. The maximum displayable amount of time that a call can be in queue is 9999 seconds or 2.78 hours. The following table summarizes the display types and field width for phones that display DWC key information.

Table 2: DWC key phone display type and field width

Display type	AAA	BBB	CCC
1 x 12	3 digits	3 digits	3 digits
1 x 16	4 digits	4 digits	4 digits
1 x 40	4 digits	4 digits	4 digits
2 x 24	4 digits	4 digits	4 digits

DWC key lamp

The DWC key lamp on a Contact Center Manager Server agent phone does not respond to calls in skillsets; it always remains dark for skillset loading and activity. However, the lamp continues to respond to the call loading and activity in any ACD queues that the agent is logged on to, as determined by the PABX configuration. Calls do not normally queue to ACD queues for Contact Center Manager Server; therefore, the primary use of this feature for agents is for the Contact Center to handle Network ACD calls or operates in default mode, and the PABX ACD features route the calls. When the agent presses the DWC key, the agent phone display shows Contact Center Manager Server skillset information.

 Note:

Risk of lost calls Avaya recommends that any maintenance work on agent phones of the Contact Center Manager Server system occurs outside office hours. Validate and test the changes before an agent logs on again. Any maintenance undertaken during office hours can result in calls lost or misdirected and ultimately abandoned.

Agent must log back on after modification

Any parameter modification that occurs with LD 10, 11, 20, 84, and 85 logs off the phone, and the agent must log on again. The following modifications trigger a logoff:

- CLID changes—Changes to LDN_Data in LD 15 have no effect on agents.
- DES (Designation) changes (LD84 for analog phones; LD 85 for digital)—You can change a DES only for the Avaya i2004 IP Deskphone and Avaya i2050 IP softphone in LD 20.
- Agent Supervisor (SPID) changes.
- Adding or deleting key features such as Auto Dial and Hunting Allowed (HTA).
- Name change (digital phones).
- Key changes.
- Class of Service changes.
- Change or add AST (Associate Set Assignment) and IAPG (ISDN/AP status message group) due to addition of a person to the performance recording tool—ETalk.

Agent queue changes (moving agents between queues)

A change in ACD queue causes the line (agent/supervisor TN) to be released in Contact Center Manager Server. If you use ECHG to change agent positions between queues, assign the ACD key a null value and then add it with the new ACD queue in another change pass. The administrator must still release the phone, make the change, and acquire the phone again.

Change only the position ID on a phone

When you attempt to change the position ID of a phone, you encounter an SCH6683 error. You cannot change the position ID while this agent is acquired. Therefore, you must release the line in Contact Center Manager Server before you can make the modification.

Name change (analog phones)

When you attempt to change the DN key of an analog phone (500), the following message appears: "SCH5039 – Already configured with AGTA." You can define the phone only as a single appearance DN. You must remove the ACD association of the phone to make the name change, and then reapply the ACD function. When you remove the ACD function, the line is released in Contact Center Manager Server.

You can change names with LD 95, which does not affect the state of the phones—the agents remain logged on and acquired in the case of analog and digital phones.

Create a new TN in Contact Center Manager Server

The Contact Center Manager administrator must create a new Terminal Number (TN) in Contact Center Manager Server.

MOV/OUT

When you attempt to MOV/OUT an acquired line, you encounter the CH1509 Cannot MOV/OUT acquired TN error. You must release the TN in Contact Center Manager Server before the MOV/OUT operation can proceed.

The administrator must delete the item from the phone section and then create a new item for the new TN. The new TN must be acquired before the agent logs on.

Avaya does not support the use of the CPY command in LD 10 or LD 11 on ACD agents.

IVR ACD-DNs

An IVR ACD-DN is a DN assigned to voice ports that provide voice-processing services. Program voice ports as ACD agents belonging to IVR ACD-DNs. Contact Center Manager Server must acquire the IVR ACD-DNs. You configure IVR ACD-DNs on the PABX, in your voice-processing system [Subsystem connections configuration](#) on page 43, and on Contact Center Manager Server.

Meridian Link Services

Meridian Link Services (MLS) is a Contact Center Manager Server based communications facility that provides an interface between a host application and the PABX. A host is any computer on which the third-party application runs. This interface integrates the computer and the Private Automatic Branch Exchange (PABX). In this integrated environment, the host processor interacts with the PABX by exchanging application layer messages.

You can also use MLS to develop applications to use information from the PABX (such as Caller ID), to connect to another application to get that customer's data, and then provide data to a PC to help agents prepare for the call.

To use CTI for your Contact Center Manager Server agents, you must use MLS.

In an environment with multiple servers on the same PABX, if you use Communication Control Toolkit to provide CTI functions, each server must have an associated Communication Control Toolkit server.

Chapter 4: Avaya Communication Server 1000 deployment

As with any interlinked configuration, you must consistently configure many parameters across subsystems.

Prerequisites for Avaya Communication Server 1000 configuration

- Create a system based on your Contact Center requirements. Consider the following information:
 - Network design (ELAN subnet, Contact Center subnet, and remote access service [RAS] IP requirements)
 - Projected peak call traffic
 - Number of agents, shifts, days of operation
 - Projected call flow and scripting requirements
 - Voice Messaging port requirements (number of ports required for Give IVR and ACCESS services, such as Voice Sessions, Give Controlled Broadcast Announcement, and collect digits sessions)
- For more information, see *Avaya Aura® Contact Center Planning and Engineering* (NN44400-210).

Navigation

- [Configuring general parameters](#) on page 32
- [Configuring Give IVR \(Avaya CallPilot®\)](#) on page 33
- [Configuring a third-party voice-processing engine](#) on page 36
- [Configuring Give Controlled Broadcast Announcement and Open Voice Session \(Avaya CallPilot\)](#) on page 37
- [Configuring NSBR networking](#) on page 40

Configuring general parameters

The checklists in this section are based on the assumption that you understand the configuration elements and how to check the status on the PABX, Avaya CallPilot®, and Contact Center Manager Administration. If you do not know how to check the status of a particular element, consult the appropriate section in this guide or the appropriate guide for the subsystem.

In the following checklists, the subsystem PABX refers to the Avaya Communication Server 1000 PABX unless otherwise indicated.

Complete the following checklist.

Subsystem	Description	Verified
PABX	Verify that Avaya Communication Server 1000 is operating correctly.	
PABX	Install required Avaya Communication Server 1000 software packages (use LD 22 to verify the packages).	
PABX	The following additional PABX requirements are available: • Avaya Communication Server 1000 dependency PEPs for Contact Center Manager Server • Avaya Communication Server 1000 dependency PEPs for the specific voice-processing subsystem you use • Provisioning of sufficient Call Registers	
PABX	Verify that the ELAN subnet connection between Contact Center Manager Server and Avaya Communication Server 1000 is functioning. Note the current ELAN subnet/VSID number using LD 48. (This is useful for troubleshooting.) Ensure that the default gateway is configured in the PABX using LD 117. For more information, see Defining a default gateway address on page 50.	
PABX	Configure Contact Center Manager Server CDNs on the PABX. For more information, see Configuring CDNs on the PABX on page 57.	
PABX	Configure the Active Directory Services (ADS) or SCB block on the PABX with LD 23. For more information, see Creating an Avaya CallPilot ACD-DN in LD 23 on page 63. Avaya recommends that you use an ADS block.	

Subsystem	Description	Verified
	! Important: Risk of ACD block corruption Before performing this task, ensure that all agents log off. Failure to do so can result in corruption of the ACD block. If you configure the ADS block, you must configure DCUS in the PARM data block with LD 17. See Defining the ELAN subnet with LD 17 on page 45.	
PABX	Configure Agent ACD queues in LD 23. For more information, see Configuring ACD queues on page 56. When you configure the ADS or SCB block, the values for HOML and RPRT are YES for existing ACD queues. If HOML=YES, agents are logged off when they replace the handset after a call. If this is not appropriate for your environment, modify the HOML parameter for existing ACD queues.	
PABX	Configure the agent and supervisor TNs on the PABX. For more information, see Associating an ACD queue with an agent on page 68.	
PABX	Configure the trunk, music, and RAN routes on the PABX. For more information, see Configuring a RAN route with LD 16 on page 73.	

Configuring Give IVR (Avaya CallPilot®)

Complete the following checklist if you use the Give IVR command with Avaya CallPilot®.

Subsystem	Description	Verified
—	Complete the general checklist.	
Avaya CallPilot®	Verify that Avaya CallPilot® is running for voice Messaging.	
PABX and Avaya CallPilot®	Verify that Avaya CallPilot® is communicating with the PABX.	
PABX	Enable the link between the PABX and the Avaya CallPilot® server for CTI operations (LD 48).	
PABX	Configure the IVR ACD-DN (for IVR voice ports) on the PABX (LD 23). For more information, see Creating an Avaya CallPilot® ACD-DN in LD 23 on page 63.	

Subsystem	Description	Verified
	When you configure the IVR ACD-DN on the PABX, configure the IVR and ALOG prompts to Yes.	
PABX	Configure the IVR voice ports on the PABX as virtual agents (LD 11). For more information, see Creating an Avaya CallPilot® voice port with LD 11 on page 64 You must dedicate voice ports to Contact Center Manager Server Give IVR voice service.	
Contact Center Manager Server	Configure the Avaya CallPilot® connection parameters for TCP voice connection as follows: • Avaya CallPilot® connection IP (ELAN network interface IP address). • Avaya CallPilot® connection port (Set TCP port to 10008).	
Avaya CallPilot®	Configure the Avaya CallPilot® server configuration and enable integration. In the CallPilot Configuration Wizard, review all Avaya CallPilot® configuration information up to and including the Switch Information page. On the Switch Information page, do the following: • Select Enable Contact Center Manager Server Integration. • Enter the customer number in the Switch Customer Number box. • Enter the Contact Center subnet address of Contact Center Manager Server in the Contact Center Manager Server CLAN (Customer Local Area Network) IP Address field.	
Avaya CallPilot®	Configure voice ports (IVR voice channels) in the Avaya CallPilot® server configuration. In the CallPilot Configuration Wizard, identify and configure the channels that provide IVR services to Contact Center Manager Server. • TN of the voice channel in Avaya CallPilot® Mail = TN of the virtual agent on the PABX = Telephony/Port Address of the phone on Contact Center Manager Server. • ACD-DN defined for the voice channel in Avaya CallPilot® = ACD-DN on the PABX = IVR ACD-DN on Contact Center Manager Server.	

Subsystem	Description	Verified
Avaya CallPilot®	Verify that the Service Directory Number (SDN) table contains the Avaya CallPilot® Primary CDN. • Application name is Voice Messaging. • Media type is Voice.	
Avaya CallPilot®	Verify the SDN table contains Contact Center Manager Server IVR ACD-DNs. Configure the IVR ACD-DNs as follows: • Application name is Contact Center Voice Services. • Media type is Voice.	
Avaya CallPilot®	Define Contact Center Manager Server Treatment DNs as Service DN entries in the SDN table. Each application is named and the media type is Voice. (Use meaningful application names because Avaya CallPilot® can store a large number of applications.) • Applications with the media type of Voice contain voice items that are announcements and menus played to callers (Give IVR script commands). • Use CallPilot Application Builder to create, record, and manage voice menus and announcements. • Complete applications before they are selected in the CallPilot SDN table.	
Contact Center Manager Server	Configure and acquire IVR ACD-DN on the Contact Center Manager Server.	
Contact Center Manager Server	Verify in Contact Center Manager Server that VSM and Meridian Link Services Manager (MLSM) services are running. (Use the VSM Status window if it is configured.)	
Contact Center Manager Server	Configure and acquire the voice ports on the Contact Center Manager Server. • IVR voice ports are defined with a TN with no channel number. • All TNs in the PABX belonging to the IVR ACD-DNs are acquired by Contact Center Manager Server as voice ports.	
Avaya CallPilot®	Verify that the acquired voice ports are idle in Avaya CallPilot®. Use the Channel Monitor in Avaya CallPilot® Manager to verify voice port status. If the voice ports are not initialized, restart Avaya CallPilot®.	

Subsystem	Description	Verified
Contact Center Manager Server	Implement and activate the Give IVR test script before you activate the remainder of the voice-processing scripts. The voice-processing script command uses both the IVR ACD-DN and a Treatment DN, as in the following examples: <pre>GIVE IVR ivr_queue WITH TREATMENT welcome_msg GIVE IVR 6000 WITH TREATMENT 1001</pre> If you do not specify the Treatment DN that is in the script, the server uses the default Treatment DN (TRDN) defined for the IVR ACD-DN on the PABX. Ensure you configure the default treatment in the SDN table in Avaya CallPilot® for proper operation.	
Contact Center Manager Server	Activate the Master Script.	
Contact Center Manager Server	Verify that the Give IVR test script is working: • Other scripts are verified and additional voice-processing scripts are implemented. • The SDN table is updated as new treatments are required.	

Configuring a third-party voice-processing engine

If you use the Give IVR command with a third-party voice-processing engine, complete the tasks in order in the following checklist.

Subsystem	Description	Verified
—	Complete the general checklist.	
PABX	Configure the IVR ACD-DN on the PABX (see the documentation provided with your IVR system). When you configure the IVR ACD-DN on the PABX, set the IVR prompt to YES.	
PABX	Configure voice ports on the PABX as analog agents (see the documentation provided with your IVR system). You must use dedicated voice ports.	
PABX	Verify that the third-party voice-processing system is up and communicating properly with the PABX.	

Subsystem	Description	Verified
Voice processing	Configure voice ports in the third-party voice-processing system.	
Contact Center Manager Server	Configure and acquire the IVR ACD-DN on the server.	
Contact Center Manager Server	Configure and acquire the voice ports on the server. • Define voice ports with a TN only (the channel number field is blank). • TN of virtual agent on PABX = Telephony/Port Address of phone on server. • The server acquires all TNs in the PABX ACD-DN as voice ports.	
Contact Center Manager Server	Verify that the script command refers to the acquired IVR ACDDN.	
Contact Center Manager Server	If an APL link exists, specify a treatment DN one of the following ways: • The script explicitly specifies a treatment DN. • You configure correctly the Default Treatment DN for the IVR ACD-DN on the PABX. • If Inter-Peripheral Message Link (IPML) exists, you specify the treatment DN in the script. If you do not specify the treatment DN in the script, the server uses the default treatment DN defined for the IVR ACD-DN on the PABX.	
Contact Center Manager Server	Verify that voice ports behind the IVR ACD-DN acquired by the server are indicated by the PABX and show the state as either Logged In or Logged Out. Ports must be Logged In for the PABX to connect calls to the port for voice processing.	

Configuring Give Controlled Broadcast Announcement and Open Voice Session (Avaya CallPilot®)

If you use the Give Controlled Broadcast Announcement or Open Voice Session commands with Contact Center Voice Services on Avaya CallPilot®, perform the following procedure.

Subsystem	Description	Verified
—	Complete the general checklist.	
Avaya CallPilot®	Verify that Avaya CallPilot® is running for voice Messaging.	
PABX and Avaya CallPilot®	Verify that Avaya CallPilot® is in communication with the PABX.	
PABX	Verify that the link between the PABX and the Avaya CallPilot® server is enabled for CTI operations.	
PABX	Configure the ACCESS ACD-DN (for ACCESS voice ports) on the PABX. For more information, see Defining IVR and ACCESS ACD-DNs on page 90. When you configure the ACD-DN on the PABX, set the IVR and ALOG prompts to YES.	
PABX	Verify that ACCESS voice ports are configured on the PABX as virtual agents. For more information, see Creating an Avaya CallPilot voice port with LD 11 on page 64. Voice ports must be dedicated to Contact Center Manager Server ACCESS voice service.	
Contact Center Manager Server	Configure the Avaya CallPilot® connection parameters for TCP voice connection as follows: • Avaya CallPilot® connection IP (CLAN network interface IP address) • Avaya CallPilot® connection port (Set the TCP port to 10008)	
Avaya CallPilot®	Verify that the Avaya CallPilot® server configuration is updated and integration is enabled. In the Avaya CallPilot® Configuration Wizard, review all Avaya CallPilot® configuration information up to and including the Switch Information page. On the Switch Information page, do the following: • Select Enable Contact Center Manager Server Integration. • Enter the customer number in the Switch Customer Number box. • Enter the Contact Center subnet address of Contact Center Manager Server in the Contact Center Manager Server CLAN IP Address field.	
Avaya CallPilot®	Verify that Voice ports (ACCESS channels) are configured in the Avaya CallPilot® server configuration.	

Subsystem	Description	Verified
	In the Avaya CallPilot Configuration Wizard, identify and configure the channels that provide ACCESS services to Contact Center Manager Server.	
Avaya CallPilot®	Verify that the SDN table contains the Avaya CallPilot® Primary CDN. • Application name is Voice Messaging. • Media type is Voice.	
Avaya CallPilot®	Update the SDN table in Avaya CallPilot® with Contact Center Manager Server ACCESS ACD-DNs. Configure ACD-DNs as follows: • Application name is Contact Center Voice Services. • Media type is Voice.	
Contact Center Manager Server	Configure and acquire ACCESS ACD-DN on the Contact Center Manager Server. For more information, see Defining IVR and ACCESS ACD-DNs on page 90.	
Contact Center Manager Server	Configure the global settings on Contact Center Manager Server to specify the following: • Default ACCESS Treatment DN and ACCESS ACD-DN (these have the same value) • Number of ACCESS ports reserved for broadcasts • Broadcast Voice Port Wait Timer • The mailbox number and password are not used—leave as default values	
Contact Center Manager Server	Configure and acquire the voice ports on Contact Center Manager Server. • ACCESS voice ports are defined with both a TN and a channel. • All TNs in the PABX belonging to the ACCESS ACD-DNs in the PABX are acquired by Contact Center Manager Server as voice ports.	
Contact Center Manager Server	Verify that the ACCESS link between Contact Center Manager Server and Avaya CallPilot® works correctly.	
Avaya CallPilot®	Verify that the acquired voice ports are idle. Use the Channel Monitor in Avaya CallPilot® Manager to check voice port status. If the voice ports are not initialized, restart Avaya CallPilot®.	

Subsystem	Description	Verified
Contact Center Manager Server, Avaya CallPilot®	Define voice segments as follows: • Update system predefined voice segments (file 1, which contains all the number prompts). Make a list of the segment IDs referenced in the Contact Center Manager Server variable table. • Define new user voice segments.	
Contact Center Manager Server	Implement and activate the ACCESS test script before you activate the remainder of the voice-processing scripts. The voice-processing script command uses both the ACCESS ACD-DN (optional) and voice segment variables that refer to the voice prompts that the caller hears.	
Contact Center Manager Server	Activate the Master Script.	
Contact Center Manager Server	Verify that the Give Controlled Broadcast Announcement and Open Voice Session test script works.	

Configuring NSBR networking

If you use Network Skill-Based Routing (NSBR), you must configure networking.

 Note:

Configure each of the individual servers in your network before you complete the following tasks.

Subsystem	Description	Verified
PABX	Install and configure the Network Automatic Call Distribution (NACD) package, and create the CDNs to which networked calls for each server are routed.	
PABX	For Universal Networking, you must provision Landing Pad CDNs or Dialed Number Identification Services (DNIS) on the PABX.	
Contact Center Manager Server	Install and configure the Network Control Center (NCC) software.	
PABX	Define the network CDNs.	
PABX	Configure the communication parameters.	

Subsystem	Description	Verified
Avaya CallPilot®	Assign agents to network skillsets.	
PABX	Modify the scripts to use the Network Skill-Based Routing feature.	

Chapter 5: Subsystem connections configuration

Configure the following links between the subsystems for the features of the Contact Center Manager Server to operate correctly:

- IP address and default gateway on the PABX.
- ELAN subnet between the PABX, Contact Center Manager Server, and Avaya CallPilot®.
- ACCESS link between Contact Center Manager Server and the voice-processing system (if you use Contact Center Voice Services on Avaya CallPilot® to provide voice-processing services). For more information about configuring the ACCESS link, see [Configuring the ACCESS link to Contact Center Manager Server from Avaya CallPilot](#) on page 51.

Prerequisites

- Ensure Avaya CallPilot® and the PABX are installed and communicating correctly.

Subsystem connections configuration procedures

About this task

This task flow shows you the sequence of procedures you perform to configure subsystem connections. To link to any procedure, see [Subsystem connections configuration navigation](#) on page 45.

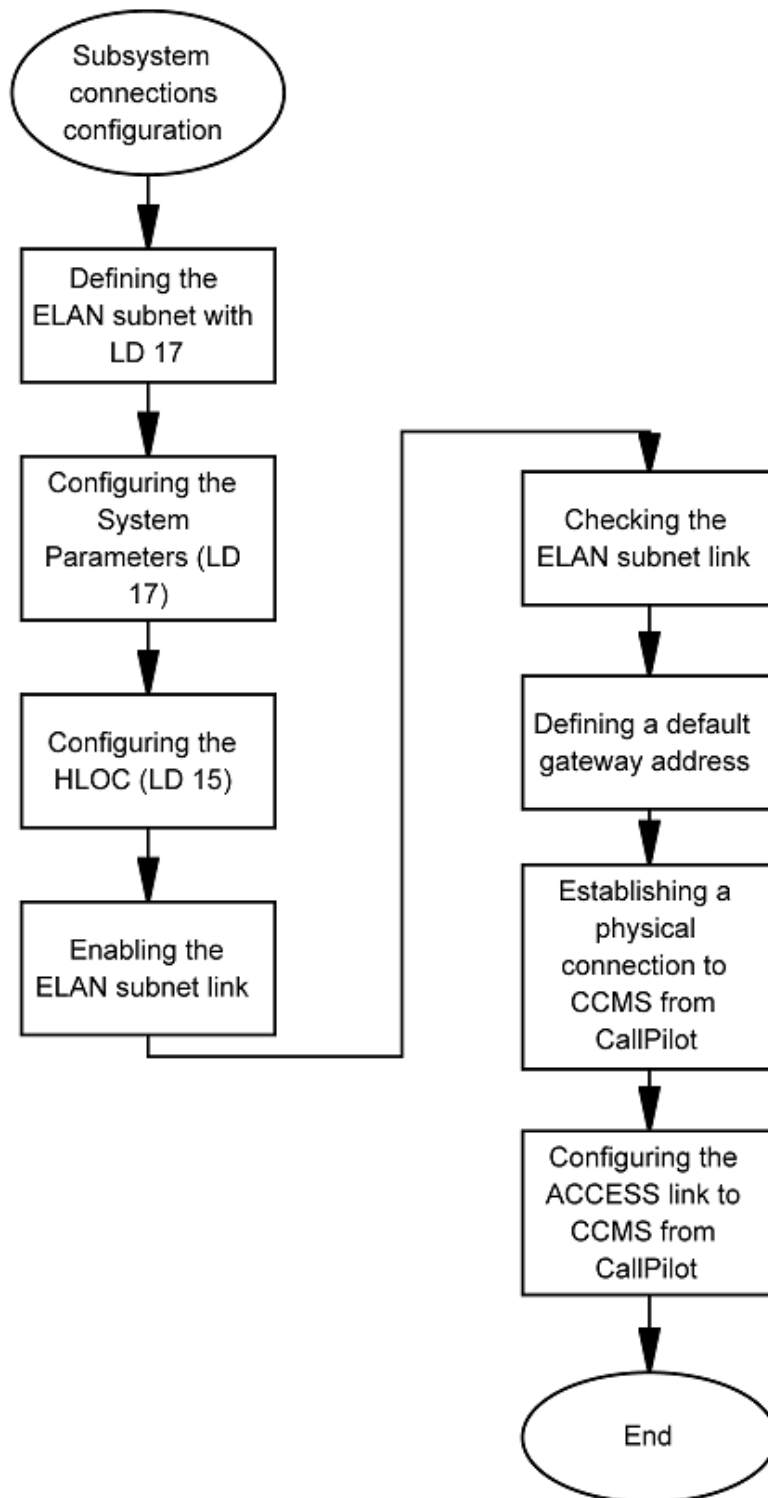


Figure 1: Subsystem connections configuration procedures

Subsystem connections configuration navigation

- [Defining the ELAN subnet with LD 17](#) on page 45
- [Configuring the System Parameters \(LD 17\)](#) on page 46
- [Configuring the Home Location Code \(LD 15\)](#) on page 47
- [Enabling the ELAN subnet link](#) on page 48
- [Checking the ELAN subnet link](#) on page 49
- [Defining a default gateway address](#) on page 50
- [Establishing a physical connection to Contact Center Manager Server from Avaya CallPilot®](#) on page 50
- [Configuring the ACCESS link to Contact Center Manager Server from Avaya CallPilot®](#) on page 51

Defining the ELAN subnet with LD 17

About this task

Define and configure the ELAN subnet for the AML link and the associated VSID in the configuration record. This provides the Ethernet connection over which AML messages are exchanged between the Avaya Communication Server 1000 system and Avaya CallPilot®. You must create a separate ELAN for Avaya CallPilot® and Contact Center for integration.

Perform this procedure only if the ELAN subnet is not provisioned.

Procedure

1. Connect to the Avaya Communication Server 1000 Call Server.
 2. Enter **LD 17**.
 3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, type four asterisks (****), and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	CHG	Change existing data block
TYPE	CFN	Configuration record
ADAN	NEW ELAN xx	Configure a new link and assign it a number, where xx is within the ELAN subnet range (16–31). You can use any number in this range as long as it is not already used. Make a note of this link number for your reference.
- CTYP	ELAN	Card type
- DES	x...x	Enter a designator of up to six characters in length to identify this ELAN subnet. Because the ELAN subnet is not dedicated to a specific application, make the designator generic.
VAS	NEW	Configure a new AML link or change the existing link configuration.
- VSID	xx	The VAS identifier can be in the range of 16–31. For convenience, this can be the same number you assigned to the new ELAN subnet link (in response to the ADAN prompt).
- ELAN	xx	Use the same number as defined in the ADAN prompt.
- SECU	YES	Security. If you have a Contact Center server connected to your PABX, enter YES (even if you are not using Contact Center's Voice Services Support).

Configuring the System Parameters (LD 17)

About this task

Perform the following procedure to configure the System Parameters.

Procedure

1. Connect to the Call Server.
 2. Enter **LD 17**.
 3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, type four asterisks (****) and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	CHG	Change existing data block
TYPE	PARM	System parameters
NCR	x...x	Number of Call Registers. The range depends on the system type. Increment the current value by 2 x the number of CallPilot DS0 channels. For example, if the current NCR value is 500 and there are 24 DS0 channels, enter 548.
CSQI	(20) to x	Maximum number of call registers for CSL input queues. Configure this parameter to the number of CallPilot DS0 channels, multiplied by two. x = 25% of NCR. For example, if you have 24 DS0 channels, enter 48.
CSQO	(20) to x	Maximum number of call registers for CSL/AML output queues. Configure this parameter to the number of CallPilot DS0 channels, multiplied by two. x = 25% of NCR. For example, if you have 24 DS0 channels, enter 48.

Configuring the Home Location Code (LD 15)

About this task

Perform the following procedure to configure the Home Location Code (HLOC) or system parameter LD 15.

Procedure

1. Connect to the Call Server.
 2. Enter **LD 15**.
 3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, type four asterisks (****) and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	CHG	Change existing data block
TYPE	NEN_Data	Networking
CUST	XX	Customer Number (where XX is your Customer number which may be 0, but depends on the site and previous customer number configuration).
HLOC	0–3999	Home Location Code For Avaya Aura® Contact Center integration, there is a limitation that the maximum value of HLOC is 3999.

Enabling the ELAN subnet link

About this task

Perform the following procedure to enable the ELAN subnet link.

Procedure

1. Connect to the call server.
 2. Enter **LD 48**.
 3. Ensure that the ELAN is enabled. For more information, see [Checking the ELAN subnet link](#) on page 49.
-

Variable definitions

Name	Description
ENL ELAN	Enable the ELAN server task.

Checking the ELAN subnet link

About this task

Perform the following procedure to check the ELAN subnet link status.

Procedure

1. Connect to the Call Server.
 2. Enter **LD 48**.
 3. Enter the commands.
 4. Check that the Server Task parameters are **ENABLED**. For more information, see [Enabling the ELAN subnet link](#) on page 48.
-

Variable definitions

Name	Description
STAT ELAN	Verify the status of all configured ELAN
STAT ELAN x	Verify the status of ELAN x

Defining a default gateway address

About this task

In Contact Center Manager Server, you must connect the ELAN subnet to the Contact Center subnet using a routed solution that adheres to the ELAN engineering requirements as specified by Avaya Communication Server 1000 documentation:

- see *Communication Server 1000 Converging the Data Network with VoIP Fundamentals* (NN43001-260) for ELAN design and engineering details.
- see *Communication Server 1000 Communication Server 1000M Large System Planning and Engineering* (NN43021-220) for Contact Center application engineering details.

In addition to the physical connection, you must configure computers on all sides of the router to acknowledge the presence of this router, also known as the default gateway.

Procedure

Enter the following commands:

```
LD 117
new route 0.0.0.0 <a.b.c.d>
```

Variable definitions

Name	Description
<a.b.c.d>	The IP address of the gateway local to the PABX.

Establishing a physical connection to Contact Center Manager Server from Avaya CallPilot®

About this task

Establish a physical connection to Contact Center Manager Server (CCMS) to enable communication between Avaya CallPilot® and CCMS over the Embedded LAN (ELAN).

Procedure

Ensure the Avaya CallPilot® server is physically connected to the ELAN subnet.

Configuring the ACCESS link to Contact Center Manager Server from Avaya CallPilot®

Before you begin

- Configure Avaya CallPilot®, and specify the Contact Center subnet IP address of Contact Center Manager Server in the CallPilot Configuration Wizard. For more information, see [Updating Avaya CallPilot configuration](#) on page 89.
- Configure Contact Center Manager Server, and specify the ELAN network interface IP address of the Avaya CallPilot® server.

About this task

Configure the ACCESS link to the Contact Center Manager Server from Avaya CallPilot® to configure the Subsystem connection.

Procedure

1. For CallPilot, choose **System Utilities, Support Tools, CallPilot Processing Utilities, Trace Viewer <nbtview>**, in Trace Control, select:
 - a. NBAPE for messages on ACCESS Link
2. Contact Center Manager Server: From **Start, Run**, enter **tsm_oam**, and then select option 3—Register with TSM Server.

For	Perform
VSM and MLSM session traces:	From the OAM menu, select option 2 , and then enter 0 at the prompt. * Note: Note the Session ID for VSM_Service CallPilot Application. Press enter to go back to the OAM menu. Select option 5 , enter the Session ID, and then respond to the prompts as appropriate.
AML trace	From the OAM menu, select option 7 . From the AML Trace menu, select option 4 .
Access Protocol trace:	From the OAM menu, select option 9 . Select option 3 to enable the trace.

For	Perform
For Access Protocol Debug trace:	From the OAM menu, select option 10 . Select option 3 to enable the trace.

Chapter 6: Avaya Communication Server 1000 Configuration

This chapter describes how to configure the required Avaya Communication Server 1000 elements. It also explains how to initialize the PABX and change resources on the PABX without causing problems on Contact Center Manager Server (CCMS).

Prerequisites to Avaya Communication Server 1000 configuration

- You have the packages required for the following features:
 - voice-processing in CCMS
 - voice-processing in Avaya CallPilot®
 - other desired PABX features
- Your PABX runs one of the following software releases:
 - Avaya Communication Server 1000 release 5.0 or later
- You applied all current Dependency Lists (DL). For information about which DLs to install on the PABX, contact your Avaya customer support representative.
- You installed the NACD package on the PABX (if you purchased the Network Skill-Based Routing option).

! Important:

CCMS cannot share PABX resources (such as CDNs, ACD-DNs, or TNs) with other applications.

Avaya Communication Server 1000 configuration procedures

About this task

This task flow shows you the sequence of procedures you perform to configure the Avaya Communication Server 1000. To link to any procedure, see [Avaya Communication Server 1000 configuration navigation](#) on page 55.

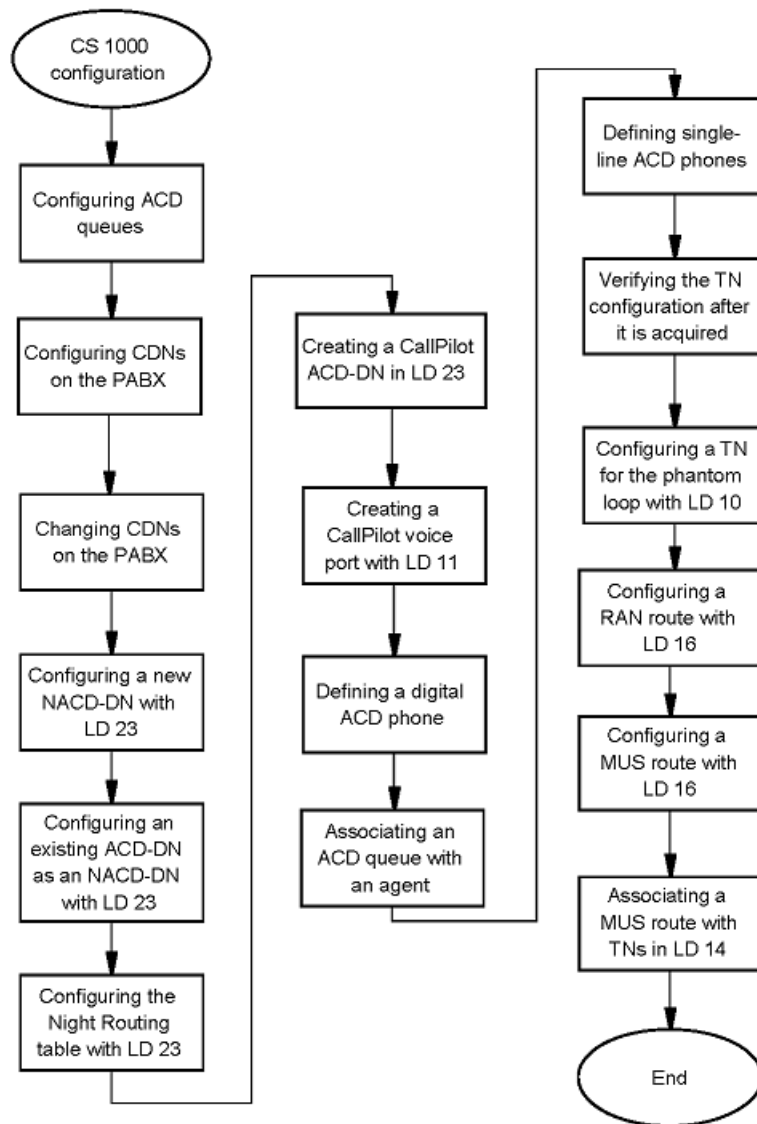


Figure 2: Avaya Communication Server 1000 configuration procedures

Avaya Communication Server 1000 configuration navigation

- [Configuring ACD queues](#) on page 56
- [Configuring CDNs on the PABX](#) on page 57
- [Changing CDNs on the PABX](#) on page 59
- [Configuring a new NACD-DN with LD 23](#) on page 60

- [Configuring an existing ACD-DN as an NACD-DN with LD 23](#) on page 61
- [Configuring the Night Routing table with LD 23](#) on page 62
- [Creating an Avaya CallPilot ACD-DN in LD 23](#) on page 63
- [Creating an Avaya CallPilot voice port with LD 11](#) on page 64
- [Defining a digital ACD phone](#) on page 66
- [Associating an ACD queue with an agent](#) on page 68
- [Defining single-line ACD phones](#) on page 68
- [Verifying the TN configuration after it is acquired](#) on page 70
- [Configuring a TN for the phantom loop with LD 10](#) on page 72
- [Configuring a RAN route with LD 16](#) on page 73
- [Configuring a MUS route with LD 16](#) on page 74
- [Associating a MUS route with TNs in LD 14](#) on page 75

Configuring ACD queues

About this task

Use LD 23 to configure the ACD-DN with which a phone is associated. If you want the agent to use the phone to enter Not Ready reason codes, perform this procedure.

Procedure

1. Connect to the Call Server.
 2. Enter **LD 23**.
 3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, enter four asterisks (****) and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	NEW	Add new data
TYPE	ACD	Automatic Call Distribution
CUST	xx	Customer number (0–99)

Prompt	Response	Description
ACDN	xxxx	ACD Directory Number, where xxxx = Default DN. Enter the DN as the Default DN (DFDN) used in the CDN configuration.
MWC	NO	Message Waiting Center. Select NO.
MAXP	1	Maximum number of agent positions. For the DFDN, MAXP must be 1.
NCFW	0	Night Call forward DN for ACD calls For DFDN, define NCFW to 0. This DN value can be up to 31 digits.

Configuring CDNs on the PABX

Before you begin

- You know the user ID and password to log on to the PABX administration terminal.
- You are familiar with PABX Change and Diagnostics overlays.
- You have a list or printout of available CDNs (overlay program 23).

Use the prompts in the following table to configure a CDN. This includes Music and RAN prompts. For prompts not listed in the table, press Enter to accept the default.

Prompt	Response	Description
REQ	aaa	Request.
TYPE	CDN	Type of data block is CDN.
CUST	xx	Customer number associated with this data block.
CDN	xxxx	Control Directory Number.
FRRT	0–511	First RAN route number for ACD.
FRT	0–2044	First RAN time.
SRRT	0–511	Second RAN route number for ACD.
SRT	0–2044	Second RAN time.
FROA	NO (YES)	First RAN on arrival.
MURT	0–511	Music route number.
DFDN	x . . . x	Local default ACD-DN.

Prompt	Response	Description
CEIL	0–(2047)	CDN ceiling value.
OVFL	(NO) YES	Force Overflow Tone to the call when the ceiling threshold is exceeded.
TDNS	(NO) YES	Is DNIS number an original called party.
RPRT	(YES) NO	Management reporting and status display.
CNTL	(NO) YES	CDN is in control.
VSID	0–15	Value Added Server ID.
HSID	0–15	Host ID.
CWTH	0–(1)–2047	Call Waiting Threshold.
BYTH	(0)–2047	Busy Threshold.
OVTH	0–(2047)	Overflow Threshold.
STIO	0, 1, 2, . . . 15	Status Input/Output devices.
TSFT	0–(20)–510	Telephone Service Factor Threshold in seconds.
ACNT	xxxx	Account.

About this task

Controlled Directory Numbers (CDN) are specialized ACD-DNs or queues on the PABX. A CDN is the entry point of a call into Contact Center Manager Server call processing. You must configure CDNs on the PABX and on Contact Center Manager Server.

❗ Important:

Risk of corruption Corruption results if you do not configure a CDN in this overlay before that CDN is acquired in Contact Center Manager Administration.

Procedure

1. Go to Contact Center Manager Administration **Launchpad, Configuration**.
2. On the system tree, double-click the server on which to add the CDN.
The server expands to reveal the resources.
3. Click the **CDNs (Route Points)** folder.
The CDNs (Route Points) window appears.
4. In the **Name** box, type the name of the CDN as you want it to appear in reports.
5. In the **Number** box, type the CDN number. This number must match the number configured on the PABX.

6. From the **Call Type** list, select whether you want the CDN to be Local calls, MCDN Network calls, or DNIS Network calls.

! Important:

To use the CDN for MCDN Network calls, the server must have Network Skills Based Routing (NSBR) enabled and be connected to an Avaya Communication Server 1000 PABX. To use the CDN for DNIS Network calls, the server must have Universal Networking enabled.

7. Click any other table row.
The system adds the CDN, and Not Acquired appears in the Status column.
8. Select the **Acquired?** check box in the row containing the CDN that you just added.
9. Click any other row in the table to acquire the CDN.
The system acquires the CDN, and the status appears in the Status column.
10. If necessary, click **Refresh Status** to view the current status of the acquisition.
11. Repeat steps 2 to 10 for each CDN to configure and acquire.
To reconfigure a CDN, you must first remove the CDN, edit the configuration parameters, and then reacquire the CDN.

Changing CDNs on the PABX

About this task

To change CDNs on the PABX, you must perform specific steps to avoid service interruptions in Contact Center Manager Server. For example, if you remove a CDN that is currently acquired by Contact Center Manager Server, you can cause some services to stop. This prevents Contact Center Manager Server from handling calls.

Procedure

1. Remove the CDN from Contact Center Manager Administration. For more information, see [Configuring CDNs on the PABX](#) on page 57.
 2. Delete, add, or change the CDN as necessary on the PABX.
 3. Acquire the CDN on Contact Center Manager Administration.
-

Configuring a new NACD-DN with LD 23

Before you begin

- You know the user ID and password to log on to the PABX administration terminal.
- You are familiar with PABX change and diagnostics overlays.

About this task

Define the default Automatic Call Distribution (ACD) DN.

Procedure

1. Connect to the Call Server.
 2. Enter **LD 23**.
 3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, enter four asterisks (****) and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	ACD	Automatic Call Distribution.
CUST	xx	Customer number (0–99).
ACDN	xxxx	ACD Directory Number, where xxxx = Default DN. Enter the DN as the Default DN (DFDN) used in the CDN configuration.
MWC	NO	Message Waiting Center. Select NO.
MAXP	1	Maximum number of agent positions. For the DFDN, MAXP must be 1.
NCFW	0	Night Call forward DN for ACD calls.

Prompt	Response	Description
		For DFDN, define NCFW to 0. This DN value can be up to 31 digits.

Configuring an existing ACD-DN as an NACD-DN with LD 23

Before you begin

- Ensure that no TNs or positions are associated with the ACD-DN. If TNs or positions are assigned to the ACD-DN, reassign them to another ACD-DN.

About this task

If you use an existing ACD-DN, ensure that no TNs or positions are associated with the ACD-DN to be configured as the NACD routing DN. If TNs or positions are assigned, reassign them to another ACD-DN. Then use LD 23 to configure the NACD-DN as described in [Configuring a new NACD-DN with LD 23](#) on page 60.

Procedure

1. Connect to the Call Server.
2. Enter **LD 23**.
3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
4. To exit the overlay, enter four asterisks (****) and press **Enter**.

Variable definitions

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	ACD	Automatic Call Distribution.
CUST	xx	Customer number (0–99).
ACDN	xxxx	ACD Directory Number.
MAXP	1	Maximum member of agent positions.

Prompt	Response	Description
NCFW	X	Night Call forward. Enter X to delete NCFW. NCFW must be blank to allow the configuration of an NACD night routing table.

Configuring the Night Routing table with LD 23

About this task

Configure an ACD-DN on the PABX with an associated Night Routing Table containing remote targets and time values. This ACD-DN has no positions assigned and, therefore, is always in Night Mode.

Procedure

1. Connect to the Call Server.
 2. Enter **LD 23**.
 3. Enter the appropriate values as. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, enter four asterisks (****), and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	NEW	Add
TYPE	NACD	Network ACD
ACDN	xxx...x	ACD directory number used
TABL	N	Night Table
TRGT	xxxx tttt	Remote target ACD-DN (xxxx) and the timer (0–1800) in seconds. Press Enter to add another target. You can add a maximum of 20 targets to the table. Press Enter twice to stop adding targets.

Prompt	Response	Description
REQ	END	Exit from overlay.

Creating an Avaya CallPilot® ACD-DN in LD 23

Before you begin

- You know the user ID and password to log on to the PABX administration terminal.
- You are familiar with PABX change and diagnostic overlays.
- You have a list or printout of available IVR ACD-DNs.

About this task

Configure IVR ACD-DNs when you use Avaya CallPilot® to play messages to callers. These messages are stored on the voice-processing system and can be announcements or voice menus.

Configure an IVR ACD-DN for each group of voice ports. For example, configure one for ACCESS voice ports, one for non-ACCESS voice ports, and one for Voice Messaging (Contact Center Manager Server does not acquire Voice Messaging ACD-DNs).

Note:

If your system uses only a MIRAN card to provide messages, you need not configure IVR ACD-DNs.

Procedure

1. Connect to the Call Server.
 2. Enter **LD 23**.
 3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, enter four asterisks (****) and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	NEW	Create a new queue.
TYPE	ACD	ACD data blocks.

Prompt	Response	Description
CUST	0–99	Customer number.
ACDN	xxxx	The DN of the ACD queue. This is the IVR ACD-DN acquired from Contact Center Manager Server.
MWC	NO	Indicates that this is not a message center.
MAXP	xx	Indicates the number of agent positions that you can assign to the queue.
IVR	YES	Indicates that you can use the queue with the Give IVR command defined in scripts.
TRDN	xxxx	Default treatment DN is used if you do not specify treatment in the script. Use treatment DN's to select the treatment that the call receives from the voice-processing system. You can also use them with Avaya CallPilot®, or any voice-processing system that connects to the PABX by means of the AML link.
ALOG	YES	ACD agents are automatically logged on. Only Avaya CallPilot® TNs are automatically logged on; analog TNs, such as those used for third-party IVR systems, are not. To log on analog TNs, you must write an application on the IVR system to log on the ports.
REQ	END	Exit from overlay.

Creating an Avaya CallPilot® voice port with LD 11

About this task

Voice ports carry speech to Avaya CallPilot®, or an IVR system. You must configure voice ports when the ports are CallPilot, or third-party IVR system ports used to play announcements or voice menus. You must configure voice ports on the PABX in CallPilot, and on Contact Center Manager Server.

Configure voice ports as virtual agent TNs for Avaya CallPilot®. For third-party IVR systems, the agent TNs are analog TNs. For Avaya CallPilot®, the class of service must be Multimedia Agent (MMA) and FLXA.

For the voice ports, ensure that the key layout matches the key configuration in Avaya CallPilot®. This matching ensure Avaya CallPilot® can answer, disconnect, originate, transfer, and conference calls.

You must have AST licenses on the PABX to support Messaging between Avaya CallPilot® and Contact Center Manager Server.

*** Note:**

Some services and applications that handle calls outside of Contact Center Manager Server control can share voice ports, while calls under Contact Center Manager Server control require dedicated voice ports to operate correctly. For more information, see [Voice port partitioning rules](#) on page 21.

Procedure

1. Connect to the Call Server.
2. Enter **LD 11**.
3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
4. To exit the overlay, enter four asterisks (****) and press **Enter**.

Variable definitions

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	2008	Type of Data block. For CallPilot, TYPE must be 2008.
TN	l s c u c u	Terminal Number. A TN is required for each agent. For Large Systems, where l s c u = loop, shelf, card, unit For Small Systems, where c u = card, unit
DES	d..d	ODAS station designator, where d..d = 1-6 alphanumeric characters
CUST	xx	Customer number (0–99)
CLS	aaaa	Class of Service. Each agent must have the VCE and MMA Class of Service. To get the VCE Class of Service on the upper 16 units (16 to 31), you must first specify the FLXA Class of Service. For CallPilot configuration, aaaa = WTA UNR VCE MMA for units 0–15 or FLXA VCE MMA for units 16–31
AST	xx yy	Associated Set Assignment (AST) A maximum of two DN keys can be controlled by the host computer. xx and yy represent the

Prompt	Response	Description
		first and second DN key controlled by the host computer. For CallPilot, xx = 00 (ACD Key) yy = 01 (SCN Key)
IAPG	1	Unsolicited Status Message Group. Range: (0)–15
KEY	xx aaa yyyy ccc zzzz	Telephone function key assignments, where xx = key number aaa = key feature or function yyyy = DN information (if required) ccc = CLID information (if required) zzzz = DN information For CallPilot, provision each agent with the following feature keys: ACD, SCN, NRD, MSB, TRN, and AO3.
	0 ACD xxxx 0 yyyy	xxx = ACD DN yyy = Agent Position DN
	1 SCN zzzz	zzzz = Non ringing DN used to make outbound calls
	2 msb	Make Set Busy (msb)
	3 nrd	Not Ready (nrd)
	4 trn	Transfer (trn)
	5 AO3	Three Party conference Key (AO3)

Defining a digital ACD phone

Before you begin

- You know the user ID and password to log on to the PABX administration terminal.
- You are familiar with PABX Change and Diagnostics overlays.

About this task

If you want a user to log on to a phone to receive or monitor incoming calls, you must configure phones (TN). You must configure agent and supervisor phones on the PABX and on Contact Center Manager Server.

While agent and supervisor phones require no special configuration for Contact Center Manager Server, they must belong to an ACD-DN. Usually, Contact Center Manager Server controls call routing and the ACD-DN does not affect the call routing.

The ACD-DN controls call routing if the Contact Center Manager Server CDN is in default mode or if incoming network ACD calls target the ACD-DN.

Procedure

1. Connect to the Call Server.
2. Enter **LD 11**.
3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
4. To exit the overlay, enter four asterisks (****) and press **Enter**.

Variable definitions

Prompt	Response	Description
REQ	NEW	Add a new phone.
TYPE	aaa	Enter phone type as appropriate.
TN	l s c u	Terminal Number
DES	Name	Enter a name for the phone.
CUST	0–99	Customer number
KLS	1–7	Number of key/lamp strips attached.
KEY 0	ACD xxxx yyyy	Where xxxx = ACD-DN, and yyyy = Agent position ID
KEY 1	NRD	Not ready
KEY 2	A06	Six-party conference
KEY 3	MSB	Make set busy
KEY 4	TRN	Transfer
KEY 8	SCR xxxx	xxxx = IDN
KEY 13	ACNT	Activity key
REQ	NEW, END	Either define another multiline ACD phone, or exit the overlay saving all of the changes entered.

Associating an ACD queue with an agent

About this task

Use the Default Queue Management (DQM) feature to associate an ACD queue with an agent. When the agent logs on to a Contact Center Manager Server acquired phone, Contact Center Manager Server determines which ACD queue is associated with the agent and the phone. If the ACD queue associated with the agent is different from the ACD queue configured on the phone, Contact Center Manager Server attempts to move the phone to the agent's associated ACD queue.

This new ACD queue information is maintained on the phone even after the agent logs off. Therefore, the ACD queue of the phone can differ from the default configuration of the phone after an agent logs on. The phone position ID, however, remains unaffected.

Procedure

1. Connect to the Call Server.
2. Enter **LD 23**.
3. Enter the appropriate values as described in the table described in [Configuring ACD queues](#) on page 56. For prompts not listed, press **Enter** to accept the default.

* Note:

If you want the agent to use the phone to enter Not Ready reason codes, ensure that NRAC for the ACD-DN is YES. To deactivate Not Ready reason codes, define the NRAC as NO.

4. To exit the overlay, enter four asterisks (****) and press **Enter**.
-

Defining single-line ACD phones

Before you begin

- You know the user ID and password to log on to the PABX administration terminal.
- You are familiar with PABX Change and Diagnostics overlays.

About this task

Use LD 10 overlay to define single-line ACD phones.

Procedure

1. Connect to the Call Server.
 2. Enter **LD 10**.
 3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, enter four asterisks (****) and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	NEW	Add a new phone
TYPE	500	Enter phone type as appropriate.
TN	l s c u	Terminal Number
CUST	0–99	Customer number
DN	xxxx	DN for the phone
CLS	AGTA	Class of service–ACD agent assignment
—	THFA	Switchhook flash allowed
—	UND	Unrestricted access
—	WTD, (WTA)	Warning Tone Denied (Allowed)
SPID	xxxx	Supervisor's position ID number
PRI	(1)–48	Priority level for agent
AACD	YES	Associated set for ACD agent. Only for AST phones.
FTR	ACD xxxx yyyy	Use the ACD feature, where xxxx=the ACD-DN yyyy=the ACD Position (POS-ID)
REQ	NEW, END	Either define another single-line ACD phone or exit the overlay saving all of the changes entered.

Verifying the TN configuration after it is acquired

Before you begin

- A phone is configured in the system using LD 11.
- The phone is acquired in CCMS.

About this task

After you configure a phone using LD 11, and CCMS acquires the phone, you can verify the TN configuration.

Procedure

Ensure that the following printout appears:

```
>ld 11
REQ prt
TYPE tnb
TN 4 0 4 2
DATE
PAGE
DES
DES agtset
TN 004 0 04 02
TYPE 3904
CDEN 8D
CUST 0
CDN 2003
AOM 0
FDN
TGAR 1
LDN no
NCOS 0
SGRP 0
RNPg 0
SCI 0
SSU
XLST
CLS CTD FBD WTA LPR MTD FND HTD ADD
MWD AAD IMD DOS XHD IRD NID OLD VCE DRG1
POD DSX VMD CMSD CCSD SWD LND CNDA
CFTD SFD MRD DDV CNID
ICDD CDMD LLCN MCTD CLBD AUTU
GPUD DPUD DNDD CFXD ARHD FITD CNTD CLTD ASCD
CPFA CPTA HSPD ABDD CFHD FICD NAID
DDGA NAMA
USMD USRD ULAD RTDD PGND OCBd FLXD
CPND_LANG ENG
HUNT
PLEV 02
AST
IAPG 0
AACS YES
ACQ AS: TN
ASID 16
SFNB 2 5 6 9 10 11 12 13 14 15 16 17 18 19
SFRB 1 2 15
USFB 1 2 3 4 5 6 7 9 10 12 13 14 15
```

```

CALB 1 3 4 5 6 8 9 11
FCTB ITNA NO
DGRP
PRI 01
MLWU_LANG 0
DNDR 0
KEY 00 ACD 2001 0 2012
    SPV
    01 NRD
    02 A06
    03 MSB
    04 TRN
    05
    06
    07
    08 SCR 4702 0 MARP
    09 RAG
    10 AAG
    AA AMG
    12 DWC 2001
    13 ACNT
    14
    15

```

Variable definitions

Prompt	Response	Description
AACS	YES	The phone is acquired by an application.
ACQ AS	TN	The TN is acquired, but no CTI application registered for the phone.
ASID	16	The application on AML 16 acquired the phone.
SFNB	—	Bitmap that controls which messages are sent to Contact Center Manager Server and is not user-definable.
SFRB	—	Bitmap that controls which messages are sent to Contact Center Manager Server and is not user-definable.
USFB	—	Bitmap that controls which messages are sent to Contact Center Manager Server and is not user-definable.
CALB	—	Bitmap that controls which messages are sent to Contact Center Manager Server and is not user-definable.

Configuring a TN for the phantom loop with LD 10

About this task

You can define and configure TNs without the required phones or line cards. An agent or supervisor can then log on to any phone.

To define and configure TNs without phones or line cards, you must have the Phantom Terminal Numbers (PHTN) feature. The PHTN feature supports Phantom Terminal Numbers to direct incoming calls to existing telephones using the Call Forward All Calls (CFW) and Remote Call Forward (RCFW) features.

You must configure phantom TNs on the PABX and on Contact Center Manager Server. Each configured TN has an associated DN. Before an agent or supervisor can log on to any phone, you must assign each associated DN to an agent or supervisor in Contact Center Manager Administration.

Procedure

1. Connect to the Call Server.
 2. Enter **LD 10**.
 3. Enter the appropriate values as described in the following table. For prompts not listed in the following table, press Enter to accept the default.
 4. To exit the overlay, enter **** and press Return.
-

Variable definitions

Prompt	Response	Description
REQ	NEW, CHG	Add, or change
TYPE	500	Telephone type
TN	l s c u	Terminal number (loop, shelf, card, and unit); if the loop is a phantom loop, "PHANTOM" is echoed to the technician
DN	xxx...x	Directory Number; must be a Single Appearance DN
SCPW	xxxx	Station Control Password
CLS	aaaa	Class of Service options, which cannot include AGTA, CCSA, MNL, or LPA

Prompt	Response	Description
FTR	DCFW nn x...x	Default Call Forward; nn = number of digits up to 23 x...x = Default Call Forward DN.

Configuring a RAN route with LD 16

Before you begin

- You know the user ID and password to log on to the PABX administration terminal.
- You are familiar with PABX Change and Diagnostics overlays.
- You obtained a list of routes using LD 21 (REQ=prt, TYPE=RDB).
- At the PABX, you ensured that physical trunks are defined for the route.

About this task

A route defines a group of trunks. Each trunk carries incoming and outgoing calls to and from the PABX. You must configure the trunk routes on the PABX. To use the Give RAN and Give Music commands in your scripts, you must also configure RAN and MUS routes.

Contact Center Manager Server supports only RAN, MUS, FGDT, TIE, DID, COT, FEX, and WATS route types.

If you do not use Contact Center Voice Services on Avaya CallPilot®, configure a MIRAN card and RAN routes to supply messages to callers waiting in queue.

Procedure

1. Connect to the Call Server.
 2. Enter **LD 16**.
 3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, enter four asterisks (****) and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	NEW	Add a new route
TYPE	RDB	Route data block

Prompt	Response	Description
CUST	nn	Customer number
ROUTE	nn	Route number
DES	x...x	Enter a description
TKTP	RAN	Recorded Announcement trunk data block requires package 7.
ASUP	YES	Answer supervisor
ACOD	nnn	Access code
REQ	END	Exit form overlay

Configuring a MUS route with LD 16

Before you begin

- You know the user ID and password to log on to the PABX administration terminal.
- You are familiar with PABX Change and Diagnostics overlays.
- You obtained a listing of routes using LD 21 (REQ=prt, TYPE=RDB).
- At the PABX, you ensured that physical trunks are defined for the route.

About this task

A route defines a group of trunks. Each trunk carries incoming and outgoing calls to and from the PABX. You must configure the trunk routes on the PABX. To use the Give RAN and Give Music commands in your scripts, you must also configure RAN and MUS routes.

Contact Center Manager Server supports only RAN, MUS, FGDT, TIE, DID, COT, FEX, and WATS route types.

If you do not use Contact Center Voice Services on Avaya CallPilot[®], configure a MIRAN card and RAN routes to supply messages to callers waiting in queue.

Procedure

1. Connect to the Call Server.
2. Enter **LD 16**.
3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
4. To exit the overlay, enter four asterisks (****) and press **Enter**.

Variable definitions

Prompt	Response	Description
REQ	NEW	Add a new route
TYPE	RDB	Route data block
CUST	nn	Customer number
ROUTE	0–511	Route number
DES	x...x	Designator field for trunk. Enter a description.
TKTP	MUS	MUSIC trunk data block requires Music package 44.
ICOG	OGT	Incoming and outgoing trunk
ACOD	nnn	Access code
REQ	END	Exit form overlay

Associating a MUS route with TNs in LD 14

Before you begin

- You know the user ID and password to log on to the PABX administration terminal.
- You are familiar with PABX Change and Diagnostics overlays.
- You have obtained a listing of routes using LD 21 (REQ=prt, TYPE=RDB).
- At the PABX, you ensured that physical trunks are defined for the route.
- You configured the trunk routes. See [Configuring a MUS route with LD 16](#) on page 74.

About this task

No special programming is required to work with Contact Center Manager Server. Use LD 16 to configure the trunk routes, and use LD 14 to associate the trunk routes with TNs.

To generate reports on trunk routes, you must configure the trunk routes on Contact Center Manager Server. For more information, see *Avaya Aura® Contact Center Server Administration* (NN44400-610).

Procedure

1. Connect to the Call Server.

2. Enter **LD 14**.
 3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, enter four asterisks (****) and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	NEW	Action request
TN	I s c u	Terminal number
TYPE	MUS	Music route
CUST	0	Customer number
RTMB	0–127 1–254	Route and member number
CFLP	0–159	Conference loop
REQ	END	Exit from overlay

Chapter 7: Meridian Link Services configuration

This chapter describes how to configure Meridian Link Services (MLS) for operation with Contact Center Manager Server (CCMS). MLS is a communications facility that provides an interface between a host application and the PABX. A host is any computer on which the third-party application runs. This interface integrates the computer and the Private Automatic Branch Exchange (PABX). In this integrated environment, the host processor interacts with the PABX by exchanging application layer messages.

You can also use MLS to develop applications to use information from the PABX (such as Caller ID), connect to another application to obtain customer information, and then provide data to a PC to help agents prepare for the call.

To use CTI functions for your Contact Center Manager Server agents, you must use MLS.

Meridian Link Services configuration procedures

About this task

This task flow shows you the sequence of procedures you perform to configure MLS. To link to any procedure, see [Meridian Link Services configuration navigation](#) on page 78

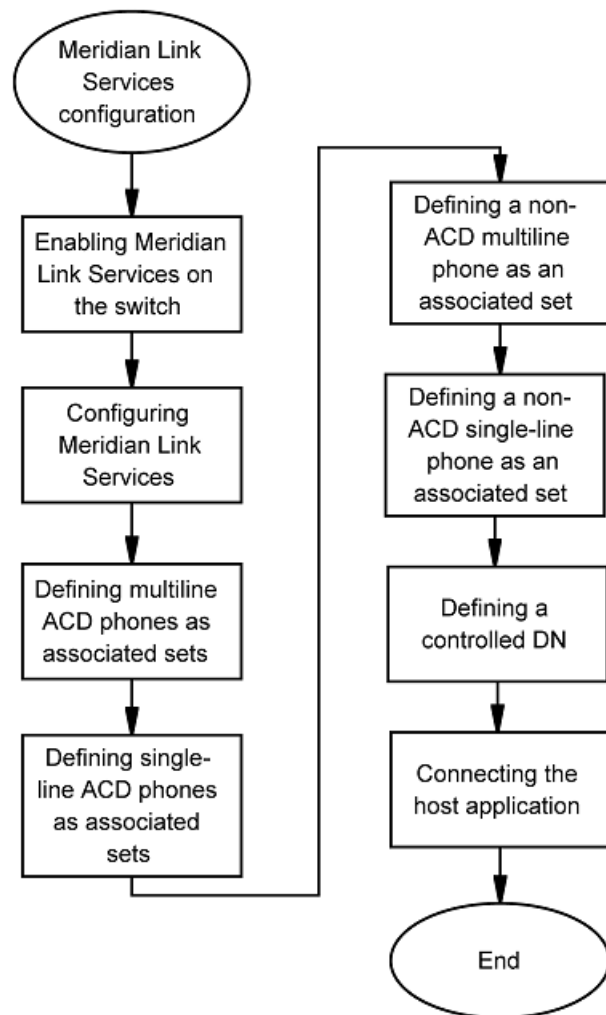


Figure 3: Meridian Link Services configuration procedures

Meridian Link Services configuration navigation

- [Enabling Meridian Link Services on the PABX](#) on page 79
- [Configuring Meridian Link Services](#) on page 79
- [Defining multiline ACD phones as associated phones](#) on page 80
- [Defining single-line ACD phones as associated phones](#) on page 81
- [Defining a non-ACD multiline phone as an associated phone](#) on page 82
- [Defining a non-ACD single-line phone as an associated phone](#) on page 83

- [Defining a controlled DN](#) on page 84
- [Connecting the host application](#) on page 86

Enabling Meridian Link Services on the PABX

Before you begin

- Install Contact Center Manager Server.
- Install the CS 1000 PABX software.

About this task

Configure your Avaya Communication Server 1000 to support Meridian Link Services (MLS)

Procedure

1. Allow CTI operations on the ELAN subnet (LD 17).
2. Configure phones for CTI (LD 11 and LD 10).
3. Configure CDNs for Host-Enhanced Routing (LD 23).

Important:

After you configure the PABX, you must connect the host applications to the Contact Center subnet.

Configuring Meridian Link Services

Before you begin

- Install Contact Center Manager Server (MLS is installed as part of the Contact Center Manager Server software).
- Install the PABX software.

About this task

Configure LD 17 to use CTI with third-party applications. Specifically, where the VAS connection for Contact Center Manager Server is defined as SECU, enter YES as the prompt response. Third-party applications can control phone functions, such as answering or initiating a call.

Procedure

1. Connect to the Call Server.

2. Enter **LD 17**.
 3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, enter four asterisks (****) and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	CHG	Change data in the database.
TYPE	VAS	Value added service
VAS	NEW	When migrating to Meridian Link Services, define the VAS connection as YES.
VSID	xx	Associate link and VSID so that the messages can be sent.
ELAN	yy	Associate VASID xx with ELAN yy.
SECU	NEW	If the same ELAN subnet link is used for the Meridian Link Services application.
INTL	1–12	Time interval for checking Meridian Link Services for overload in increments of 5 seconds.
MCNT	5–100000	Message count threshold for number of Meridian Link Services messages for each time interval.

Defining multiline ACD phones as associated phones

About this task

To enable a multiline ACD phone as an associated set (AST), configure it by using LD 11.

Procedure

1. Connect to the Call Server.
2. Enter **LD 11**.

3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, enter four asterisks (****) and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	NEW	Add new phone.
TYPE	aaa	Enter telephone type as appropriate.
TN	l s c u	Terminal Number
DES	Name	Enter a name for the phone.
CUST	0–99	Customer number
KLS	1–7	Number of key/lamp strips attached
KEY 0	ACD xxxx yyyy	Where xxxx = ACD-DN, and yyyy = Agent position ID
KEY 1	NRD	Not ready
KEY 2	A06	Six-party conference
KEY 3	MSB	Make set busy
KEY 4	TRN	Transfer
KEY 8	SCR xxxx	xxxx=IDN
KEY 13	ACNT	Activity key
REQ	NEW, END	Either define another multiline ACD phone, or exit the overlay saving all of the changes entered.

Defining single-line ACD phones as associated phones

About this task

To enable a single-line ACD phone as an associated set (AST), configure it by using LD 10. Avaya IVR voice ports are configured as analog voice ports.

For more information, see [Avaya Communication Server 1000 Configuration](#) on page 53.

Procedure

1. Connect to the Call Server.
 2. Enter **LD 10**.
 3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, enter four asterisks (****) and press **Enter**.
-

Variable definitions

Prompt	Response	Description
AST	YES	Associated phone
AACD	YES	Associated phone for ACD agent

Defining a non-ACD multiline phone as an associated phone

About this task

To enable a multiline non-ACD phone as an associated set (AST), configure it by using LD 11. Avaya CallPilot® voice ports are configured as digital voice ports.

Procedure

1. Connect to the Call Server.
 2. Enter **LD 11**.
 3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, enter four asterisks (****) and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	NEW	Add new phone.
TYPE	aaa	Enter telephone type as appropriate.
TN	l s c u	Terminal Number
CDEN	SD, (DD), 4D	Card density
CUST	0–99	Customer number
KLS	1–7	Number of key or lamp strips attached
		Press Enter until the AST prompt appears.
AST	00	DN key with AST telephone assignment (Host controllable. You can assign up to two DN keys as AST). This example shows that key 0 is an AST DN.

Defining a non-ACD single-line phone as an associated phone

About this task

To enable a non-ACD single-line phone as an associated set (AST), configure it by using LD 10.

Procedure

1. Connect to the Call Server.
 2. Enter **LD 10**.
 3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, enter four asterisks (****) and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	NEW	Add new phone.
TYPE	500	Single-line telephone type
TN	I s c u	Terminal Number
CDEN	SD, (DD), 4D	Card density
CUST	0–99	Customer number
DN	xxxx	DN for the telephone number
AST	YES	Phone is designated as an associated set.
IAPG	Enter	AML link status message group, defined in LD 15, is not used for MLS.
		Press Enter until the REQ prompt appears.
REQ	NEW, END	Either define another single-line ACD phone as an AST, or exit the overlay and save all changes.

Defining a controlled DN

About this task

Define a controlled DN (CDN) for host-enhanced routing using LD 23.

Host-enhanced routing is an MLS feature that a third-party application uses to control calls that wait at a Controlled DN (CDN).

MLS cannot use a CDN that is already acquired (controlled) by Contact Center Manager Server for host-enhanced routing; however, an application can register for a CDN so that it can receive messages about calls handled at the CDN.

Procedure

1. Connect to the Call Server.
2. Enter **LD 23**.

3. Enter the appropriate values. For prompts not listed, press **Enter** to accept the default.
 4. To exit the overlay, enter four asterisks (****) and press **Enter**.
-

Variable definitions

Prompt	Response	Description
REQ	aaa	Request.
TYPE	CDN	Type of data block is CDN.
CUST	xx	Customer number associated with this data block.
CDN	xxxx	Control Directory Number.
FRRT	0–511	First RAN route number for ACD.
FRT	0–2044	First RAN time.
SRRT	0–511	Second RAN route number for ACD.
SRT	0–2044	Second RAN time.
FROA	NO (YES)	First RAN on arrival.
MURT	0-511	Music route number.
DFDN	x . . . x	Local default ACD-DN.
CEIL	0–(2047)	CDN ceiling value.
OVFL	(NO) YES	Force Overflow Tone to the call when the ceiling threshold is exceeded.
TDNS	(NO) YES	Is DNIS number an original called party.
RPRT	(YES) NO	Management reporting and status display.
CNTL	(NO) YES	CDN is in control.
VSID	0–15	Value Added Server ID.
HSID	0–15	Host ID.
CWTH	0–(1)–2047	Call Waiting Threshold.
BYTH	(0)–2047	Busy Threshold.
OVTH	0–(2047)	Overflow Threshold.

Prompt	Response	Description
STIO	0, 1, 2, . . . 15	Status Input/Output devices.
TSFT	0–(20)–510	Telephone Service Factor Threshold in seconds.
ACNT	xxxx	Account.

Connecting the host application

The host application connects on the Contact Center subnet. You must configure the application to access the Contact Center subnet IP address. The application must point to a CLAN IP address.

Chapter 8: Avaya CallPilot® configuration

To use Avaya CallPilot® as your voice-processing system, you must update the Avaya CallPilot® server configuration:

- Identify channels that provide ACCESS and IVR services to the server in Contact Center Manager Server.
- Define Service Directory Numbers (SDNs) for use by Contact Center Voice Services.

! Important:

You must configure the channels that provide ACCESS and IVR services to Contact Center Manager Server as dedicated Voice channels. Do not use Fax or Speech Recognition channels.

Avaya CallPilot® configuration procedures

About this task

This task flow shows you the sequence of procedures you perform to configure Avaya CallPilot®. To link to any procedure, see [Avaya CallPilot® configuration navigation](#) on page 88.

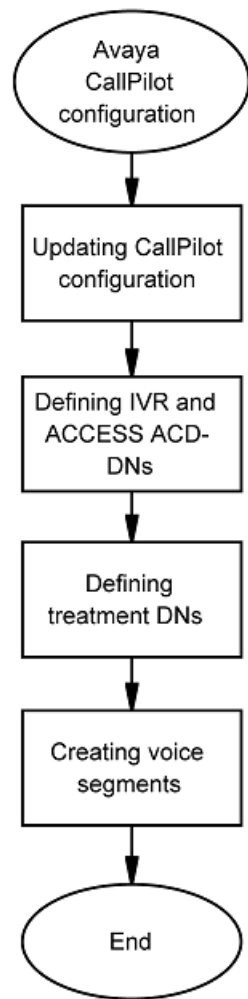


Figure 4: Avaya CallPilot® configuration procedures

Avaya CallPilot® configuration navigation

- [Updating Avaya CallPilot® configuration](#) on page 89
- [Defining IVR and ACCESS ACD-DNs](#) on page 90
- [Defining treatment DNs](#) on page 90
- [Creating voice segments](#) on page 91

Updating Avaya CallPilot® configuration

Before you begin

- Read *Communication Server 1000 System and CallPilot Server Configuration* (NN44200-312).
- Read *Avaya CallPilot® Planning and Engineering Guide* (NN44200-200).

About this task

Update the Avaya CallPilot® configuration to ensure proper function of the upgraded version.

In the CallPilot Configuration Wizard, perform the following tasks:

- Define the Contact Center subnet address of Contact Center Manager Server in the Contact Center Server CLAN IP Address box.
- Identify channels that provide ACCESS and Give IVR services to Contact Center Manager Server.

Procedure

1. On the Avaya CallPilot® server, start the Configuration Wizard.
 2. Advance to the Switch Information page.
 3. In the Contact Center **Server CLAN IP Address** box, type the Contact Center subnet address of Contact Center Manager Server. The Contact Center subnet address was previously known as the CLAN address.
 4. On the left side of the page, click the link for the channels you want to configure. The Channel Name column displays the channels on the selected link.
 5. In the **Channel Name** column, click the first channel that you want to configure. The Channel Detail Information page appears.
 6. For each TN used to provide IVR services to Contact Center Manager Server, select the **IVR** check box.
 7. For each TN used to provide ACCESS services, select the **ACCESS** check box, and specify a class ID.
The class ID is used to communicate between the server and CallPilot over the ACCESS link. When you define the TN as a voice port on Contact Center Manager Server, make sure that the channel number you assign to the voice port matches the class ID for the TN.
 8. Click **Fill**.
 9. Click **OK**.
-

Defining IVR and ACCESS ACD-DNs

About this task

You must define Service DNs for use by Contact Center Voice Services. Define the Contact Center Manager Server IVR and ACCESS ACD-DNs, as well as any treatment DNs.

Procedure

1. Start CallPilot Manager.
 2. Choose **System, Service Directory Number**.
 3. Click **New**.
The SDN Details window appears.
 4. In the **Service DN** box, enter the Contact Center Manager Server IVR and ACCESS ACD-DN numbers, as defined on the PABX.
 5. From the **Application Name** list, select Contact Center **Voice Services**.
 6. From the **Media Type** list, select **Voice**.
 7. Click **Save**.
 8. Go to Contact Center Manager Administration **Launchpad, Configuration**.
 9. On the system tree, double-click the appropriate server.
The server expands to reveal the resources.
 10. Select the **Global Settings** folder.
The **Global Settings** window appears.
 11. Select the **Default Access IVR DN**.
 12. Click **Submit**.
-

Defining treatment DNs

About this task

You must define Service DNs for use by Contact Center Voice Services. Define the Contact Center Manager Server IVR and ACCESS ACD-DNs, as well as any treatment DNs.

Procedure

1. Start CallPilot Manager.

2. Choose **System, Service Directory Number**.
 3. Click **New**.
 4. In the **Service DN** box, enter the treatment DN.
 5. From the **Application Name** list, select the Application Builder application name.
 6. From the **Media TypeMedia Type** list, select **Voice**.
 7. Click **Save**.
-

Creating voice segments

About this task

Use voice segments as building blocks to create powerful, flexible voice applications. If you use voice-processing commands (specifically, the Play Prompt element), you must define voice segments.

Voice segment file names are case-sensitive. Enter voice segments included in scripts exactly as they appear in the Voice Prompt Editor.

Procedure

1. Log on to Contact Center Manager Server.
2. Go to **Server Utility, System Administrator, Voice Prompt Editor, Create a Voice File**.
3. Start the Voice Prompt Editor and create the appropriate prompts.
4. Go to Contact Center Manager Administration **Launchpad, Scripting**.
5. In the system tree, click the server on which you want to view scripts. The server expands to reveal a series of folders.
6. Click the Script Variables folder.
The Script Variables tree expands to show all script variable types.
7. Right-click on the **VOICE_SEGMENT** folder, and select **New**.
8. Below the table, click **Script Variable Properties**.
The heading expands to reveal the General and Attribute tabs.
9. On the **General** tab, in the **Name** box, type the name of the new variable.
10. Click **Call Variable** if the variable is for one specific call only; otherwise, click **Global Variable** to use the variable in all scripts.
11. In the **Comment** box, type any comments you want to save with the variable.
12. Click the **Attribute** tab.

13. From the **Language** list, select a language.
 14. Type in a value in the format filename:segment number.
 15. Click **Add** to add it to your list of values.
 16. Click **Submit** to save your data.
The new variable appears in the system tree in the folder corresponding to the variable type.
-

Part II: Avaya Communication Server 1000 — SIP

- [Avaya Communication Server 1000 configuration for SIP-enabled Contact Centers](#) on page 95
- [Avaya Communication Server 1000 Hotdesking configuration](#) on page 105
- [Avaya Aura[®] System Manager configuration](#) on page 107
- [Avaya Aura[®] Session Manager configuration](#) on page 111

Chapter 9: Avaya Communication Server 1000 configuration for SIP-enabled Contact Centers

This section describes how to configure the Avaya Communication Server 1000 for a SIP-enabled contact center.

For more information about configuring Session Manager to support Avaya Communication Server 1000 Release 7.5, see [Avaya Aura Session Manager configuration](#) on page 111.

Prerequisites to Avaya Communication Server 1000 configuration for SIP-enabled Contact Centers

- Ensure you are familiar with IP Telephony, including SIP Trunking and TR87 CTI on the Avaya Communication Server 1000.
- Engineer the Avaya Communication Server 1000 system so it can support SIP, in particular engineer DSP hardware resources to support TDM/IP transcoding. Incoming PSTN calls require one DSP for each call. Agents with TDM phones each require another DSP. For more information, see *Communication Server 1000M Large System Planning and Engineering* (NN43021-220) and *Communication Server 1000E Planning and Engineering* (NN43041-220).
- Obtain suitable licenses and packages for SIP trunking, T87A Class of Service (CLS), and SIP CTI services.
- Configure SIP trunking and SIP CTI in the same way you configure Converged Office. For more information, see *Converged Office Fundamentals — Microsoft Office Communications Server 2007* (NN43001-121).
- Configure a dialing plan (CDP or UDP) using Element Manager or Avaya Communication Server 1000 overlays. A dialing plan is required to route calls from the Avaya Communication Server 1000 towards the SIP-enabled Contact Center Manager Server. Typically, this configuration occurs through the SIP trunk towards the Network Routing Service (NRS).

Avaya Communication Server 1000 configuration procedures for SIP-enabled Contact Centers

About this task

This task flow shows the sequence of procedures you perform to configure the Avaya Communication Server 1000 for a SIP-enabled contact center.

To link to any task, go to [Avaya Communication Server 1000 configuration navigation for SIP-enabled Contact Centers](#) on page 98.

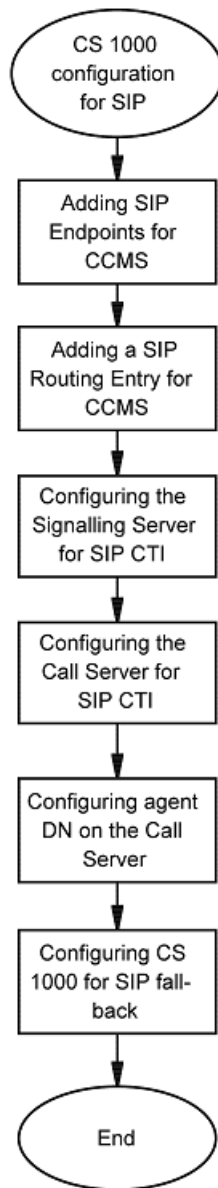


Figure 5: Avaya Communication Server 1000 configuration procedures

Avaya Communication Server 1000 configuration navigation for SIP-enabled Contact Centers

- [Adding SIP endpoints for Contact Center Manager Server](#) on page 98
- [Adding a SIP Routing Entry for Contact Center Manager Server](#) on page 99
- [Configuring the Signaling Server for SIP CTI \(Remote Call Control\)](#) on page 100
- [Configuring the Call Server for SIP CTI \(Remote Call Control\)](#) on page 101
- [Configuring Agent DN with SIP CTI \(RCC\) on the Call Server](#) on page 102

Adding SIP endpoints for Contact Center Manager Server

About this task

Add Contact Center Manager Server as a static endpoint to the Network Routing Services (NRS). Routing entries enable the NRS decide, based on received REQUEST URIs, when to forward calls to the new Contact Center Manager Server endpoint.

Various types of Network Routing Services are available for the Avaya Communication Server 1000, such as SIP Redirect Server (SRS) or SIP Proxy Server (SPS).

Procedure

1. Log on to the Network Routing Services server.
2. In the navigation pane, expand the **Numbering Plan** option.
3. Click **Endpoints > Gateway Endpoints**.
4. Under Managing, click **Standby database**.
5. In the **Limit results to Domain**, select the appropriate domain from the **All service domains** list.
6. In the **Limit results to Domain**, select the appropriate domain from the **All L1 domains** list.
7. In the **Limit results to Domain**, select the appropriate domain from the **All L0 domains** drop-down list. The new Endpoint is added to this domain and the Add button is enabled.
8. Click **Add**.
9. In the **End point name** box, type the name of the Contact Center Manager Server. If you are using the Avaya Aura® Contact Center – High Availability feature and if you have a pair of configured High Availability CCMS servers, type a name for the High Availability campus Contact Center here.

10. In the **Description** box, type a descriptive name for the CCMS in your Contact Center.
 11. In the **Endpoint authentication enabled** list, select **Authentication off**.
 12. Scroll down to find the Static endpoint address field.
 13. In the **Static endpoint address** box, type the IP address of the Contact Center Manager Server. If you are using the Avaya Aura® Contact Center – High Availability feature and if you have a pair of configured High Availability CCMS servers, type the Managed IP address of the High Availability active CCMS server here.
 14. From the **SIP support** list, select Static **SIP** endpoint.
 15. Select **TCP transport enabled**. (UDP may also be selected, TLS is not yet supported).
 16. Click **Save**.
 17. In the left pane, click **System > Database**.
 18. Click **Cut over**.
 19. Click **Commit**.
-

Adding a SIP Routing Entry for Contact Center Manager Server

Before you begin

- Know the dialing plan entry.

About this task

Routing entries ensure the Network Routing Services (NRS) can decide when to forward calls to the Contact Center Manager Server endpoint.

This procedure outlines how to add a SIP Routing Entry for Contact Center Manager Server on the NRS.

Procedure

1. Log on to the Network Routing Services server.
2. In the navigation pane, expand the **Numbering Plan** option.
3. Click **Routes, Gateway Endpoint**.
4. Under **Managing** click on the **Standby database**.
5. In the **Limit results to Domain**, select the appropriate domain from the **All service domains** list.

6. In the **Limit results to Domain**, select the appropriate domain from the **All L1 domains** list.
 7. In the **Limit results to Domain**, select the appropriate domain from the **All L0 domains** list.
 8. Click **Add**.
 9. In the **DN type** list, select the appropriate Dial Plan.
 10. In the **DN prefix** box, type the required prefix to integrate your Contact Center solution into your Dial Plan.
 11. Enter a **Route cost** to support least-cost routing, usually 1.
 12. Click **Save**.
 13. In the left pane, click **System, Database**.
 14. Click **Cut over**.
 15. Click **Commit**.
-

Configuring the Signaling Server for SIP CTI (Remote Call Control)

Before you begin

- The Signaling Server must be reset for the following changes to take effect.

About this task

You can use SIP CTI to control and monitor an Avaya Communication Server 1000 DN, if that DN has the TR87 class of service (CLS) and key 0 or 1 has AST applied.

This procedure outlines configuring the SIP Signaling Server for SIP CTI (Remote Call Control) and enabling the SIP CTI services configuration.

Procedure

1. Log on to Element Manager of the Signaling Server.
2. In the left pane, click **System > IP Network > Nodes: Servers > Media Cards**.
3. Click **Edit**.
4. In the right pane, expand the **SIP CTI Services** heading.
5. Select the **Service Enabled** check box.
6. Clear the **Support TLS Endpoints Only** check box.

7. Under **CTI Settings**, from the **Maximum Associations per DN** list, select the maximum number of associations for each DN.

! Important:

Avaya recommends that you configure this value as 2 or greater.

8. Click **Save** and **Transfer**.

Configuring the Call Server for SIP CTI (Remote Call Control)

About this task

The following example demonstrates how to configure the Call Server for SIP CTI remote call control.

This procedure outlines how to add a new elan and vas for SIP CTI using Avaya Communication Server 1000 overlay 17.

Procedure

```
>ld 17
REQ chg
TYPE adan
ADAN new elan 36 <- Must be over 35 because it is virtual
CTYP elan
DES elan
LCTL
...
ADAN DATA SAVED
ADAN
...
REQ chg
TYPE vas
VAS new
VSID 36
ELAN 36
SECU yes
```

INTL
MCNT
VSID
VAS

Configuring Agent DN with SIP CTI (RCC) on the Call Server

Before you begin

- Create Agent DNs with T87A class of service (CLS) and AST key.
- Multiple DN keys are not supported on any SIP CC installation.

About this task

This procedure outlines how to configure agent DN with SIP CTI remote call control on the Call Server.

SIP CTI is used to control and monitor an Avaya Communication Server 1000 DN with T87A CLS, specified with key 0 or 1 via the AST prompt.

Procedure

Configure the Call Server as shown in the following example:

```
>ld 11
REQ: new
TYPE: 2004p1
TN 160 0 15 29
DES 2004p1
CUST 0
NUID
.....
SFLT
CAC_MFC
CLS t87a <- Use t87a Class of Service for SIP
HUNT
SCI
PLEV
```

DANI

AST 1 <- **May also be 0**

IAPG

ITNA

MLWU_LANG

MLNG

DNDR

KEY 00 scr 50509 <- **Key zero is SCR**

MARP

CPND

VMB

KEY

Chapter 10: Avaya Communication Server 1000 Hotdesking configuration

This section outlines how to configure an Avaya Communication Server 1000 based and SIP-enabled contact center to support agent hotdesking using Virtual Office.

The Avaya Communication Server 1000 Virtual Office feature allows agents to log on to any designated phone and use their individual phone configurations at that phone. Phone calls to the agent are then routed to the phone where the agent is logged on.

For more information on configuring Avaya Communication Server 1000 Virtual Office, see *Avaya Communication Server 1000 Telephones and Consoles Fundamentals* (NN43001-567).

Navigation

- [Logging on to Virtual Office remotely](#) on page 105
- [Logging off of Virtual Office](#) on page 106

Logging on to Virtual Office remotely

Before you begin

- Apply packages 382 and 387 to your Avaya Communication Server 1000.
- On the Avaya Communication Server 1000, Station Control Password Length (SCPL) must be configured in LD 15 to enable the Station Control Password (SCPW) prompt.
- On the Avaya Communication Server 1000, the remote IP phone must have an Station Control Password (SCPW) configured in LD 11.
- The Remote Call Server must have the Virtual Office User Allowed (VOUA) Class of Service (CLS) configured in LD 11.
- The Host Call Server must have the Virtual Office Login Allowed (VOLA) Class of Service (CLS) configured in LD 11.
- You must have a working IP peer configuration and know how to dial the remote phone from the host phone.

About this task

Log on to Virtual Office remotely to use the features of your home office IT phone.

Procedure

1. On the phone, click the **Applications** button.
 2. From the menu, select **Virtual Office**.
 3. At the **Enter User ID** prompt, enter the diallable DN of the target remote phone.
 4. At the **Enter Password** prompt, enter the SCPW of the target remote phone.
If successful, the host phone resets and comes back online with the features of the remote IP phone, including the DN, autodial numbers, feature keys, and voice mail indication and access.
-

Logging off of Virtual Office

About this task

Log off from Virtual Office to remove the features transferred to the phone when you logged on.

Procedure

From the **Options** menu, select **Virtual Office Logout**.
After you log off, the features that were transferred to that phone are removed.

Chapter 11: Avaya Aura® System Manager configuration

Avaya Aura® System Manager delivers a set of shared, secure management services and a common console across multiple products. System Manager includes the following Central Management services:

- User Management: Allows for the administration of users and user groups.
- Communication System Management: Allows for the administration of individual and group stations and mailboxes.
- Routing: Allows for the administration of routing policies for all Session Manager instances within an enterprise.
- Alarm Management Service: Supports alarm monitoring, acknowledgement, configuration, clearing, and retiring.
- Logging Service: Receives log events formatted in the common log format.
- Enterprise Licensing Management Service.

You use Avaya Aura® System Manager to manage Session Manager.

You can also use System Manager to access the Avaya Unified Communications Management (UCM) Web interface. This grants System Manager users access to the Avaya Communication Server 1000 Elements manager.

A central database that resides on the System Manager server stores all the System Manager central data, the Session Manager administration data, and the Central Data Distribution Service information. The Central Data Distribution Service detects changes to the System Manager central database and distribute these changes to the Session Manager instances. All communication between System Manager and Session Manager instances is done over secure links.

Prerequisites

- Ensure that your Avaya Aura® platform meets the minimum template requirements for integration with Contact Center.

Navigation

- [Logging on to the System Manager Web interface](#) on page 108
- [Adding Avaya Unified Communications Management](#) on page 109

Logging on to the System Manager Web interface

Before you begin

- A user account to log on to the System Manager Web interface. If you do not have a user account, contact your system administrator to create your account.

About this task

The System Manager Web interface is the main interface of Avaya Aura® System Manager. You must log on to the System Manager Web console before you can perform any tasks.

Procedure

1. On the browser, type the Avaya Aura® System Manager URL (https://<SERVER_NAME>/SMGR) and press the **Enter** key. Where SERVER_NAME is the name or IP address of your Session Manager server.
 2. In the **User ID** box, type the user name.
 3. In the **Password** box, type the password.
 4. Click **Log On**.
-

Procedure job aid

The System Manager home page displays the main navigation menu. The tasks you can perform using System Manager depends on your user role.

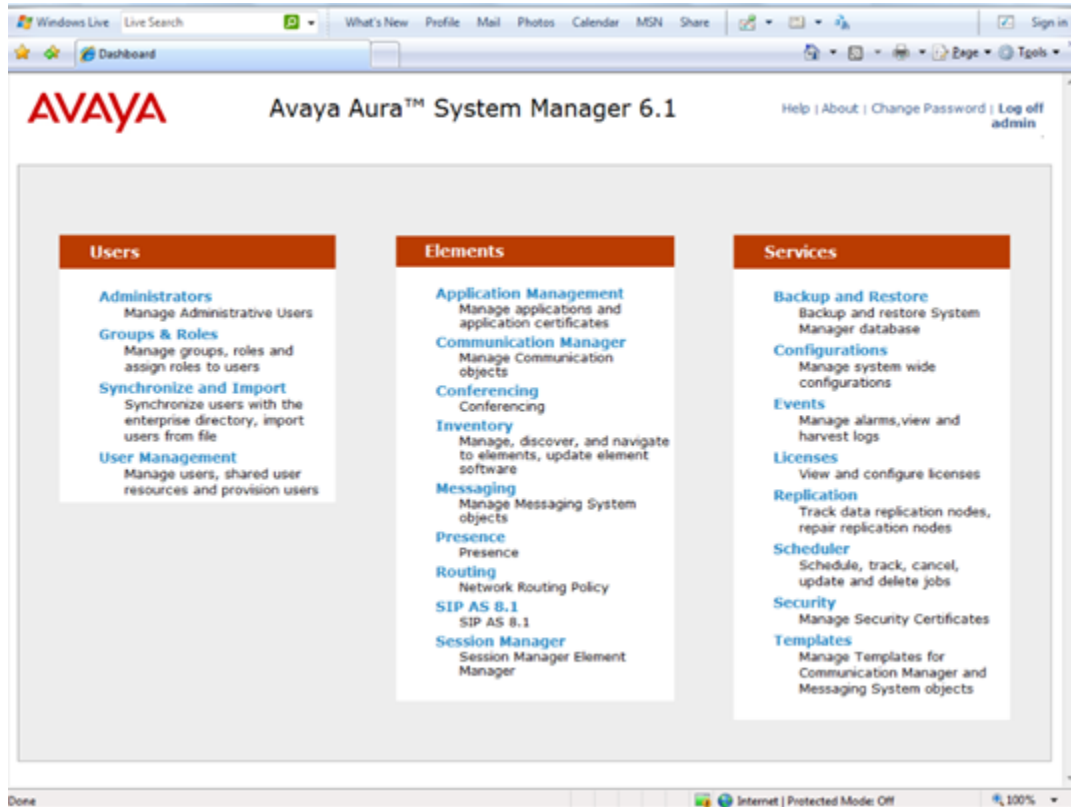


Figure 6: Example of the System Manager Web page

Adding Avaya Unified Communications Management

Before you begin

- Log on to the System Manager Web interface. For more information about logging on to System Manager, see [Logging on to the System Manager Web interface](#) on page 108.

About this task

Add the Unified Communications Management Web interface to the Avaya Aura® System Manager Web interface. This grants System Manager users access to the Avaya Communication Server 1000 Elements manager.

Procedure

1. In System Manager, navigate to **Services, Configurations, Settings, SMGR, Common Console**.
2. In the **UCM Configured** box, type **true**.

3. Click **Done**.

Procedure job aid

The following shows the Unified Communications Management (UCM) services on the bottom right-hand corner of the Avaya Aura® System Manager Web interface.

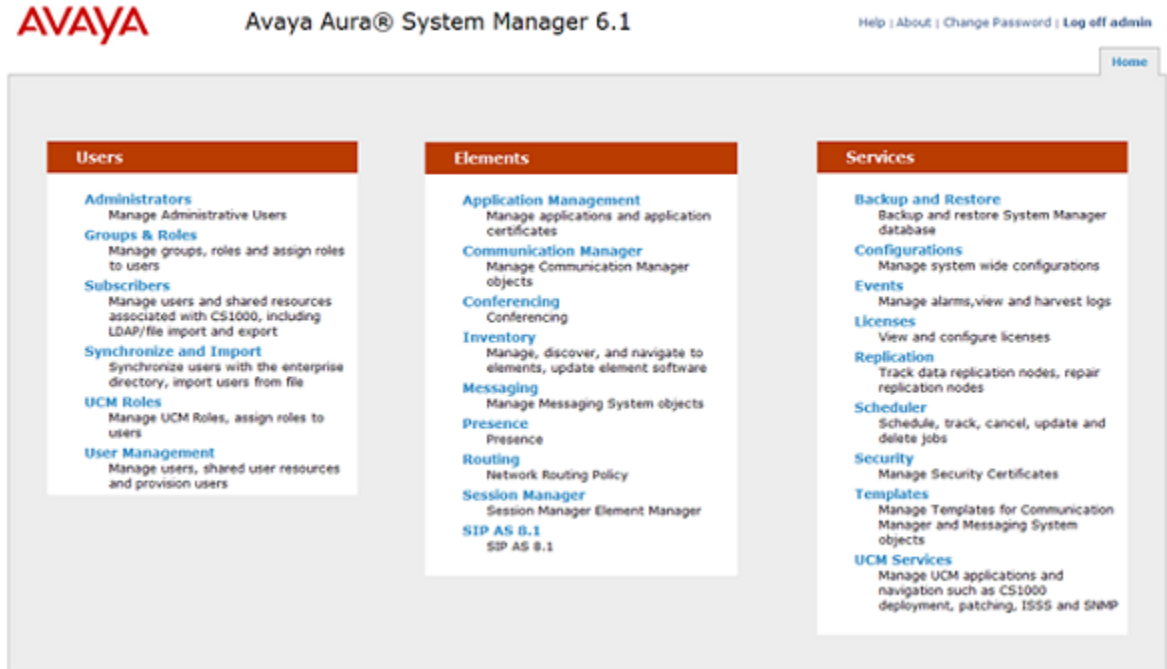


Figure 7: Example of the System Manager Web page with UCM

Chapter 12: Avaya Aura® Session Manager configuration

This section describes how to configure Avaya Aura® Session Manager Release 6.1 or Release 6.2 for use with Avaya Aura® Contact Center Release 6.2 and Avaya Communication Server 1000 Release 7.5 and later.

Session Manager is a SIP routing and integration tool. It integrates all the SIP entities across the entire enterprise network within a company.

Session Manager provides the following functionality:

- Fully SIP redundant core provides active-active N+1 redundancy with up to six Session Manager instances in an enterprise.
- Support for up to 250 SIP survivable branches.
- Enhanced SIP feature set and connects to any IETF compliant SIP solution.
- Support for mixed H.323, DCP, and SIP endpoints at the redundant core and at the survivable branch.
- Simplified network-wide feature deployment.
- Centralized routing, SIP trunking, and user profiles.
- Cost-effective scalability (from small to very large deployments).
- High availability with geographic redundancy.

Session Manager offers a core communication service that builds on existing equipment but adds a SIP-based architecture. Session Manager connects to and acts as a system-wide dial plan for call processing applications such as:

- Avaya Communication Server 1000 Release 7.5 SIP-enabled PBX.
- Communication Manager using direct SIP connections.

You use Avaya Aura® System Manager to configure Session Manager.

SIP devices Session Manager can support up to 50,000 Avaya SIP devices. Session Manager provides SIP proxy, registrar, location services, and more to this initial set of devices.

The following diagram shows a typical Session Manager deployment and configuration. The procedures in this section are based on this typical routing configuration.

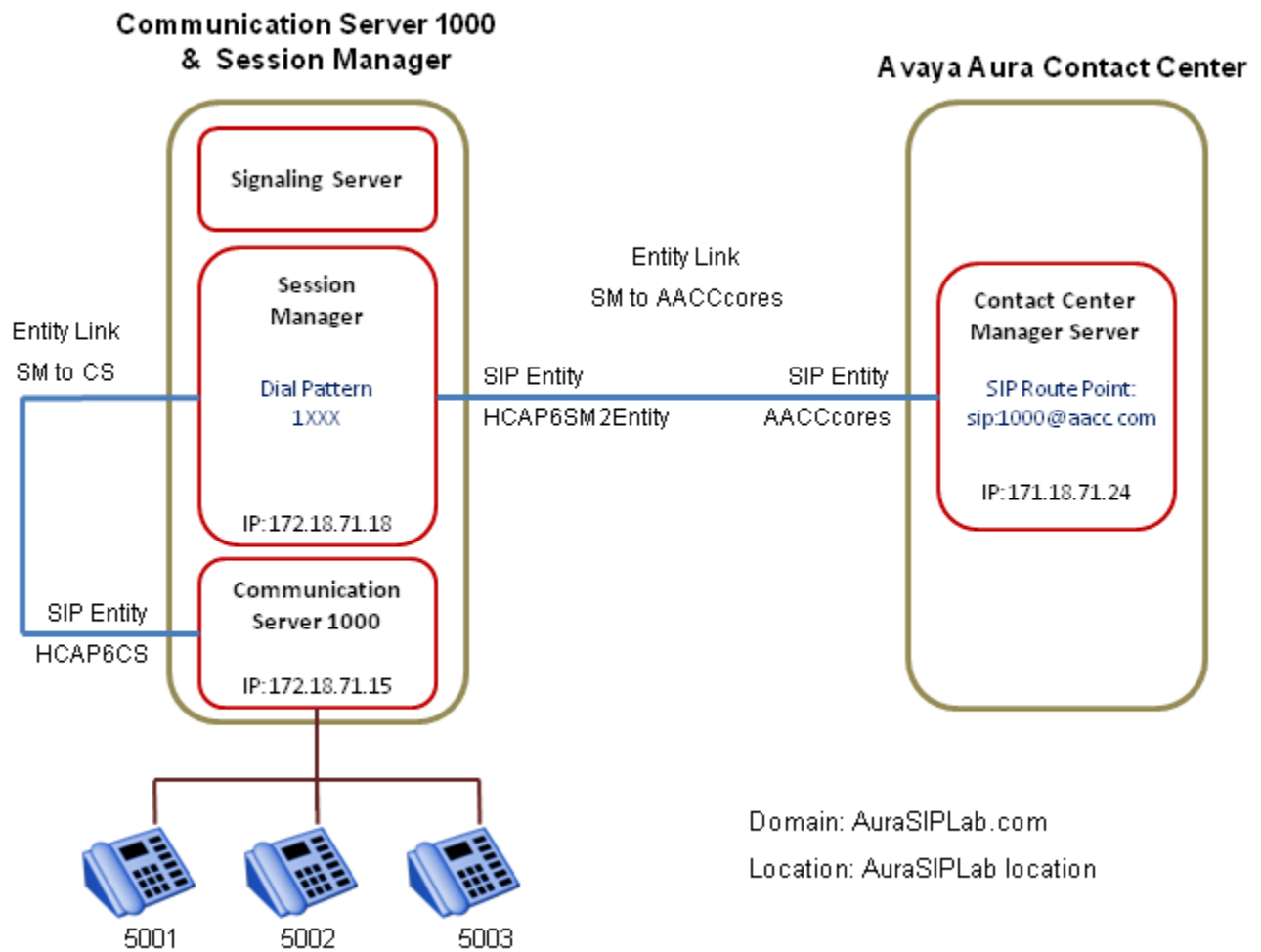


Figure 8: Example of a typical Session Manager routing solution

This typical routing configuration shows how the Session Manager routes calls to Avaya Aura® Contact Center.

Prerequisites

- Ensure that your Session Manager is Release 6.1 or later.

Avaya Aura[®] Session Manager configuration procedures

About this task

This task flow shows the sequence of procedures you perform to configure the Avaya Aura[®] Session Manager.

To link to any task, go to [Navigation](#) on page 114.

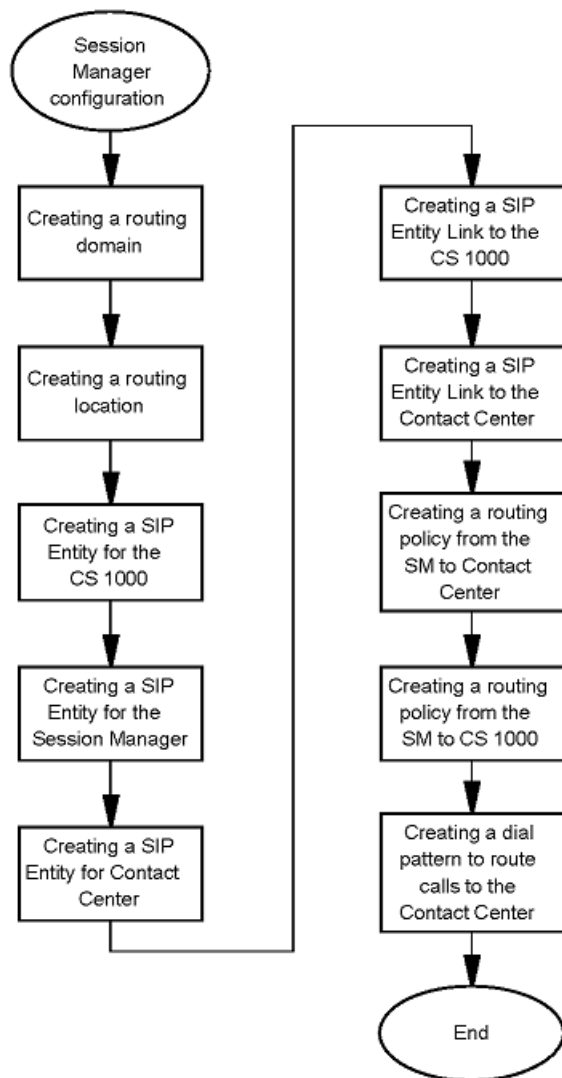


Figure 9: Avaya Aura® Session Manager configuration procedures

Navigation

- [Creating a routing domain](#) on page 115
- [Creating a routing location](#) on page 116
- [Creating a SIP Entity for the Avaya Communication Server 1000](#) on page 118
- [Creating a SIP Entity for the Session Manager](#) on page 121

- [Creating a SIP Entity for the Avaya Aura® Contact Center](#) on page 124
- [Creating a SIP Entity Link to the Avaya Communication Server 1000](#) on page 128
- [Creating a SIP Entity Link to the Avaya Aura® Contact Center](#) on page 130
- [Creating a routing policy from the Session Manager to Contact Center](#) on page 132
- [Creating a routing policy from the Session Manager to Avaya Communication Server 1000](#) on page 133
- [Creating a dial pattern to route calls to the Contact Center](#) on page 135

Creating a routing domain

Before you begin

- Configure Session Manager using Avaya Aura® System Manager. For more information about logging on to Session Manager, see [Logging on to the System Manager Web interface](#) on page 108.

About this task

Create a domain or set of domains. Domains determine if the Session Manager dial plan can be used to route a particular call.

Procedure

1. On the System Manager console, select **Routing, Domains**.
2. Click **New**.
3. On the **Domain Management** page, in the **Name** box, type the domain name. Avaya recommends that you type a descriptive name for your domain.
4. From the **Type** list, select **sip**.
5. In the **Notes** box, type your notes about this domain.
6. Click **Commit**.

Procedure job aid

Example of a Session Manager domain.

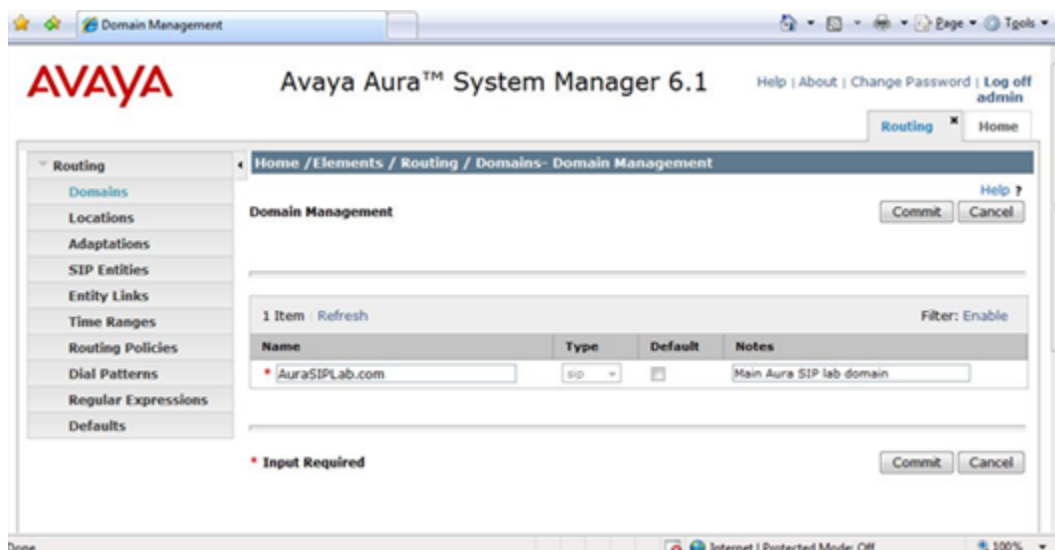


Figure 10: Example of configuring a routing domain

Creating a routing location

Before you begin

- Configure Session Manager using Avaya Aura® System Manager. For more information about logging on to Session Manager, see [Logging on to the System Manager Web interface](#) on page 108.

About this task

Configure the location of your Session Manager. Session Manager uses the origination location to determine which dial pattern to use when routing calls. Locations are also used to limit the number of calls coming out of or going to a physical location.

Procedure

1. On the System Manager console, select **Routing, Locations**.
2. Click **New**.
3. In the **Name** box, type the location name. Avaya recommends that you type a descriptive name for your location.
4. In the **Notes** box, type your notes about this location.
5. From the **Managed Bandwidth Units** list, select **kbit/sec**.
6. In the **Total Bandwidth** box, type your required bandwidth.
7. To add a location pattern, click **Add** under **Location Pattern**.
8. In the **IP Address Pattern** box, type the pattern string to match your system.

9. In the **Name** box, type your notes about this pattern.
10. Click **Commit**.

Procedure job aid

Example of a Session Manager routing location. If your Avaya Communication Server 1000 has an IP address of 172.18.71.15 and if your Session Manager has an IP address of 172.18.71.18, then your IP Address Patterns may be 172.18.71.*. This pattern must cover the addresses that you deem part of this routing location.

The screenshot shows the Avaya Aura System Manager 6.1 web interface. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The 'Locations' option is selected, leading to the 'Location Details' page. The page title is 'Home / Elements / Routing / Locations - Location Details'. The main content area is titled 'Location Details' and includes a 'Help' link, 'Commit', and 'Cancel' buttons. A note states: 'Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth. See Session Manager -> Session Manager Administration -> Global Setting'. The 'General' section contains a 'Name' field with the value 'AuraSIPLab location' and a 'Notes' field with the value 'Main Aura SIP Lab location'. The 'Overall Managed Bandwidth' section shows 'Managed Bandwidth Units' set to 'Kbit/sec' and a 'Total Bandwidth' field. The 'Per-Call Bandwidth Parameters' section shows 'Default Audio Bandwidth' set to '80 Kbit/sec'. The 'Location Pattern' section includes 'Add' and 'Remove' buttons, a 'Filter: Enable' button, and a table with one item: 'IP Address Pattern' with the value '172.18.71.*' and a note 'IP range for main Aura SIP lab'. Below the table is a 'Select: All, None' option. At the bottom, there is a red asterisk indicating 'Input Required' and 'Commit' and 'Cancel' buttons.

Figure 11: Example of configuring a routing location

Creating a SIP Entity for the Avaya Communication Server 1000

Before you begin

- Configure Session Manager using Avaya Aura® System Manager. For more information about logging on to Session Manager, see [Logging on to the System Manager Web interface](#) on page 108.
- Ensure you have created a routing domain. For more information about creating a routing domain, see [Creating a routing domain](#) on page 115.
- Ensure you have create a routing location. For more information about creating a routing location, see [Creating a routing location](#) on page 116.

About this task

Create a SIP Entity for the Avaya Communication Server 1000 (CS 1000). To administer minimal routing via Session Manager, you need to configure a SIP Entity of type SIP Trunk and the Session Manager.

Procedure

1. On the System Manager console, select **Routing, SIP Entities**.
2. Click **New**.
3. In the **Name** box, type the name of the Avaya Communication Server 1000 SIP Entity. Avaya recommends that you type a descriptive name for your CS 1000 SIP Entity.
4. In the **FQDN or IP address** box, type the IP address of the CS 1000.
5. From the **Type** list, select **SIP Trunk**.
6. If you need to specify an Adaptation Module for the CS 1000 SIP entity, from the **Adaptation** list, select an adaptation value.
7. In the **Location** box, select the location for this CS 1000.
8. Enter a regular expression string in the Credential name field. The **Credential name** is used for TLS connection validation by searching for this string in the SIP entity identity certificate.
9. From the SIP Link Monitoring list, select one of the following:
 - Use Session Manager Configuration - Use the settings under Session Manager
 - Session Manager Administration.
 - Link Monitoring Enabled - Enables link monitoring on this SIP entity.
 - Link Monitoring Disabled - Link monitoring is turned off for this SIP entity.

10. If you need to specify the Port parameters, under **Port** click **Add**. When Session Manager receives a request where the host-part of the request-URI is the IP address of the Session Manager, it associates one of the administered domains with the port on which the request was received.
 11. Enter the necessary **Port** and **Protocol** parameters.
 12. Click **Commit**.
-

Variable definitions

Name	Description
Name	SIP entity name. This name must be unique and can have between 3 and 64 characters.
FQDN or IP Address	Fully qualified domain name or IP address of the Avaya Communication Server 1000.
Type	SIP entity type, such as an Avaya Communication Server 1000.
Notes	Additional notes about the SIP entity.
Adaptation	Adaptation to be used for the SIP entity. Select from already defined adaptations.
Location	Avaya Communication Server 1000 SIP entity location. Select from previously defined locations.
Time Zone	Time zone for the SIP entity.
Override Port & Transport with DNS SRV	Specify if you wish to use DNS routing. SIP uses DNS procedures to allow a client to resolve a SIP URI into the IP address, port, and transport protocol of the next hop to contact. It also uses DNS routing to allow a server to send a response to a backup client if the primary client fails.
SIP Timer B/F (in seconds)	Amount of time the Session Manager waits for a response from the SIP entity.
Credential name	Enter a regular expression string in the Credential name. Credential name is used for TLS connection validation by searching this string in the SIP entity identity certificate.

Name	Description
Call Detail Recording	Select or clear the check box to turn SIP monitoring on or off.
Proactive cycle time (Seconds)	Enter a value between 120 and 9000 seconds. The default is 900. This specifies how often the entity is monitored when the link to the entity is up or active.
Reactive cycle time (Seconds)	Enter a value between 30 and 900 seconds. The default is 120. This specifies how often the entity is monitored when a link to the entity is down or inactive.
Number of retries	Enter a value between 0 and 15. The default is 1. This specifies the number of times Session Manager tries to ping or reach the SIP entity before marking it as down or unavailable.
Port	Add a listening port for the SIP entity.
Protocol	Protocol that the SIP entity uses.
SIP Domain	The domain of the SIP entity.
Notes	Additional notes about the port and port parameters.

Procedure job aid

Example of an Avaya Communication Server 1000 SIP Entity.

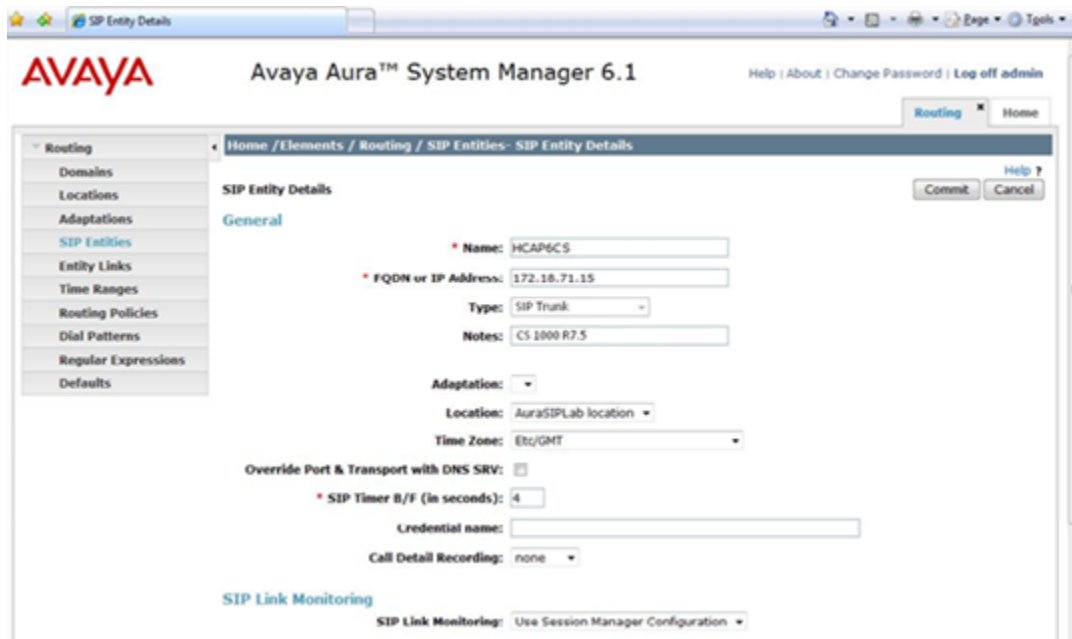


Figure 12: Example of configuring an Avaya Communication Server 1000 SIP Entity

Creating a SIP Entity for the Session Manager

Before you begin

- Configure Session Manager using Avaya Aura® System Manager. For more information about logging on to Session Manager, see [Logging on to the System Manager Web interface](#) on page 108.
- Ensure you have created a routing domain. For more information about creating a routing domain, see [Creating a routing domain](#) on page 115.
- Ensure you have create a routing location. For more information about creating a routing location, see [Creating a routing location](#) on page 116.
- Ensure you have created a SIP Entity for the Avaya Communication Server 1000. For more information about creating an Avaya Communication Server 1000 SIP Entity, see [Creating a SIP Entity for the Avaya Communication Server 1000](#) on page 118.

About this task

Create a SIP Entity for the Session Manager. To administer minimal routing via Session Manager, you need to configure two SIP Entities, a SIP Entity of type SIP Trunk and a SIP Entity of type Session Manager.

Procedure

1. On the System Manager console, select **Routing, SIP Entities**.

2. Click **New**.
 3. In the **Name** box, type the name of the Session Manager SIP Entity. Avaya recommends that you type a descriptive name for your Session Manager SIP entity.
 4. In the **FQDN or IP address** box, type the IP address of the Session Manager.
 5. From the **Type** list, select Session Manager.
 6. If you need to specify an Adaptation Module for the Session Manager SIP entity, from the **Adaptation** list, select an adaptation value.
 7. In the **Location** box, select the location for this Session Manager.
 8. If the SIP entity Type is Session Manager and you need to specify an Outbound Proxy for the SIP entity, click the drop-down selector for the **Outbound Proxy** field.
 9. Enter a regular expression string in the Credential name field. The **Credential name** is used for TLS connection validation by searching for this string in the SIP entity identity certificate.
 10. Under **SIP Link Monitoring**, use the drop-down menu to select one of the following:
 - Use Session Manager Configuration - Use the settings under Session Manager
 - Session Manager Administration.
 - Link Monitoring Enabled - Enables link monitoring on this SIP entity.
 - Link Monitoring Disabled - Link monitoring is turned off for this SIP entity.
 11. If you need to specify the Port parameters, under **Port** click **Add**. When Session Manager receives a request where the host-part of the request-URI is the IP address of the Session Manager, it associates one of the administered domains with the port on which the request was received.
 12. Enter the necessary **Port** and **Protocol** parameters.
 13. Click **Commit**.
-

Variable definitions

Name	Description
Name	SIP entity name. This name must be unique and can have between 3 and 64 characters.
FQDN or IP Address	Fully qualified domain name or IP address of the Session Manager SIP Entity.

Name	Description
Type	SIP entity type, such as a Session Manager.
Notes	Additional notes about the SIP entity.
Adaptation	Adaptation to be used for the SIP entity. Select from already defined adaptations.
Location	SIP entity location. Select from previously defined locations.
Outbound Proxy	Outbound proxy if the entity type is Session Manager, and you wish to specify a proxy.
Time Zone	Time zone for the SIP entity.
Override Port & Transport with DNS SRV	Specify if you wish to use DNS routing. SIP uses DNS procedures to allow a client to resolve a SIP URI into the IP address, port, and transport protocol of the next hop to contact. It also uses DNS routing to allow a server to send a response to a backup client if the primary client fails.
SIP Timer B/F (in seconds)	Amount of time the Session Manager waits for a response from the SIP entity.
Credential name	Enter a regular expression string in the Credential name. Credential name is used for TLS connection validation by searching this string in the SIP entity identity certificate.
Call Detail Recording	Select or clear the check box to turn SIP monitoring on or off.
Proactive cycle time (Seconds)	Enter a value between 120 and 9000 seconds. The default is 900. This specifies how often the entity is monitored when the link to the entity is up or active.
Reactive cycle time (Seconds)	Enter a value between 30 and 900 seconds. The default is 120. This specifies how often the entity is monitored when a link to the entity is down or inactive.
Number of retries	Enter a value between 0 and 15. The default is 1. This specifies the number of times Session Manager tries to ping or reach the SIP entity before marking it as down or unavailable.
Port	Add a listening port for the SIP entity.
Protocol	Protocol that the SIP entity uses.

Name	Description
SIP Domain	The domain of the SIP entity.
Notes	Additional notes about the port and port parameters.

Procedure job aid

Example of a Session Manager SIP Entity.

The screenshot shows the Avaya Aura System Manager 6.1 web interface. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities (selected), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'SIP Entity Details' and 'General'. It contains the following fields:

- Name:** HCAF6SM2Entity
- FQDN or IP Address:** 172.18.71.18
- Type:** Session Manager
- Notes:** Session Manager
- Location:** AuraSIPLab location
- Outbound Proxy:** (empty)
- Time Zone:** Etc/GMT
- Credential name:** (empty)
- SIP Link Monitoring:** Use Session Manager Configuration

Buttons for 'Commit' and 'Cancel' are visible at the top right of the configuration area.

Figure 13: Example of configuring a Session Manager SIP Entity

Creating a SIP Entity for the Avaya Aura® Contact Center

Before you begin

- Configure Session Manager using Avaya Aura® System Manager. For more information about logging on to Session Manager, see [Logging on to the System Manager Web interface](#) on page 108.
- Ensure you have created a routing domain. For more information about creating a routing domain, see [Creating a routing domain](#) on page 115.
- Ensure you have create a routing location. For more information about creating a routing location, see [Creating a routing location](#) on page 116.

- Ensure you have created a SIP Entity for the Avaya Communication Server 1000. For more information about creating an Avaya Communication Server 1000 SIP Entity, see [Creating a SIP Entity for the Avaya Communication Server 1000](#) on page 118.
- Ensure you have created a SIP Entity for the Session Manager. For more information about creating a Session Manager SIP Entity, see [Creating a SIP Entity for the Session Manager](#) on page 121.

About this task

Create a SIP Entity for the Avaya Aura® Contact Center. To administer minimal routing to Avaya Aura® Contact Center, you must configure a SIP Entity of type Session Manager and the Avaya Aura® Contact Center must be configured as type Other.

If your Avaya Aura® Contact Center supports High Availability (HA) then the IP address for the Avaya Aura® Contact Center SIP Entity is the Avaya Aura® Contact Center HA cluster IP address.

If your Avaya Aura® Contact Center does not support High Availability (HA) then the IP address for the Avaya Aura® Contact Center SIP Entity is the Contact Center Manager Server IP Address.

Procedure

1. On the System Manager console, select **Routing, SIP Entities**.
2. Click **New**.
3. In the **Name** box, type the name of the Avaya Aura® Contact Center SIP Entity. Avaya recommends that you type a descriptive name for your Avaya Aura® Contact Center SIP entity.
4. In the **FQDN or IP address** box, type the IP address of the Avaya Aura® Contact Center HA cluster or the IP address of the Contact Center Manager Server.
5. From the **Type** list, select **Other**.
6. If you need to specify an Adaptation Module for the Avaya Aura® Contact Center SIP entity, from the **Adaptation** list, select an adaptation value.
7. In the **Location** box, select the location for this Avaya Aura® Contact Center.
8. Enter a regular expression string in the Credential name field. The **Credential name** is used for TLS connection validation by searching for this string in the SIP entity identity certificate.
9. Under SIP Link Monitoring, use the drop-down menu to select one of the following:
 - Use Session Manager Configuration - Use the settings under Session Manager – Session Manager Administration.
 - Link Monitoring Enabled - Enables link monitoring on this SIP entity.
 - Link Monitoring Disabled - Link monitoring is turned off for this SIP entity.
10. If you need to specify the Port parameters, under **Port** click **Add**. When Session Manager receives a request where the host-part of the request-URI is the IP

address of the Session Manager, it associates one of the administered domains with the port on which the request was received.

11. Enter the necessary **Port** and **Protocol** parameters.
12. Click **Commit**.

Variable definitions

Name	Description
Name	SIP entity name. This name must be unique and can have between 3 and 64 characters.
FQDN or IP Address	Fully qualified domain name or IP address of the Avaya Aura® Contact Center SIP entity. If your Avaya Aura® Contact Center supports High Availability (HA) then the IP address for the Avaya Aura® Contact Center SIP Entity is the Avaya Aura® Contact Center HA cluster IP address. If your Avaya Aura® Contact Center does not support High Availability (HA) then the IP address for the Avaya Aura® Contact Center SIP Entity is the Contact Center Manager Server IP Address.
Type	SIP entity type, such as a Other.
Notes	Additional notes about the SIP entity.
Adaptation	Adaptation to be used for the SIP entity. Select from already defined adaptations.
Location	SIP entity location. Select from previously defined locations.
Outbound Proxy	Outbound proxy if the entity type is Session Manager, and you wish to specify a proxy.
Time Zone	Time zone for the SIP entity.
Override Port & Transport with DNS SRV	Specify if you wish to use DNS routing. SIP uses DNS procedures to allow a client to resolve a SIP URI into the IP address, port, and transport protocol of the next hop to contact. It also uses DNS routing to allow a server to send a response to a backup client if the primary client fails.

Name	Description
SIP Timer B/F (Seconds)	Amount of time the Session Manager waits for a response from the SIP entity.
Credential name	Enter a regular expression string in the Credential name. Credential name is used for TLS connection validation by searching this string in the SIP entity identity certificate.
Call Detail Recording	Select or clear the check box to turn SIP monitoring on or off.
Proactive cycle time (Seconds)	Enter a value between 120 and 9000 seconds. The default is 900. This specifies how often the entity is monitored when the link to the entity is up or active.
Reactive cycle time (Seconds)	Enter a value between 30 and 900 seconds. The default is 120. This specifies how often the entity is monitored when a link to the entity is down or inactive.
Number of retries	Enter a value between 0 and 15. The default is 1. This specifies the number of times Session Manager tries to ping or reach the SIP entity before marking it as down or unavailable.
Port	Add a listening port for the SIP entity.
Protocol	Protocol that the SIP entity uses.
SIP Domain	The domain of the SIP entity.
Notes	Additional notes about the port and port parameters.

Procedure job aid

Example of a SIP Entity for an Avaya Aura® Contact Center.

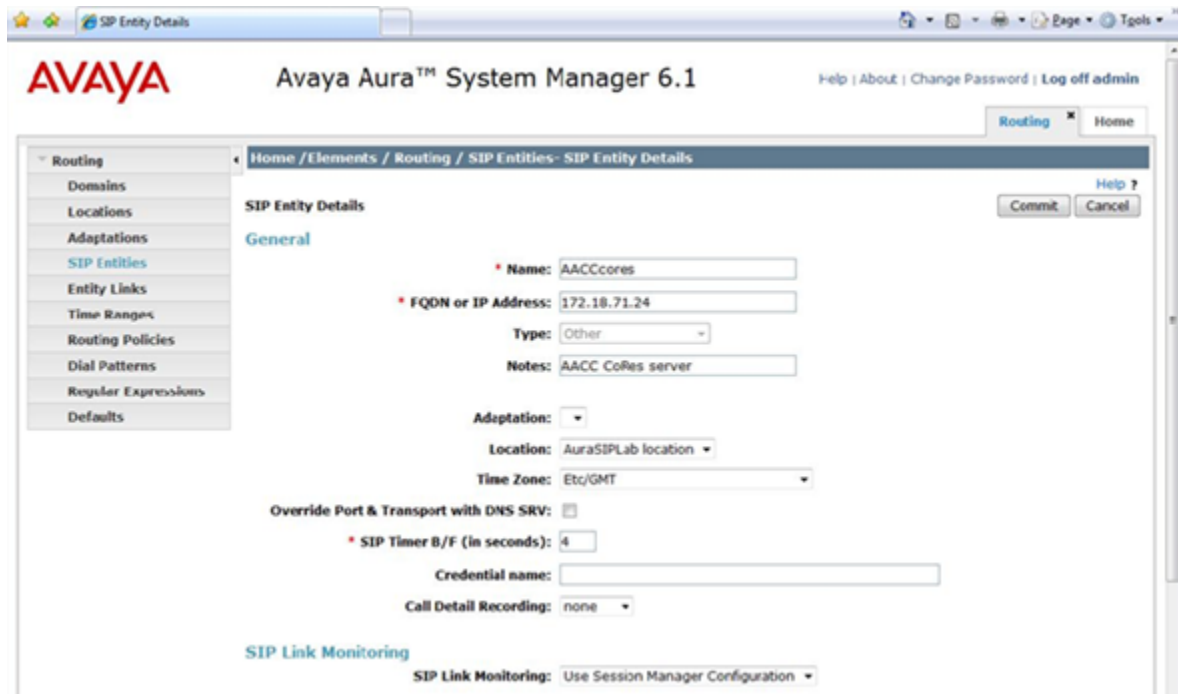


Figure 14: Example of configuring a SIP Entity for an Avaya Aura® Contact Center

Creating a SIP Entity Link to the Avaya Communication Server 1000

Before you begin

- Configure Session Manager using Avaya Aura® System Manager. For more information about logging on to Session Manager, see [Logging on to the System Manager Web interface](#) on page 108.
- Ensure you have a SIP Entity for the Avaya Communication Server 1000. For more information about creating an Avaya Communication Server 1000 SIP Entity, see [Creating a SIP Entity for the Avaya Communication Server 1000](#) on page 118.
- Ensure you have a SIP Entity for the Session Manager. For more information about creating a Session Manager SIP Entity, see [Creating a SIP Entity for the Session Manager](#) on page 121.

About this task

Create a SIP entity link to the Avaya Communication Server 1000. Session Manager enables you to create an entity link between the Session Manager and any other administered SIP entity. You must configure an entity link between a Session Manager and any entity that you have administered if you want Session Manager to be able to send or receive messages from that entity directly. To be able to communicate with other SIP entities, each Session Manager

instance must know the port and the transport protocol of its entity link to these SIP entities in the network.

Procedure

1. On the System Manager console, select **Routing, Entity Links**.
2. Click **New**.
3. In the **Name** box, type the name for this SIP Entity Link. Avaya recommends that you type a descriptive name for your SIP Entity Link.
4. Under **SIP Entity 1**, select the required Session Manager SIP entity from the drop-down list and provide the required port number.
SIP entity 1 must always be an Session Manager instance.
The default port for TCP and UDP is 5060. The default port for TLS is 5061.
5. Under **SIP Entity 2**, select the required Avaya Communication Server 1000 SIP entity from the drop-down list and provide the required port number. The port number is the port on which you have configured the remote entity to receive requests for the specified transport protocol.
6. If the Avaya Communication Server 1000 SIP entity is trusted, select the **Trusted** check box. Session Manager does not accept SIP connection requests or SIP packets from untrusted SIP entities.
7. Click **Commit**.

Procedure job aid

Example of a SIP Entity Link from Session Manager to an Avaya Communication Server 1000.

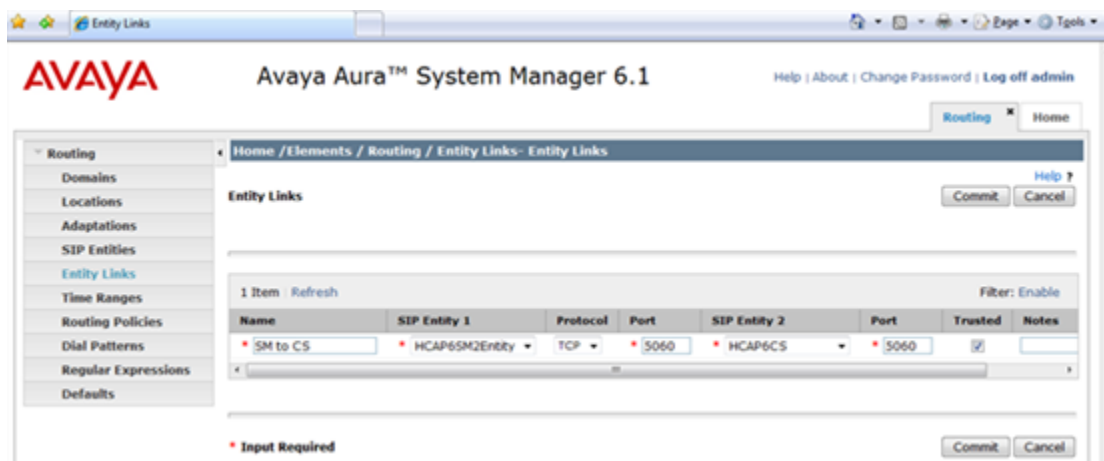


Figure 15: Example of a SIP Entity Link from Session Manager to an Avaya Communication Server 1000

Creating a SIP Entity Link to the Avaya Aura® Contact Center

Before you begin

- Configure Session Manager using Avaya Aura® System Manager. For more information about logging on to Session Manager, see [Logging on to the System Manager Web interface](#) on page 108.
- Ensure you have a SIP Entity for the Session Manager. For more information about creating a Session Manager SIP Entity, see [Creating a SIP Entity for the Session Manager](#) on page 121.
- Ensure you have a SIP Entity for the Avaya Aura® Contact Center. For more information about creating an Avaya Aura® Contact Center SIP Entity, see [Creating a SIP Entity for the Avaya Aura® Contact Center](#) on page 124.

About this task

Create a SIP entity link from Session Manager to the Avaya Aura® Contact Center. Session Manager enables you to create an entity link between the Session Manager and any other administered SIP entity. You must configure an entity link between a Session Manager and any entity that you have administered if you want Session Manager to be able to send or receive messages from that entity directly. To be able to communicate with other SIP entities, each Session Manager instance must know the port and the transport protocol of its entity link to these SIP entities in the network.

Procedure

1. On the System Manager console, select **Routing, Entity Links**.

2. Click **New**.
3. In the **Name** box, type the name for this SIP Entity Link. Avaya recommends that you type a descriptive name for your SIP Entity Link.
4. Under **SIP Entity 1**, select the required Session Manager SIP entity from the drop-down list and provide the required port number.
SIP entity 1 must always be an Session Manager instance.
The default port for TCP is 5060.
5. From the Protocol list, select TCP.
6. Under **SIP Entity 2**, select the required Avaya Aura® Contact Center SIP entity from the drop-down list and provide the required port number. The port number is the port on which you have configured Avaya Aura® Contact Center to receive requests for the specified transport protocol. By default this is port 5060.
7. If the Avaya Aura® Contact Center SIP entity is trusted, select the **Trusted** check box. Session Manager does not accept SIP connection requests or SIP packets from untrusted SIP entities.
8. Click **Commit**.

Procedure job aid

Configure your Session Manager and Avaya Aura® Contact Center as SIP entities and then create a SIP Entity Link between them.

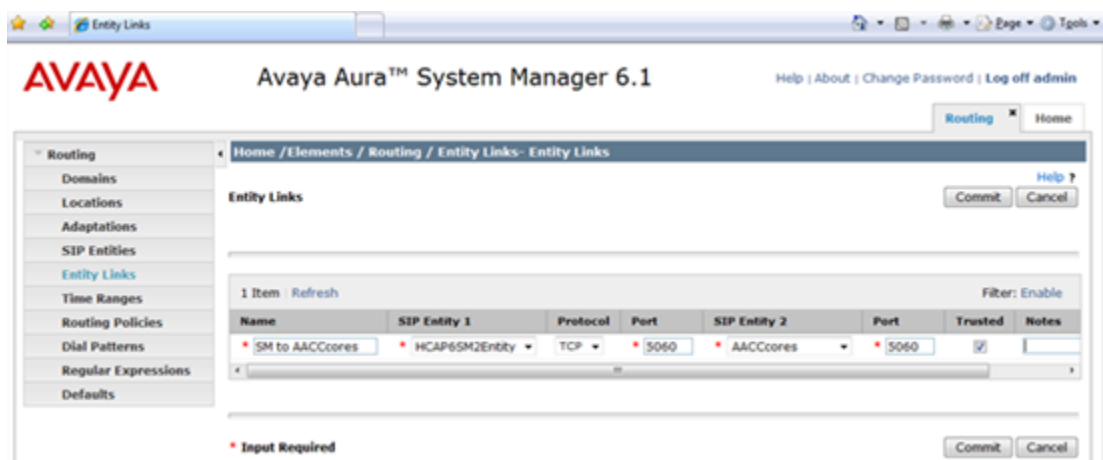


Figure 16: Example of configuring a SIP Entity Link to the Avaya Aura® Contact Center

Creating a routing policy from the Session Manager to Contact Center

Before you begin

- Configure Session Manager using Avaya Aura® System Manager. For more information about logging on to Session Manager, see [Logging on to the System Manager Web interface](#) on page 108.

About this task

Create a routing policy from Session Manager to Avaya Aura® Contact Center. Routing policies can include the “Origination of the caller”, the “dialed digits” of the called party, the “domain” of the called party and the actual time the call occurs. Optionally, instead of “dialed digits” of the called party and the “domain” of the called party a “regular expression” can be defined.

Session Manager uses the data configured in the Routing Policy to find the best match against the number (or address) of the called party.

Procedure

1. On the System Manager console, select **Routing, Routing Policies**.
2. Click **New**.
The Routing Policy Details screen is displayed.
3. In the **General** section, in the Name box, type the name for the Routing Policy.
Avaya recommends that you type a descriptive name for your Routing Policy.
4. In the **Notes** box, type your notes about this Routing Policy.
5. In the **SIP Entities as Destination** section, click **Select**.
6. From the list of SIP Entities, select the SIP Entity for your Avaya Aura® Contact Center, click **Select**.
7. If you need to associate the Time of Day routing parameters with this Routing Policy, click **Add** from the **Time of Day** section.
8. Select the Time of Day patterns that you want to associate with this routing pattern and click **Select**.
9. Click **Commit**.

Procedure job aid

Example of creating a routing policy from Session Manager to Avaya Aura® Contact Center.

Avaya Aura™ System Manager 6.1

Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Routing Policies- Routing Policy Details

Routing Policy Details

General

* Name: SM to AACCcores

Disabled: ☐

Notes: From SM to AACC coresident serve

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
AACCcores	172.18.71.24	Other	

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	☒	☒	☒	☒	☒	☒	☒	00:00	23:59	Time Range 24/7

Select : All, None

Figure 17: Example of configuring a routing policy to Contact Center

Creating a routing policy from the Session Manager to Avaya Communication Server 1000

Before you begin

- Configure Session Manager using Avaya Aura® System Manager. For more information about logging on to Session Manager, see [Logging on to the System Manager Web interface](#) on page 108.

About this task

Create a routing policy from Session Manager to Avaya Communication Server 1000. Routing Policies can include the “Origination of the caller”, the “dialed digits” of the called party, the “domain” of the called party and the actual time the call occurs. Optionally, instead of “dialed digits” of the called party and the “domain” of the called party a “regular expression” can be defined.

Session Manager uses the data configured in the Routing Policy to find the best match against the number (or address) of the called party.

Procedure

1. On the System Manager console, select **Routing, Routing Policies**.
 2. Click **New**.
The Routing Policy Details screen is displayed.
 3. In the **General** section, in the Name box, type the name for the Routing Policy.
Avaya recommends that you type a descriptive name for your Routing Policy.
 4. In the **Notes** box, type your notes about this Routing Policy.
 5. In the **SIP Entities as Destination** section, click **Select**.
 6. From the list of SIP Entities, select the SIP Entity for your Avaya Communication Server 1000, click **Select**.
 7. If you need to associate the Time of Day routing parameters with this Routing Policy, click **Add** from the **Time of Day** section.
 8. Select the Time of Day patterns that you want to associate with this routing pattern and click **Select**.
 9. Click **Commit**.
-

Procedure job aid

Create a routing policy from Session Manager to Avaya Communication Server 1000.

Avaya Aura™ System Manager 6.1

Help | About | Change Password | Log off admin

Routing | Home

Home / Elements / Routing / Routing Policies - Routing Policy Details

Routing Policy Details

Commit Cancel Help ?

General

Name: SM to CS

Disabled: ☐

Notes: From SM to CS

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
HCAP6CS	172.18.71.15	SIP Trunk	

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	☒	☒	☒	☒	☒	☒	☒	00:00	23:59	Time Range 24/7

Select : All, None

Figure 18: Example of configuring a routing policy to Avaya Communication Server 1000

Creating a dial pattern to route calls to the Contact Center

Before you begin

- Configure Session Manager using Avaya Aura® System Manager. For more information about logging on to Session Manager, see [Logging on to the System Manager Web interface](#) on page 108.

About this task

Create a dial pattern using the Session Manager to Avaya Aura® Contact Center Routing Policy. Session Manager uses this dial pattern to route calls to the Contact Center for processing.

A dial pattern specifies which routing policy or routing policies are used to route a call based on the digits dialed by a user which match that pattern. The originating location of the call and the domain in the request-URI also determine how the call gets routed.

Session Manager tries to match the request-URI of a request to a row in the dial pattern table. The rows considered for the match are all rows where:

- The domain in the dial pattern table matches the domain in the request-URI, and
- The originating location in the dial pattern table row matches the originating location of the request, or, if there are no rows matching the originating location, the originating

location in the table is set to -ALL-, or, if there was no originating location, the originating location in the table is -ALL-, and

- The digit pattern in the row matches the user-part of the request-URI, ignoring any parameters that are in the user part of the request-URI

Procedure

1. On the System Manager console, select **Routing, Dial Patterns**.
2. Click **New**.
The Dial Pattern Details screen is displayed.
3. In the **General** section, type the Dial Pattern General information. Note that a **Domain** can be provided to restrict the Dial Pattern to the specified Domain.
4. Under the **Originating Locations and Routing Policies** section, click **Add**.
5. Select all the required Locations and Routing Policies that you want associated with the Dial Pattern by selecting the check box in front of each item.
6. From **Routing Policy Name**, select the Session Manager to Avaya Aura® Contact Center Routing Policy.
7. From the **Routing Policy Destination**, select the Avaya Aura® Contact Center SIP Entity.
8. Click **Select** to indicate that you have completed your selections.
9. If you need to deny calls from a specified location, click **Add** under the **Denied Originating Locations** section.
10. Select all the Locations that are to be denied and click **Select** to indicate that you have completed your selections.
11. Click **Commit**.

Variable definitions

Name	Description
Pattern	Dial pattern to match. The pattern can have between 1 and 36 characters. Roll over the field for the valid pattern.
Min	Minimum number of digits to be matched.
Max	Maximum number of digits to be matched.
Emergency Call	Indicate if it is an emergency call.  Note:

Name	Description
	Some of the important constraints on the use of this feature are as follows — Each location must be assigned to only one emergency dial number. — This emergency dial number must match the emergency dial number in the 96xx Deskphone settings file for all SIP phones in the identified location.
SIP Domain	Domain for which you want to restrict the dial pattern.
Notes	Other details that you wish to add.
Select check box	Use this check box to select and use the digit conversion for the incoming calls.
Location Name	Name of the location to be associated to the dial pattern.
Location Notes	Notes about the selected location.
Routing Policy Name	Name of the routing policy to be associated to the dial pattern.
Routing Policy Disabled	Name of the routing policy that must not be used for the dial pattern.
Routing Policy Destination	Destination of the routing policy.
Routing Policy Notes	Any other notes about the routing policy that you wish to add.

Procedure job aid

Example of creating a dial pattern using the Session Manager to Avaya Aura[®] Contact Center routing policy.

Avaya Aura™ System Manager 6.1
Help | About | Change Password | Log off admin

Routing
Home

Home / Elements / Routing / Dial Patterns - Dial Pattern Details

Dial Pattern Details
Help ↑
Commit
Cancel

General

* Patterns: 10000
* Min: 5
* Max: 5
Emergency Call: ☐
SIP Domain: AuraSIPLab.com
Notes:

Originating Locations and Routing Policies
Add Remove

1 Item Refresh Filter: Enable

☐	Originating Location Name 1 ..	Originating Location Notes	Routing Policy Name	Rank 2 ..	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
☒	AuraSIPLab location		SM to AACCCores	0	☐	AACCCores	

Select : All, None

Denied Originating Locations
Add Remove

0 Items Refresh Filter: Enable

☐	Originating Location	Notes
--------------------------	----------------------	-------

* Input Required

Commit
Cancel

Figure 19: Example of configuring a dial pattern to route calls to the Contact Center

Index

A

ACCESS voice ports	22
ACD phones	25
ACD queue	68
agent phones	25
agent queue changes	28
Avaya CallPilot® configuration	87
Avaya Mentor videos	10

B

Broadcast voice port wait timer	15
broadcasts ports	14

C

CDNs	24
configuring ACD queues	56
controlled DN	84

D

dial pattern	135
digit pattern	135
digital ACD phone	66

E

ELAN subnet link	49
------------------------	--------------------

G

gateway address	50
general parameters	32

H

high timer value	15
home location code	47

I

interactive	19
-------------------	--------------------

interruptible operation	20
IVR ACD-DNs	29

L

LD overlays	25
listen only	19

M

manual caller interaction	16
meridian link services	79
MOV/OUT	29
MUS	75

N

night routing table	62
non-ACD multiline phone	82
non-ACD single-line phone	83
non-interruptible operation	20

P

position ID	28
-------------------	--------------------

R

related resources	10
Avaya Mentor videos	10
routing domain	115
routing location	116

S

session manager support	8
SIP Entity	121
SIP entity link	130
supervisor phone display	26
support	10
contact	10
system parameters	46, 47

T	
treatment DNS	90
U	
user-defined voice segments	22
V	
videos	10
	Avaya Mentor 10
	voice port 64
	voice port partitioning 22
	voice segment variables 22
	voice segments 91
	voice-processing commands 17
	voice-processing methods 16