Communication Server 1000 to Multimedia Communication Server 5100 Converged Desktop Type 2 Configuration Guide Avaya Communication Server 1000

2   Communication Server 1000 to Multimedia Communication Server 5100 Converged Desktop Type 2 Configuration Guide   August 2011

Comments? infodev @avaya.com

Contents

Chapter 1: New in this Release

The following sections detail what's new in *Avaya Communication Server 1000 to Avaya Multimedia Communication Server 5100 Converged Desktop Type 2 Configuration Guide* (NN43001-321) for Avaya Communication Server 1000 (Avaya CS 1000) Release 7.5:

Features

There are no updates to the feature descriptions in this document.

Other changes

See the following sections for information about changes that are not feature-related:

Revision history

August 2011	Standard 05.03. This document was up-issued to support the removal of content for outdated features, hardware, and system types.
November 2010	Standard 05.01 and 05.02. These documents were issued to support Avaya Communication Server 1000 Release 7.5.
June 2010	Standard 04.01. This document is up-issued to support Communication Server 1000 Release 7.0.
May 2009	Standard 03.01. This document is up-issued to support Communication Server 1000 Release 6.0.
December 2007	Standard 02.02. This document is up-issued to support Communication Server 1000 Release 5.5.

June 2007	Standard 01.02. This document is up-issued to remove the Nortel Networks Confidential statement.
May 2007	Standard 01.01. This document is issued to support Communication Server (CS) 1000 Release 5.0. This document contains information previously contained in the following legacy document, now retired: CS 1000 to MCS 5100 Converted Desktop Type 2 Configuration Guide (553-3001-521) () . No new content has been added for Communication Server 1000 Release 5.0. All references to Communication Server 1000 Release 4.5 are applicable to Communication Server Release 5.0. In addition following changes are included: In the section "Feature implementation" a note is added regarding call forwarding on a converged telephone. A note is added regarding the default configuration on-hook Handsfree. A sentence is added regarding a virtual trunk call queued on a CDN, explaining why a DSP resource is used. Two options are added to the section "Configuring MCS 5100 in the NRS for CDS". MCS 5100 is configured as a Gateway Endpoint in the first option and as a Collaborative Server in the second.
October 2006	Standard 3.00. This document is up-issued to add prompt and response to LD-17 Configuration of Value Added Server (VSID) table, to set the VSID security for the SIP CD AML link to No. This prompts the Signaling Server to send a Cancel message to convergeddesktop so that the PC Client does not continue to ring after call is answered by Call Pilot.
April 2006	Standard 2.00. This document is up-issued to update the available MCS 5100 documentation.
September 2005	Standard 1.00. This document is a new document issued to support Communication Server 1000 Release 4.5, and Multimedia Communication Server 5100 Release 3.5.

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Chapter 2: Customer service

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Chapter 3:   Introduction

This document is a global document. Contact your system supplier or your Avaya representative to verify that the hardware and software described are supported in your area.

Subject

This document describes Session Internet Protocol Converged Desktop System (SIP CDS) and its configuration on Avaya Communication Server 1000 (Avaya CS 1000) and Avaya Multimedia Communication Server 5100 (Avaya MCS 5100) systems. It also provides examples of the configuration of a dialing plan and numbering plan for a mixed CS 1000/MCS 5100 network.

This document is describes the design and configuration of a Converged Desktop environment between CS 1000 and MCS 5100 systems. The examples provided in this document represent only one example of design and configuration. <u>Description of the network configured in this document</u> on page 54 provides the outline of the network used as an example only.

Note on legacy products and releases

This technical document contains information about systems, components, and features that are compatible with CS 1000 and MCS 5100 software. For more information on legacy products and releases, click the **Technical Documentation** link under **Support & Training** on the Avaya home page: <u>http://www.avaya.com</u>

Applicable systems

This document applies to the following systems:

- Communication Server 1000M Single Group (CS 1000M SG)
- Communication Server 1000M Multi Group (CS 1000M MG)
- Communication Server 1000E (CS 1000E)

Intended audience

This document is intended for individuals responsible for administering CS 1000 and Meridian 1 systems.

Conventions

Terminology

In this document, the following systems are referred to generically as "system":

- Communication Server 1000E (CS 1000E)
- Communication Server 1000M (CS 1000M)
- Meridian 1

The following systems are referred to generically as "Large System":

- Communication Server 1000M Single Group (CS 1000M SG)
- Communication Server 1000M Multi Group (CS 1000M MG)
- Meridian 1 PBX 61C CP PII, CP PIV
- Meridian 1 PBX 81C CP PII, CP PIV

Related information

This section lists information sources that relate to this document.

Technical documents

The following technical document is referenced in this document:

IP Peer Networking Installation and Commissioning (NN43001-313)

Refer to the following MCS 5100 documentation for additional information about CDS:

- *Provisioning Client User Guide (NN42020-105)*

- *Feature Description Guide (NN42020-125)*

- *Interworking Fundamentals (NN42020-127)*

Online

To access Avaya documentation online, go to:

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Chapter 4: Converged Desktop Services

Contents

This section contains information about the following topics:

Introduction

This section provides an overview of Converged Desktop Services (CDS) to give the user a perspective on what CDS is, how it works, and what it requires. This section also provides an

overview of dialing plan and numbering plan issues for mixed Avaya Communication Server 1000 (Avaya CS 1000) and Avaya Multimedia Communication Server 5100 (Avaya MCS 5100) networks.

Scope of this document

This document is intended to assist technical administrators in designing and configuring a Converged Desktop environment between CS 1000 and MCS 5100 systems. The examples provided in this document represent only one example of design and configuration. Description of the network configured in this document on page 54 provides the outline of the network used as an example only.

How Converged Desktop Services works

Session Initiation Protocol Converged Desktop Service (SIP CDS) is a CS 1000 and MCS 5100 feature. SIP CDS allows users to have simultaneous access to multimedia features on MCS 5100, and to business grade telephony features on CS 1000 systems. The Converged Desktop feature gives users access to business grade telephony features not supported by the SIP standard, while also allowing users to take advantage of the multimedia functionality of MCS 5100. The user's existing telephone is used for telephony functionality while the PC Client delivers the multimedia features.

A Converged Desktop consists of a telephone and multimedia PC Client (PCC) software (see Figure 1: Converged Desktop on page 17). The supported telephones include analog (500 or 2500-type) telephones, digital telephones, and IP Phones. A SIP Phone cannot be configured as a Converged Desktop. A SIP telephone can be configured on the MCS 5100 as a stand-alone device or be tied to a PC Client. In both instances, the SIP telephone receives only features associated with the MCS 5100.
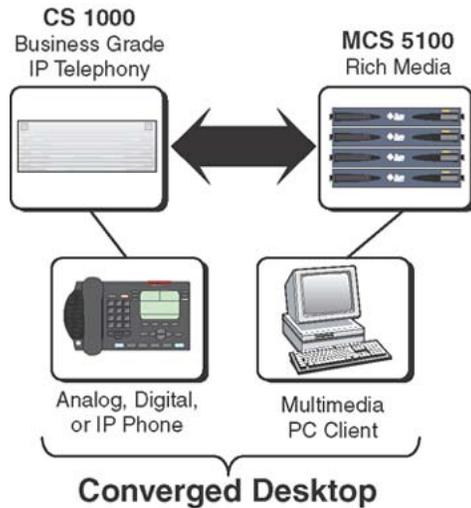
Figure 1: Converged Desktop

In Converged Mode, both the CS 1000 telephone and the MCS 5100 PC Client are integrated to provide both feature sets. In Converged Mode, the CS 1000 telephone effectively becomes the audio device for the MCS 5100 PC Client, so that any audio requirement from the MCS 5100 user (for example, Click-to-Call) is directed to the CS 1000 telephone.

A user may selectively deactivate Converged Mode, so both the MCS 5100 client and the CS 1000 telephone become stand-alone devices. This allows remote users to utilize the MCS 5100 client for both multimedia and voice functionality (for example, using a USB headset with the MCS 5100 PC Client). The voice functionality on a MCS 5100 client in Unconverged Mode is limited to the voice features supported on the MCS 5100. The CS 1000 telephone maintains its full feature set as programmed.

The Converged Desktop feature utilizes the CDS software on the Signaling Server and the SIP Trunking Application to integrate the two systems. These functions became available in CS 1000 Release 4.0.

CDS is enabled between CS 1000 and MCS 5100 using several components, including:

- SIP Trunking Application — SIP Gateway and Virtual Trunk Route

- CDS on the Signaling Server

- Application Module Link (AML) and Value Added Server (VAS) assigned to the CDS application over the common ELAN subnet

- Control Directory Number (CDN) (Converged Services Directory Number [CSDN]) to trigger messaging over the AML/VAS link associated with CDS

- Automatic Call Distribution (ACD) queue with agents — used for the Personal Call Assistant (PCA) application associated with CDS

- telephone Class of Service — used to enable Converged Desktop messaging

- telephone option for CSDN — redirects inbound calls directed to the telephone to CSDN to enable Converged Desktop signaling

- optional Recorded Announcement (RAN) route — used to play progress or delay indication to caller while waiting for PCA agent
- Converged Desktop as a part of the Service Package for the MCS Client user
- dialing plan for CS 1000 and MCS 5100

In general, when a CS 1000 user is designated as a Converged Desktop user, they receive a CS 1000 telephone and an MCS 5100 PC Client. The telephone is programmed with a Class of Service to allow CDS messaging (the options described in CDS configuration: overview on page 53).

When a call terminates to the CS 1000 telephone, the CDMV or CDMO Class of Service redirects the call to the CSDN defined on the telephone, before ringing the CS 1000 telephone. The CSDN is a CDN used for CDS. This CDN number is also configured on the Signaling Server so that it can be acquired by the CDS application when the Signaling Server starts.

When the call terminates to the CDN a message is sent over AML/VAS to the CDS as notification of the incoming call. If the call is a virtual trunk call, and queued on a CDN, a Digital Signal Processor (DSP) resource is used to play the ringback to the caller because CDN is a TDM resource. (A DSP resource is required to handle media between a TDM resource and an IP resource.) The CDS cannot determine where the call terminates, either on the CS 1000 telephone or in the SIP network; the termination location is based on the user's Advanced Screening options in the MCS 5100 Client. Therefore, the CDS instructs the Call Server to make a Virtual Trunk call to a predetermined number, as defined on the Signaling Server (the Service DN used for making a VTRK call from an Agent) and the CS 1000 dialing plan. This call reserves a Virtual Trunk in case it is required to terminate the call to a SIP device (if so configured in the user's Advanced Screening options). After the Virtual Trunk call is established, CDS sends a SIP INVITE message to the MCS 5100 using the SIP Gateway Application.

The MCS 5100 receives the INVITE message and processes it on the user's PC Client. The user's Advanced Screening options are checked for any rules that determine the destination. If no rules exist and the user is in Converged Mode, the MCS 5100 notifies the CDS application to notify the converged telephone. The CDS bridges the call and the Virtual Trunk is dropped. An MCS 5100 call notification screen-pop appears, and the CS 1000 telephone rings. The call is answered on the CS 1000 telephone and call logs are generated on the MCS 5100 PC Client. If the caller is another SIP device or Converged Desktop user, all MCS 5100 features become available for the call.

For outbound calls, the user may use their MCS client in a Click-to-Call scenario or use the keypad of their CS 1000 telephone. When using the MCS 5100 Client, the user has access to all directories available and can process a call. When the user selects a directory entry or dials a telephone number, the MCS 5100 sends an INVITE message to the CS 1000, and CDS processes the call to the CS 1000 telephone. The telephone must be answered before CDS can process the second leg of the call. When the call is answered at the CS 1000 telephone, the destination called and the two calls are bridged together. The PCA application is used for making these calls. The MCS 5100 Client activates call logs. If the destination is another SIP device, the MCS 5100 Client allows multimedia features.

In Converged Mode, incoming and outgoing calls to and from a converged telephone notify the MCS 5100. Upon notification, the MCS 5100 then provides different SIP services and features to the user. The features include the following, depending on where the call is destined, or from where it originated.

All calls to or from a Converged Desktop user receive:

- Call log — Allows a user to see who has called and when the call occurred.

- Real-time call state update — Provides the presence (status) of a user. For example, "active on the phone" or "active available".

- Presence update — Displays the status of other users if the user is added in the Friends list. The Friends list is a list of friends/users that were added (by the user) from the Personal Directory Book, Global Directory Book, or from the call logs. This feature is provided by the MCS 5100 to all PC Clients (including the Converged PCC). Activity on the telephone can update the presence on the PC Client. For example, establishing a voice call to the telephone can update your presence for all registered MCS 5100 users to a presence update such as "Active on the phone". Other presence updates include "Available", "Unavailable", or "Offline".

- Popup window on the converged PC Client — Opens a window when receiving a call. The user sees the caller's name and picture ID (if a picture is available). A popup window also opens when the user calls another Converged Desktop user or calls a stand-alone SIP user.

- Click-to-Call — Allows a user to click a contact from their PC Client and make a call. The call is initiated from the PC Client but the call is placed from the telephone. For more information, see Click-to-Call on page 27.

The following features are available if the call if originated on, or destined for, a SIP client:

- Application sharing — Allows users to share applications such as whiteboard functionality, clipboard transfers, instant messaging, and file transfers.

- Video — Allows users to have a video conversation. The voice portion of the conversation is provided by the telephone and the video portion is provided by the PC Client. It is possible to add video to an already initiated telephone conversation. If in Converged Mode, the video can be started manually from the PC Client. The Converged Desktop application knows from where to pick up the video because the requested video session is bound with the voice conversation. If the voice conversation is dropped, then the video session is automatically closed.

- Web co-browsing — Allows users to share a web browser. If both users have the web co-browsing feature, then one user can automatically control (or drive) the other user's web browser.

All of these features are achieved by sending originating and terminating call notifications to the MCS 5100.

Users in a mixed CS 1000/MCS 5100 network

CS 1000 and MCS 5100 systems can interoperate directly using SIP. Both systems can be deployed in a single mixed customer network. Table 1: Types of users in a mixed CS 1000 and MCS 5100 customer network on page 20 shows the three types of users in this mixed CS 1000 and MCS 5100 customer network.

Table 1: Types of users in a mixed CS 1000 and MCS 5100 customer network

User	Description
CS 1000 user	A CS 1000 user has an account created on the CS 1000 system. The only client available to the CS 1000 user is a client under CS 1000 control. This user does not have an account on the MCS 5100 system and the user cannot use clients under MCS 5100 control.
MCS 5100 user	An MCS 5100 user has an account created on the MCS 5100 system. The only client available to the MCS 5100 user is a client under MCS 5100 control. This user does not have an account on the CS 1000 system. The user cannot use clients under CS 1000 control.
Converged Desktop (CD) user	A Converged Desktop user has two accounts: • a CS 1000 account • an MCS 5100 account The Converged Desktop user also has two available clients: • a client under CS 1000 control (for example, an IP Phone 2004) • a client under MCS 5100 control (the Multimedia PC Client)

A Converged PC Client always uses the CS 1000 client as an endpoint for audio streams (in this configuration, the Multimedia PC Client cannot terminate/originate audio streams). A call itself can be initiated on the CS 1000 client or on the Multimedia PC Client. The Multimedia PC Client uses the "Click-to-Call" mechanism to engage the CS 1000 client in the call.

Note:

When a Converged Desktop user is not in a converged state, the user effectively becomes an MCS 5100 user and the CS 1000 device becomes a stand-alone voice device.

Converged Mode and Unconverged Mode

An MCS 5100 client can be in one of the two following modes:

• Converged Mode — The MCS 5100 client uses the CS 1000 voice path. It also uses the AML link for CDS signaling.

• Unconverged Mode — The MCS 5100 client uses the MCS 5100 voice path. It does not use the AML link for CDS signaling.

The mode is defined when each Client is configured.

Note:
This document assumes that the MCS 5100 client is in Converged Mode. Exceptions for Unconverged Mode are noted.

CDS originating and terminating call handling

The implementation of the Converged Desktop Service includes two parts:

- outgoing call (or a telephone-originated call) notification to the MCS 5100
- incoming call (or a call terminated on a converged telephone) notification to the MCS 5100

Users in Converged Mode can be defined with two Classes of Services for Converged Desktop:

- CDMO = Converged Desktop Multimedia Only — The user has an MCS 5100 PC Client and a CS 1000 desktop telephone. The user has access to all MCS 5100 multimedia features (such as video), except Personal (PA) routing.
- CDMV = Converged Desktop Multimedia and Voice — The user has an MCS 5100 PC Client and a CS 1000 desktop telephone. The user has access to all MCS 5100 multimedia features (such as video), including PA routing.

Making a call

Users can make an outgoing Converged Desktop call in two ways:

- Dial from a CS 1000 telephone.
- "Dial" from an MCS 5100 PC Client (Click-to-Call) in one of the following ways:

  - Press the **Make a call** button.
  - Select from a Call Log.

    A call from the inbox to the PSTN is not supported.
  - Select from a Friends list or Directory book.

In both instances the following functionality is automatically provided:

- The Call Log is updated after the call.
- A pop-up window is generated if another CD user or SIP user was called.
- The users' Presences are updated after the call is answered.

shows the call flow and messaging that occurs when a call is made from a CS 1000 desktop telephone.
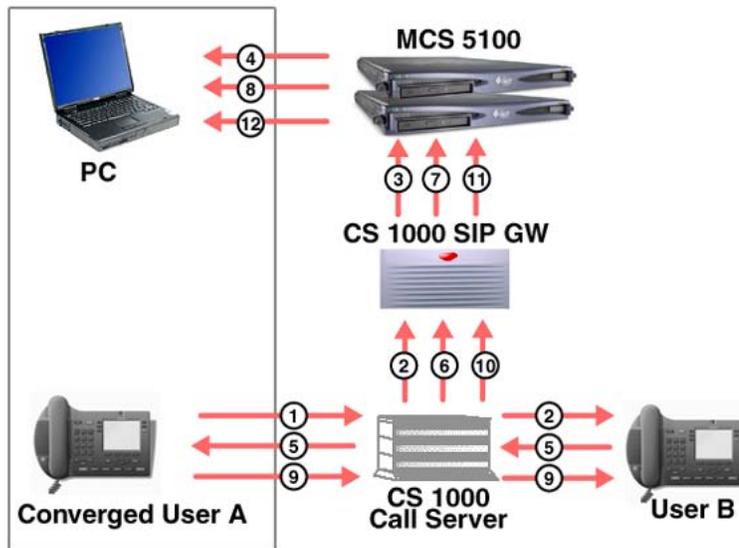
Figure 2: Call flow: making a call from a CS 1000 desktop telephone

Legend:

1   Converged user A dials user B.

2   User B rings; Off-hook message sent.

3   Calling message sent.

4   PCC Call Log updated.

5   User B answers call.

6   Active message sent.

7   Active message sent.

8   Call timer started.

9   Converged user A releases.

10   Call disconnected; Disconnect message sent.

11   Release message sent.

12   Call timer stopped, Call Log updated.

Figure 3: Call flow: Click-to-Call on page 23 shows the call flow and messaging that occurs during a Click-to-Call session.
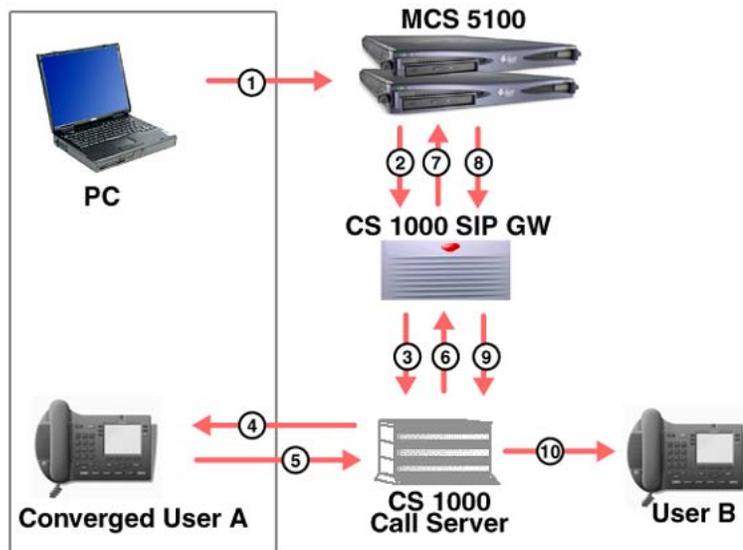
Figure 3: Call flow: Click-to-Call

Legend:

1   Click-to-Call action initiated.

2   Click-to-Call Invite message sent.

3   One idle PCA selected to call own phone.

4   Converged user A rings.

5   User A answers.

6   Initial call is answered.

7   Call is answered.

8   Call made to user B.

9   Call between PCA and converged user A disconnected; call made from converged user A to user B.

10   Call placed to user B.

Receiving a call

Depending on the Class of Service (CDMV or CDMO) of the converged user, the call flow will differ slightly. To receive a call, note the following:

- A CD user in CS 1000 can be defined as Multimedia Only user or Multimedia and Voice user.
- All calls to CD users are sent first to a predefined CDN queue.
- The MCS 5100 forks the call based on the user's PA setting.
- The new Class of Service (CDMO or CDMV) distinguishes the CD user from other users.
- The CSDN prompt in LD10/11 defines the CDN queue.
- A group of PCAs are defined to serve as the agents servicing CDN.
- Use AML messages for call notification and call control.

Figure 4: Call flow: call terminating to a user with a CDMV Class of Service on page 24 shows the call flow and messaging that occurs for a call terminating to a user with a CDMV Class of Service.
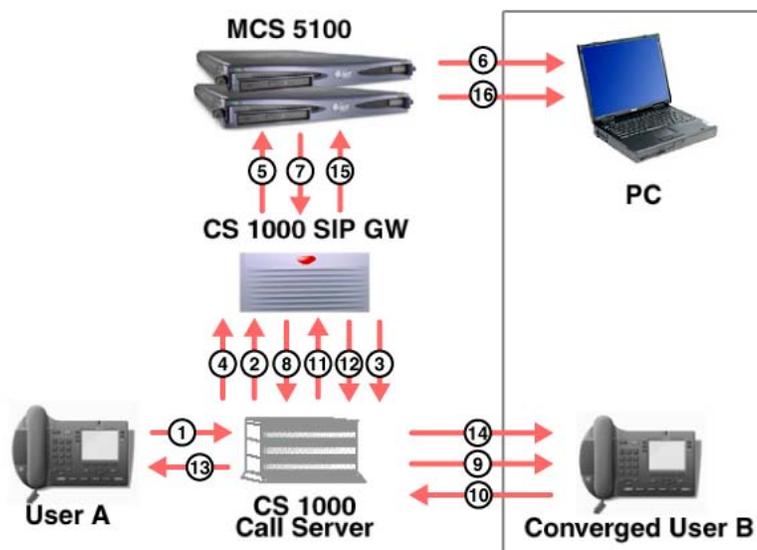
Figure 4: Call flow: call terminating to a user with a CDMV Class of Service

Legend:

1   User A dials user B; call is forwarded to CDN.

2   Incoming call to CDMV user sent to CDS.

3   PCA selected; call made to Service DN to select a Virtual Trunk.

4   Virtual Trunk reserved.

5   INVITE message sent to CDS URL.

6   INVITE PC Client

    • Call Forking based on PA route

    • Call Log updated

    • Pop-up window generated

If PA settings allow a call to converged user B, then:

7   INVITE message sent to converged user B.

8   Select one idle PCA to call converged user B.

9   Calling converged user B.

10   Converged user B answers.

11   Call Answered message sent to CDS.

12   Message sent to Call Server to release Virtual Trunk and to connect user A and converged user B.

13,14   Call is established between user A and converged user B.

15   User B is active.

16   Call Log updated; call timer started.

shows the call flow and messaging that occurs for a call terminating to a user with a CDMO Class of Service.

Figure 5: Call flow: call terminating to a user with a CDMO Class of Service

Legend:

1   User A dials user B; call is forwarded to CDN.

2   Incoming call to CDMO user sent to CDS.

3   PCA selected; call made to Service DN to select a Virtual Trunk.

4   Virtual Trunk reserved.

5   INVITE message sent to CDS URL.

6   INVITE PC Client

   • No Call Forking

   • Call Log updated

   • Pop-up window generated

7   302 Moved Temporarily message sent to CDS.

8   Message sent to Call Server to release Virtual Trunk and to connect user A and
     converged user B.

9   Converged user B called.

10   Converged user B answers call.

11   Call is established between user A and converged user B.

When receiving a call, the functionality of the PC Client is different from that when making a
call. With CDMO, a call to a converged telephone always rings the converged telephone. It

does not invoke PA routing. But with CDMV, a call to a converged telephone on MCS 5100 invokes PA routing to determine if it is to ring the converged telephone or another telephone.

AML ELAN numbers and VAS numbers

The messaging required for CDS to operate between the CS 1000 Call Server and the Signaling Server is achieved by defining two functions:

- Application Module Link (AML) — AML is the definition of a link to provide connectivity between two applications.
- Value Added Server (VAS) — VAS is the Application Layer link carried through the AML link.

A new range of AML/VAS IDs has been created for CDS. To distinguish this AML link from existing AML ELAN links, the AML ELAN link numbers and VAS numbers are:

a range of 32–127 (inclusive) on Large Systems

with numbers greater than 32 reserved for the "logical" AML ELAN link.

These numbers enable multiple CDS gateways and multiple MCS 5100 systems to be supported on CS 1000 systems in addition to other Meridian Link products (such as Symposium, Avaya CallPilot, or IP Call Recording), which also use ELAN link numbers.

The 16 existing AML links are used for physical connections to other applications.

Click-to-Call

The Click-to-Call feature enables users to originate voice calls from their PC Client when in Converged Mode. This feature is not available to CS 1000 telephones not configured for CDS.

✴ Note:

Click-to-Call operation for MCS 5100 Clients in Unconverged Mode may differ from that described.

When the call is established, a voice path is set up between the telephone of the originating converged user and the terminating side. The PC Client is not used to establish the voice path.

The user originates a Click-to-Call call by clicking on a number or address from such things as an address book, a call log, a buddies list, an inbox, an outbox, or by entering an address in an address field on the PC Client.

The following steps must be performed when using Click-to-Call:

- The caller makes a call from the PC Client software and the caller's telephone rings.

- The caller must answer their telephone when it rings if the desktop telephone is an analog (500/2500-type) telephone or is part of a Multiple Appearance DN (MADN) group. Answering the telephone then directs the call to the called party. However, the call is automatically answered if the desktop telephone is a digital telephone or an IP Phone and is a Single Appearance DN.

> ✱ **Note:**
> If the caller is in an MADN group, and the answered phone is configured with the default on-hook path Handsfree, the answered phone rings the called party using handsfree, even if the call is answered using a headset. This occurs because the answered phone makes the new call to the called party using the default on-hook path.

Operating parameters and feature interactions

The following items apply to all Converged Desktop users (including both incoming and outgoing Converged Desktop calls):

- When a Converged Desktop user is a member of a Multiple Appearance Directory Number (MADN) group, the user must be Multiple Appearance Redirection Prime (MARP) to receive the Converged Desktop Service. Or, all users in the same MADN group must be configured with the same Class of Service, so that they receive the same Converged Desktop Service.

- The CDS feature applies only to a primary DN key. Calls made or received on any other keys do not receive the Converged Desktop Service.

- If a call is answered by any non-Converged Desktop telephone, then the popup window, call log, and presence of the Converged Desktop user are removed after a few seconds. The timeout value for this setting is configurable (default value is 2 seconds).

- A Converged Desktop user does not receive a popup window if the caller is an attendant and is using the Attendant Monitor feature. An attendant call terminating to the Converged Desktop does not notify the MCS 5100.

- Group Call does not apply to a Converged Desktop caller. Initiating Group Call from a Converged Desktop user does not notify the MCS 5100.

- MCS 5100 clients can be in Converged Mode or Unconverged Mode. The selected mode determines which functionality is available, as described throughout this document.

- In the Converged Desktop Environment, voice mail is supported only on the CS 1000 telephone, not the MCS 5100 Client.

Call walk-throughs

The figures in this section show messages exchanged between a CS 1000 system and an MCS 5100 system with different configurations. Although two CS 1000 systems appear in the figures, they collapse into one if the call is internal.

The different configurations are:

Figure 6: Outgoing Converged Desktop call
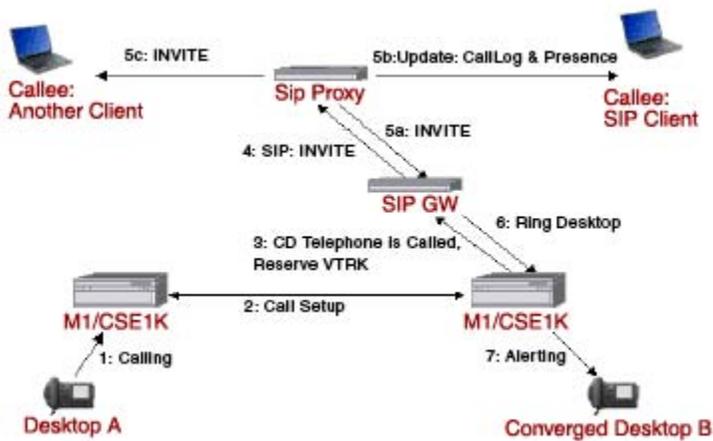
Figure 7: Incoming call to CDMV user with forking
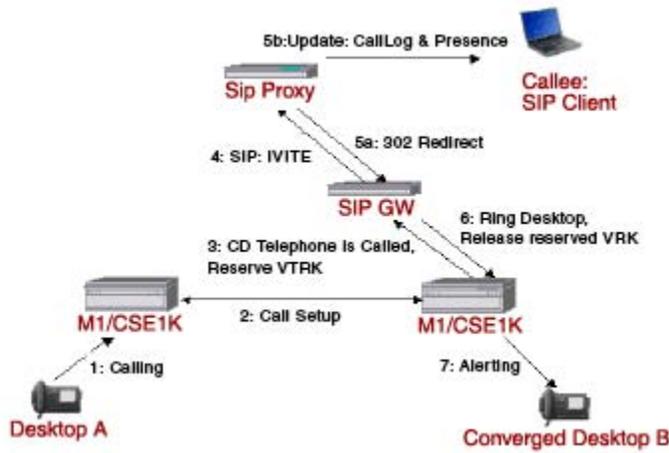
Figure 8: Incoming call to CDMV user with NO forking
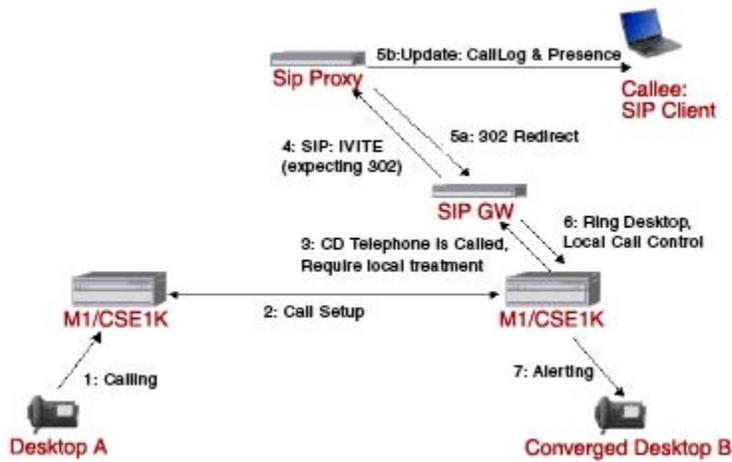
Figure 9: Incoming call to CDMO user
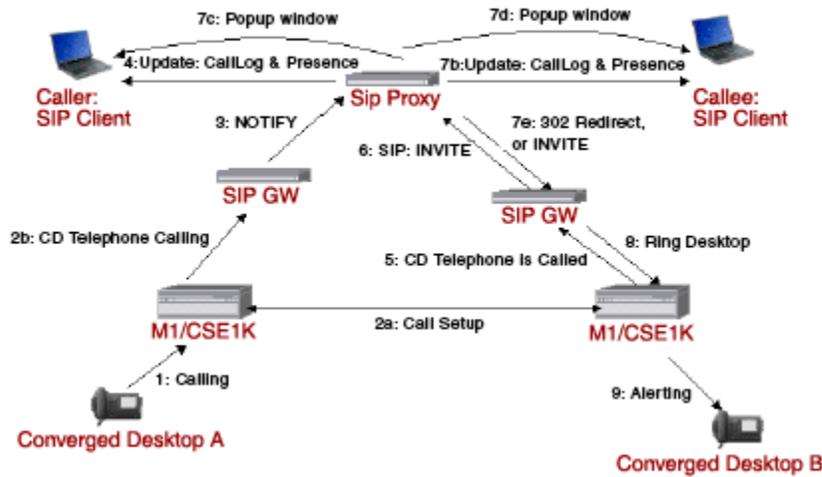
Figure 10: Combined incoming and outgoing Converged Desktop call

Feature requirements

CS 1000 systems

CS 1000 systems require:

- CS 1000 Release 4.0 or later software
- Symposium Call Center (NGCC) (also referred to as Next Generation Call Center) package 311 (Level 2 or 3b)
- SIP Gateway and Converged Desktop Package (SIP) package 406 (Level 2)
- ACD and PCA licenses for PCA Agents
- SIP Access Ports for the CDS application; may also need other SIP Access Ports for MCS 5100 users calling in Unconverged Mode, depending on traffic requirements
- Signaling Server with CDS and SIP Trunk applications enabled

MCS 5100 systems

MCS 5100 systems require MCS 5100 Release 3.5 or later software.

Product Enhancement Packages and Patches

In addition, you must ensure that all required Product Enhancement Packages (PEP) and patches are installed. For a list of the required PEPs and patches, go to http://www.avaya.com.

Feature implementation

For CDS to operate, you must have the following conditions in place:

- MCS 5100 clients must assume the location identifier (LOC) of a CS 1000 system. This enables one MCS 5100 system to provide CDS to many CS 1000 systems, having the MCS 5100 clients assume the location identifier (LOC) of each CS 1000 system.

- The dialing plan and numbering plan between the CS 1000 systems and the MCS 5100 system must be compatible. See Dialing and numbering plan issues for the mixed network on page 33, and Examples of dialing and numbering plan configuration for mixed networks on page 41.

Note:

If the original converged phone is call forwarded to another number, the forwarded phone is used as the terminator, however the Converged Multimedia PC Client call window for the original Converged phone continues to appear.

Implementation summary

In order to use SIP Converged Desktop Services, the SIP Trunk Gateway must first be configured. See *IP Peer Networking Installation and Commissioning (NN43001-313)*.

Configuration of the SIP CDS requires configuration on both the CS 1000 system and the MCS 5100 system. For CS 1000 configuration, refer to Configuring Avaya Communication Server 1000 on page 55. For MCS 5100 configuration, refer to Configuring Avaya Multimedia Communication Server 5100 on page 75.

Dialing and numbering plan issues for the mixed network

In a mixed CS 1000/MCS 5100 network, special attention must be placed on addressing.

If a call is originated by a CS 1000 user and the call must terminate at an MCS 5100 user, the destination address given to the MCS 5100 system must make sense for the MCS translation

engine. The same requirement applies to the call in the opposite direction — a destination address passed to the CS 1000 system by the MCS 5100 must make sense to the CS 1000 translation engine.

Digit-based addresses and alphanumeric addresses

The CS 1000 translation engine supports only digit-based addresses. An example is 5573.

The MCS 5100 translation engine supports both digit-based and alphanumeric addresses. Examples are 3435573 and asmith.

 Important:

Rule 1: All users in a mixed CS 1000/MCS 5100 network must be accessible by all systems through a digit-based address.

On the MCS 5100 system, each username must have an alias that is all digits, or each username must be all digits.

Dialing plan issues

The term "dialing plan" is used in a very broad sense in this context. It includes all the arrangements that determine an exact digit sequence that the user must dial to access a destination. To access a destination, a user must know an appropriate access code (for the type of destination being called) and the unique address of the destination (within a certain address space).

In most cases, a set of the following access codes may be needed:

- access code for private/UDP numbers (for example, 6)

- access code for public/local numbers (for example, 9 used for the NXX in North America and for local exchange Special Numbers (SPN) outside North America)

- access code for public/national numbers (for example, 61)

- access code for public/international numbers (for example, 6011)

No access code is needed for private/CDP numbers.

Special numbers used to access a service (for example, 911 and 411) are a special case. Depending on which system is providing the special number service (CS 1000, MCS 5100, or PSTN), the special number can be accessed as follows:

- without dialing an access code (for example, with an emergency service the user dials 911)

- with an access code (for example, with the PSTN directory service the user may be required to dial 9411).

❗ Important:
Rule 2: In a mixed CS 1000/MCS 5100 network, both systems must have the dialing plan set up in the same manner. MCS 5100 clients in Converged Mode must share the UDP LOC of the CS 1000 client with which it is converged.

For example, at a certain location, a user must be able to make a call to a UDP number by dialing the same digit string on a CS 1000 client as on an MCS 5100 client.

Rule 2 is important for a CD user. Although a CD user can enter a destination number on the Multimedia PC Client, the call itself is originated under CS 1000 control.

DNs as MCS aliases versus DNs as MCS usernames

This section describes how to assign a digit-based address (that is, directory number [DN]) to an MCS 5100 user.

The following options are available to associate a DN with an MCS user:

- Provision the user's username in the form of a DN.

- Provision the user's alias in the form of a DN.

You must also do the following as part of the setup:

- Provision the user's CD alias in the form of a DN.

- Provision the user's public and private charge ID in the form of a DN.

The existing MCS practice is to provision a user with an alias in the DN form. This works for non-converged MCS 5100 users but presents problems to CD users when they initiate calls from the converged Multimedia PC Client in the Click-to-Call mode, specifically from the following locations:

- Call Logs – Inbox window

- Friends Online window

- Directory window

In the cases, if the item represents an MCS 5100 user, the call is initiated towards the destination identified by the MCS username. In the case of a CD user, the call is actually handled by the CS 1000 Call Server that can route based on DN addresses only. To ensure

that the CS 1000 Call Server receives a DN-based address, the MCS 5100 replaces the username with the user's private charge ID.

> **Important:**
> Rule 3: In a mixed CS 1000/MCS 5100 network with Converged Desktop users, MCS 5100 private charge IDs must be in the form of a UDP number (LOC + DN).

Directory Number qualification

An additional aspect of using digit-based addressing is the directory number qualification.

Existing telephony systems support multiple address spaces. The address spaces include the following:

- Public address space — which is subdivided into the following categories:

    - Local numbers

    - National numbers

    - International numbers

    - Special numbers

- Private address space — which is subdivided into the following categories:

    - Level 0 regional numbers (in CS 1000, these are known as Coordinated Dialing Plan [CDP] numbers)

    - Level 1 regional numbers (in CS 1000, these are known as Universal Dialing Plan [UDP] numbers)

To be useful, a DN must be scoped within an appropriate address space. One way of achieving this is by qualifying DNs.

Both CS 1000 and MCS 5100 systems have implemented a phone-context parameter of the userinfo-field of the telephony-based SIP URI as a means to fully qualify a DN.

- A SIP URI with the user=phone-context parameter is known as a telephony-based SIP URI. For more information, see Parameter configuration for a SIP URI on page 111.

- A SIP URI without the user=phone-context parameter is known as a non-telephony-based SIP URI or a regular SIP URI.

The format of a telephony-based SIP URI is as follows:

sip:[directorynumber];phone-context=[phonecontext]@[domainame]; user=phone

sip:+[directorynumber]@[domainname];user=phone (for international PSTN numbers)

The format of a regular SIP URI is as follows:

sip:[username]@[domainame]

> 🛈 **Important:**
> Rule 4: In a mixed CS 1000/MCS 5100 network, all DNs passed between the two systems must be in the form of a telephony-based SIP URI.

If Rule 4 is not followed, a CDP DN may be incorrectly interpreted as a UDP number or a public/local number as a UDP number.

Non-telephony-based SIP URIs

In accordance with the preceding Rule 4, non-telephony-based, or regular, SIP URIs must not be passed between systems in a mixed CS 1000/MCS 5100 network.

There is one exception to this rule — regular SIP URIs can be passed between systems in the Refer-to header.

> 🛈 **Important:**
> Rule 5: In a mixed CS 1000/MCS 5100 network, for routing purposes, regular SIP URIs can be passed between the two systems only within a Refer-to header.

The refer operation is not initiated by CS 1000 SIP Gateways. It is used by the MCS 5100 in the following scenarios:

- Call redirection (for example, a call transfer) initiated by the MCS 5100 — The SIP URI in the Refer-to header is analyzed only by the MCS 5100 system and has no impact on the CS 1000 system.

- Click-to-Call operation of a CD user — The SIP URI in the Refer-to header is analyzed by the CS 1000 system and has to be understood by the system. Therefore, the Username field of the SIP URI must be in the form of a DN.

> 🛈 **Important:**
> Rule 6: In a mixed CS 1000/MCS 5100 network with Converged Desktop users, the CS 1000 SIP Gateway must support regular SIP URIs within the Refer-to header.

If the Username field of the SIP URI is not in the form of a DN, the CS 1000 SIP Gateway handles the refer operation according to normal SIP processing. In other words, it lets the MCS system route the call to the destination indicated in the Refer-to header.

If the Username field of the SIP URI is in the form of a DN, the CS 1000 SIP Gateway configures NPI = Private and TON = CDP (the default settings; the latter having the same effect as TON = Unknown). The CS 1000 SIP Gateway then passes the processing of the refer operation to the CS 1000 Call Server.

Converged Desktop implications

The biggest impact of the adopted addressing scheme on the Converged Desktop feature in a mixed CS 1000/MCS 5100 network is on Click-to-Call.

With Click-to-Call, Converged Desktop users initiate a call using their converged Multimedia PC Client user interface, but use their CS 1000 client to conduct the voice conversation when in Converged Mode.

The converged Multimedia PC Client provides several ways for a user to initiate Click-to-Call, including:

- using the Directory window
- using the Friends Online window
- using the Call Logs window (Outbox and Inbox tabs)
- using the Make a Call window (Dialpad, Recent, and Directory tabs)

These four user interfaces retrieve an address. The retrieved address has several origins, including:

- The address is an MCS 5100 username (as defined by an MCS 5100 administrator).
- The address was delivered by an incoming call over a route.
- The address was entered directly by the user (personal directory).
- The address was entered by an MCS 5100 administrator or retrieved from a directory (global directory).

It is very difficult for the CS 1000 system to route calls based on the retrieved address alone. The address must be either:

- explicitly qualified (in terms of phone-context) by the MCS 5100 system, or
- implicitly qualified (using a proper prefix) by the MCS 5100 system

Problematic Click-to-Call scenarios

Scenario 1: Click-to-Call using an incoming call log when the call has come from a route

In this scenario, the calling party number stored in the call log must be explicitly or implicitly qualified. The MCS 5100 system must store the original phone-context in the call log.

Scenario 2: Click-to-Call using a directory entry (Call at...)

In this scenario, the number stored in the directory entry must include the proper prefix.

The problem arises because directory entries are created by users themselves (PA, Contact Info), MCS 5100 system administrators, or an external Directory Service. Directory entries may not include the required access codes.

Users must be warned to ensure that all directory entries they intend to use include correct access codes. An MCS 5100 or CD user may edit their personal directory through the PA.

Chapter 5: Examples of dialing and numbering plan configuration for mixed networks

Contents

This section contains information about the following topics:

Introduction

This section provides examples of a dialing plan setup for a mixed Avaya Communication Server 1000 (Avaya CS 1000) and Avaya Multimedia Communication Server 5100 (Avaya MCS 5100) network. The following examples are provided:

- Uniform Dialing Plan (UDP)
- Coordinated Dialing Plan (CDP)

> ⓘ **Important:**
>
> The values given in the following examples are examples only. Actual values may differ between applications.

Mixed network example: UDP

The following is an example of a dialing plan setup for a mixed CS 1000/MCS 5100 network. UDP numbers are used to dial among sites.

Sample network

The sample network has the following characteristics:

- The network includes several sites.
- Each site has a mixture of CS 1000 users, MCS 5100 users, and Converged Desktop users.
- Dialing within a site is done using CDP numbers (no access code, 4-digit numbers).
- Dialing among the sites is done by dialing 6 + 7-digit UDP number.
- Local PSTN hop-off is done by dialing 9 + local number.
- AC1 = 9 is used for calls on the public network; AC2 = 6 is used for calls on the private network.
    - National dialing is done by dialing 9 + 1 + national number.
    - International dialing is done by dialing 9 + 011 + international number.

Figure 11: Network diagram: mixed UDP on page 43 shows this network.

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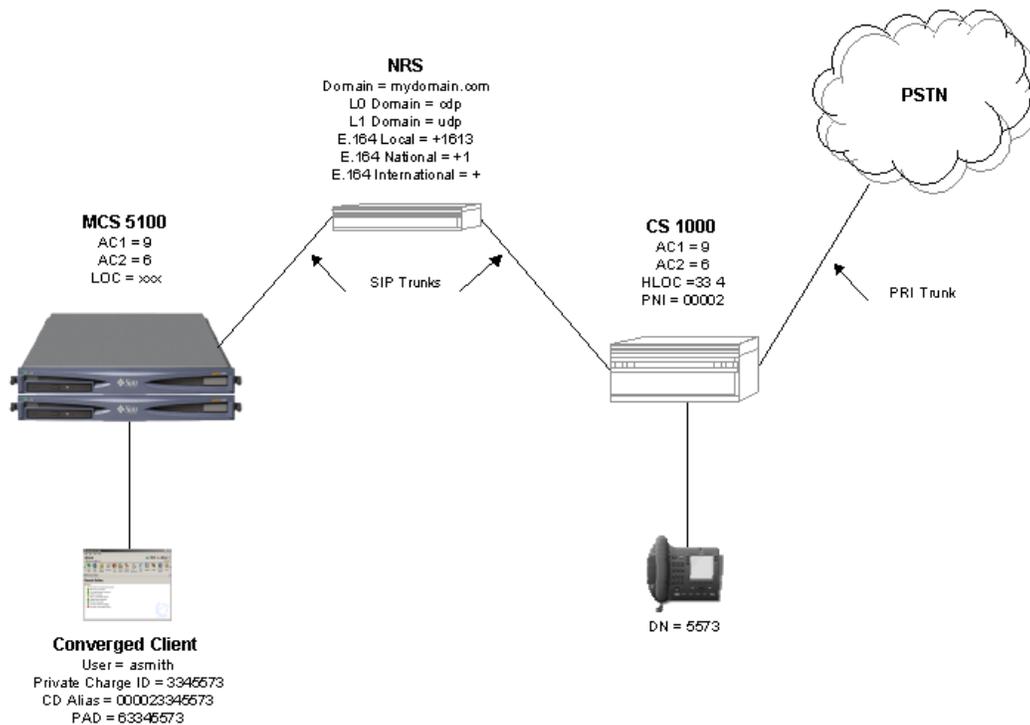
Figure 11: Network diagram: mixed UDP

CS 1000 setup

The CS 1000 setup includes both a user setup and translation setup.

User setup

The CS 1000 user setup involves both the general user setup and the Converged Desktop user setup.

General user setup

The general user setup applies to both stand-alone and Converged Desktop users. The general user setup involves the following:

- Configure the user's extension as a 4-digit number. For example: extension = 5573

- Configure the 3-digit HLOC codes for user groups (that is, customers). For example: HLOC = 334

- The HLOC and user's extension creates the user's UDP number. The number is unique within an Enterprise: HLOC + Extension. For example: UDP Number = 3345573

- Configure the Private Network Identifier (PNI) for user groups (that is, customers). For example: PNI = 00002

- PNI, HLOC, and the user's extension create the Global user ID that is globally unique: PNI + HLOC + Extension. For example: Global ID: 000023345573

Converged Desktop user setup

The Converged Desktop user setup involves the following:

- Configure the user's Class of Service (CLS) to include CDMO or CDMV.

- Configure the Converged Desktop Service (CDS) parameters.

- CS 1000 SIP Services use the user's global ID within CD-related signaling (for example: INVITE and NOTIFY).

Translation setup

The translation setup involves the following:

- Configure CDP routing (Distant Steering Codes [DSC] and Local Steering Codes [LSC]) (based on the 4-digit CDP numbers).

- Configure UDP routing (AC2, HLOCs, and LOCs) (based on the 6 + 7-digit UDP number).

- Configure PSTN routing (AC1, NXX, NPA, SPN, and so on) according to the preceding access codes.

- Configure the CS 1000 SIP Trunk Gateways at each site with appropriate phone-context strings:

    - UDP phone-context For example: phone-context="udp"

    - CDP phone-context For example: phone-context="cdp.udp"

    - Public/Local phone-context For example: phone-context="+1613"

    - Public/National phone-context For example: phone-context="+1"

MCS 5100 setup

The MCS 5100 setup involves includes both a user setup and translation setup.

User setup

The MCS 5100 user setup involves both the general user setup, stand-alone user setup, and the Converged Desktop user setup.

General user setup

The general user setup applies to both stand-alone and Converged Desktop users. The general user setup involves the following:

- Configure the user's username as an alphanumeric address. For example: Username = asmith
- Configure the user's private charge ID as the UDP number. For example: Private Charge ID = 3345573

Standalone user setup

The stand-alone user setup involves the following:

- Do not include the CDS in the service package.
- Configure the user's alias as the Global ID. For example: Alias = 000023345573

> **Note:**
> The alias for a stand-alone user is unique to the Network Routing Service (NRS) of the CS 1000 system. It cannot be a DN within any DN range used in the CS 1000 NRS Level0 or Level1 Domain configuration.

Converged Desktop user setup

The Converged Desktop user setup involves the following:

- Include the CDS in the service package.
- There is no need to define the user's "regular" alias.
- Configure the user's CD alias as the Global ID: For example: CD Alias = 000023345573
- Configure the user's Preferred Audio Device (PAD) as: 6 + UDP number. For example: PAD = 63345573

    The number 6 is the access code configured for UDP calls as configured on CS 1000. The configured values must match in both configurations.

Translation setup

The MCS 5100 translation setup includes pretranslation setup, private telephony routes setup, and gateway telephony routes setup.

Pretranslation setup

Configure pretranslations to support CS 1000-to-MCS 5100 calls to do the following:

- If the UDP number qualifier is received, insert 6. For example, if number qualifier="udp", insert 6.

- If the CDP number qualifier is received, insert 6 + HLOC. For example, if number qualifier="cdp.udp", insert 6334.

- If the Public/Local number qualifier is received, insert 9. For example, if number qualifier="+1613", insert 9.

- If the Public/National number qualifier is received, insert 91. For example, if number qualifier="+1", insert 91.

- If the Public/International number is received, insert 9011. For example, DN starts with a plus sign (+), insert 9011.

Private telephony routes

Configure private telephony routes to support CS 1000-to-MCS 5100 calls and MCS 5100-to-MCS 5100 calls as follows:

- Local termination using UDP dialing (to/from digits 6 + HLOC, length 8, remove 1, prefix PNI) For example: to/from digits 6334, length 8, remove 1, prefix 00002

- Local termination for CDP dialing (to/from digits LSC, length 4, remove 0, prefix PNI + HLOC) For example: to/from digits 5, length 4, remove 0, prefix 00002334

Gateway telephony routes

Configure gateway telephony routes to support MCS 5100-to-CS 1000 calls as follows:

- Outgoing routes using UDP dialing (to/from digits 6 + LOC, length 8, remove 1, use UDP number qualifier) For example: to/from digits 6334, length 8, remove 1, number qualifier="udp", Gateway Route Type = private

- Outgoing routes using CDP dialing (to/from digits DSC, length 4, remove 0, use CDP number qualifier) For example: to/from digits 2 to 7, length 4, remove 0, number qualifier="cdp.udp", Gateway Route Type = private

Mixed network example: CDP

The following is an example of a dialing plan setup for a mixed CS 1000/MCS 5100 network. CDP numbers are used to dial among sites.

Sample network

The sample network has the following characteristics:

- The network includes several sites.

- Each site has a mixture of CS 1000 users, MCS 5100 users, and Converged Desktop users.

- Dialing within a site and between sites is done using CDP numbers (no access code, 5-digit numbers).

- Local PSTN hop-off is done by dialing 9 + local number.

- AC1 = 9 is used for calls on the public network; AC2 = 6 is used for calls on the private network.

    - National dialing is done by dialing 9 + 1 + national number.

    - International dialing is done by dialing 9 + 011 + international number.

Figure 12: Network diagram: mixed CDP

CS 1000 setup

The CS 1000 setup includes both a user setup and translation setup.

User setup

The CS 1000 user setup involves both the general user setup and the Converged Desktop user setup.

General user setup

The general user setup applies to both stand-alone and Converged Desktop users. The general user setup involves the following:

- Configure the user's extension as a 5-digit number. For example: extension = 35573
- The user's extension creates the user's CDP number. The number is unique within an Enterprise. For example: CDP Number = 35573
- Configure the HLOC codes for user groups (that is, customers). For example: HLOC = 34

⊛ Note:

Although the network is CDP-only, some UDP configuration is necessary to support MCS 5100 private charge ID, which is always presented as a UDP number.

- Configure the Private Network Identifier (PNI) for user groups (that is, customers). For example: PNI = 00002
- PNI, HLOC, and the user's extension create the Global user ID that is globally unique: PNI + HLOC + Extension. For example: Global ID: 000023435573

Converged Desktop user setup

The Converged Desktop user setup involves the following:

- Configure the user's Class of Service (CLS) to include CDMO or CDMV.
- Configure the Converged Desktop Service (CDS) parameters.
- CS 1000 SIP Services use the user's global ID within CD-related signaling (for example: INVITE and NOTIFY).

Translation setup

The translation setup involves the following:

- Configure CDP routing (Distant Steering Codes [DSC] and Local Steering Codes [LSC]) based on the 5-digit CDP numbers.

- Configure minimal UDP routing (AC2, HLOCs, and so on) to support incoming calls using UDP numbers.

- Configure PSTN routing (AC1, NXX, NPA, SPN, and so on) according to the preceding access codes.

- Configure the CS 1000 SIP Trunk Gateways at each site with appropriate phone-context strings:

  - CDP phone-context For example: phone-context="cdp.udp"

  - UDP phone-context For example: phone-context="udp"

  - Public/Local phone-context For example: phone-context="+1613"

  - Public/National phone-context For example: phone-context="+1"

MCS 5100 setup

The MCS 5100 setup includes both a user setup and translation setup.

User setup

The MCS 5100 user setup involves both the general user setup, stand-alone user setup, and the Converged Desktop user setup.

General user setup

The general user setup applies to both stand-alone and Converged Desktop users. The general user setup involves the following:

- Configure the user's username as an alphanumeric address. For example: Username = asmith

- Configure the user's private charge ID as the UDP number. For example: Private Charge ID = 3435573

Standalone user setup

The stand-alone user setup involves the following:

- Do not include the CDS in the service package.

- Configure the user's alias as the Global ID. For example: Alias = 000023435573

Converged Desktop user setup

The Converged Desktop user setup involves the following:

- Include the CDS in the service package.

- There is no need to define the user's "regular" alias.

- Configure the user's CD alias as the Global ID: PNI + HLOC + CDP number. For example: CD Alias = 000023435573

- Configure the user's Preferred Audio Device (PAD) as: CDP number. For example: PAD = 35573

Translation setup

The MCS 5100 translation setup includes pretranslation setup, private telephony routes setup, and gateway telephony routes setup.

Pretranslation setup

Configure pretranslations to support CS 1000-to-MCS 5100 calls to do the following:

- If the CDP number qualifier is received, ignore the phone-context. For example, if number qualifier="cdp.udp", leave the digits as is.

- If the UDP number qualifier is received, insert 6. For example, if number qualifier="udp", insert 6.

- If the Public/Local number qualifier is received, insert 9. For example, if number qualifier="+1613", insert 9.

- If the Public/National number qualifier is received, insert 91. For example, if number qualifier="+1", insert 91.

- If the Public/International number is received, insert 9011. For example, DN starts with a plus sign (+), insert 9011.

Private telephony routes setup

Configure private telephony routes to support CS 1000-to-MCS 5100 calls and MCS 5100-to-MCS 5100 calls as follows:

Local termination for CDP dialing (to/from digits LSC, length 5, remove 0, prefix PNI + HLOC) For example: to/from digits 3, length 5, remove 0, prefix 0000234

Gateway telephony routes setup

Configure gateway telephony routes to support MCS 5100-to-CS 1000 calls as follows:

Outgoing routes using CDP dialing (to/from digits DSC, length 5, remove 0, use CDP number qualifier) For example: to/from digits 2, length 5, remove 0, number qualifier="cdp.udp", Gateway Route Type = private

Chapter 6: CDS configuration: overview

Contents

This section contains information about the following topics:

Introduction

In order to use SIP Converged Desktop Services, the SIP Trunk Gateway must first be configured. See *IP Peer Networking Installation and Commissioning (NN43001-313)* for detailed instructions.

Configuration of SIP CDS on Avaya Communication Server 1000 (Avaya CS 1000) requires configuration on both the Call Server and the Signaling Server. See Configuring Avaya Communication Server 1000 on page 55 for detailed instructions on configuring these elements.

SIP CDS must also be configured on Avaya Multimedia Communication Server 5100 (Avaya MCS 5100). Refer to Configuring Avaya Multimedia Communication Server 5100 on page 75 for detailed instructions.

Configuration notes

You must configure UDP/Home Location (HLOC) codes using an Access Code (either AC 1 or AC2) and the ESN Data Block with Local Termination Feature (LTER). You must also configure a Route List Block entry (RLI) to delete the HLOC when the CS 1000 recognizes a user has dialed within its own customer (Home Location) or when a converged MCS 5100 user is dialing within the CS 1000 Home Location. This feature is needed for the MCS 5100 PC Clients in a Click-to-Call call from a Friends list or call log.

Description of the network configured in this document

The user will encounter one of many CDS configuration situations. This section describes the basic characteristics of the network to which the configuration described in the rest of this document applies. The intent is to provide a configuration example which the user can then adapt to their particular situation.

The network configured in this document has the following characteristics:

- The CS 1000 system is a CS 1000M system.

- The NRS is used to route all Converged Desktop calls.

- The network supports both pure UDP and CDP calls dialing plans, as seen from the users' perspective.

- The dialing range of MCS overlaps that of CS 1000 in a "split-range" manner — half of the overlap is directed to the MCS side and the other half is directed to the CS 1000 side. DSCs are used to direct the calls accordingly.

The following entities are used in this network:

- SIP Domains:

  - Service Domain — mydomain.com

  - Level 1 Domain — udp

  - Level 0 Domain — cdp

- SIP Gateway route — myGWRoute

- SIP Gateway route list — my RouteList

- SIP Trunk group — my TrkGroup

- CDS Class of Service — cdCOS

- MCS Application Server — MyMCSAppSrv

The sample configuration uses the following IP addresses:

- IP Telephony node — 12.123.1.123

- MCS Application server — 12.234.3.234

Chapter 7: Configuring Avaya Communication Server 1000

Contents

This section contains information about the following topics:

Before you begin

> ● **Important:**
>
> Some values in these instructions are taken from the values in Description of the network configured in this document on page 54, and from example numbering plans described in Examples of dialing and numbering plan configuration for mixed networks on page 41. Note that actual values may differ between applications.

Before you begin configuring CDS on the Avaya Communication Server 1000 (Avaya CS 1000), you must have a SIP Trunk Gateway configured. The SIP Trunk must have Insert ESN Access Code (INAC) configured for incoming LOC calls. Refer to *IP Peer Networking Installation and Commissioning (NN43001-313)* for details.

Ensure that CDS calls are routed over the SIP Trunk Gateway that is associated with the CDS application.

CS 1000 SIP configuration and UDP location dialing

Configuring the Call Server

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

1. Table 2: LD 15 Configure CLID and HLOC. on page 56.
2. Table 3: LD 17 Configure the Application Module Link (AML). on page 57.
3. Table 4: LD 17 Configure Value Added Server (VAS). on page 58.
4. Table 5: LD 23 Configure Automatic Call Distribution (ACD) Directory Number (DN). on page 58.
5. Table 6: LD 23 Configure Control Directory Number (CDN). on page 58.
6. Table 7: LD 86 Configure ESN Data Block with AC1 and AC2 (if not already defined). on page 59.
7. Table 8: LD 90 Configure Home Location Code (HLOC). on page 59.
8. Table 9: LD 86 Configure Access Code (AC1 or AC2). on page 60.
9. Table 10: LD 86 Configure Digit Manipulation. on page 60.
10. Table 11: LD 86 Configure Home Location Code. on page 61.
11. One of the following:
    - Table 12: LD 90 Configure Service DN for Converged Desktop. on page 61.
    - Table 13: LD 87 Configure Steering Code to be used as the Service DN for making Virtual Trunk call from Agent. on page 62.

    In the case where the customer has used all Special Numbers (SPN), a CDP Distance Steering Code (DSC) can be used as a Service DN.
12. Table 14: LD 11 Configure Personal Call Assistant (PCA). on page 62.

Table 2: LD 15 Configure CLID and HLOC.

Prompt	Response	Description
REQ	CHG	Change existing data block.
CUST		Customer number

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Prompt	Response	Description
	0-99	Range for Large Systems and Avaya Communication Server 1000E (Avaya CS 1000E) system
	0-31	Range for Media Gateway 1000 B (Avaya MG 1000B)
TYPE	NET	Networking
AC2	LOC	Access Code 2
ISDN	YES	Integrated Services Digital Network allowed.
-PNI	2	Private Network Identifier
		PNI cannot be zero (0).
-HLOC	334	Home Location Code (ESN) (1-7 digits)
		This must be the same HLOC as configured under CLID.
-CLID	Yes	Allow Calling Line Identification option.
--ENTRY	0	CLID entry to be configured (Table 0 for CLID)
---HLOC	334	Home Location Code (ESN)
-PFX1	613	Prefix 1
-PFX2	962	Prefix 2
DITI	Yes	DID to TIE connections allowed.
		To connect external DID and internal network TIE trunks for the customer (which allows an external trunk to be transferred across an internal network TIE trunk), set DITI=YES.

Table 3: LD 17 Configure the Application Module Link (AML).

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device And Number
- ADAN	NEW ELAN x	Action Device And Number
		Where x is the ELAN Link number:
		x = 32 to 127 (inclusive) for Large Systems
		AML link number within the preceding range implies that the transport is over TCP/RUDP link.
CTYP	ELAN	AML over Ethernet card type

Table 4: LD 17 Configure Value Added Server (VAS).

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	VAS	Value Added Server configuration
VAS	NEW	New Value Added Server data block
- VSID	32-500	VAS Identifier
		= 32 to 127 for Large Systems
		Avaya recommends that the VAS Identifier match the ELAN Link number configured in the last step (LD 17 — Configuring AML).
- ELAN	x	Associate Value Added Server ID (VSID) x with Application Module Link over Ethernet (ELAN) x
		Note: The AML ELAN link number must match the number provisioned in the last step (LD 17 — Configuring AML).

Table 5: LD 23 Configure Automatic Call Distribution (ACD) Directory Number (DN).

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	ACD	Automatic Call Distribution data block
CUST	xx	Customer number as defined in LD 15. (Customer number associated with this data block.)
ACDN	5300	ACD Directory Number

Table 6: LD 23 Configure Control Directory Number (CDN).

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	CDN	Control Directory Number data block
		This is a special DN created to specify a destination ACD DN to which incoming calls are directed. Multiple CDNs can direct calls to the same ACD DN providing different treatments based on the CDN parameters.
CUST	xx	Customer number as defined in LD 15. (Customer number associated with this data block.)
CDN	5200	Control DN

Prompt	Response	Description
CDSQ	YES	CDN queue is used for Converged Desktop Service.
DFDN	x...x	Local Default ACD DN
		IMPORTANT: The Local Default ACD DN must match ACDN configured in the previous step (LD 23 – Configuring ACD DN).
		Up to 4 digits; up to 7 digits with Directory Number Expansion (DNXP) package 150.
CNTL	YES	Control DN is in control.
VSID	<cr>	Value Added Service ID

Table 7: LD 86 Configure ESN Data Block with AC1 and AC2 (if not already defined).

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
CUST	xx	Customer number
FEAT	ESN	Electronic Switched Network
AC1	9	NARS/BARS Access Code 1
AC2	6	NARS/BARS Access Code 2

Configuring HLOC is necessary to properly handle a UDP number sent by the Avaya Multimedia Communication Server 5100 (Avaya MCS 5100) based on the Private Charge ID of the MCS 5100 user profile. The MCS 5100 always qualifies this number as a UDP number. HLOC will already have been configured if UDP is used. Even if only CDP is used, you must still configure HLOC.

Table 8: LD 90 Configure Home Location Code (HLOC).

Prompt	Response	Description
REQ	NEW	Add new data.
CUST	xx	Customer number as defined in LD 15.
FEAT	NET	Network translation tables
TRAN	AC1 or AC2	Translator
TYPE	HLOC	Home Location Code. **Note:** HLOC must match what MCS 5100 is sending.
HLOC	334	Home Location code.
DMI	1-255	Digit Manipulation Index numbers

Prompt	Response	Description
		⊛ **Note:**
		DMI range is dependent on which Package is used. (0) No digit manipulation required (0)-31CDP (0)-255 NARS and BARS (0)-1999 NARS and BARS with FNP
		DMI is only prompted when the Directory Number Expansion (DNXP) package 150 is equipped and SDRR = LDID.
		The maximum number of Digit Manipulation tables is defined by prompt MXDM.

Configuring AC1 or AC2 is necessary to properly handle a UDP number sent by the MCS 5100 based on the Private Charge ID of the MCS 5100 user profile. The MCS 5100 always qualifies this number as a UDP number. AC1 or AC2 will already have been configured if UDP is used. Even if only CDP is used, you must still configure AC1 or AC2.

Table 9: LD 86 Configure Access Code (AC1 or AC2).

Prompt	Response	Description
REQ	CHG	Change existing data block.
CUST	xx	Customer number as defined in LD 15.
FEAT	ESN	Electronic switched network
...		
AC1 or AC2	xxxx	1-4 digit Flexible Numbering Plan Access Code 1. The access code cannot conflict with the numbering plan.

Removal of the HLOC digits is necessary so that local termination is possible. Configuring digit manipulation is necessary to properly handle a UDP number sent by the MCS 5100 based on the Private Charge ID of the MCS 5100 user profile. The MCS 5100 always qualifies this number as a UDP number. Digit manipulation will already have been configured if UDP is used. Even if only CDP is used, you must still configure digit manipulation.

Table 10: LD 86 Configure Digit Manipulation.

Prompt	Response	Description
REQ	NEW	Add new data.
CUST	xx	Customer number as defined in LD 15.
FEAT	DGT	Digit manipulation data block
DMI	3	Digit Manipulation Index numbers
DEL	3	Number of leading digits to be Deleted (HLOC)

Table 11: LD 86 Configure Home Location Code.

Prompt	Response	Description
REQ	CHG	Change existing data.
CUST	xx	Customer number
FEAT	NET	Feature
HLOC	334	Home Location Code
-DMI	3	Digit Manipulation Index

Table 12: LD 90 Configure Service DN for Converged Desktop.

Prompt	Response	Description
REQ	NEW	Add new data.
CUST	xx	Customer number as defined in LD 15. (Customer number associated with this telephone.)
FEAT	NET	Network translation tables
TRAN	AC1 or AC2	Access code, where:
		AC1 = NARS/BARS AC2 = NARS
TYPE:	SPN	Special Number translation (Special code translation block)
		Configure the CDS Service DN as an SPN, and point it to the SIP trunk. This number is only internal between the Call Server and the Signaling Server for reserving the Virtual Trunk when routing the terminating CD call out to the MCS 5100.
SPN	5100	Special Number.
		The same DN must be used in the config.ini file in the Service DN field on the Signaling Server.
- FLEN	xx	Flexible length number of digits
		Flexible length is the exact number of digits the system expects to receive before accessing a trunk and outpulsing those digits.
-RLI	xxx	Route List Index for Converged Desktop Service Class of Service (CLS)
		The RLI number must point to the SIP Virtual Trunk on the desired Signaling Server. For example, if the CDN and SPN are configured on Signaling Server 1, then this RLI must use the route configured on Signaling Server 1.

In the case where the customer has used all Special Numbers (SPN), a CDP Distant Steering Code (DSC) can be used as a Service DN.

> ⓘ **Important:**
> Do this only if the Service DN has not already been configured as a Special Number in LD 90 on Table 12: LD 90 Configure Service DN for Converged Desktop. on page 61.

Table 13: LD 87 Configure Steering Code to be used as the Service DN for making Virtual Trunk call from Agent.

Prompt	Response	Description
REQ:	NEW	Add new data block.
CUST	xx	Customer number as defined in LD 15. (Customer number associated with this telephone.)
FEAT:	CDP	Coordinated Dialing Plan
TYPE	DSC	Distant Steering Code
DSC	x...x	Service DN
-FLEN	(0)-24	Flexible length number of digits
		Flexible length is the exact number of digits the system expects to receive before accessing a trunk and outpulsing those digits.
-DSP	aaa	Display, where aaa = LSC, LOC, DN.
-RRPA	(NO) YES	Remote Radio Paging Access
-RLI	xxx	Route List to be accessed for Distant Steering Code.
		Virtual Route RLI with no DMI changes.
-CCBA	(NO) YES	Collect Call Blocking
-NPA	xxxxxxx	NPA code, maximum 7 digits
-NXX	xxxxxxx	NXX code, maximum 7 digits

Table 14: LD 11 Configure Personal Call Assistant (PCA).

Prompt	Response	Description
REQ:	NEW	Add new data block to the system.
TYPE:	PCA	Personal Call Assistant
TN		Terminal Number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit
CUST	xx	Customer number as defined in LD 15. (Customer number associated with this telephone.)

Prompt	Response	Description
TGAR	xx	Trunk Group Access Restriction
		Make sure this value is non-blocking.
NCOS	(0)-99	Network Class of Service Group.
		IMPORTANT: Ensure this value is high enough to allow calls on SIP trunks.
CLS	UNR	Unrestricted Class of Service.
KEY		Telephone function key assignments
	0 ACD 5308 cccc zzzz	Automatic Call Distribution (ACD) key
		Where:
		• yyyy = ACD DN configured in LD 23 for Converged Desktop use
		• cccc = CLID table entry of (0)-N, where N = the value entered at the SIZE prompt in LD 15 minus 1
		zzzz = agent's position ID (zzzz can be up to 4 digits; up to 7 digits with Directory Number Expansion [DNXP] package 150).
	2 MCN 3604	Multiple Call Non-Ringing key
		Where yyyy = DN
		The DN can be up to 4 digits; up to 7 digits with Directory Number Expansion (DNXP) package 150. Once the MCN key has been defined, the MARP prompt appears.
		Since MCN and DN are used by the feature to originate calls, the same DN can be applied to all PCAs.

Configure the Signaling Server using CS 1000 Element Manager

To configure SIP Converged Desktop Services using CS 1000 Element Manager, follow the steps in

> ⊛ **Note:**
>
> The SIP Trunk Gateway must be configured for SIP CDS. Refer to *IP Peer Networking Installation and Commissioning (NN43001-313)* for detailed instructions.

Configuring SIP CDS on the Signaling Server

1. Log in to Element Manager.

2. Select **IP Telephony > Nodes: Servers, Media Cards > Configuration** from the navigator.

   The Node Configuration web page opens, as shown in <u>Figure 13: Node configuration web page</u> on page 64.

Managing: **207.179.153.99**
    IP Telephony » Nodes: Servers, Media Cards » Node Configuration

Node Configuration

New Node [   ] [to Add]

[Import Node Files]

+ **Node: 8   Node IP: 192.168.253.7**   [Edit] [Transfer / Status] [Delete]

Figure 13: Node configuration web page

3. Click **Edit** next to the node for which SIP CDS is to be configured.

   The Edit web page opens, as shown in <u>Figure 14: Edit web page</u> on page 65.

Managing: **207.179.153.99**
IP Telephony » Nodes: Servers, Media Cards » Node Configuration » IP Telephony: Node ID 8 » Edit

Edit

Save and Transfer	Cancel

– Node

Node ID **8**

Voice LAN (TLAN) Node IP address	12.123.3.123	*
Management LAN (ELAN) gateway IP address	207.179.153.97	
Management LAN (ELAN) subnet mask	255.255.255.224	
Voice LAN (TLAN) subnet mask	255.255.255.0	

+ SNMP	Add
+ VGW and IP phone codec profile	
+ QoS	
+ LAN configuration	
+ SNTP	
+ H323 GW Settings	
+ Firmware	
+ SIP GW Settings	
+ SIP URI Map	
+ SIP CD Services	
+ Cards	Add
+ Signaling Servers	Add

Save and Transfer	Cancel

Figure 14: Edit web page

4. Select **SIP CD Services** to expand the section, as shown in Figure 15: SIP CD Services on page 66.

SIP Gateway Services

SIP Converged Desktop: ☑ Enable CD service

Service DN:		Used for making VTRK call from agent.
Converged telephone call forward DN:		
RAN route for announce:		(route number 0 - 511)
Wait time before RAN queue:	1	(-1 - 32767 msec)
Timeout for ringing indication:	10	(5 - 60 seconds)
Timeout for CD server:	5	(1 - 30 seconds)
Timeout for non-CD server:	2	(2 - 60 seconds)
User information fields		
Invite message for MO set:	sip:convergeddesktop@name;avayaconverged=continueforce	
Invite message for MV set:	sip:convergeddesktop@name;avayaconverged=conditionalfork	
Notify message for converged desktop:	sip:convergeddesktop@name	

SIP CTI Service: ☐ Enable CTI service

☐ TLS endpoints only

Figure 15: SIP CD Services

5. Enter the necessary information in the following fields:

   a. **Service Enabled:** Select whether the Converged Desktop Service application is enabled or disabled for this node. CDS is enabled by default (the check box is selected).

   😊 **Note:**

   If CDS is not enabled, no calls are sent to the MCS 5100, including both originating and terminating calls either to or from the Converged Desktop telephone.

   b. **Service DN used for making VTRK call from Agent:** Enter the Service DN. The Service DN is a number for CS 1000 to call the Signaling Server CDS application over a SIP Virtual Trunk. This number can be composed of:

      • LOC + extension

      • Extension

      • SPN

   As described in How Converged Desktop Services works on page 16, upon receipt of an incoming call notification over the AML/VAS links, the CDS application must reserve a Virtual Trunk until the destination of the call is determined. The CDS application instructs the Call Server to make a call to this Service DN. The Call Server's numbering plan directs the call to the Virtual Trunk route based on translations. The CDS answers and reserves the Virtual Trunk in case it is needed to terminate the incoming call.

   The example shown uses an SPN and is defined as follows:

Service DN = AC1 (from LD 86) + SPN (from LD 90)

For example, the value entered for the Service DN is 65100, where 6 is Access Code 1 (AC1) (configured in LD 86 on Table 9: LD 86 Configure Access Code (AC1 or AC2). on page 60) and 5100 is the Special Number (SPN) (configured in LD 90 on Table 12: LD 90 Configure Service DN for Converged Desktop. on page 61).

c. **Converged Telephone Call Forward DN:** This field is required if the SIP Trunk Gateway application is enabled and the Converged Desktop Service will be used. The Converged Telephone Call Forward DN is the CDN defined in LD 23 for the CDS application. This entry designates the CDN that CDS will acquire to obtain messaging for call activity. CS 1000 telephones with CDMO or CDMV Class of Service redirect all calls to the CDN. The acquisition of the CDN identified in this field allows messaging for call activity to be processed to the CDS Application on the Signaling Server.

For example, the value entered is 5100, which is the Control DN (CDN) configured in LD 23 on Table 6: LD 23 Configure Control Directory Number (CDN). on page 58.

d. **User Info. field for Invite message on the Converged Desktop MO Set:** Verify the field against the following string:

sip:convergeddesktop@SIPdomainname;nortelconverged= continueforce

Ensure that the SIPdomainname is part of the string. The SIP Domain name should have been configured during the SIP Trunk Gateway configuration.

If the string is missing the SIP Domain name (that is, the string is sip:convergeddesktop@;nortelconverged=continueforce), then the SIP Domain name was not configured during the SIP Trunk Gateway configuration. The **SIP Domain name** is found under **Signaling Server > Signaling Server xxx.xxx.xxx.xxx Properties** on the Edit web page (Figure 14: Edit web page on page 65).

This field is used in the INVITE message of Converged Desktop terminating call handling if the telephone is a multimedia-only telephone.

e. **User Info. field for Invite message on the Converged Desktop MV Set:** Verify the field against the following string:

sip:convergeddesktop@SIPdomainname;nortelconverged= conditionalfork

Ensure that the SIPdomainname is part of the string. The SIP Domain name should have been configured during the SIP Trunk Gateway configuration.

If the string is missing the SIP Domain name (that is, the string is sip:convergeddesktop@;nortelconverged=conditionalfork), then the SIP Domain name was not configured during the SIP Trunk Gateway configuration. The **SIP Domain name** is found under **Signaling Server**

> **Signaling Server xxx.xxx.xxx.xxx Properties** on the Edit web page (Figure 14: Edit web page on page 65).

This field is used in the INVITE message for Converged Desktop terminating call handling if the telephone is both a multimedia and a voice telephone.

f. **User Info. field in the notify message for Converged Desktop:** Verify the field against the following string:

sip:convergeddesktop@SIPdomainname

Ensure that the SIPdomainname is part of the string. The SIP Domain name should have been configured during the SIP Trunk Gateway configuration.

If the string is missing the SIP Domain name (that is, the string is sip:convergeddesktop@), then the SIP Domain name was not configured during the SIP Trunk Gateway configuration. The **SIP Domain name** is found under **Signaling Server > Signaling Server xxx.xxx.xxx.xxx Properties** on the Edit web page (Figure 14: Edit web page on page 65).

This field is used in the NOTIFY message for Converged Desktop.

g. **RAN route for Announce:** Enter the Recorded Announcement (RAN) route number for call announce. The route was configured in LD 16. The range is 0 to 511 for Large Systems.

The RAN definition in this field can be used in two distinct ways:

- Call progress – A customer may want to announce to a caller that their call is proceeding and to please wait. RAN can be used when delays may be longer than expected, as PCA makes calls to other devices. If RAN is not used, callers will hear only ringback tone.

- PCA busy – If PCA agents are busy processing a call and the caller must wait for a PCA so CDS can process the call, the user can be directed to RAN. The RAN typically will say, for example, "Your call is proceeding, please wait..."

h. **Wait time before a caller is sent to RAN Queue:** Enter the time (in seconds) that the caller waits before the called is directed to the RAN queue. The range is -1 to +32767 seconds, where:

Value	Meaning
-1	Never send to RAN queue. This is the default.
0	Immediately send to RAN queue.
+1 to +32767	Wait n seconds before sending to RAN queue.

i. **Timeout for Ringing indication of the CD set:** Enter the time to wait for ringing indication on the Converged Desktop telephone. If a timeout occurs, the system considers the telephone as busy or call-forwarded to

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other devices. As a result, the Converged Desktop call is canceled. The timeout range is 5 to 60 seconds (default is 10 seconds).

j. **Timeout for CD Server:** Enter a timeout value. This value is used when the MCS 5100 is not available or there are network problems. An incoming Converged Desktop call is terminated directly to Converged Desktop telephone without waiting for the MCS 5100. The timeout range is 1 to 30 seconds (default is 5 seconds).

 Note:

Avaya recommends that the timeout value not be configured as a small value. If the time is too short, then a delayed packet can eventually reach the MCS 5100. This can cause a "race condition" between the MCS 5100 routing the call and the CS 1000 routing the call. If the administrator knows of a persistent network problem, or if the MCS 5100 is out-of-service for a period of time, then Avaya recommends that the Converged Desktop Service be disabled instead of having a small timeout value.

k. **Timeout for call answered by other than CD phone set:** Enter a timeout value. If a call is answered on any telephone (within the CS 1000 network) other than Converged Desktop telephone itself, then the SIP session is torn down automatically after this timeout. The purpose is to remove "lone" screen pop-ups on the Converged Desktop PC Client (PCC). The timeout range is 2 to 60 seconds (default is 2 seconds).

6. Select **SIP URI Map** to expand the section, as shown in

 Note:

The fields require a character string that is less than 128 characters in length. The valid characters include: alphanumeric (a-z, 0-9), period (.), dash (-), underscore (_), and plus sign (+). These fields must be completed if the SIP Trunk Gateway application is enabled.

Figure 16: CS 1000 SIP URI Map

7. Fill in the fields as follows:

    a. Type **+1** in the **Public E.164/National domain name** text box.

    b. Type **+1613** in the **Public E.164/Subscriber domain name** text box.

    c. Type **publicUnknown** in the **Public E.164/Unknown domain** name text box.

    d. Type **publicSpecial** in the **Public E.164/Special Number domain** name text box.

    e. Type **udp** in the **Private/UDP domain name** text box.

    f. Type **cdp.udp** in the **Private/CDP domain name** text box.

    g. Type **special.udp** in the **Private/Special Number domain name** text box.

    h. Type **privateUnknown** in the **Private/Unknown (vacant number routing) domain name** text box.

    i. Type **unknown.unknown** in the **Unknown/Unknown domain name** text box.

⊛ **Note:**

The values entered in these fields must agree with values entered in MCS 5100. See

8. Click **Save and Transfer**.

Verifying configuration

After configuring the Virtual IP D-channel, SIP Route, Converged Desktop AML, and the other elements, log in to the Leader Signaling Server of the node. At the Command Line Interface

(CLI), use the commands **cdsShow** and **cdsAgentShow** to verify the configuration. Output of the commands will be similar to that shown in Figure 17: Verifying configuration: cdsShow on page 72 and Figure 18: Verifying configuration: cdsAgentShow on page 72.

The two diagrams show that the AML link is Up and your PCA agents have been acquired by the ACD queue you configured for CD2.

> **Note:**
>
> In Figure 17: Verifying configuration: cdsShow on page 72 and Figure 18: Verifying configuration: cdsAgentShow on page 72, CDN is the Converged Telephone Call Forward DN.

```
[admin2@ibmzeus ~]$ cdsShow

=== VTRK ===
CDS Application Current State: ServiceEnabled

hAppBlk       TaskName         Tid      LinkState      NumRetry   LinkNum   Trace
0xfd3080      CDS              0xf800   Down           0          -1        0

AML Link Timer Next Timeout: 3 sec
Call Audit Timer Next Timeout: 0 sec

CDS Service Enabled : Yes
Service DN         : 965
CDN               : 45
MO Service URI    : sip:convergeddesktop@;  avaya converged=continueforce
MV Service URI    : sip:convergeddesktop@;  avaya converged=conditionalfork
Notify URI        : sip:convergeddesktop@
PNI + HLOC        : 00000   (null)
AnnounceRAN       : 211
AnnounceRANSeconds : 1
CD Server Timeout  : 5
CD Ringing Timeout : 10
Non-PAD Ans Timeout : 2

CDS Call Trace: Off
Calling Number: (null)
Called Number : (null)
[admin2@ibmzeus ~]$
```

Figure 17: Verifying configuration: cdsShow

```
oam> cdsAgentShow

The active CDN: 5200
The default ACD DN: 5300
Round Robin index: 1
-----------------------------------------------------------------
No  TN      PosID    ACD_DN    DN    dnStatus    loginStatus    acdStatus
1 0x703c  510300    5300     3604    Idle         Ready          Idle
2 0x703d  510300    5300     3604    Idle         Ready          Idle
3 0x703e  510300    5300     3604    Idle         Ready          Idle
4 0x703f  510300    5300     3604    Idle         Ready          Idle
5 0x707c  510300    5300     3604    Idle         Ready          Idle
-----------------------------------------------------------------
Number of CDS Agents: 5, 5
```

Figure 18: Verifying configuration: cdsAgentShow

⊛ **Note:**

If your Agents do not show Up, you may not have Symposium Call Center (NGCC) package 311 equipped. You need CS 1000 PEP MPLR19786 if the package is not equipped and you are using CS 1000 Release 4.0 software. This PEP is not required on CS 1000 Release 4.5 or later software.

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Configuring the Converged Desktop User

The Converged Desktop user is configured in LD 10 for analog (500/2500-type) telephones or LD 11 for digital telephones.

Table 15: LD 10/11 Configure Converged Desktop User.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	a..a	Type of data block
TN		Terminal Number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit
CUST	xx	Customer number as defined in LD 15.
CLS		Converged Desktop Service Class of Service (CLS)
	(CDMR) CDMV CDMO	Converged Desktop Multimedia Restricted (default) Converged Desktop Multimedia and Voice Converged Desktop Multimedia Only ✳ **Note:** Avaya recommends CDMV to give users all features.
CSDN	5200	Converged Service Directory Number
		Converged Desktop Service Control Directory Number (CDN) configured in LD 23.
		CSDN is only prompted if CLS is defined as CDMV or CDMO.
KEY		Telephone function key assignments
	0 SCR yyyy	Single Call Ringing key
		Where yyyy = DN
		The DN can be up to 4 digits; up to 7 digits with Directory Number Expansion (DNXP) package 150. Use a single appearance DN to terminate Voice Call (VCC) or Signaling (SIG) calls. Once the SCR key has been defined, the MARP prompt appears.

Chapter 8: Configuring Avaya Multimedia Communication Server 5100

Contents

This section contains information about the following topics:

Before you begin

> 🛈 **Important:**
>
> Some values in these instructions are taken from the example numbering plans described in Description of the network configured in this document on page 54 and Examples of dialing and numbering plan configuration for mixed networks on page 41. Note that actual values may differ between applications.

Before you begin configuring CDS on the Avaya Multimedia Communication Server 5100 (Avaya MCS 5100), you must have a Service Domain configured in MCS with the same name

as in the Avaya Communication Server 1000 (Avaya CS 1000). For this example, the domain name is mydomain.com.

Implementation summary

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

- Add IP address of IP Telephony node as an Authenticated Server on page 76.
- Configure the SIP Gateway, trunk, and trunk group on page 77.
- Configure number qualifiers on page 80.
- Configure a route list on page 83.
- Configure routes for each dialing plan entry on page 85.

Implementation details

Add IP address of IP Telephony node as an Authenticated Server

Use Adding the IP Address of the IP Telephony node as an Authenticated Server on page 76 to add the IP address of the IP Telephony node to the Application Server as an Authenticated Server.

Adding the IP Address of the IP Telephony node as an Authenticated Server

1. Access the **MCS 5100 Management Console** on the MCS Client:

   a. Choose **MCP** from the **Start** menu.

   b. Click **Start console**.

   c. Log in using your username and password.

   d. Select **Sites**.

   e. Select **Servers**.

   f. Select **Application Server**.

   g. Select **Components**.

   h. Select the application server.

2. Right-click on **Authentication** (or **Authent...**).

3. Select **Modify**.

4. Enter `12.123.1.123`, the IP address of the IP Telephony node, in the `Authorized Node IP Address` field.

5. Click **Add**.

6. Click **Apply**.

7. Verify the address appears in the list under the **Authentication** tab, as shown in

   a. Right-click on the application server.

   b. Select **Query**.
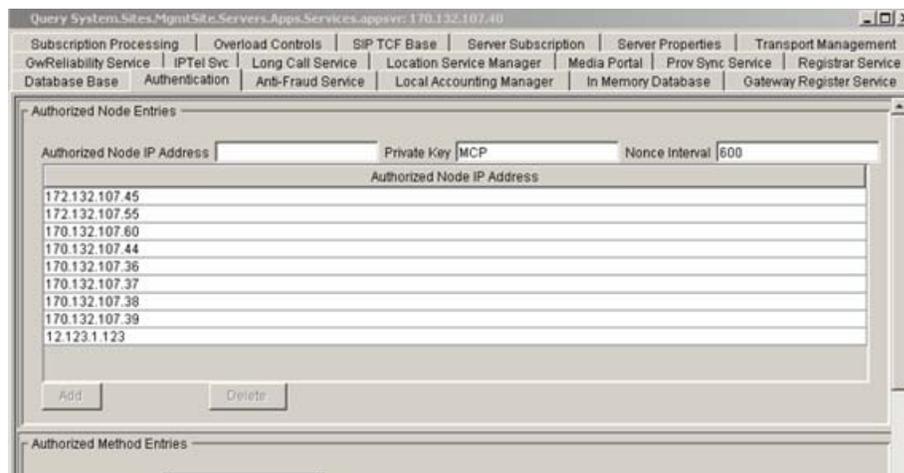
Figure 19: IP address of IP Telephony node added as an Authenticated Server

Configure the SIP Gateway, trunk, and trunk group

Use to configure the CS 1000 Node IP address as a SIP Gateway, and configure the corresponding trunk and trunk group.

Configuring the SIP Gateway, trunk, and trunk group

1. Enter the IP address of the MCS Provisioning Client in a web browser.

   The Provisioning Client Login web page opens, as shown in

Figure 20: Provisioning Client Login

2. Enter your **Username** and **Password**.

   The **Provisioning Client** web page opens, as shown in Figure 21: Provisioning Client on page 78. The Provisioning Client navigator is in the section on the left of the web page.

Figure 21: Provisioning Client

3. Add a new gateway, as follows:

   a. Select **Provisioning > Gateways > Add Gateway** in the Provisioning Client navigator.

      The **Add a new gateway** web page opens, as shown in Figure 22: Provisioning Client: Add a new gateway on page 78.

Figure 22: Provisioning Client: Add a new gateway

   b. Enter **mydomain.com:maddr=12.123.1.123** in the **Gateway host** text box.

   c. Select **CS 1000** in the **Gateway type** drop-down list.

    d. Select **Other** from the **Location** drop-down list.

    e. Select **TRUE** for **Trusted Node**.

    f. Select **TRUE** for **Is Gateway**.

    g. Click **Submit**.

4. Add a new route, as follows:

    a. Select **Provisioning > Gateways > Add Route** in the Provisioning Client navigator.

    The **Create new gateway route** web page opens, as shown in

Figure 23: Provisioning Client: Create new gateway route

    b. Enter **myGWRoute** in the **Route Name** text box.

    c. Select **mydomain.com** in the **Domain** drop-down list.

    d. Click **Save**.

5. Add a trunk group directed to the gateway and route just created, as follows:

    a. Select **Provisioning > Gateways > Add TrunkGroup** in the Provisioning Client navigator.

    The **Trunkgroup Provisioning** web page opens, as shown in

Figure 24: Provisioning Client: Trunkgroup Provisioning

b. Select **mydomain.com:maddr=12.123.1.123** from the **Gateway** drop-down list.

c. Select **myGWRoute** from the **Route** drop-down list.

d. Enter **myTrkGroup** in the **Trunk Group** edit box.

e. Click **Save**.

Configure number qualifiers

Use Configuring number qualifiers on page 82 to configure number qualifiers in MCS 5100 so they match the values configured in the CS 1000 NRS SIP URI phone-context Level 1 and Level 0 Domains, and the CS 1000 SIP URI Map.

The following three sets of provisioned information must match:

- CS 1000 SIP URI Map (see Figure 25: CS 1000 SIP URI Map on page 81) as configured in Configuring SIP CDS on the Signaling Server on page 64.

- CS 1000 NRS SIP URI phone-context in the Endpoint Routes (see Figure 26: CS 1000 NRS Endpoint Routing Entries on page 81).

- MCS 5100 Telephony Routes Number Qualifiers (see Figure 27: MCS 5100 Number Qualifiers on page 82).

where:

- cdp.udp is the CDP/Level 0 Domain, with NPI = private and TON = Level0

- udp is the UDP Level 1 Domain, with NPI = private and TON = Level1

– SIP URI Map

Public E.164/National domain name	+1
Public E.164/Subscriber domain name	+1613
Public E.164/Unknown domain name	publicUnknown
Public E.164/Special Number domain name	publicSpecial
Private/UDP domain name	udp
Private/CDP domain name	cdp.udp
Private/Special Number domain name	special.udp
Private/Unknown (vacant number routing) domain name	privateUnknown
Unknown/Unknown domain name	unknown.unknown

Figure 25: CS 1000 SIP URI Map

#	DN Prefix	DN Type	Route Cost	SIP URI Phone Context
1	011	E.164 international	1	+
2	1	E.164 national	1	+1
3	334	Level1 regional	1	udp
4	4-7	Level0 regional	1	cdp.udp
5	411	Special	1	special.udp
6	962	E.164 local	1	+1613
7	966	E.164 local	1	+1613
8	967	E.164 local	1	+1613

Figure 26: CS 1000 NRS Endpoint Routing Entries

Current Number Qualifiers

Name	Description	Delete
udp	My UDP phone context	Delete
cdp.udp	My CDP phone context	Delete
special.udp	My Special Numbers phone context	Delete
+1	My National Numbers phone context	Delete
+1613	My Local Numbers phone context	Delete
local	NPI=public, TON=Subscriber	Delete
national	NPI=public, TON=National	Delete
international	NPI=public, TON=International	Delete
publicUnknown	NPI=public, TON=Unknown	Delete
cdpDomain	NPI=private, TON=Level 0	Delete
subscriber	NPI=private, TON=Subscriber	Delete
privateUnknown	NPI=private, TON=Unknown	Delete
unknown	NPI=Unknown, TON=Unknown	Delete

Figure 27: MCS 5100 Number Qualifiers

Configuring number qualifiers

1. Enter the IP address of the MCS Provisioning Client in a web browser.

    The **Provisioning Client Login** web page opens, as shown in Figure 20: Provisioning Client Login on page 78.

2. Enter your **Username** and **Password**.

    The **Provisioning Client** web page opens, as shown in Figure 21: Provisioning Client on page 78.

3. Add a new gateway, as follows:

    Select **Provisioning > Domains > mydomain.com > Telephony Routes > Number Qualifiers** in the Provisioning Client navigator.

    The **Number Qualifiers** web page opens, as shown in Figure 28: Provisioning Client: Number Qualifiers on page 83. The Current Number Qualifiers section shows the number qualifiers configured so far. Those listed in Figure 28: Provisioning Client: Number Qualifiers on page 83 are

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configured by default. You can use or **Delete** any of all of them as you prefer.

4. Add a number qualifier for each entry in the SIP URI map (shown in Figure 25: CS 1000 SIP URI Map on page 81) as follows:

   a. Enter the name in the **Name** text box, for example **cdp.udp**.

   b. Enter a description in the **Description** text box, for example **My CDP phone context**.

   c. Click **Add**.

Number Qualifiers

Add a new Number Qualifier

Name	[ ] * ❓
Description	[ ] * ❓

[Add]

Current Number Qualifiers

Name	Description	Delete
local	NPI=public, TON=Subscriber	Delete
national	NPI=public, TON=National	Delete
international	NPI=public, TON=International	Delete
publicUnknown	NPI=public, TON=Unknown	Delete
cdpDomain	NPI=private, TON=Level 0	Delete
udp	NPI=private, TON=Level 1	Delete
subscriber	NPI=private, TON=Subscriber	Delete
privateUnknown	NPI=private, TON=Unknown	Delete
unknown	NPI=Unknown, TON=Unknown	Delete

Figure 28: Provisioning Client: Number Qualifiers

Configure a route list

Use Configuring a route list on page 84 to add a route list.

Configuring a route list

1. Enter the IP address of the MCS Provisioning Client in a web browser.

   The **Provisioning Client Login** web page opens, as shown in .

2. Enter your **Username** and **Password**.

   The **Provisioning Client** web page opens, as shown in .

3. Configure the Class of Service, as follows:

   a. Select **Provisioning > Domains > mydomain.com > Telephony Routes > Routing COS** in the Provisioning Client navigator.

   The **Class of Service Information** web page opens, as shown in .

Class of Service Information for mydomain.com

Add or Modify Class of Service Information

Name: cdCOS

Description: CD COS

Save

Reorder or Delete Class of Service

Current Choices Available

UP

DOWN

Save

Delete

Figure 29: Provisioning Client: Class of Service Information

   b. Enter a name for the Class of Service, such as **cdCOS**, in the **Name** text box.

c. Enter a description for the Class of Service, such as **CD COS**, in the **Description** text box.

d. Click **Save**.

4. Add a new route list, as follows:

a. Select **Provisioning > Domains > mydomain.com > Telephony Routes > Add RouteList** in the Provisioning Client navigator.

The **Create a New Route List** web page opens, as shown in Figure 30: Provisioning Client: Create a New Route List on page 85.

Create a New Route List in mydomain.com domain

Name: myRouteList
Description: myRouteList
Incoming Other Domain Tree Call Routing: ALLOW
Incoming Same Domain Tree Call Routing: ALLOW
Class of Service: cdCOS

Telephony Route(s):

Save   Clear

Figure 30: Provisioning Client: Create a New Route List

b. Enter a list name, such as **myRouteList**, in the **Name** text box.

c. Enter a list description, such as **myRouteList**, in the **Description** text box.

d. Select **ALLOW** from the **Incoming Other Domain Tree Call Routing** drop-down list.

e. Select **ALLOW** from the **Incoming Same Domain Tree Call Routing** drop-down list.

f. Select **cdCOS** from the **Class of Service** drop-down list.

5. Click **Save**.

Configure routes for each dialing plan entry

For each dialing plan entry, you must configure the following routes:

• private CDP route

• private UDP route

- Gateway CDP route

- Gateway UDP route

- private PSTN route

- route for local Dial-9 National (NPA) dialing using the Gateway CS 1000 PSTN

- route for national Dial-9 National (NPA) dialing using the Gateway CS 1000 PSTN

Use Configuring routes for each dialing plan entry on page 86 to configure these routes.

Configuring routes for each dialing plan entry

1. Enter the IP address of the MCS Provisioning Client in a web browser.

   The **Provisioning Client Login** web page opens, as shown in Figure 20: Provisioning Client Login on page 78.

2. Enter your **Username** and **Password**.

   The **Provisioning Client** web page opens, as shown in Figure 21: Provisioning Client on page 78.

3. Add the new routes, as follows:

   a. Select **Provisioning > Domains > mydomain.com > Telephony Routes > Add Telephony Route** in the Provisioning Client navigator.

   The **Create New Telephony Route** web page opens, as shown in Figure 31: Provisioning Client: Create New Telephony Route on page 87.

Create New Telephony Route

Name:		❓
Description:		❓
From Digits:		❓
To Digits:		❓
Min Number of Digits:		❓
Max Number of Digits:		❓
Route Type:	Private ▼	
	Private	
Remove:	SIP	❓
Prefix:	Gateway	❓
Recursive:	No ▼	
Route List:	myRouteList	

Save   Clear

Figure 31: Provisioning Client: Create New Telephony Route

b. Configure each new route, as follows:

      i. Enter the values for the appropriate fields, as listed in .

      Do not use blank spaces in text fields.

      ✳ **Note:**
      In the Name field, only the following characters are valid: A-Z, a-z, 0-9, comma (,), period (.), hyphen (-), and underscore (_). Any other character in the route Name causes an error message.

      ii. Click **Save**.

Table 16: Values for configuring Routes

Entry Field	Route Type						
	Private CDP	**Private UDP**	**Private PSTN**	**Gateway CDP**	**Gateway UDP**	**Public Local**	**Public National**
Name	myCDP Route Private	myUDP Route Private	myPSTN Route Private	myCDP Route GW	myUDP Route GW	myLocal Route Public	my National Route Public
Description	myCDP Route Private	myUDP Route Private	myPSTN Route Private	myCDP Route GW	myUDP Route GW	myLocal Route Public	my National Route Public
From Digits	7100	6334	9	7100	6334	9	9961
To Digits	7199	6334	9	7199	6334	9	9969
Min Number of Digits	4	8	8	4	8	11	8
Max Number of Digits	4	8	22	4	8	22	8
Route Type	Private	Private	Private	Gateway	Gateway	Gateway	Gateway
Remove	0	1	1	0	1	1	1
Prefix	00002334	00002	Not applicable				
Recursive	No	No	Not applicable				
Gateway Route	Not applicable		myGWRoute				
Gateway	Not applicable		Private	Private	Public	Public	

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	Route Type						
Entry Field	**Private CDP**	**Private UDP**	**Private PSTN**	**Gateway CDP**	**Gateway UDP**	**Public Local**	**Public National**
Route Type							
Number Qualifier	Not applicable			cdp.udp	udp	+1	+1613
Route List	myRouteList						

4. Configure the Pretranslation table:

   a. Select **Provisioning > Domains > mydomain.com > Telephony Routes > Pretranslation Table** in the Provisioning Client navigator.

   The **Pre translations** web page opens, as shown in . The **Current Pretranslations** section shows any pretranslations currently configured.

Figure 32: Provisioning Client: Pre translations

   b. Configure pretranslations for each of the number qualifiers in the **Add New Pretranslations** section, as follows:

      i. Select the appropriate number qualifier from the **Number Qualifier** drop-down list.

      ii. Enter a number in the **Length** text box.

      iii. Enter a prefix in the **Prefix** text box.

      iv. Click **Add**.

Configuring a Converged Desktop user

Configure a Service Package

Before you configure any Converged Desktop user in MCS, you must configure a Service Package for the Domain which includes the SIP CDS feature. Use Configuring a Service Package on page 90 to configure a Service Package.

Configuring a Service Package

1. Select **Provisioning > Domains > mydomain.com > Service Package > Create Package** in the Provisioning Client navigator.

   The **Create new package** web page opens, as shown in Figure 33: Provisioning Client: Create new package on page 91.

2. Enter a package name in the **Name of the Packages** text box.

3. Select services and their appropriate parameters as appropriate, including the following:

   a. Select **Converged Desktop**.

   b. Select **Converged Desktop** from the **Setup** drop-down list.

   c. Select **Yes** from the **Converged Desktop Enabled** drop-down list.

4. Click **Save**.

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Create new package for domain mydomain.com

Name of the Packages	[ ]	*
Set package as default for domain	NO ▾	

Select Service(s)

☐ **Choose All Services**

☐ **Ad Hoc Conferencing**
    Maximum Number of Ports   4 ▾

☐ **Advanced Addressbook**
    Maximum Number of Addressbook Entries Allowed   50 ▾

☐ **Advanced Screening**
    Maximum Number of Ringlists   3 ▾
    Maximum Number of Telephone Numbers per Ringlist 3 ▾
    Presence Based Routing   ☐

☐ **Call Waiting Disable**

☐ **Converged Desktop**
    Setup   ConvergedDesktop ▾
    Converged Desktop Enabled   No ▾

☐ **IM Chatroom**

☐ **Meet Me Conferencing**
    Maximum Number of Participants   50
    Premium Conferencing Enabled   ☐

Figure 33: Provisioning Client: Create new package

Configuring the Converged User on the MCS 5100

Use <u>Configuring a Converged Desktop User in MCS 5100</u> on page 91 to configure each Converged Desktop user in MCS 5100.

Configuring a Converged Desktop User in MCS 5100

1. Select **Provisioning > Domains > mydomain.com > Users > Add Users** in the Provisioning Client navigator.

   The **Add a new user** web page opens, the upper portion of which is shown in <u>Figure 34: Provisioning Client: Add a new user</u> on page 92.

Add new user to mydomain.com

User Name: cduser1 *
First Name: CD
Last Name: User1
Password: **** *
Confirm Password: **** *
Service package: myPKG
Aliases:
Status Reason: ACTIVE
email:
Business Phone:
Home Phone:
Cell Phone:
Pager:
Fax:
Private Charge ID:
Public Charge ID:

Figure 34: Provisioning Client: Add a new user

2. Enter appropriate values in the fields, including the following:

   a. Enter **3345573** in the **Private Charge ID** text box.

   This is the HLOC + DN of the converged telephone. This number is used for Click-to-Call in Converged Mode, and must be a UDP number.

   b. Enter **6139625573** in the **Public Charge ID** text box.

   This is the NPA + NXX + DN of the converged telephone. It is the caller identification information sent on a public call. This value is required, and may be need even in Unconverged Mode.

   c. Select **cdCOS** from the **Class of Service** drop-down list.

   This is the same Class of Service configured in Configuring a route list on page 84.

> ⊛ **Note:**
>
> You do not have to configure an alias, but if you do, ensure that it does not conflict with the alias properties configured in 3 on page 93.

    d. Click **Save**.

3. Set up the converged properties, as follows:

    a. Click **Converged Desktop properties**.

       The **Converged Desktop Data** web page opens, as shown in Figure 35: Provisioning Client: Converged Desktop Data on page 93.

Figure 35: Provisioning Client: Converged Desktop Data

    b. Enter **000023345573** in the **Converged Desktop Alias** text box.

       This is the PNI + HLOC + DN of the converged telephone.

    c. Enter one of the following values in the **Converged Desktop Preferred Audio Device** text box:

- If you are configuring UDP, enter the Access Code (AC1 or AC2) + HLOC + DN of the converged telephone.

- If you are configuring CDP, enter the CDP number.

    d. Select **Enterprise Converged User** from the **Converged Desktop User Type** drop-down list.

Chapter 9:   Avaya Communication Server 1000 NRS operation

Contents

This section contains information about the following topics:

Introduction

In a mixed Avaya Communication Server 1000 (Avaya CS 1000) and Avaya Multimedia Communication Server 5100 (Avaya MCS 5100) network, the following is expected to occur:

- The CS 1000 SIP Redirect Server routes calls to SIP endpoints (gateways and SIP lines) under CS 1000 control.

- The MCS 5100 routes calls to SIP endpoints (gateways and SIP clients) under MCS 5100 control.

 Note:
In the context of SIP, the usage of the phrase "endpoint under XXX control" indicates that entity XXX is the only entity that knows the contact information for the endpoint.

Provisioning is impacted in the following manner:

- The CS 1000 SIP Redirect Server represents all CS 1000 endpoints in the MCS 5100 provisioning.
- The MCS 5100 Session Manager represents all MCS 5100 endpoints in the CS 1000 provisioning.

This network configuration requires the CS 1000 SIP Redirect Server and MCS 5100 to interwork seamlessly. Therefore, this section describes the basics of the redirect server logic.

Operation

> ✷ **Note:**
>
> Some values in these instructions are taken from the example numbering plans described Examples of dialing and numbering plan configuration for mixed networks on page 41. Note that actual values may differ between applications.

Domains

The following domains are created in the CS 1000 NRS.

- Service Domain = mydomain.com (see Figure 36: Service Domain on page 97)
- L1 Domain (UDP) = udp (see Figure 37: Level1 Domain on page 97)
- L0 Domain (CDP) = cdp (see Figure 38: Level0 Domain on page 97)

These domains must match in both CS 1000 and in MCS 5100.

Service Domains

Add...

#	ID	Description	# of L1 domains	# of L0 domains	# of gateway endpoints
1	mydomain.com	My domain	1	1	5

Figure 36: Service Domain

L1 Domains (UDP)

Show L1 Domains for (Service Domain):

mydomain.com ▼

Add...

#	ID	Description	# of L0 domains	# of gateway endpoints	# of routing entries
1	udp	My UDP domain	1	5	21

Figure 37: Level1 Domain

L0 Domains (CDP)

Show L0 Domains for (Service Domain / L1 Domain):

mydomain.com ▼ / udp ▼

Add...

#	ID	Ancestor Path	Description	# of gateway endpoints	# of routing entries
1	cdp	mydomain.com/udp	My CDP domain	5	21

Figure 38: Level0 Domain

Gateway Endpoints

The Gateway endpoint is configured with the following properties:

- ID — Node134Loc334

  This matches the SIP Endpoint name configured in CS 1000.

- Call Signaling IP address — 12.123.1.123

  This is the IP address of the CS 1000 IP Telephony node.

- SIP Support — Dynamic SIP endpoint

- SIP Transport Protocol — UDP

- SIP port — 5060

This effectively configures the CS 1000 as an NRS endpoint. See Figure 39: Gateway Endpoints on page 98.

Gateway Endpoints

Show Gateway Endpoints for (Service Domain / L1 Domain / L0 Domain):

mydomain.com ▼ / udp ▼ / cdp ▼

Showing 1 - 5 of 5   < Previous | Next >

#	ID	Support Protocol(s)	Call Signaling IP	Description	# of routing entries
1	Node102Fox3	RAS H.323	47.11.250.102	Fox3 Rls3.0 Nod . . .	4
2	Node109Loc309	RAS H.323 / Dynamic SIP	47.11.250.109 / 47.11.250.109	Fox6 w/Call Pil . . .	8
3	Node122	RAS H.323	47.11.250.122	Rls3 IP-TRUNK 6 . . .	1
4	Node134Loc334	RAS H.323 / Dynamic SIP	12.123.1.123 / 12.123.1.123	Option 81F#3	7

Figure 39: Gateway Endpoints

Routing entries

The routing entries configured in CS 1000 NRS for Gateway Endpoint Node134Loc334 are shown in Figure 40: Routing Entries on page 99.

⊛ **Note:**

SIP URI phone-contexts are configured in CS 1000 NRS as part of the Level1 Domain (udp).

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Routing Entries

Show Routing Entries for (Service Domain / L1 Domain / L0 Domain / Endpoint):

mydomain.com ▼ / udp ▼ / cdp ▼ / Node134Loc334   **Look up**

Showing 1 - 7 of 7   < Previous | Next >

#	DN Prefix	DN Type	Route Cost	SIP URI Phone Context
1	011	E.164 international	1	+
2	1	E.164 national	1	+1
3	334	Level1 regional	1	udp
4	9	E.164 national	1	+1
5	962	E.164 local	1	+1613
6	966	E.164 local	1	+1613
7	968	E.164 local	1	+1613

Figure 40: Routing Entries

Configuring MCS 5100 in the NRS for CDS

To implement CDS in the NRS, configure the MCS 5100 system in one of two ways:

- Configure the MCS 5100 as a Gateway Endpoint.

  This is the preferred option and is the only option when multiple Collaborative Servers are required in the system. When configuring the MCS 5100 as a Gateway Endpoint in the NRS, note the following requirements:

    - endpoint name = convergeddesktop
    - call signaling IP address = 12.234.3.234

      This is the application server logical IP address on the MCS 5100.

    - SIP support = static SIP endpoint
    - SIP transport = UDP

      SIP transport protocol must always be UDP.

    - SIP port = 5060

      Configure at least two routing entries against this Gateway Endpoint, so the Gateway Endpoint can be added to the lookup table.

- Configure the MCS 5100 as a Collaborative Server. See Figure 41: MCS 5100 configured as a Collaborative Server on page 100.

⊛ **Note:**

This option is not recommended and can only be used when no other Collaborative Servers exist in the system.

When you configure the MCS 5100 system in the NRS, note the following requirements:

- domain = 12.234.3.234

- alias name = MyMCSAppSrv

- domain type = L1

- SIP support = yes

Indicate yes for SIP support with a check mark.

- SIP transport = UDP

SIP transport protocol must always be UDP.

- SIP port = 5060

Figure 41: MCS 5100 configured as a Collaborative Server

To route calls from MCS 5100 to CS 1000 when MCS is configured as a Collaborative Server, choose one of two configuration options:

1. Configure MCS 5100 as a Gateway Endpoint for the NRS without routing entries for calls from CS 1000 to MCS 5100. See Figure 42: MCS 5100 configured as a Gateway Endpoint without routing entries on page 102.

   When configuring MCS 5100 in NRS without routing entries, use the following entries:

   - endpoint name = Converged Desktop
   - call signaling IP address = 12.234.3.234

     This is the application server logical IP address on the MCS 5100.

   - SIP support = static SIP endpoint
   - SIP transport = UDP

     SIP transport protocol must be UDP.

   - SIP port = 5060

Figure 42: MCS 5100 configured as a Gateway Endpoint without routing entries

2. Configure telephony routes on MCS 5100 to contact the CS 1000 SIP Gateway directly (not through NRS). See Figure 43: Configure telephony routes on MCS 5100 on page 102. For more information about configuring MCS 5100, see Configuring Avaya Multimedia Communication Server 5100 on page 75.

Figure 43: Configure telephony routes on MCS 5100

# Configuring Meet Me Audio Conferencing in the NRS for CDS

To configure Meet Me Audio Conferencing for CDS, you must point the meetme DID to the convergeddesktop gateway end point as a CDP entry.

> ✳ **Note:**
>
> You must create the convergeddesktop endpoint before beginning this procedure. For more information, see Gateway Endpoints on page 97.

Configure Meet Me Audio Conferencing

1. Log into the Network Routing Service Manager as admin.

2. Select Configuration > Routing Entries.

3. In the top toolbar, select the set Standby DB view radio button.

4. In the Routing Entries section, from the pulldown lists, select the <Domain name>, the <L1 level>, and the <L0 level>.

5. In the Gateway Endpoint field, select convergeddesktop.

6. In the With DN Type field, select the DN Type and click Show.
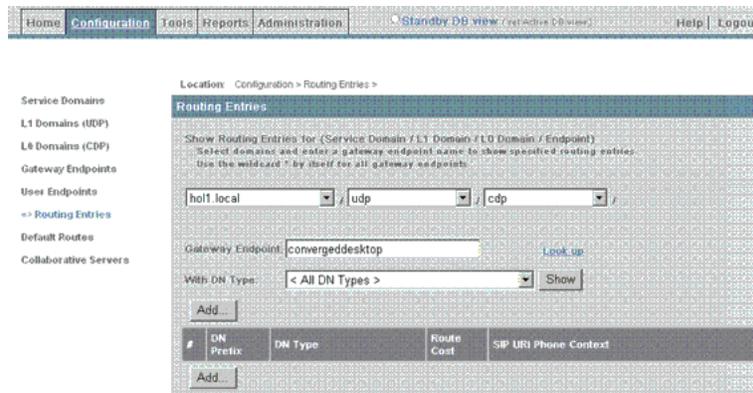
7. Click Add.

8. Enter the DN Type, DN Prefix, and Route Cost.

9. Click Save.

10. Navigate to Tools > Database Actions.

11. Under Select database action, choose Cut over & Commit.
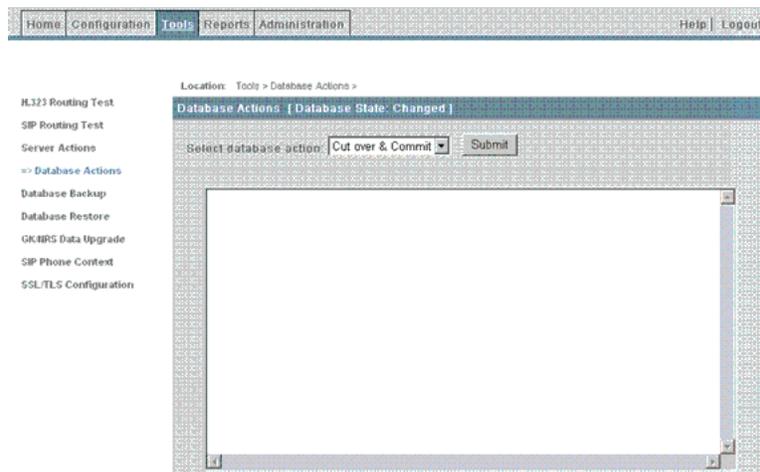
12. Click Submit.

Operational logic

The following items describe the operational logic of the CS 1000 SIP Redirect Server:

- When an INVITE is received, the Public Address is extracted from the request URI.

  For example: INVITE <sip:dn;phone-context=pc@domain;user=phone> SIP/2.0

- When the SIP Redirect Server searches the location database, it tries to find all the endpoints that meet the following requirements:

  - The endpoint belongs to the specified domain.

  - The endpoint has associated routing entries (with both of the following) :

    - exact phone-context as the one received

• longest DN-prefix that prefixes the received DN

Table 17: Search results of location database on page 105 shows the possible outcomes of the location database search.

Table 17: Search results of location database

If...	Then...
The resolution fails (that is, no endpoints are found), the SIP Redirect Server redirects to a collaborating server. ⊛ **Note:** For the purpose of this discussion, the collaborating server can be an MCS 5100 Session Manager.	• Send the 302 response (Moved Temporarily) • Build the Contact Address of the 302 response to include the collaborating servers IP address For example: <sip:dn;phone-context=pc@domain:5060; maddr=47.11.160.010;transport=TCP;user=phone>
A single endpoint is found.	• Send the 302 response (Moved Temporarily) • Build the Contact Address of the 302 response to include the IP address of the endpoint. (This can be IP address of the MCS 5100 Session Manager, for the purposes of this discussion) For example: <sip:dn;phone-context=pc@domain:5060; maddr=47.11.182.170;transport=TCP;user=phone>
Multiple endpoints are found.	• Send the 300 response (Multiple Choices) • Include multiple Contact-headers in the 300 response • Individual Contact-header URIs are built • Contact-headers are ordered according to route costs

Chapter 10:   Maintenance

Contents

This section contains information about the following topics:

Introduction

This section contains Avaya Communication Server 1000 (Avaya CS 1000) Command Line Interface (CLI) commands that can be used with CDS. The CS 1000 CLI commands are used from the Signaling Server CLI.

This section also contains a diagnostic tool available on Avaya Multimedia Communication Server 5100 (Avaya MCS 5100) for verifying translations.

CS 1000 CLI commands

The CS 1000 Signaling Server provides a Command Line Interface (CLI) through a serial port or a Telnet session. This section contains the CLI commands available at that interface that are applicable to CDS.

Signaling Server CLI commands are available at three levels:

- Level One — Operations, Administration, and Maintenance (OAM) shell for basic technician support and general status system checking (**oam>prompt**)

- Level Two — Problem Determination Tool (PDT) shell for expert support; also includes all Level One (OAM) commands (**pdt>prompt**)

- Level Three — Avaya proprietary vxWorks[a] shell for advanced debugging and design support (**prompt**)

> **Note:**
> This section describes the Level One (OAM) and Level Two (PDT) CLI commands. Level Three commands are considered expert support and design level commands, and are not documented here.

You must log in to the Signaling Server to use the CLI commands. Refer to *Avaya Signaling Server IP Line Applications Fundamentals (NN43001-125)* for this procedure.

Table 18: CDS OAM commands on page 108 shows the Converged Desktop CLI commands available on the Signaling Server from the OAM shell.

Table 18: CDS OAM commands

Command	Description
cdsShow	Displays the current Converged Desktop configuration. The command is used to check the basic Converged Desktop application status.
cdsAgentShow	Displays the Personal Call Assistance (PCS) agent's information and status.

Table 19: CDS PDT commands on page 108 shows the Converged Desktop CLI commands available on the Signaling Server from the PDT shell.

Table 19: CDS PDT commands

Command	Description
cdsAmlTrace <level>	Sets the Application Module Link (AML) trace level for the CDS application. Turns tracing on/off for troubleshooting message flows between the Call Server and the Signaling Server. Where <level> is: 0 – turns off trace 1 – prints all input/output AML data buffer 2 – prints all input/output AML data buffer except POLLING message 3 – prints all input/output AML data buffer except POLLING message, with Information Element (IE)-type decoding 4 – prints all input/output AML data buffer except POLLING message, with IE type and data decoding

Command	Description
	⊛ **Note:** This command prints the AML messages to/from the Call Server at the transport layer. Because sending and receiving AML messages is based on each AML link instead of each DN or TN, no good solution exists for filtering on this AML trace tool. Avaya recommends that tracing not be turned on in a busy system. Instead, Avaya recommends the use of the cdsCallTrace tool or vtrkCallTrace tool.
cdsCallTraceSetAll	Enables CDS call trace for all calls. All messages to or from the Call Server and to or from the Signaling Server that relate to a Converged Desktop call are traced.
cdsCallTraceOff	Turns off CDS call trace for all calls.
cdsCallTraceSetDN "callingNum", "calledNum"	Sets CDS call trace for a particular calling party or called party DN. Where: • "callingNum" is the calling party DN. If set to 0, it traces all calls to a non-zero "calledNum". • "calledNum" is the called party DN. If set to 0, it traces all calls from a non-zero "callingNum".

MCS 5100 tools

Translation Verification tool

Use the MCS 5100 Translation Verification tool to check the configuration of CSD telephony route. Use Using the MCS 5100 Translation Verification tool on page 109 to access the Translation Verification tool.

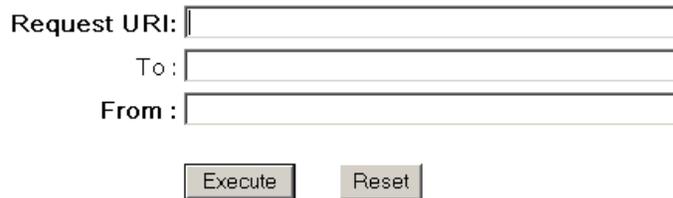**Using the MCS 5100 Translation Verification tool**

1. Enter the IP address of the MCS Provisioning Client in a web browser.

   The Provisioning Client Login web page opens, as shown in Figure 20: Provisioning Client Login on page 78.

2. Enter your **Username** and **Password**.

   The Provisioning Client web page opens, as shown in Figure 21: Provisioning Client on page 78.

3. Select **Provisioning > Domains > mydomain.com > Telephony Routes > Translation Tool** in the MCS Provisioning Client navigator.

The Translation Verification Tool web page opens, as shown in

Figure 44: Provisioning Client: Translation Verification Tool

4. Enter the SIP URI to be traced in the `Request URI` text box.

5. Enter the destination DN in the `To` text box.

6. Enter the originating DN in the `From` text box.

7. Click **Execute**.

Appendix A: Parameter configuration for a SIP URI

Contents

This section contains information about the following topics:

Introduction

The SIP phone-context parameter qualifies a DN beyond the usual Numbering Plan Identifier (NPI) and Type of Number (TON). In addition to the NPI/TON, the phone-context parameter includes unique identification of the scope for which the number is valid.

Example

If a CDP number is dialed within the Site1 CDP domain of the COMPANYABC network and the call must be routed at the network level (by a network router such as an Avaya Communication Server 1000 Redirect Server or Avaya Multimedia Communication Server

Proxy Server), then the number must be qualified with a phone-context parameter that incorporates the following:

- NPI = Private
- TON = CDP
- Enterprise ID = companyabc.com
- CDP Domain ID = Site1

These elements must be combined such that the resulting string is globally unique.

Global uniqueness of phone-context strings is a prerequisite for being able to serve a domain (for example, CDP-domain or an area code) by several systems (such as Avaya CS 1000 and Avaya MCS 5100).

If the Site1 CDP domain is divided between a CS 1000 and an MCS 5100 system, then a globally unique phone-context for Site1 CDP domain must be defined and recognized by both systems.

Note:
The CS 1000 Redirect Server only routes calls to the CS 1000 users and the MCS 5100 only routes calls to MCS 5100 users.

Phone-context strings

Phone-context strings for public numbers

Table 20: Phone-context strings for public numbers (example) on page 112 provides an example for creating phone-context strings for public numbers.

Table 20: Phone-context strings for public numbers (example)

NPI/TON	Scope	Hierarchical domain-based schema	Global prefixed-based schema	CS 1000 SIP Redirect Server implementation
NPI=Public, TON=International	Global	international.e164. myrootdomain	Not applicable DN starts with +.	Not applicable DN starts with +.

NPI/TON	Scope	Hierarchical domain-based schema	Global prefixed-based schema	CS 1000 SIP Redirect Server implementation
NPI=Public, TON=National	National For example, mynation	mynation.national.e164.myrootdomain	+1 (+"country code")	+1 (+"country code")
NPI=Public, TON=Local	Area within nation For example, myarea, mynation	myarea.mynation.local.e164.myrootdomain	+1-613 (+"countrycode"- "area code")	+1-613 (+"country code"-"area code")
NPI=Public, TON=Special	Area within nation For example, myarea, mynation	myarea.mynation.special.e164.myrootdomain	Not supported	Not supported
NPI=Public, TON=Unknown	Area within nation For example, myarea, mynation	myarea.mynation.unknown.e164.myrootdomain	Not supported	Not supported

Phone-context strings for unknown numbers

Table 21: Phone-context strings for unknown numbers (example) on page 113 provides an example for creating phone-context strings for unknown numbers.

Table 21: Phone-context strings for unknown numbers (example)

NPI/TON	Scope	Hierarchical domain-based schema	Global prefixed-based schema	CS 1000 SIP Redirect Server implementation
NPI=Unknown, TON=Unknown	Location For example, mylocation	mylocation.unknown.unknown.myrootdomain	Not supported	Not supported

Phone-context strings for private numbers

Table 22: Phone-context strings for private numbers (example) on page 114 provides an example for creating phone-context strings for private numbers.

Table 22: Phone-context strings for private numbers (example)

NPI/TON	Scope	Hierarchical domain-based schema	Global prefixed-based schema	CS 1000 SIP Redirect Server implementation
NPI=Private, TON=Level 1 Regional (UDP)	Enterprise For example, myenterprise	level1.private. myenterprise	+1-800-466-7 835 (The 1 800 number of the enterprise.)	myl1domain
NPI=Private, TON=Level 0 Regional (CDP)	Location within Enterprise For example, mylocation, myenterprise	mylocation.level0. private.myenterpris e	+1-613-966-0 100 (The listed number of the location.)	myl0domain. myl1domain
NPI=Private, TON=Special	Location within Enterprise For example, mylocation, myenterprise	mylocation.special. private.myenterpris e	+1-613-966-0 100-1 (The listed number of the location extended with 1.)	myspeciallabel. myl1domain
NPI=Private, TON=Unknow n (see Note)	Location within Enterprise For example, mylocation, myenterprise	mylocation.unknow n. private.myenterpris e	+1-613-966-0 100-2 (The listed number of the location extended with 2.)	myunknownlabel.m yl0domain. myl1domain

> **Note:**
> CS 1000 SIP Redirect Server does not support NPI=Public/TON=Unknown numbers.

Appendix B: PC Client windows

on page 115 describes pop-up and other windows that may open and close on the PC Client when two users make a call on an Enterprise switch.

Table 23: PC Client: user expectations

	Network Call Log	Popup Window	Video	PCC Call Log/ Auto Presence	Terminating PCC's call window closes when originating PCC closes call window	Originating PCC's call window closes when terminating PCC closes call window
CD1 to CD2	CD1 no CD2 yes	CD1 no CD2 yes	No	CD1 no CD2 yes	N/A	N/A
CD2 to CD1	CD1 yes CD2 yes	CD1 yes CD2 yes	No	CD1 call log yes auto presence no CD2 yes	No	No
CD2 to SIP	CD2 yes SIP yes	CD2 yes SIP yes	Yes	CD2 yes SIP yes	No	Yes
SIP to CD2	SIP yes CD2 yes	SIP yes CD2 yes	Yes	SIP yes CD2 yes	Yes	No
CD2 to CD2	CD2 yes CD2 yes	CD2 yes CD2 yes	Yes	CD2 yes CD2 yes	Yes	Yes
CD2 to CD2 with ring list	CD2 yes CD2 yes	CD2 yes CD2 yes	Yes	CD2 yes CD2 yes	No	No
CD2 to PSTN	CD2 yes PSTN n/a	CD2 no PSTN n/a	No	CD2 yes PSTN n/a	N/A	N/A
PSTN to CD2	PSTN n/a CD2 yes	PSTN n/a CD2 yes	No	PSTN n/a CD2 yes	N/A	N/A