Complete Reference Guide for Cisco Unity Connection Survivable Remote Site Voicemail (SRSV)

Release 9.1(1)
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Preface

Audience and Use

The *Cisco Unity Connection Survivable Remote Site Voicemail Guide for Cisco Unity Connection* contains information and instructions for working with Cisco Unity Connection Survivable Remote Site Voicemail.

Documentation Conventions

Table 1 Conventions in the System Administration Guide for Cisco Unity Connection

Convention	Description
boldfaced text	Boldfaced text is used for: • Key and button names. (Example: Select **OK**.) • Information that you enter. (Example: Enter **Administrator** in the Username box.)
< > (angle brackets)	Angle brackets are used around parameters for which you supply a value. (Example: In your browser, go to **https://<Cisco Unity Connection server IP address>/cuadmin**.)
- (hyphen)	Hyphens separate keys that must be pressed simultaneously. (Example: Press **Ctrl-Alt-Delete**.)
> (right angle bracket)	A right angle bracket is used to separate selections that you make in the navigation bar of Cisco Unity Connection Administration. (Example: In Cisco Unity Connection Administration, expand **Contacts > System Contacts**.)

The *Cisco Unity Connection Survivable Remote Site Voicemail Guide for Cisco Unity Connection* also uses the following conventions:

Note Means reader take note. Notes contain helpful suggestions or references to material not covered in the document.

Tip Means the following information may help you solve a problem.

Caution Means reader be careful. In this situation, you might do something that could result in equipment damage or loss of data.

Cisco Unity Connection Documentation

For descriptions and URLs of Cisco Unity Connection documentation on Cisco.com, see the *Documentation Guide for Cisco Unity Connection*. The document is shipped with Connection and is available at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/8x/roadmap/8xcucdg.html.

Documentation References to Cisco Unified Communications Manager Business Edition

The name of the product known as Cisco Unified Communications Manager Business Edition in versions 8.0 and earlier has been changed to Cisco Unified Communications Manager Business Edition 5000 in versions 8.5 and later.

In the Cisco Unity Connection 8.x documentation set, references to "Cisco Unified Communications Manager Business Edition" and "Cisco Unified CMBE" apply to both Business Edition version 8.0 and to Business Edition 5000 versions 8.5 and later. The references do not apply to Business Edition 6000.

Obtaining Documentation and Submitting a Service Request

For information on obtaining documentation, submitting a service request, and gathering additional information, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html

Subscribe to the *What's New in Cisco Product Documentation* as a Really Simple Syndication (RSS) feed and set content to be delivered directly to your desktop using a reader application. The RSS feeds are a free service and Cisco currently supports RSS Version 2.0.

Cisco Product Security Overview

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

Further information regarding U.S. export regulations can be found at http://www.access.gpo.gov/bis/ear/ear_data.html.

CHAPTER 1

Overview of Cisco Unity Connection SRSV in Unity Connection 9.1(1) and Later

Cisco Unity Connection Survivable Remote Site Voicemail (Unity Connection SRSV) is a backup voicemail solution that allows you to receive voice messages during WAN outages. It works in conjunction with Cisco Unified Survivable Remote Site Telephony (SRST) for providing voicemail service to a branch when the connectivity with the central Unity Connection voicemail service is lost.

Unity Connection SRSV is used in the centralized Cisco Unified Communications Manager and Cisco Unity Connection environment with multiple branch offices or small sites. It provides limited voicemail and auto-attendant features that remain in synchronization with the central Unity Connection voicemail service so that when the WAN outage or failure occurs, the Unity Connection SRSV solution can provide voicemail service to the subscribers at the branch. However, as soon as the network is restored, all the voicemails received by the branch subscribers are automatically uploaded to the central Unity Connection voicemail server.

Unity Connection SRSV solution requires the following two components:

- Cisco Unity Connection: It is deployed at the central site alongside with Cisco Unified CM to deliver powerful integrated messaging and voicemail services.
- Unity Connection SRSV: The SRSV component is natively a part of Unity Connection which is deployed at the branch site alongside with Cisco Unified CM Express or Cisco Unified Survivable Remote Site Telephony (SRST). Unity Connection SRSV is hosted on Cisco Integrated Service Routers Generation 2 (ISR G2) platform by using Services Ready Engine Virtualization.

See the following sections:

- Working with Unity Connection SRSV, page 1-1
- Workflow in Cisco Unity Connection SRSV, page 1-4

Working with Unity Connection SRSV

Unity Connection SRSV becomes active during WAN outages and acts as a backup of voice messaging system at the branch sites. It allows the users at branch offices to receive the voice messages during WAN outages.

In Unity Connection 9.1(1), Unity Connection SRSV involves provisioning of Unity Connection through the command line interface (CLI) to run it in SRSV mode, SRST/E-SRST references Cisco Unified Communications Manager, and all the SRSV related functionalities, such as user(s) provisioning and voicemail upload, are managed by the central Unity Connection server. Once the entire Unity Connection SRSV system is deployed and provisioned, it remains in the idle state at the branch site and is ready to receive calls from the SRST system (either SRST or CUCME-as-SRST). The SRST component also remains idle and wait for IP phones to register with it. When the WAN outage occurs, the branch office IP phones that are registered to the central Cisco Unified Communications Manager detect the loss of connectivity and re-home to the SRST. Now, all the incoming calls to the branch are handled by the SRST. For calls that are either no-answer or reach a busy line, SRST forwards the call to the CUC-SRSV voicemail server that allows the caller to leave a voice message for the branch user. As a result, the branch office voicemail is supported during WAN outages when the central office voicemail system is unreachable.

However, when the WAN connection is restored, the IP phones automatically re-home to the central Cisco Unified Communications Manager. All the calls are then managed by Cisco Unified Communications Manager and the no-answer / busy calls are forwarded to the central Unity Connection voicemail system and all the voicemails stored on the branch get automatically synchronized with the central Unity Connection voicemail.

Supported SRSV Topologies

Unity Connection SRSV supports several topologies based on the configuration of the router. You can deploy either original SRST or CUCME-as-SRST (also known as SRST Fallback Mode) at branch.

Note If you are running SRST at the branch site, you cannot also deploy the E-SRST feature.

Following figures show three topolgies supported by Unity Connection SRSV:

Figure 1-1: Shows a topology in which SRST is deployed at the branch site. If the WAN outage occurs or PSTN goes down, Unity Connection SRSV at the branch site provides limited voicemail support in the failover mode.

Figure 1-2: Shows a topology where CUCME-as-SRST (also known as SRST Fallback Mode) is providing call control at the branch site.

Figure 1-3: Shows a topology where multiple CUCME-as-SRST and SRSV-CUE devices are paired for load balancing at the survivable branch site. In this scenario, the administrator uses Cisco Unified Communications Manager to divide the branch users between CUCME-SRST-1 and CUCME-SRST-2. The central Unity Connection server detects that and then sends the appropriate configuration to SRSV-1 and SRSV-2 at the branch site. In the event of WAN failure, each SRSV device will handle calls directed to it from the paired CUCME-as-SRST device.

Figure 1-1 Topology 1

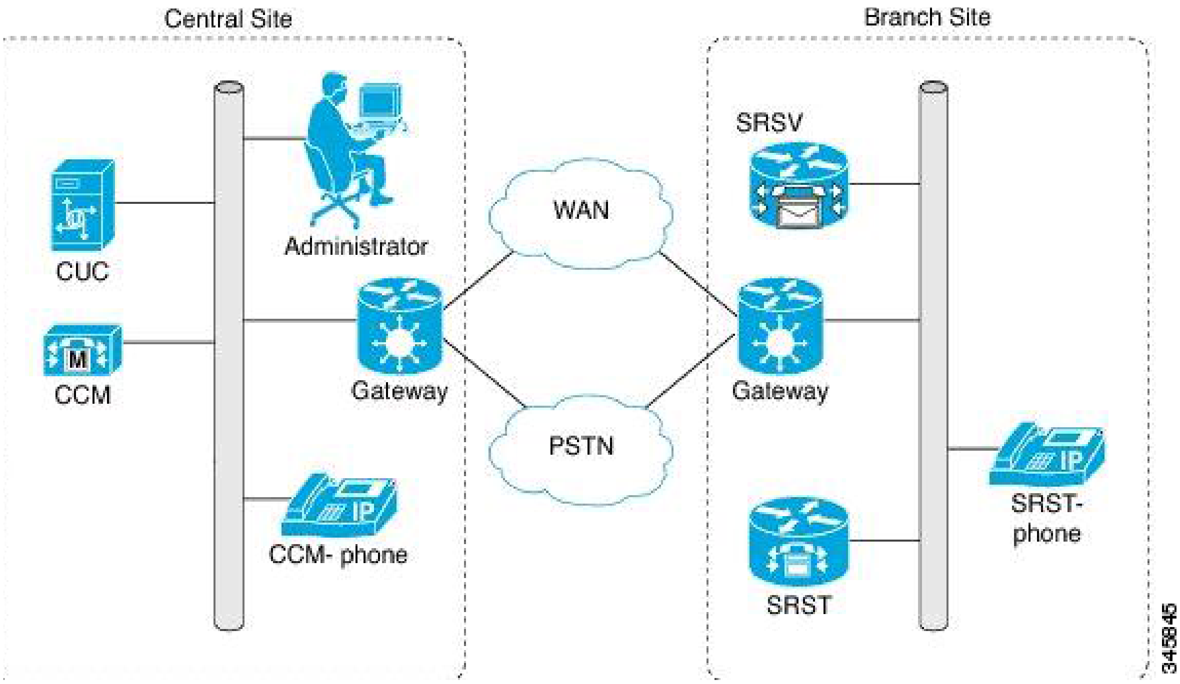


Figure 1-2 Topology 2

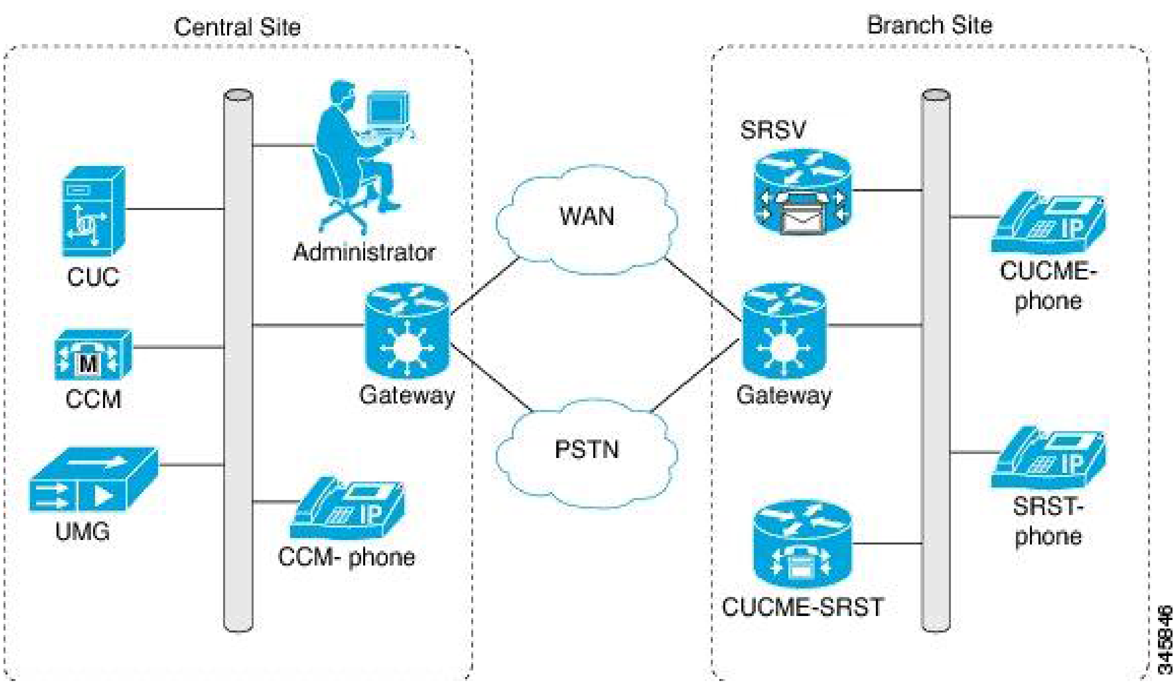
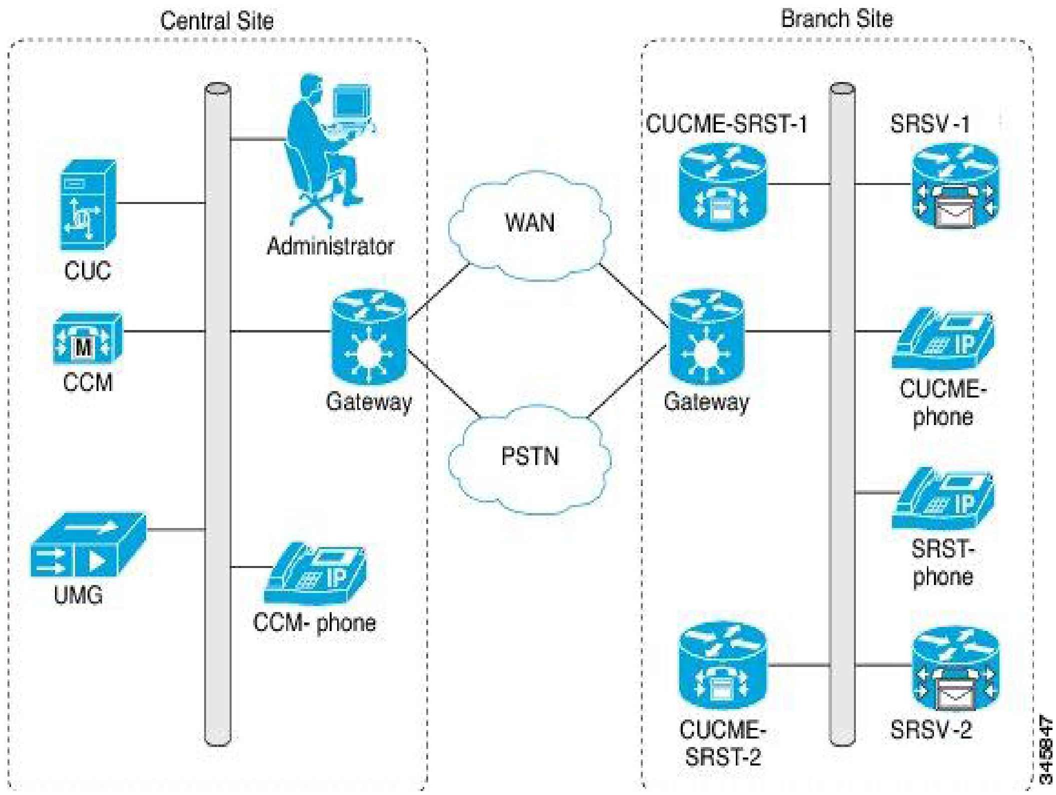


Figure 1-3 Topology 3



Workflow in Cisco Unity Connection SRSV

1. The administrator installs Cisco Unity Connection on SRE-900/SRE-910 series blade or MCS 7845/MCS 7825. For more information refer to http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/installation/guide/9xcucigx.html .
2. Unity Connection starts in the Demo mode. Run the CLI command **utils cuc activate CUSRSV** to convert standalone Unity Connection server to Unity Connection SRSV server. For more information on installation of Unity Connection SRSV, refer to the "Overview of Mandatory Tasks for Installing a Cisco Unity Connection SRSV System"chapter of this guide.
3. The Unity Connection SRSV server disables some of the Unity Connection components and displays only the following Unity Connection components within the Connection SRSV Administration:
   - Users, with the list of administrators and subscribers of the branch. For more information on the user settings of Unity Connection SRSV, refer to the "Cisco Unity Connection SRSV Administration - User Settings Interface" chapter of this guide.
   - Templates, with only the Call Handler templates. For more information on the template settings of Unity Connection SRSV, refer to the "Cisco Unity Connection SRSV Administration - Template Settings Interface" chapter of this guide.

- Distribution Lists, with only System Distribution Lists. For more information on the distribution lists, refer to the "Cisco Unity Connection SRSV Administration - Template Settings Interface" chapter of this guide.
- Call Management, with only System Call Handlers and Directory Handlers. For more information on call management for Unity Connection SRSV, refer to the "Cisco Unity Connection SRSV Administration - Call Management Settings Interface" chapter of this guide.
- Networking, with central server configuration. For more information on the user settings of Unity Connection SRSV, refer to the "Cisco Unity Connection SRSV Administration - Networking Settings Interface" chapter of this guide.
- System Settings, with only Schedules, Conversations, Enterprise Parameters, Plugins. For more information on the user settings of Unity Connection SRSV, refer to the "Cisco Unity Connection SRSV Administration - System Settings Interface" chapter of this guide.
- Telephony Integrations, with only Phone System, Port Group, Port, Security. For more information on telephony integrations, refer to the "Cisco Unity Connection SRSV Administration - Telephony Integration Settings Interface" chapter of this guide.
- Tools, with only Custom Keypad Mapping. For more information on the user settings of Unity Connection SRSV, refer to the "Cisco Unity Connection SRSV Tool Settings" chapter of this guide.

4. The administrator logs into the Connection Administration page and navigates to the Branch Management page. For more information on how to configure the central Unity Connection server for Unity Connection SRSV, refer to the "Configuring Cisco Unity Connection SRSV Settings in Cisco Unity Connection 9.1(1) and Later" chapter of this guide.
5. The administrator enters the Fully Qualified Domain Name (FQDN), administrator username, and password for the branch connection node. Unity Connection and Unity Connection SRSV verifies registration and associates the branch to the Unity Connection server. For more information on how to configure the central Unity Connection server for Unity Connection SRSV, refer to the "Configuring Cisco Unity Connection SRSV Settings in Cisco Unity Connection 9.1(1) and Later" chapter of this guide.
6. You must set a method to provision the users from the central Unity Connection server to the branch system. For more information refer to the "Configuring Cisco Unity Connection SRSV Settings in Cisco Unity Connection 9.1(1) and Later" chapter of this guide.
7. The administrator imports the subscribers by searching on users details, such as extension/phone number, already existing in Unity Connection and selects those users. For more information on how to configure the central Unity Connection server for Unity Connection SRSV, refer to the "Configuring Cisco Unity Connection SRSV Settings in Cisco Unity Connection 9.1(1) and Later" chapter of this guide.
8. The administrator selects the "Sync Provisioning" button to push the subscribers to Unity Connection SRSV. The provisioned status is displayed on the Connection SRSV Administration page. For more information on how to configure the central Unity Connection server for Unity Connection SRSV, refer to the "Configuring Cisco Unity Connection SRSV Settings in Cisco Unity Connection 9.1(1) and Later" chapter of this guide.

CHAPTER 2

Compatibility Matrix, Software Requirements, and Licensing Requirements

This document lists the supported version combinations for Cisco Unity Connection SRSV and Cisco Unified SRST when they are integrated through a SIP trunk or SCCP. In addition, the document has also software and licensing requirements for Cisco Unity Connection SRSV. It contains the following sections:

- Supported Version Combinations of Cisco Unity Connection SRSV and Cisco Unified SRST, page 2-1
- Software Requirements—Administrator Workstations (Cisco Unity Connection SRSV), page 2-2
- Licensing Requirements for Cisco Unity Connection SRSV, page 2-2

Supported Version Combinations of Cisco Unity Connection SRSV and Cisco Unified SRST

Table 1 Supported Version Combinations of Cisco Unity Connection SRSV and Cisco Unified SRST

Cisco Unity Connection SRSV	Cisco Unified SRST/CME-SRST	Cisco Unified E-SRST
9.1(1)	8.6 and higher	8.6 and higher

Software Requirements—Administrator Workstations (Cisco Unity Connection SRSV)

Table 2 Supported Operating Systems and Browsers on Administrator Workstations for Cisco Unity Connection SRSV

Operating System on Administrator Workstation	Browser on Administrator Workstation
Microsoft Windows Vista (32 bit and 64 bit)	• Microsoft Internet Explorer 7.0, 8.0 and 9.0(32 bit) • Mozilla Firefox 3.6, 10
Microsoft Windows XP	• Microsoft Internet Explorer 7.0 and 8.0 • Mozilla Firefox 3.6, 10.
Mac OS X 10.4 and later	• Mozilla Firefox 3.6, 10 • Safari 5.1
Red Hat Enterprise Linux	Mozilla Firefox 3.6, 10
Microsoft Windows 7 (32 and 64 bit)	• Microsoft Internet Explorer 7.0, 8.0 and 9.0 • Mozilla Firefox 3.6, 10

Licensing Requirements for Cisco Unity Connection SRSV

Connection SRSV is a licensed feature for which you need to install the SRSV specific license on the Enterprise License Manager (ELM) server for the central Connection. For more information on installing licenses on the ELM server, refer to the ELM user guide available at http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/elmuserguide/9_0_1/CUCM_BK_E596FD72_00_enterprise-license-manager-user-90.html.

The number of licenses installed for Connection users and Connection SRSV feature is reflected under the **CUC_EnhancedMessaging** tag on the **License** page of Cisco Unity Connection Administration. For more information on licenses installed on the central Connection, refer to the "Managing Licenses in Cisco Unity Connection 9.x" chapter of the *System Administration Guide for Cisco Unity Connection Release 9.x* at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/administration/guide/9xcucsagx.html.

When the central Connection license status is "**Compliance**" or "**Violation**", all the functionalities (user provisioning and voicemail upload) related to Connection SRSV work as expected in a normal scenario. However, when the central Connection server license status is "**Expire**", the synchronization of users from the central Connection to the branch stops working. However, the voicemail and auto-attendant functionalities still work at the branch.

CHAPTER 3

Overview of Mandatory Tasks for Installing a Cisco Unity Connection SRSV System

The Cisco Unity Connection SRSV (branch) can be installed on SRE blade (mounted on Cisco Unified SRST router), virtual machine and MCS 7845/7825. For SRST blade, user needs to virtualize SRE blade and over that install Branch. SRE virtualization for the branch install is supported with VMware ESXi v5.0 and 5.1. For more information, refer to the SRE virtualization documentation.

The following three Cisco Unified SRST configurations are supported:

- SRST: Survivable Remote Site Telephony
- E-SRST: Enhanced SRST
- CME-SRST: Call Manager Express as SRST

Note For installing Cisco Unity Connection on Virtual Machine and MCS 7845/7825, refer to the *Installation Guide for Cisco Unity Connection* available at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/installation/guide/9xcucigx.html.

The tasks reference instructions in Cisco Unity Connection documentation as noted. Follow the documentation for a successful installation.

Note Install Cisco Unity Connection 9.1(1), either as a cluster setup or as part of digital networking.

Some of the tasks apply only to specific situations, and are noted as such. If a task does not apply to your situation, skip it.

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Note The Cisco Unity Connection SRSV system can have only one locale installed.

1. *If you want Cisco Unity Connection SRSV Administration to be localized to Japanese locale:* Download and install the Cisco Unity Connection Japanese locale. See the "Locale Installation" section in the "Software Upgrades" chapter of the applicable *Cisco Unified Communications Operating System Administration Guide* at http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_maintenance_guides_list.html.

2. Secure the communication between central and branch after the installation is complete. This can be done by either uploading signed certificates to the Central server and to the or by allowing the Central server and the to use self-signed certificates. For more information, refer to the "Securing Connections in Cisco Unity Connection Survivable Remote Site Voicemail 9.1(1)" chapter of this guide.
3. *If you installed additional languages and you want the Cisco Personal Communications Assistant to be localized:* Download and install the corresponding Cisco Unified Communications Manager locales. See the "Locale Installation" section in the "Software Upgrades" chapter of the *Cisco Unified Communications Operating System Administration Guide* at http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_maintenance_guides_list.html.

Once completed with the mandatory tasks, follow the given steps:

1. Download and install the Real-Time Monitoring Tool software on administrator workstations. See the "Installing and Configuring Real-Time Monitoring Tool" chapter of the *Cisco Unified Real-Time Monitoring Tool Administration Guide*
2. Store all of the software that was shipped with together in a location that is safe and can be readily accessed.

CHAPTER 4

Upgrading to Cisco Unity Connection SRSV 9.1 Version

This chapter contains the following sections:

About Upgrades to Connection SRSV 9.1

At the start of the upgrade, you choose whether to restart to the inactive partition. If you choose to restart, when the upgrade is complete, the Connection SRSV automatically restarts, running the upgraded version of Connection SRSV. If you choose not to restart, after the upgrade is complete, you must manually switch to the upgraded version.

Note the following considerations about upgrading a Connection SRSV to version 9.1:

- If you have never upgraded the Connection SRSV before, the upgrade copies the new version to an empty partition.
- Upgrading to Connection SRSV 9.1 requires approximately four hours.
- Switching to the upgraded software requires approximately two hours.
- During the switchversion of the Cisco Unity Connection SRSV, it is recommended to stop the automatic provisioning/vmupload feature at the central Cisco Unity Connection.

Note To stop the automatic provisioning/vmupload feature, ucheck the **Enabled** checkbox onthe Branch Listing page in Cisco Unity Connection Administration.

Status of Connection Features During the upgrade to Connection SRSV 9.1

During the switch version to the upgraded software, all the telephone user interface (touchtone conversation) features and web features gets completely disabled for approximately 1 hour.

Task List for Upgrading to Connection SRSV 9.1 Version

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Do the following tasks to upgrade an existing Connection SRSV 9.1 to the shipping Connection SRSV9.1 version when no Connection cluster is configured.

1. *If you are upgrading Connection SRSV on a Cisco MCS 7825-H3 server or the equivalent HP DL320G5:* Confirm that you have a 128 GB or larger USB flash drive or external hard disk.

   During the upgrade, disk drives in the Connection server are converted from hardware-based RAID to software-based RAID. Before the RAID conversion, the USB drive is reformatted, and data and voice messages on the Connection server are copied to the drive. After the RAID reconfiguration, data and voice messages are copied back to the disk drives in the Connection server.

Caution Do not use a USB drive that contains data that you want to keep. During the upgrade, the USB drive is reformatted, and all existing data on the drive is destroyed.

2. Review the list of features that are disabled or that have limited functionality during the upgrade. See the "Status of Connection Features During the upgrade to Connection SRSV 9.1" section on page 4-2.
3. *If you are upgrading from Connection SRSV 9.1 to Connection SRSV 9.1:* Run the CLI command **run cuc preupgrade test** to verify the prerequisites before running the upgrade.
4. See the applicable version of *Release Notes for Cisco Unity Connection* for any additional information on upgrading to the shipping version. In particular, note the items in the "Installation and Upgrade Information" section. Release notes are available at http://www.cisco.com/en/US/products/ps6509/prod_release_notes_list.html.
5. *If you do not have a backup from replacing hard disks or replacing the server:* Back up the server by using the Disaster Recovery System. For more information, see the applicable *Disaster Recovery System Administration Guide for Cisco Unity Connection* at http://www.cisco.com/en/US/products/ps6509/prod_maintenance_guides_list.html.
6. Upgrade the Connection SRSV software. See the applicable section:
   - Upgrading Connection SRSV 9.1 Software to the Shipping Connection SRSV 9.1 Version from a Local DVD, page 4-3
   - Upgrading Connection SRSV 9.1 Software to the Shipping Connection SRSV 9.1 Version from a Network Location, page 4-4

Note If you have the Japanese locale already installed on your Connection server, make sure to uninstall, uc-locale-ja_JP-9.1.0.1-xx.cop.sgn, the default Japanese cop file before upgrading to Connection SRSV server. After the Connection SRSV server is installed, you can install, uc-locale-SRSV-ja_JP-9.1.0.1-xx.cop.sgn, the SRSV specific Japanese cop file to have the Japenese locale.

7. Switch to the upgraded software on the Connection SRSV. See the Switching to the Upgraded Version of Connection 9.x Software section of the *Reconfiguration and Upgrade Guide for Cisco Unity Connection* at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/upgrade/guide/9xcucrugx.html.

Upgrading Connection SRSV 9.1 Software to the Shipping Connection SRSV 9.1 Version from a Local DVD

To upgrade Connection SRSV from a local DVD, you can do either of the following:

- Use a DVD shipped from Cisco.
- Download a signed .iso file from Cisco.com, and burn a disc image of the downloaded software. Burning a disc image extracts the files from the .iso file that you downloaded and writes them to a DVD.

To follow the upgrade process through the CLI interface, see the utils system upgrade command at the *Command Line Interface Reference Guide for Cisco Unified Communications Solutions* at http://www.cisco.com/en/US/products/ps6509/prod_maintenance_guides_list.html.

To Upgrade to the Connection SRSV 9.1 Version from a Local DVD

Step 1 Insert the DVD that contains Connection SRSV into the disc drive on the Connection SRSV.

Step 2 Sign in to Cisco Unified Operating System Administration.

Step 3 From the Software Upgrades menu, select **Install/Upgrade**.

Step 4 On the Software Installation/Upgrade page, in the Source field, select **DVD/CD**.

Step 5 In the Directory field, enter a forward slash (**/**).

Step 6 Select **Next**.

Step 7 Select the upgrade version that you want to install, and select **Next**. The upgrade file is copied to the hard disk on the Connection SRSV. When the file is copied, a screen displays the checksum value.

Step 8 Verify the checksum.

Step 9 On the next page, monitor the progress of the upgrade.

If you lose your Connection SRSV with the remote server or close your browser during this step, you may see the following message when you try to view the Software Installation/Upgrade page again:

Warning: Another session is installing software, click Assume Control to take over the installation.

To continue monitoring the upgrade, select **Assume Control**.

You can also monitor the upgrade by using the Real-Time Monitoring Tool.

Step 10 Select **Next**.

Note

- For without a Connection cluster, you can select either the manual switch version or the automatic switch version option.
- For a Connection cluster, you need to select the manual switch version option,

During the initial phase of the upgrade, the Installation Log text box in Cisco Unified Operating System Administration is updated with information on the progress of the upgrade, but updates stop after the server automatically restarts for the first time. To determine when the upgrade is complete, view the console for the Connection SRSV: the console screen displays a message indicating that the installation is complete, and the login prompt for the command-line interface appears.

Step 11 To verify the success of the upgrade, run the CLI command **show cuc version**. The upgrade succeeded if the active partition has the upgraded version and the inactive partition has the old version.

Upgrading Connection SRSV 9.1 Software to the Shipping Connection SRSV 9.1 Version from a Network Location

To upgrade Connection SRSV from a network location, you must download a signed .iso file from Cisco.com, and copy the .iso file to an FTP or SFTP server. Connection SRSV does not allow you to upgrade by copying either the contents of a DVD shipped from Cisco or the extracted contents of a downloaded .iso file to an FTP or SFTP server. This helps prevent someone from attempting to upgrade by using software that has been tampered with.

To follow the upgrade process through the CLI interface, see the utils system upgrade command at the *Command Line Interface Reference Guide for Cisco Unified Communications Solutions* at http://www.cisco.com/en/US/products/ps6509/prod_maintenance_guides_list.html.

To Upgrade to the Connection SRSV 9.1 Version from a Network Location

Step 1 Copy the upgrade file to a folder on an FTP or SFTP server that the Connection SRSV can access.

Step 2 Sign in to Cisco Unified Operating System Administration.

Step 3 From the Software Upgrades menu, select **Install/Upgrade**.

Step 4 On the Software Installation/Upgrade page, in the Source field, select **Remote Filesystem**.

Step 5 In the **Directory** field, enter the path to the folder that contains the upgrade file.

If the upgrade file is located on a Linux or Unix server, you must enter a forward slash (/) at the beginning of the folder path. (For example, if the upgrade file is in the upgrade folder, you must enter **/upgrade**.)

If the upgrade file is located on a Windows server, you must use the applicable syntax for an FTP or SFTP server such as:

- The path must begin with a forward slash (/) and contain forward slashes throughout instead of backward slashes (\).
- The path must start from the FTP or SFTP root folder on the server and must not include a Windows absolute path, which starts with a drive letter (for example, C:).

Step 6 In the **Server** field, enter the server name or IP address.

Step 7 In the **User Name** field, enter the alias that will be used to sign in to the remote server.

Step 8 In the **User Password** field, enter the password that will be used to sign in to the remote server.

Step 9 In the **Transfer Protocol** field, select the applicable transfer protocol.

Step 10 Select **Next**.

Step 11 Select the upgrade version that you want to install and select **Next**. The upgrade file is copied to the hard disk on the Connection SRSV server. When the file is copied, a screen displays the checksum value.

Step 12 Verify the checksum.

Step 13 On the next page, monitor the progress of the upgrade.

If you lose your Connection SRSV with the remote server or close your browser during this step, you may see the following message when you try to view the Software Installation/Upgrade page again:

Warning: Another session is installing software, click Assume Control to take over the installation.

To continue monitoring the upgrade, select **Assume Control**.

You can also monitor the upgrade by using the Real-Time Monitoring Tool.

Step 14 Select **Next**.

Note

- For without a Connection cluster, you can select either the manual switch version or the automatic switch version option.
- For a Connection cluster, you need to select the manual switch version option,

During the initial phase of the upgrade, the Installation Log text box in Cisco Unified Operating System Administration is updated with information on the progress of the upgrade, but updates stop after the server automatically restarts for the first time. To determine when the upgrade is complete, view the console for the Connection SRSV: the console screen displays a message indicating that the installation is complete, and the login prompt for the command-line interface appears.

Step 15 To verify the success of the upgrade, run the CLI command **show cuc version**. The upgrade succeeded if the active partition has the upgraded version and the inactive partition has the old version.

CHAPTER 5

Configuring Cisco Unity Connection SRSV Settings in Cisco Unity Connection 9.1(1) and Later

See the following sections:

Methods of Provisioning and Voicemail Upload

Connection 9.1(1) supports two methods for provisioning and voicemail upload between Connection and Connection SRSV. You can use either of the following methods to provision the users from the central Connection server to the branch system:

- **Manual Synchronization of Provisioning**: To manually provision the users on the branch system, navigate to **Branch Management** > **Edit Branch** on Cisco Unity Connection Administration (CUCA) and select **Sync Provisioning**.
- **Automatic Synchronization of Provisioning**: To automatically enable the provisioning of the users, navigate to **Tools** > **Task Management** on Cisco Unity Connection Administration (CUCA) and schedule **Branch Provisioning Synchronization Task**. For more information on scheduling a task, refer to the "Cisco Unity Connection 9.x Tool Settings" chapter of *Interface Reference Guide for Cisco Unity Connection Administration Release 9.x* guide at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/gui_reference/guide/9xcucgrgx.html.

You can use either of the following methods to upload voicemails from the branch system to the central Connection server:

- **Manual Synchronization of Voicemail Messages**: To manually upload voicemails from the branch to the central Connection server, navigate to **Branch Management** > **Edit Branch** on Cisco Unity Connection Administration (CUCA) and select **Voicemail Upload**.

- **Automatic Synchronization of Voicemail Messages**: To automatically enable the uploading of voice messages from the branch to the central Connection server, navigate to **Tools** > **Task Management** on Cisco Unity Connection Administration (CUCA) and schedule **Branch Voice mail polling task**. For more information on scheduling a task, refer to the "Cisco Unity Connection 9.x Tool Settings" chapter of *Interface Reference Guide for Cisco Unity Connection Administration Release 9.x* guide at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/gui_reference/guide/9xcucgrgx.html.

Configuring an SRSV User

You can either create a new user or update an existing user to provide access to the SRSV feature. Before creating an SRSV user, make sure that all the required services, such as Connection REST Service and Connection Branch Sync Service are started on the central Connection server and on the branch system. For more information on services required for SRSV feature, refer to the *Administration Guide for Cisco Unity Connection Serviceability Release 9.x* at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/serv_administration/guide/9xcucservagx.html.

Task List to Create a Connection SRSV User

Create a Connection SRSV user

Step 1 *Creating a Partition*: Create a partition on the Connection server. For more information on how to create a partition, refer to the "Managing Partitions in Cisco Unity Connection 9.x" section of the "Managing Partitions and Search Spaces in Cisco Unity Connection 9.x" chapter of the *System Administration Guide for Cisco Unity Connection Release 9.x* guide at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/administration/guide/9xcucsagx.html.

Step 2 *Creating a Branch*: Create a branch on the Connection server with details of the Cisco Unity Connection SRSV server, which corresponds to the partition created above. For more information on how to create a branch, refer to the Managing Branches in Cisco Unity Connection 9.1(1), page 5-5 section.

Step 3 *Creating a New User or Assigning the Partition to an Existing User*: Assign the partition created above to an existing user or create a new user and assign the partition to the user for providing access to the SRSV feature. For more information on how to create a user, refer to the "Creating Cisco Unity Connection 9.x User Accounts in Cisco Unity Connection Administration" section of the "Adding Cisco Unity Connection 9.x Accounts Individually" chapter of the *User Moves, Adds, and Changes Guide for Cisco Unity Connection Release 9.x* guide at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/user_mac/guide/9xcucmacx.html.

Branch Listing

Table 5-1 Branch Listing Page

Field	Description
Display Name	*(Display only)* Displays the name of the branch.
Server Address	*(Display only)* Displays the IP address/Fully Qualified Domain Name (FQDN) of the branch.
Enabled	*(Display only)* Displays whether the branch is active or not.
Status	*(Display only)* Displays whether the branch is connected with the central Connection server or not.

New Branch

Table 5-2 New Branch Page

Field	Description
Display Name	Enter a name for the branch.
Server Address	Enter the IP address/FQDN of the branch.
User Name	Enter the username of the branch administrator.
Password	Enter the password of the branch administrator.
SMTP Domain	Enter the Fully Qualified Domain Name (FQDN) of the branch.
PAT Port Number	Enter a PAT port number that the central Connection server uses to communicate with the branch. This port number specifies the port on the public side of the NAT, which further maps to port 443 for communicating with the branch.
Partition	Select the partition that you want to assign to the branch from the dropdown list. **Note** You cannot assign same partition to multiple branches.
Operator	Select the subscriber that must be used as operator to manage the operator messages received at the branch during WAN outages.
Provisioning Sync Options	
Sync voice name for users	Check this check box to synchronize the recorded voice names of the branch users from central Cisco Unity Connection to the branch.
Sync greetings for users	Check this check box to synchronize the greetings of the branch users from central Connection to the branch.
Save	Select this option to save the branch with the specified settings. **Note** As soon as you save the branch details, buttons, such as Sync Provisioning and Voicemail Upload get visible. For more information, refer to the Edit Branch, page 5-4 section.

Edit Branch

Table 5-3 Edit Branch Page

Field	Description
Enabled	Check this check box to enable or activate the branch for provisioning and voicemail upload operations.
Display Name	Enter a name for the branch.
Server Address	Enter the IP address/FQDN of the branch.
User Name	Enter the username of the branch administrator.
Password	Enter the password of the branch administrator.
SMTP Domain	Enter the Fully Qualified Domain Name (FQDN) of the branch.
PAT Port Number	Enter a PAT port number that the central Connection server uses to communicate with the branch. This port number specifies the port on the public side of the NAT, which further maps to port 443 for communicating with the branch.
Partition	Select the partition that you want to assign to the branch from the dropdown list. **Note** You cannot assign same partition to multiple branches.
Operator	Select the subscriber that must be used as operator to manage the operator messages received at the branch during WAN outages.
Provisioning Sync Options	
Sync voice name for users	Check this check box to synchronize the recorded voice names of the branch users from central Connection to the branch.
Sync greetings for users	Check this check box to synchronize the greetings of the branch users from central Connection to the branch.
Save	Select this option to save the branch with the specified settings.
Delete	Select this option to delete the branch.
Test	Select this option to check the connectivity of the central Connection sever with the branch.
Sync Provisioning	Select this option to synchronize the user(s) that you created on the central Connection server with the branch. **Note** Before you start the provisioning of the user(s), the state of provisioning on the **Edit Branch** page remains “Idle”. As soon as you select the **Sync Provisioning** option, the state changes to “Scheduled” and then to “In Progress”. After the completion of provisioning, the state of provisioning again changes to “Idle”.
Voicemail Upload	Select this option to upload voicemails of the user(s) from the branch to the central Connection server. **Note** Before you start uploading voicemails on the central Connection server, the state of voicemail upload on the **Edit Branch** page remains “Idle”. As soon as you select the **Voicemail Upload** option, the state changes to “Scheduled” and then to “In Progress”. After the completion of voicemail upload, the state of voicemail upload again changes to “Idle”.

Branch Sync Results

Table 5-4 Branch Sync Results Page

Field	Description
Branch	*(Display Only)* Displays the name of the branch.
Sync Type	*(Display Only)* Displays the type of synchronization activity, such as Voicemail Upload or Provisioning, performed on the branch.
Result	*(Display Only)* Displays the status or result of the synchronization activity performed. The result of synchronization can be any one of the following: • **In Progress**: Signifes that the synchronization activity is in progress. • **Success**: Signifies that the synchronization activity is completed successfully. • **Partial Success**: Signfies that the synchronization activity is partially compeleted. For Example, if you have initiated synchronization of users from the central Connection to the branch, it might be possible that out of the 4 users those need to be synchronized, only 2 users are synchronized successfully and the other 2 are failed. In such scenarios, you need to check the error logs to find out the reason of the failure and take the appropriate action to resolve the problem. For more information on troubleshooting, refer to the “Troubleshooting Cisco Unity Connection SRSV in Connection SRSV 9.x” chapter of *Troubleshooting Guide for Cisco Unity Connection Release 9.x at* • **Failed**: Signifies that the synchronization activity is failed. To check the reason for the failure, refer to the error logs.
Start Date	*(Display Only)* Displays the start date and time of the synchronization activity.
End Date	*(Display Only)* Displays the end date and time of the synchronization activity. The text of the **End Date** appears as a hyperlink. When you click on the hyperlink, the “**Task Execution Results**” pop-up window appears specifying the detailed status of the synchronization activity. For example, if the **Details** column has "**Total=1 Processed=1 Success=1 Failed=0**" description with the synchronization activity as "**Voicemail**", it means that there was only 1 voicemail to be synchronized and it is processed and synchronized successfully without any failure.

Managing Branches in Cisco Unity Connection 9.1(1)

You can create a branch in Cisco Unity Connection Administration, which is further associated with a partition of users having access to the SRSV functionality.

To Create a Branch

Step 1 In Cisco Unity Connection Administration, expand **Networking** > **Branch Management**, then select **Branches**.

Step 2 On the **Branch Listing** page, select **Add New**.

Step 3 On the **New Branch** page, enter basic settings, as applicable. (For field information, on the **Help** menu, select **This Page**.)

Note Fields marked with * (an asterisk) are required.

Step 4 Select **Save**.

Step 5 On the **Edit Branch** page, continue entering the applicable settings.

Step 6 When you have finished entering settings on the **Edit Branch** page, select **Save**.

CHAPTER 6

Cisco Unity Connection SRSV Administration - User Settings Interface

See the following sections:

Search Administrators

The Search Administrators page lists the administrators those can login in Cisco Unity Connection SRSV Administration. You can also search a particular administrator name from the list. The search results, by default, return all administrators detail. By default, the administrator can view 25 records per page and can select rows per page from the dropdown list, the max limit being 250 search results returned on one page. The administrator can perform custom search on the name field using the following options:

- Begins with
- Contains
- Ends with
- Is Exactly
- Is Empty
- Is Not Empty

Table 6-1 Search Administrators Page

Field	Description
Alias	A unique text name for the administrator. Select the Alias to go to the specific page for the administrator.

Table 6-1 Search Administrators Page (continued)

Field	Description
First Name	*(Display only)* The first name of the administrator.
Last Name	*(Display only)* The last name of the administrator.
Display Name	*(Display only)* The name of the administrator.
Delete Selected	Check the check box to the left of the administrator display name, and select Delete Selected. You can delete multiple administrators at once.
Add New	Select the Add New button to add an administrators. A new page opens, on which you enter data applicable to the new administrator.

Add New Administrator

Table 6-2 Add New Administrator Page

Field	Description
Alias	A unique text name for the administrator.
First Name	The first name of the administrator.
Last Name	The last name of the administrator.
Display Name	Enter a descriptive name for the administrator.

Edit Administrator Basics

Table 6-3 Edit Administrator Basics Page

Field	Description
Alias	A unique text name for the administrator.
First Name	*(Optional)* The first name of the administrator.
Last Name	*(Optional)* The last name of the administrator.
Display Name	*(Optional)* Enter a descriptive name for the administrator.
Initials	*(Optional)* Enter the initials of the administrator.
Title	*(Optional)* Enter a title for the administrator.
Employee ID	*(Optional)* Enter an ID for the administrator.
Address	*(Optional)* Enter the administrator address.
Building	*(Optional)* Enter the building the administrator is located in.
City	*(Optional)* Enter the city.
State	*(Optional)* Enter the state.
Postal Code	*(Optional)* Enter the postal code.
Country	*(Optional)* Enter the country.

Table 6-3 Edit Administrator Basics Page (continued)

Field	Description
Use System Default Time Zone	Check this check box to have branch apply the system default time zone to the hours selected in the active schedule. When this check box is not checked, you select a Time Zone from the list.
Time Zone	Select the desired time zone for the administrator, or check the Use System Default Time Zone check box. The default time zone is the time zone set on the branch. Change this setting only for those administrators who are located in a different time zone than the Connection SRSV server.
Language	Select the language in which the conversation plays instructions to administrators. Select Use System Default Language or select a language from the list. Note that this setting does not apply to the voice-recognition conversation. **Note** Depending on your license settings, United States English may not be available.
Department	*(Optional)* Enter the administrator department.
Manager	*(Optional)* Enter the name of the manager.
Billing ID	*(Optional)* Billing ID can be used for organization-specific information, such as accounting information, department names, or project codes. This information can be included in user reports.

Change Password

Table 6-4 Change Password Page

Field	Description
Password	Enter a Web application password that can have any combination of alphanumeric characters, and the following special characters: ~!@#$%^&*()-_+={}|][:”’;<>?/ \., To help protect Cisco Unity Connection from unauthorized access and toll fraud, enter a long and non-trivial password (eight or more characters for passwords). The maximum length for passwords is 80 characters.
Confirm Password	Enter the new password again to confirm the entry.

Edit Roles

Table 6-5 Edit Roles Page

Field	Description
Assigned Roles	Use in conjunction with the Available Roles setting to assign roles to users who administer the branch. Select the up and down arrows to move the applicable roles from the Available Roles box to the Assigned Roles box. Select from the following pre-defined roles: • Audio Text Administrator • Audit Administrator • Greeting Administrator • Help Desk Administrator • Remote Administrator • System Administrator • Technician • User Administrator
Available Roles	Use in conjunction with the Assigned Roles setting to assign roles to users who administer the branch. Select the up and down arrows to move the applicable roles from the Available Roles box to the Assigned Roles box. Select from the following pre-defined roles: • Audio Text Administrator • Audit Administrator • Greeting Administrator • Help Desk Administrator • Remote Administrator • System Administrator • Technician • User Administrator

Adding an Administrator Account

To Add an Administrator Account

Step 1 In Cisco Unity Connection SRSV Administration, select **Users > Administrators**.

Step 2 On the Search Administrators page, select **Add New**. The New Administrator page opens.

Step 3 In the Alias field, enter an alias for the account.

Step 4 Enter information in the optional fields, as applicable. (For field information, on the Help menu, select **This Page**.)

Note that the SMTP Address field is optional in the sense that if you do not enter a value, Connection uses the alias to form the SMTP address. However, the SMTP address cannot include non-ASCII characters. Thus, if the user alias contains non-ASCII characters, you must provide an acceptable SMTP address.

Step 5 Select **Save**. The administrator account is created, and the Edit Administrator page opens.

Step 6 On the Edit Administrator page, enter additional information, as applicable. If you change any settings on the page, select **Save**.

Step 7 On the Edit menu, select **Roles**.

Step 8 On the Edit Roles page, select a role name in the Assigned Roles or Available Roles fields, then select the **Up** or **Down** arrow to move the role to the applicable field.

Step 9 When the **Assigned Roles** field contains all of the applicable roles for the administrator, select **Save**.

Step 10 On the Edit Menu, select **Change Password**.

Step 11 On the Change Password page, enter a password in the Password field. Note that the password must meet the following requirements for password complexity:

- A minimum length requirement (as set on the Edit Authentication Rule page, in the Minimum Credential Length field)
- Inclusion of at least one character from each of the following categories: upper-case letter, lower-case letter, number, and symbol (~ ! @ # $ % ^ & * “ ‘ , . : ; ? - _ ( ) [ ] < > { } + = / \ |)
- No characters repeated consecutively more than three times (for example, aaaaB1C9 is invalid)
- No inclusion of the alias or name of the administrator

Step 12 Enter the password again in the Confirm Password field.

Step 13 Select **Save**.

Search Subscribers

The Search Subscribers page lists all the subscribers available in Cisco Unity Connection SRSV.

The search results, by default, return all subscribers detail those can access Cisco Unity Connection SRSV. By default, the administrator can view 25 records per page and can select rows per page from the dropdown list, the max limit being 250 search results returned on one page. The administrator can perform custom search on the name field using the following options:

- Begins with
- Contains
- Ends with
- Is Exactly
- Is Empty
- Is Not Empty

Table 6-6 Search Subscribers Page

Field	Description
Alias	A unique text name for the subscriber. Select the Alias to go to the specific page for the subscriber.
Extension	*(Display only)* An extension of the subscriber.
First Name	*(Display only)* The first name of the subscriber.
Last Name	*(Display only)* The last name of the subsciber.
Display Name	*(Display only)* The name of the subscriber.

You cannot edit any subscribers related information on the branch. If any update is required in the subscribers information, then it should be done at the central Connection location.

CHAPTER 7

Cisco Unity Connection SRSV Administration - Template Settings Interface

See the following sections:

Search Call Handler Templates

The Search Call Handler Templates page displays the total number of call handler templates created by an administrator.

The administrator can search the call handler templates that are used for sending HTML notifications. The search results, by default, return all templates including default and custom templates. By default, the administrator can view 25 records per page and can select rows per page from the dropdown list, the max limit being 250 search results returned on one page. The administrator can perform custom search on the template name field using the following options:

- Begins with
- Contains
- Ends with
- Is Exactly
- Is Empty
- Is Not Empty

Table 7-1 Search Call Handler Templates Page

Field	Description
Limit Search To	Select the criteria by which to limit the display of search results: • All—Display all search results, regardless of the Cisco Unity Connection SRSV location or partition to which they belong. • Partition—Display only results that belong to a particular partition. When you select this option, choose the name of the partition from the Where Name Is list. • Location—*(Applicable to Cisco Unity Connection SRSV configurations only)* Display only results that belong to a particular Connection location. When you select this option, choose the name of the location from the Where Name Is list.
Delete Selected	To delete a call handler template, check the check box to the left of the display name, and select Delete Selected. You can delete multiple call handler templates at once.
Display Name	The name of the call handler template. Select the Display Name to go to the specific page for the call handler template.

New Call Handler Template

Table 7-2 New Call Handler Template Page

Field	Description
Display Name	Enter a descriptive name for the call handler template.
Message Recipient	Select the user or distribution list that receives messages left for the call handler. Select a recipient type by selecting the applicable button, and then select from the options available in the list. When you select a distribution list, each member of the list receives the call handler messages. Remember to enter this information when you create individual call handlers, unless all call handlers that are created from a template will have an identical message recipient, in which case you can enter the information on the template. Check the Mark for Dispatch Delivery check box to have messages sent to the distribution list as a dispatch message. When sent as a dispatch message, only one user in the group needs to act on the message.
Active Schedule	Select a schedule from the list to specify the days and times that the standard and closed greetings play, as well as the action that Connection SRSV takes after the greeting.
Partition	*(Display only)* Shows the partition to which the object belongs. Partitions are grouped together into search spaces, which are used to define the scope of objects (for example, users and distribution lists) that a user or outside caller can reach while interacting with Connection SRSV. Most objects can belong only to one partition; the exception is users, who can have their primary extension in one partition and alternate extensions in other partitions. A partition can belong to more than one search space. Note that extensions must be unique within a partition, and that partitions can contain objects that do not have an associated extension (for example, some contacts and system distribution lists).

Table 7-2 New Call Handler Template Page (continued)

Field	Description
Use System Default Time Zone	Check this check box to have Connection SRSV apply the system default time zone to the hours selected in the active schedule. When this check box is not checked, you select a Time Zone from the list.
Time Zone	Select the desired time zone for the call handler, or check the Use System Default Time Zone check box to have Connection SRSV use the system default time zone defined on the System Settings > General Configuration page. The call handler time zone setting is applied to the selected Active Schedule to determine when standard, closed, or holiday greetings are played for callers, and when standard or closed transfer rules apply.
Language	Select the language in which Connection SRSV plays the handler system prompts to the caller: • Use System Default Language— uses the system default language for the language that callers hear when they call your organization. • Inherit Language from Caller—Select this option to use the language that was applied to the caller by a previous call handler or by a routing rule. If the language is set to Inherited for every rule and handler that processes a call, then the system prompts are played in the system default language. Alternatively, you can select a specific language from the list. **Note** Depending on your license settings, United States English may not be available. The TTY language allows TTY users to read Connection SRSV prompts and to record messages by using a TTY device. TTY functionality is supported only when G.711 is selected as the systemwide message recording and storage codec.

Edit Call Handler Template Basics

Table 7-3 Edit Call Handler Template Basics Page

Field	Description
Display Name	Enter a descriptive name for the call handler template.
Creation Time	*(Display only)* Shows the date and time when the call handler template was created.
Phone System	*(Display only)* Shows the phone system that the template uses.
Active Schedule	Select a schedule from the list to specify the days and times that the standard and closed greetings play, as well as the action that Connection SRSV takes after the greeting.
Use System Default Time Zone	Check this check box to have Connection SRSV apply the system default time zone to the hours selected in the active schedule. When this check box is not checked, you select a Time Zone from the list.
Time Zone	Select the desired time zone for the call handler, or check the Use System Default Time Zone check box to have Connection SRSV use the system default time zone defined on the System Settings > General Configuration page. The call handler time zone setting is applied to the selected Active Schedule to determine when standard, closed, or holiday greetings are played for callers, and when standard or closed transfer rules apply.

Table 7-3 Edit Call Handler Template Basics Page (continued)

Field	Description
Language	Select the language in which Connection SRSV plays the handler system prompts to the caller: • Use System Default Language—Connection SRSV uses the system default language for the language that callers hear when they call your organization. • Inherit Language from Caller—Select this option to use the language that was applied to the caller by a previous call handler or by a routing rule. If the language is set to Inherited for every rule and handler that processes a call, then the system prompts are played in the system default language. Alternatively, you can select a specific language from the list. **Note** Depending on your license settings, United States English may not be available. The TTY language allows TTY users to read Connection SRSV prompts and to record messages by using a TTY device. TTY functionality is supported only when G.711 is selected as the systemwide message recording and storage codec.
Partition	*(Display only)* Shows the partition to which the object belongs. Partitions are grouped together into search spaces, which are used to define the scope of objects (for example, users and distribution lists) that a user or outside caller can reach while interacting with Cisco Unity Connection SRSV. Most objects can belong only to one partition; the exception is users, who can have their primary extension in one partition and alternate extensions in other partitions. A partition can belong to more than one search space. Note that extensions must be unique within a partition, and that partitions can contain objects that do not have an associated extension (for example, some contacts and system distribution lists).
Search Scope	Select the search scope that is applied to match extensions that callers dial from the call handler to objects in a particular search space: • Search Space—Select a specific search space from the list. • Inherit Search Space from Call—Select this option to use the search space that was applied to the call by a previous call handler or by a routing rule.

Call Handler Templates Transfer Rules

Table 7-4 Call Handler Templates Transfer Rules Page

Field	Description
Enabled	Check or uncheck this check box and select Save to enable or disable one or more transfer rules. By design, the standard transfer rule cannot be disabled.
Rule Name	The name of the transfer rule. Select the Rule Name to go to the specific page for the transfer rule.
Extension	*(Display only)* The extension that the phone system uses to connect to the object.
End Date	*(Display only)* Indicates the date and time at which the rule is disabled, if it has been enabled until a specific end date.

Call Handler Templates Edit Transfer Rules

Table 7-5 Call Handler Templates Edit Transfer Rules Page

Field	Description
Rule Name	*(Display only)* The name of the transfer rule.
Status	*(Display only)* Indicate whether the transfer option is enabled and for how long: • Disabled—The transfer option is not in effect. • Enabled With No End Date and Time—The transfer option is enabled until you disable it. • Enabled Until—Connection SRSV performs the selected transfer option until the specified date and time arrives. Select Enabled Until, and then select the month, day, year, and time at which Connection will automatically disable the transfer option. **Note** By design, the standard transfer rule cannot be disabled.
Transfer Calls To	Select one of the following settings: • Greeting—When this option is selected, the call is transferred as follows: – For user settings—to the user greeting, without ringing the user phone. – For call handler settings—to the call handler greeting. • Extension—Enter an extension to which the call is forwarded.
Transfer Type	Select how Connection SRSV transfers calls. Use this setting with caution and only if you understand its implications on the phone and voice messaging systems. • Release to Switch—Connection SRSV puts the caller on hold, dials the extension, and releases the call to the phone system. When the line is busy or is not answered, the phone system forwards the call to the user or handler greeting. This transfer type allows Connection SRSV to process incoming calls more quickly. Use Release to Switch only when call forwarding is enabled on the phone system. • Supervise Transfer—Connection SRSV acts as a receptionist, handling the transfer. If the line is busy or the call is not answered, Connection SRSV forwards the call to the user or handler greeting. You can use supervised transfer whether or not the phone system forwards calls. The Transfer Type option is unavailable when Transfer Incoming Calls is set to the My Personal Greeting option. Transfer options apply only to indirect calls; they do not apply when an unidentified caller or another user dials a user extension directly.
Rings to Wait For	Select the number of times the extension rings before Cisco Unity Connection SRSV plays the user or handler greeting. Set this value to at least three to give users a chance to answer. Avoid setting to more than four, especially if the call may be transferred to another extension, where the caller might have to wait for another set of rings. This value should be at least two rings fewer than the phone system setting for forwarding calls. This option is unavailable when Transfer Incoming Calls is set to the Greeting option or when Release to Switch is selected. **Note** Transfer options apply only to indirect calls; they do not apply when an unidentified caller or another user dials a user extension directly.

Table 7-5 Call Handler Templates Edit Transfer Rules Page (continued)

Field	Description
Play the "Wait While I Transfer Your Call" Prompt	Check this check box to have Cisco Unity Connection SRSV play "Wait while I transfer your call" to callers while performing the transfer. This option is unavailable when Transfer Incoming Calls is set to the Greeting option. Default setting: Check box checked.
If Extension Is Busy	Indicate how Cisco Unity Connection SRSV handles calls when the phone is busy. You may want to use holding options sparingly, because having calls on hold can tie up ports. • Send Callers to Voicemail—Connection SRSV plays the busy greeting and allows the caller to leave a voice message. • Put Callers on Hold Without Asking—Connection SRSV puts callers on hold. • Ask Callers to Hold—Connection SRSV gives the caller the option of holding These options are unavailable when Release to Switch is selected or when Transfer Calls To is set to the Greeting option. **Note** Transfer options apply only to indirect calls; they do not apply when an unidentified caller or another user dials a user extension directly.
Tell Me When the Call Is Connected	Check this check box to have Cisco Unity Connection SRSV say "transferring call" when the user answers the phone. This option is unavailable when Release to Switch is selected or when the Transfer Calls To setting is set to the Greeting option. **Note** Transfer options apply only to indirect calls; they do not apply when an unidentified caller or another user dials a user extension directly. Default setting: Check box not checked.
Tell Me Who the Call Is For	Check this check box to have Cisco Unity Connection SRSV say "call for <recorded name of user or call handler>" or "call for <dialed extension number>" when the user answers the phone. Use this setting when users share a phone or a user takes calls from more than one dialed extension. This option is unavailable when Release to Switch is selected or when the Transfer Calls To setting is set to the Greeting option. **Note** Transfer options apply only to indirect calls; they do not apply when an unidentified caller or another user dials a user extension directly. Default setting: Check box not checked.
Ask Me If I Want to Take the Call	Check this check box to have Cisco Unity Connection SRSV ask users whether they want to take a call before transferring the call. This option is unavailable when Release to Switch is selected or when the Transfer Calls To setting is set to the Greeting option. **Note** Transfer options apply only to indirect calls; they do not apply when an unidentified caller or another user dials a user extension directly. Default setting: Check box not checked.

Table 7-5 Call Handler Templates Edit Transfer Rules Page (continued)

Field	Description
Ask for Caller's Name	Check this check box to have Cisco Unity Connection SRSV prompt callers to say their names. When answering the phone, the user hears “Call from...” before Connection SRSV transfers the call. This option is unavailable when Release to Switch is selected or when the Transfer Calls To setting is set to the Greeting option. **Note** Transfer options apply only to indirect calls; they do not apply when an unidentified caller or another user dials a user extension directly. Default setting: Check box not checked.

Call Handler Templates Caller Input

Table 7-6 Call Handler Templates Caller Input Page

Field	Description
Key	To edit caller input settings, select the applicable key. The Edit Caller Input page opens for that key.
Action	*(Display only)* Indicates the action that Cisco Unity Connection SRSV takes when a caller presses this key. If the key is configured for a call action such as “Ignore” or “Take Message,” the action is displayed; if the key is configured to send calls to a call handler, interview handler, directory handler, conversation, or user, “Send Caller To” is displayed, and the Target field shows the object that receives the call.
Target	*(Display only)* Indicates the object that receives the call if the key is configured to send calls to a call handler, interview handler, directory handler, conversation, or user. Otherwise, this field is blank.
Status	*(Display only)* Indicates whether Cisco Unity Connection SRSV allows additional input (Unlocked) or ignores additional input (Locked) when a caller presses this key.
Wait for Additional Digits ____ Milliseconds	Indicate the amount of time that Cisco Unity Connection SRSV waits for additional input after callers press a single key that is not locked. If there is no input within this time, Connection SRSV performs the action assigned to the single key. We recommend a value of 1,500 milliseconds (one and one-half seconds). **Note** This option is unavailable if Ignore Caller Input is enabled on the Greetings page. Default setting: 1,500 milliseconds.
Prepend Digits to Dialed Extensions—Enable	Check this check box to simulate abbreviated extensions by using prepended digits for call handlers and user mailboxes. When such digits are defined, they are prepended to any extension that a caller dials while listening to the greeting for the call handler or user mailbox. Cisco Unity Connection SRSV first attempts to route the call to the prepended extension. If the prepended extension is not valid, Connection SRSV attempts to route the call to the dialed extension. For example, a call handler named Sales is configured with the prepended digits 123. When a caller dials 1000 while listening to the greeting for the Sales call handler, Connection SRSV attempts to route the call to extension 1231000; if the prepended extension is not valid, Connection SRSV attempts to route the call to extension 1000.
Digits to Prepend	Enter the digits that are prepended to any extension that a caller dials while listening to the greeting of the user or call handler.

Call Handler Templates Edit Caller Input

Table 7-7 Call Handler Templates Edit Caller Input Page

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Key</td><td>(Display only) Indicates the phone keypad key to which the settings on the page apply.</td></tr>
<tr><td>Ignore Additional Input (Locked)</td><td>Check this check box to have Cisco Unity Connection SRSV ignore additional input after callers press the key; Connection SRSV performs the action assigned to the key. To create efficient caller input menus, lock all keys except those that begin extensions on your system. You also can lock a key to block calls to extensions that begin with that key.

To lock the actions for all keys, check the Ignore Caller Input check box on the Edit Greeting page.

Default setting: Check box not checked.</td></tr>
<tr><td>Action</td><td>Select from the following, to indicate the action that Cisco Unity Connection SRSV performs when the caller presses the applicable key:

Call Action—Select the applicable action from the list:

Hang Up—Connection SRSV terminates the call when a caller presses the applicable phone key.
Ignore—Connection SRSV ignores the key press and continues playing the greeting. Use when you want only certain key presses to be responded to.
Restart Greeting—Connection SRSV plays the greeting from the beginning.
Route from Next Call Routing Rule—Connection SRSV continues processing the call according to the call routing table (direct or forwarded, depending on how the call was received from the phone system) starting at the next rule after the rule that Connection SRSV previously applied to the call.
Skip Greeting—Connection SRSV skips the greeting and performs the after-greeting action.
Take Message—Connection SRSV records a message from the caller.
Transfer to Alternate Contact Number—Connection SRSV transfers the call to the phone number that you specify in the Extension field, for example to a mobile phone or other external number. You can also specify whether Connection SRSV transfers the call by releasing it to the phone system or by supervising the transfer. If you select Supervise Transfer as the transfer type, you can also specify the number of rings to wait before Connection SRSV ends the attempt to transfer.

Call Handler—Sends the call to the system call handler that you specify. Specify whether the call should transfer to the call handler extension or go directly to the greeting of the handler.
Directory Handler—Sends the call to the directory handler that you specify.
Conversation—Sends the call to the conversation that you specify:

Sign-In—A conversation that prompts the caller to enter an ID and a PIN.
User System Transfer—A conversation that allows users to transfer to a number that they specify. Users are prompted to sign in and then can enter numbers that are not associated with Connection users—such as lobby phones or phone numbers outside the organization. Connection performs the transfer only when the user restriction table permits it.

</td></tr>
</table>

Call Handler Templates Greetings

Table 7-8 Call Handler Templates Greetings Page

Field	Description
Enabled	Check this check box and select Save to enable a greeting indefinitely. When a greeting is enabled, Cisco Unity Connection SRSV plays the greeting in the applicable situation until the end date and time, or, if no end date and time is specified, until you disable the greeting.
Greeting	*(Display only)* The name of the greeting. Select the Greeting name to go to the specific page for that greeting.
End Date	*(Display only)* Indicates the date and time at which the greeting is disabled, if it has been enabled until a specific end date.
Source	*(Display only)* Indicates the type of recording that callers hear when the greeting plays: • Blank—Callers hear nothing. • Recording—Callers hear a personally recorded greeting. • System—Callers hear the System Default Greeting.

Call Handler Templates Edit Greeting

Table 7-9 Edit Call Handler Templates Greeting Page

Field	Description
Status	Indicate whether the selected greeting is enabled and for how long: • Disabled—The applicable greeting is not in effect. • Greeting Enabled with No End Date and Time—The greeting is enabled until you disable it. • Enabled Until—Cisco Unity Connection SRSV plays the applicable greeting until the specified date and time arrives. Select Enabled Until, and then select the month, day, year, and time at which Connection automatically disables the greeting.
Callers Hear	Indicate the source for the selected greeting: • System Default Greeting—Select to use the prerecorded system default greeting. Cisco Unity Connection SRSV plays a prerecorded greeting along with the recorded name of the user (for example, “Sorry, <user name> is not available”). If the user does not have a recorded name, Connection SRSV plays the user extension instead. When a greeting is enabled but not recorded, Connection SRSV plays a prerecorded system greeting. **Note** Recording a greeting does not enable it. • My Personal Recording—Select to use the personal recording of the user. • Nothing—Select to have no recording. When the greeting source is left blank, Connection SRSV immediately performs the after-greeting action.

Table 7-9 Edit Call Handler Templates Greeting Page (continued)

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Play the “Record Your Message at the Tone” Prompt</td><td>Check this check box to have Cisco Unity Connection SRSV prompt callers to wait for a tone before recording their message.

Default setting: Check box checked.</td></tr>
<tr><td>During Greeting</td><td>Indicate the actions that Cisco Unity Connection SRSV performs during the greeting:

• Ignore Caller Input—Check this check box to ignore caller input during the greeting. When this check box is not checked, Connection SRSV responds to key presses the caller makes while the greeting plays.

Default setting: Check box not checked.

• Allow Transfers to Numbers Not Associated with Users or Call Handlers—Check this check box to allow callers to transfer to extensions that are not assigned to other users or call handlers. Connection SRSV attempts a release transfer as long as the number entered by the caller is allowed by the Default System Transfer restriction table.

Default setting: Check box not checked.

• Times to Re-Prompt Caller—Enter the number of times Connection SRSV reprompts the caller for input. When the caller does not press any key after being reprompted, Connection SRSV asks for confirmation that the caller is still there. If there is no response, Connection SRSV performs the action selected in the If Caller Exits Send To field.

Default setting: Zero.

• Delay Between Re-Prompts—Indicate the number of seconds after prompting a caller for input that Connection waits before prompting the caller again.

Default setting: 2 seconds.</td></tr>
</table>

Table 7-9 Edit Call Handler Templates Greeting Page (continued)

Field	Description
After Greeting	Indicate the action that Cisco Unity Connection SRSV performs after the greeting plays: • Call Action—Select the applicable action from the list. – Hang Up—Connection SRSV immediately terminates the call when a caller presses the applicable key on the phone. Use carefully; unexpected hang-ups can appear rude to callers. – Restart Greeting—Connection SRSV replays the greeting. This option is typically used for the error greeting. – Route from Next Call Routing Rule—Connection SRSV continues processing the call according to the applicable call routing table (direct or forwarded, depending on how the call was received from the phone system) starting at the next rule after the rule that Connection SRSV previously applied to the call. – Take Message—Connection SRSV records a message from the caller. The greeting should indicate that a message will be recorded. • Call Handler—Sends the call to the system call handler that you specify. Specify whether the call should transfer to the call handler extension or go directly to the greeting of the handler. • Directory Handler—Sends the call to the directory handler that you specify. • Conversation—Sends the call to the conversation that you specify. – Sign-In—Sends the call to the user sign-in conversation, which prompts the caller to enter an ID. – User System Transfer—Sends the call to a conversation that allows users to transfer to a number that they specify. Users are prompted to sign in and then can enter numbers that are not associated with Connection SRSV users—such as lobby and conference room phones, and even phone numbers outside the organization. Connection SRSV performs the transfer only when the user restriction table permits it.

Call Handler Templates Message Settings

Table 7-10 Call Handler Templates Message Settings Page

Field	Description
Maximum Message Length	Set the recording length (in seconds) allowed for messages left by unidentified callers. Users may want to limit the length of messages from unidentified callers. Some departments, such as Customer Service, may want to permit much longer messages. If enabled, callers hear a warning tone before the maximum message length is reached. **Note** The maximum recording length for messages left by other users is set on the applicable Edit Class of Service page. The maximum recording length for broadcast messages that users record (if applicable) is set on the System Settings > Advanced > Conversations page. Default setting: 300 seconds.

Table 7-10 Call Handler Templates Message Settings Page (continued)

Field	Description
Callers Can Edit Messages	Check this check box to allow callers to be prompted to listen to, add to, rerecord, or delete their messages. Balance giving callers the additional control of editing messages with having voice messaging ports tied up for the additional time. Default setting: Check box checked.
Message Urgency	Indicate the action that Cisco Unity Connection SRSV allows when a message has been left by an unidentified caller or by a user who has not explicitly signed in: • Mark Normal—Messages left by unidentified callers are never marked urgent. • Mark Urgent—All messages left by unidentified callers are marked urgent. This may be useful for Sales or Technical Support calls. • Ask Callers—Connection SRSV asks unidentified callers whether to mark their messages urgent.
Message Sensitivity	Indicate the action that Cisco Unity Connection SRSV allows when a message has been left by an unidentified caller or by a user who has not explicitly signed in: • Mark Normal—Messages left by unidentified callers are never marked private. • Mark Private—All messages left by unidentified callers are marked private. • Ask Callers—Connection SRSV asks unidentified callers whether to mark their messages private.
Message Security—Mark Secure	Check this check box to have Cisco Unity Connection SRSV mark messages as secure that are left for this user by unidentified callers or by users who have not explicitly signed in (when identified user messaging is enabled).
Message Recipient	Select the user or distribution list that receives messages left for the call handler. Select a recipient type by selecting the applicable button, and then select from the options available in the list. When you select a distribution list, each member of the list receives the call handler messages. Remember to enter this information when you create individual call handlers, unless all call handlers that are created from a template will have an identical message recipient, in which case you can enter the information on the template. Check the Mark for Dispatch Delivery check box to have messages sent to the distribution list as a dispatch message. When sent as a dispatch message, only one user in the group needs to act on the message.

Table 7-10 Call Handler Templates Message Settings Page (continued)

Field	Description
After Message Action	Indicate the action that Cisco Unity Connection SRSV performs after a caller leaves a message: • Call Action—Select the applicable action from the list: – Hang Up—Connection SRSV immediately terminates the call when a caller presses the applicable key on the phone. – Route from Next Call Routing Rule—Connection SRSV continues processing the call according to the applicable call routing table (direct or forwarded, depending on how the call was received from the phone system) starting at the next rule after the rule that Connection SRSV previously applied to the call. • Call Handler—Sends the call to the system call handler that you specify. Specify whether the call should transfer to the call handler extension or go directly to the greeting of the handler. • Directory Handler—Sends the call to the directory handler that you specify. • Conversation—Sends the call to the conversation that you specify: – Sign-In—Sends the call to the user sign-in conversation, which prompts the caller to enter an ID. – User System Transfer—Sends the call to a conversation that allows users to transfer to a number that they specify. Users are prompted to sign in and then can enter numbers that are not associated with Connection SRSV users—such as lobby and conference room phones, and even phone numbers outside the organization. Connection SRSV performs the transfer only when the user restriction table permits it.

CHAPTER 8

Managing System Distribution Lists in Cisco Unity Connection SRSV

System distribution lists are used to send voice messages to multiple users. The members of a system distribution list typically are users who need the same information on a regular basis, such as employees in a department or members of a team. At branch, sending a message to a distribution list is supported only when the distribution list is selected in the **Message Recipient** property of the call handler.

When you send a message to a distribution list, you receive the notification confirming that the message has been sent to the distribution list. However, the members of the distribution list will receive the message only after the restoration of WAN outage and successful upload of the message on the central Connection server.

Note Composing a new message that address to a distribution list is not supported at branch.

Search Distribution Lists

Table 8-1 Search Distribution Lists Page

Field	Description
Alias	*(Display only)* A unique text name of the distribution list. Select the Alias to go to the specific page for the distribution list.
Extension	*(Display only)* The extension that the phone system uses to connect to the object.
Display Name	*(Display only)* The name of the distribution list.

Edit Distribution List Basics

See Table 8-2 for information about the fields on the Edit Distribution List Basics page.

Table 8-2 Edit Distribution List Basics Page

Field	Description
Alias	*(Display Only)* Displays the text name of the distribution list.
Display Name	*(Display Only)* Displays the descriptive name for the distribution list.
Extension	*(Display Only)* Displays the extension that the phone system uses to connect to the distribution list.
Partition	*(Display Only)* Displays the partition to which the object belongs.

CHAPTER 9

Cisco Unity Connection SRSV Administration - Call Management Settings Interface

See the following sections:

Search Call Handlers

The Search Call Handlers page displays lists of all available call handlers.

The search results, by default, return all call handlers. By default, the administrator can view 25 records per page and can select rows per page from the dropdown list, the max limit being 250 search results returned on one page. The administrator can perform custom search on the call handler name field using the following options:

- Begins with
- Contains
- Ends with

- Is Exactly
- Is Empty
- Is Not Empty

Table 9-1 Search Call Handlers Page

Field	Description
Limit Search To	Select the criteria by which to limit the display of search results: • All—Display all search results, regardless of the partition to which they belong. • Partition—Display only results that belong to a particular partition. When you select this option, choose the name of the partition from the Where Name Is list.
Display Name	The name of the call handler. Select the Display Name to go to the specific page for the call handler.
Extension	*(Display only)* The extension that the phone system uses to connect to the call handler.
Delete Selected	To delete a call handler, check the check box to the left of the display name, and select Delete Selected. You can delete multiple call handlers at once. Note that the default system call handlers can not be deleted.
Add New	To add a call handler, select the Add New button. A new page opens, on which you enter data applicable to the new call handler.

New Call Handler

Table 9-2 New Call Handler Page

Field	Description
Display Name	Enter a descriptive name for the call handler.
Extension	Enter the extension that the phone system uses to connect to the call handler.
Call Handler Template	Select the template on which to base the new call handler. The template affects most call handler settings.

Edit Call Handler Basics

Table 9-3 Edit Call Handler Basics Page

Field	Description
Display Name	Enter a descriptive name for the call handler.
Creation Time	*(Display only)* Shows the date and time when the call handler was created.
Phone System	*(Display only)* Select the phone system that the call handler uses.
Active Schedule	Select a schedule from the list to specify the days and times that the standard and closed greetings play, as well as the action that Cisco Unity Connection SRSV takes after the greeting.

Table 9-3 Edit Call Handler Basics Page (continued)

Field	Description
Use System Default Time Zone	Check this check box to have Cisco Unity Connection SRSV apply the system default time zone to the hours selected in the active schedule. When this check box is not checked, you select a Time Zone from the list.
Time Zone	Select the desired time zone for the call handler, or check the Use System Default Time Zone check box to have Connection SRSV use the system default time zone defined on the System Settings > General Configuration page. The call handler time zone setting is applied to the selected Active Schedule to determine when standard, closed, or holiday greetings are played for callers, and when standard or closed transfer rules apply. Note that if you change the time zone setting for a user, the standard and closed greetings are also played as per the user's time zone settings.
Language	Select the language in which Connection SRSV plays the handler system prompts to the caller: • Use System Default Language—Connection SRSV uses the system default language for the language that callers hear when they call your organization. • Inherit Language from Caller—Select this option to use the language that was applied to the caller by a previous call handler or by a routing rule. If the language is set to Inherited for every rule and handler that processes a call, then the system prompts are played in the system default language. Alternatively, you can select a specific language from the list. **Note** Depending on your license settings, United States English may not be available.
Extension	Enter the extension that the phone system uses to connect to the call handler.
Partition	Select the partition to which the object belongs. Partitions are grouped together into search spaces, which are used to define the scope of objects (for example, users and distribution lists) that a user or outside caller can reach while interacting with Connection SRSV. Most objects can belong only to one partition; the exception is users, who can have their primary extension in one partition and alternate extensions in other partitions. A partition can belong to more than one search space. Note that extensions must be unique within a partition, and that partitions can contain objects that do not have an associated extension (for example, some contacts and system distribution lists).
Recorded Name	This is the recorded name of the user, contact, distribution list, or handler. You can record the name here, or a user can record the name by using the self-enrollment conversation, the setup options, or by using the Cisco Unity Connection Messaging Assistant. To record the name here, use the Media Master. Use the Open File option on the Options menu of the Media Master to use a prerecorded WAV file as the recording.
Search Scope	Select the search scope that is applied to match extensions that callers dial from the call handler to objects in a particular search space: • Search Space—Select a specific search space from the list. • Inherit Search Space from Call—Select this option to use the search space that was applied to the call by a previous call handler or by a routing rule.

Call Handler Transfer Rules

Table 9-4 Call Handler Transfer Rules Page

Field	Description
Enabled	Check or uncheck this check box and select Save to enable or disable one or more transfer rules. By design, the standard transfer rule cannot be disabled.
Rule Name	The name of the transfer rule. Select the Rule Name to go to the specific page for the transfer rule.
Extension	*(Display only)* The extension that the phone system uses to connect to the object.
End Date	*(Display only)* Indicates the date and time at which the rule is disabled, if it has been enabled until a specific end date.

Call Handler Edit Transfer Rule

Table 9-5 Call Handler Edit Transfer Rule Page

Field	Description
Rule Name	*(Display only)* The transfer option being edited.
Status	Indicate whether the transfer option is enabled and for how long: • Disabled—The transfer option is not in effect. • Enabled With No End Date and Time—The transfer option is enabled until you disable it. • Enabled Until—Connection SRSV performs the selected transfer option until the specified date and time arrives. Select Enabled Until, and then select the month, day, year, and time at which Connection SRSV will automatically disable the transfer option. **Note** By design, the standard transfer rule cannot be disabled.
Transfer Calls To	Select one of the following settings: • Greeting—When this option is selected, the call is transferred as follows: – For user settings—to the user greeting, without ringing the user phone. – For call handler settings—to the call handler greeting. • Extension—Enter an extension to which the call is forwarded.

Table 9-5 Call Handler Edit Transfer Rule Page (continued)

Field	Description
Transfer Type	Select how Cisco Unity Connection SRSV transfers calls. Use this setting with caution and only if you understand its implications on the phone and voice messaging systems. • Release to Switch—Connection SRSV puts the caller on hold, dials the extension, and releases the call to the phone system. When the line is busy or is not answered, the phone system—not Connection SRSV—forwards the call to the user or handler greeting. This transfer type allows Connection SRSV to process incoming calls more quickly. Use Release to Switch only when call forwarding is enabled on the phone system. • Supervise Transfer—Connection SRSV acts as a receptionist, handling the transfer. If the line is busy or the call is not answered, Connection SRSV—not the phone system—forwards the call to the user or handler greeting. You can use supervised transfer whether or not the phone system forwards calls. The Transfer Type option is unavailable when Transfer Incoming Calls is set to the My Personal Greeting option. Transfer options apply only to indirect calls; they do not apply when an unidentified caller or another user dials a user extension directly.
Rings to Wait For	Select the number of times the extension rings before Cisco Unity Connection SRSV plays the user or handler greeting. Set this value to at least three to give users a chance to answer. Avoid setting to more than four, especially if the call may be transferred to another extension, where the caller might have to wait for another set of rings. This value should be at least two rings fewer than the phone system setting for forwarding calls. This option is unavailable when Transfer Incoming Calls is set to the Greeting option or when Release to Switch is selected. **Note** Transfer options apply only to indirect calls; they do not apply when an unidentified caller or another user dials a user extension directly.
Play the "Wait While I Transfer Your Call" Prompt	Check this check box to have Cisco Unity Connection SRSV play "Wait while I transfer your call" to callers while performing the transfer. This option is unavailable when Transfer Incoming Calls is set to the Greeting option. Default setting: Check box checked.
If Extension Is Busy	Indicate how Cisco Unity Connection SRSV handles calls when the phone is busy. You may want to use holding options sparingly, because having calls on hold can tie up ports. • Send Callers to Voicemail—Connection SRSV plays the busy greeting and allows the caller to leave a voice message. • Put Callers on Hold Without Asking—Connection SRSV puts callers on hold. • Ask Callers to Hold—Connection SRSV gives the caller the option of holding These options are unavailable when Release to Switch is selected or when Transfer Calls To is set to the Greeting option. **Note** Transfer options apply only to indirect calls; they do not apply when an unidentified caller or another user dials a user extension directly.

Table 9-5 Call Handler Edit Transfer Rule Page (continued)

Field	Description
Tell Me When the Call Is Connected	Check this check box to have Cisco Unity Connection SRSV say “transferring call” when the user answers the phone. This option is unavailable when Release to Switch is selected or when the Transfer Calls To setting is set to the Greeting option. **Note** Transfer options apply only to indirect calls; they do not apply when an unidentified caller or another user dials a user extension directly. Default setting: Check box not checked.
Tell Me Who the Call Is For	Check this check box to have Cisco Unity Connection SRSV say “call for <recorded name of user or call handler>” or “call for <dialed extension number>” when the user answers the phone. Use this setting when users share a phone or a user takes calls from more than one dialed extension. This option is unavailable when Release to Switch is selected or when the Transfer Calls To setting is set to the Greeting option. **Note** Transfer options apply only to indirect calls; they do not apply when an unidentified caller or another user dials a user extension directly. Default setting: Check box not checked.
Ask Me If I Want to Take the Call	Check this check box to have Cisco Unity Connection SRSV ask users whether they want to take a call before transferring the call. This option is unavailable when Release to Switch is selected or when the Transfer Calls To setting is set to the Greeting option. **Note** Transfer options apply only to indirect calls; they do not apply when an unidentified caller or another user dials a user extension directly. Default setting: Check box not checked.
Ask for Caller's Name	Check this check box to have Cisco Unity Connection SRSV prompt callers to say their names. When answering the phone, the user hears “Call from...” before Connection SRSV transfers the call. This option is unavailable when Release to Switch is selected or when the Transfer Calls To setting is set to the Greeting option. **Note** Transfer options apply only to indirect calls; they do not apply when an unidentified caller or another user dials a user extension directly. Default setting: Check box not checked.

Call Handler Caller Input

Table 9-6 Call Handler Caller Input Page

Field	Description
Key	To edit caller input settings, select the applicable key. The Edit Caller Input page opens for that key.

Table 9-6 Call Handler Caller Input Page (continued)

Field	Description
Action	*(Display only)* Indicates the action that Cisco Unity Connection SRSV takes when a caller presses this key. If the key is configured for a call action such as "Ignore" or "Take Message," the action is displayed; if the key is configured to send calls to a call handler, interview handler, directory handler, conversation, or user, "Send Caller To" is displayed, and the Target field shows the object that receives the call.
Target	*(Display only)* Indicates the object that receives the call if the key is configured to send calls to a call handler, interview handler, directory handler, conversation, or user. Otherwise, this field is blank.
Status	*(Display only)* Indicates whether Cisco Unity Connection SRSV allows additional input (Unlocked) or ignores additional input (Locked) when a caller presses this key.
Wait for Additional Digits ____ Milliseconds	Indicate the amount of time that Cisco Unity Connection SRSV waits for additional input after callers press a single key that is not locked. If there is no input within this time, Connection SRSV performs the action assigned to the single key. We recommend a value of 1,500 milliseconds (one and one-half seconds). **Note** This option is unavailable if Ignore Caller Input is enabled on the Greetings page. Default setting: 1,500 milliseconds.
Prepend Digits to Dialed Extensions—Enable	Check this check box to simulate abbreviated extensions by using prepended digits for call handlers and user mailboxes. When such digits are defined, they are prepended to any extension that a caller dials while listening to the greeting for the call handler or user mailbox. Cisco Unity Connection SRSV first attempts to route the call to the prepended extension. If the prepended extension is not valid, Connection SRSV attempts to route the call to the dialed extension. For example, a call handler named Sales is configured with the prepended digits 123. When a caller dials 1000 while listening to the greeting for the Sales call handler, Connection SRSV attempts to route the call to extension 1231000; if the prepended extension is not valid, Connection SRSV attempts to route the call to extension 1000.
Digits to Prepend	Enter the digits that are prepended to any extension that a caller dials while listening to the greeting of the call handler.

Call Handler Edit Caller Input

Table 9-7 Call Handler Edit Caller Input Page

Field	Description
Key	*(Display only)* Indicates the phone keypad key to which the settings on the page apply.
Ignore Additional Input (Locked)	Check this check box to have Cisco Unity Connection SRSV ignore additional input after callers press the key; Connection SRSV performs the action assigned to the key. To create efficient caller input menus, lock all keys except those that begin extensions on your system. You also can lock a key to block calls to extensions that begin with that key. To lock the actions for all keys, check the Ignore Caller Input check box on the Edit Greeting page. Default setting: Check box not checked.

Table 9-7 Call Handler Edit Caller Input Page (continued)

Field	Description
Action	Select from the following, to indicate the action that Cisco Unity Connection SRSV performs when the caller presses the applicable key: • Call Action—Select the applicable action from the list: – Hang Up—Connection SRSV terminates the call when a caller presses the applicable phone key. – Ignore—Connection SRSV ignores the key press and continues playing the greeting. Use when you want only certain key presses to be responded to. – Restart Greeting—Connection SRSV plays the greeting from the beginning. – Route from Next Call Routing Rule—Connection SRSV continues processing the call according to the call routing table (direct or forwarded, depending on how the call was received from the phone system) starting at the next rule after the rule that Connection SRSV previously applied to the call. – Skip Greeting—Connection SRSV skips the greeting and performs the after-greeting action. – Take Message—Connection SRSV records a message from the caller. – Transfer to Alternate Contact Number—Connection SRSV transfers the call to the phone number that you specify in the Extension field, for example to a mobile phone or other external number. You can also specify whether Connection SRSV transfers the call by releasing it to the phone system or by supervising the transfer. If you select Supervise Transfer as the transfer type, you can also specify the number of rings to wait before Connection SRSV ends the attempt to transfer. • Call Handler—Sends the call to the system call handler that you specify. Specify whether the call should transfer to the call handler extension or go directly to the greeting of the handler. • Directory Handler—Sends the call to the directory handler that you specify. • Conversation—Sends the call to the conversation that you specify: – Sign-In—A conversation that prompts the caller to enter an ID and a PIN. – User System Transfer—A conversation that allows users to transfer to a number that they specify. Users are prompted to sign in and then can enter numbers that are not associated with Connection SRSV users—such as lobby phones or phone numbers outside the organization. Connection SRSV performs the transfer only when the user restriction table permits it.

Call Handler Greetings

Table 9-8 Call Handler Greetings Page

Field	Description
Enabled	Check this check box and select Save to enable a greeting indefinitely. When a greeting is enabled, Cisco Unity Connection SRSV plays the greeting in the applicable situation until the end date and time, or, if no end date and time is specified, until you disable the greeting.
Greeting	*(Display only)* The name of the greeting. Select the Greeting name to go to the specific page for that greeting.

Table 9-8 Call Handler Greetings Page (continued)

Field	Description
End Date	*(Display only)* Indicates the date and time at which the greeting is disabled, if it has been enabled until a specific end date.
Source	*(Display only)* Indicates the type of recording that callers hear when the greeting plays: • Blank—Callers hear nothing. • Recording—Callers hear a personally recorded greeting. • System—Callers hear the System Default Greeting.

Call Handler Edit Greeting

Table 9-9 Edit Call Handler Greeting Page

Field	Description
Status	Indicate whether the selected greeting is enabled and for how long: • Disabled—The applicable greeting is not in effect. • Greeting Enabled with No End Date and Time—The greeting is enabled until you disable it. • Enabled Until—Cisco Unity Connection SRSV plays the applicable greeting until the specified date and time arrives. Select Enabled Until, and then select the month, day, year, and time at which Connection SRSV automatically disables the greeting.
Callers Hear	Indicate the source for the selected greeting: • System Default Greeting—Select to use the prerecorded system default greeting. Cisco Unity Connection SRSV plays a prerecorded greeting along with the recorded name of the user (for example, “Sorry, <user name> is not available”). If the user does not have a recorded name, Connection SRSV plays the user extension instead. When a greeting is enabled but not recorded, Connection SRSV plays a prerecorded system greeting. **Note** Recording a greeting does not enable it. • My Personal Recording—Select to use the personal recording of the user. • Nothing—Select to have no recording. When the greeting source is left blank, Connection SRSV immediately performs the after-greeting action.
Play the “Record Your Message at the Tone” Prompt	Check this check box to have Cisco Unity Connection SRSV prompt callers to wait for a tone before recording their message. This check box is enabled only when Call Action is set to "Take Message" in After Greeting field.When the option is set to "System Default Greeting", the checkbox will remain disabled and checked Default setting: Check box checked.

Table 9-9 Edit Call Handler Greeting Page (continued)

Field	Description
During Greeting	Indicate the actions that Cisco Unity Connection SRSV performs during the greeting: • Ignore Caller Input—Check this check box to ignore caller input during the greeting. When this check box is not checked, Connection SRSV responds to key presses the caller makes while the greeting plays. Default setting: Check box not checked. • Times to Re-Prompt Caller—Enter the number of times Connection SRSV reprompts the caller for input. When the caller does not press any key after being reprompted, Connection SRSV asks for confirmation that the caller is still there. If there is no response, Connection SRSV performs the action selected in the If Caller Exits Send To field. Default setting: Zero. • Delay Between Re-Prompts—Indicate the number of seconds after prompting a caller for input that Connection waits before prompting the caller again. Default setting: 2 seconds.
After Greeting	Indicate the action that Cisco Unity Connection SRSV performs after the greeting plays: • Call Action—Select the applicable action from the list. – Hang Up—Connection SRSV immediately terminates the call when a caller presses the applicable key on the phone. Use carefully; unexpected hang-ups can appear rude to callers. – Restart Greeting—Connection SRSV replays the greeting. This option is typically used for the error greeting. – Route from Next Call Routing Rule—Connection SRSV continues processing the call according to the applicable call routing table (direct or forwarded, depending on how the call was received from the phone system) starting at the next rule after the rule that Connection SRSV previously applied to the call. – Take Message—Connection SRSV records a message from the caller. The greeting should indicate that a message will be recorded. • Call Handler—Sends the call to the system call handler that you specify. Specify whether the call should transfer to the call handler extension or go directly to the greeting of the handler. • Directory Handler—Sends the call to the directory handler that you specify. • Conversation—Sends the call to the conversation that you specify. – Sign-In—Sends the call to the user sign-in conversation, which prompts the caller to enter an ID. – User System Transfer—Sends the call to a conversation that allows users to transfer to a number that they specify. Users are prompted to sign in and then can enter numbers that are not associated with Connection users—such as lobby and conference room phones, and even phone numbers outside the organization. Connection SRSV performs the transfer only when the user restriction table permits it.

Table 9-9 Edit Call Handler Greeting Page (continued)

Field	Description
Recordings	If more than one language is installed on Cisco Unity Connection SRSV, greetings can be recorded in multiple languages. The Recorded Languages field displays each language in which the greeting has been recorded. To play or record the greeting here, select the language for the greeting that you will be recording, then select the Play/Record button to open the Media Master. On the Options menu of the Media Master, select Open File to use a prerecorded WAV file as the recording. Note that when a greeting recording is available in multiple languages, the recording that plays to a caller depends on the language that is set for the call. You can set the language via the Language setting on the Edit Call Handler Basics page for the call handler. When the Inherit Language from Caller option is selected for this setting, Connection SRSV determines the language to use on a per-call basis, depending on the language set by the call routing rule or handler that most recently processed the call. (If the language is set to inherited for every rule and handler that processes a call, when the call reaches the call handler greeting, the greeting that corresponds to the system default language is played.)

Call Handler Message Settings

Table 9-10 Call Handler Message Settings Page

Field	Description
Maximum Message Length	Set the recording length (in seconds) allowed for messages left by unidentified callers. Users may want to limit the length of messages from unidentified callers. Some departments, such as Customer Service, may want to permit much longer messages. If enabled, callers hear a warning tone before the maximum message length is reached. **Note** The maximum recording length for messages left by other users is set on the applicable Edit Class of Service page. The maximum recording length for broadcast messages that users record (if applicable) is set on the System Settings > Advanced > Conversations page. Default setting: 300 seconds.
Callers Can Edit Messages	Check this check box to allow callers to be prompted to listen to, add to, rerecord, or delete their messages. Balance giving callers the additional control of editing messages with having voice messaging ports tied up for the additional time. Default setting: Check box checked.
Message Urgency	Indicate the action that Cisco Unity Connection SRSV allows when a message has been left by an unidentified caller or by a user who has not explicitly signed in: • Mark Normal—Messages left by unidentified callers are never marked urgent. • Mark Urgent—All messages left by unidentified callers are marked urgent. This may be useful for Sales or Technical Support calls. • Ask Callers—Connection asks unidentified callers whether to mark their messages urgent.

Table 9-10 Call Handler Message Settings Page (continued)

Field	Description
Message Sensitivity	Indicate the action that Cisco Unity Connection SRSV allows when a message has been left by an unidentified caller or by a user who has not explicitly signed in: • Mark Normal—Messages left by unidentified callers are never marked private. • Mark Private—All messages left by unidentified callers are marked private. • Ask Callers—Connection asks unidentified callers whether to mark their messages private.
Message Security—Mark Secure	Check this check box to have Cisco Unity Connection SRSV mark messages as secure that are left for this user by unidentified callers or by users who have not explicitly signed in (when identified user messaging is enabled).
Message Recipient	Select the user or distribution list that receives messages left for the call handler. Select a recipient type by selecting the applicable button, and then select from the options available in the list. When you select a distribution list, each member of the list receives the call handler messages. Remember to enter this information when you create individual call handlers, unless all call handlers that are created from a template will have an identical message recipient, in which case you can enter the information on the template. Check the Mark for Dispatch Delivery check box to have messages sent to the distribution list as a dispatch message. When sent as a dispatch message, only one user in the group needs to act on the message.
After Message Action	Indicate the action that Cisco Unity Connection SRSV performs after a caller leaves a message: • Call Action—Select the applicable action from the list: – Hang Up—Connection SRSV immediately terminates the call when a caller presses the applicable key on the phone. – Route from Next Call Routing Rule—Connection SRSV continues processing the call according to the applicable call routing table (direct or forwarded, depending on how the call was received from the phone system) starting at the next rule after the rule that Connection SRSV previously applied to the call. • Call Handler—Sends the call to the system call handler that you specify. Specify whether the call should transfer to the call handler extension or go directly to the greeting of the handler. • Directory Handler—Sends the call to the directory handler that you specify. • Conversation—Sends the call to the conversation that you specify: – Sign-In—Sends the call to the user sign-in conversation, which prompts the caller to enter an ID. – User System Transfer—Sends the call to a conversation that allows users to transfer to a number that they specify. Users are prompted to sign in and then can enter numbers that are not associated with Connection users—such as lobby and conference room phones, and even phone numbers outside the organization. Connection SRSV performs the transfer only when the user restriction table permits it.

Search Directory Handlers

The Search Directory Handlers page displays the list of all available directory handlers.

The search results, by default, return all directory handlers. By default, the administrator can view 25 records per page and can select rows per page from the dropdown list, the max limit being 250 search results returned on one page. The administrator can perform custom search on the directory handler name field using the following options:

- Begins with
- Contains
- Ends with
- Is Exactly
- Is Empty
- Is Not Empty

Table 9-11 Search Directory Handlers Page

Field	Description
Limit Search To	Select the criteria by which to limit the display of search results: • All—Display all search results, regardless of the partition to which they belong. • Partition—Display only results that belong to a particular partition. When you select this option, choose the name of the partition from the Where Name Is list.
Display Name	*(Display only)* The name of the directory handler.
Extension	*(Display only)* The extension that the phone system uses to connect to the directory handler.
Voice Enabled	*(Display only)* Indicates whether the directory handler is voice-enabled; for voice-enabled directory handlers, callers say the first name and last name of the Cisco Unity Connection SRSV user that they want to reach.

Edit Directory Handler Basics

Table 9-12 Edit Directory Handler Basics Page

Field	Description
Display Name	Enter a descriptive name for the directory handler.
Creation Time	*(Display only)* Shows the date and time when the directory handler was created.
Language	*(Not applicable to voice-enabled directory handlers.)* Select the language in which Cisco Unity Connection SRSV plays the handler system prompts to the caller: • Use System Default Language—Connection SRSV uses the system default language for the language that callers hear when they call your organization. • Inherit Language from Caller—Select this option to use the language that was applied to the caller by a previous call handler or by a routing rule. If the language is set to Inherited for every rule and handler that processes a call, then the system prompts are played in the system default language. Alternatively, you can select a specific language from the list. **Note** Depending on your license settings, United States English may not be available.
Extension	Enter the extension that the phone system uses to connect to the directory handler.

Table 9-12 Edit Directory Handler Basics Page (continued)

Field	Description
Partition	*(Display only)* Shows the partition to which the object belongs. Partitions are grouped together into search spaces, which are used to define the scope of objects (for example, users and distribution lists) that a user or outside caller can reach while interacting with Cisco Unity Connection SRSV. Most objects can belong only to one partition; the exception is users, who can have their primary extension in one partition and alternate extensions in other partitions. A partition can belong to more than one search space. Note that extensions must be unique within a partition, and that partitions can contain objects that do not have an associated extension (for example, some contacts and system distribution lists).
Play All Names	*(Not applicable to voice-enabled directory handlers.)* Check this check box to play the names of users in the directory for caller selection, rather than requiring the caller to search by spelled name. Cisco Unity Connection SRSV plays the names of all users in the directory when either of the following conditions are true: • One to five usernames are listed in the directory. • The caller chooses to play all names listed in the directory. When there are more than five (but fewer than 51) usernames listed in the directory, the Connection SRSV conversation allows callers the choice of either searching for a user in the directory by spelled name or having Connection play all names listed in the directory. When a directory handler includes more than 50 usernames, Connection SRSV requires the caller to search for a user by spelled name. When there are no usernames listed in the directory, Connection SRSV sends the caller to the call handler specified on the Caller Input page. Default setting: Check box not checked.
Recorded Name	This is the recorded name of the user, contact, distribution list, or handler. You can record the name here, or a user can record the name by using the self-enrollment conversation, the setup options, or by using the Cisco Unity Connection Messaging Assistant. To record the name here, use the Media Master. Use the Open File option on the Options menu of the Media Master to use a prerecorded WAV file as the recording.

Table 9-12 Edit Directory Handler Basics Page (continued)

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Search Scope</td><td>Select the scope for directory handler searches:
Entire Server—Restricts directory handler searches to users and contacts who are associated with the entire Cisco Unity Connection server that the caller dialed.
Class of Service (Not applicable to voice-enabled directory handlers.)—Restricts directory handler searches to users who are assigned to the selected class of service on the local Connection server.
System Distribution List (Not applicable to voice-enabled directory handlers.)—Restricts directory handler searches to members of the selected system distribution list. Note that all system distribution lists are presented, including lists that may contain members who are not Connection users.
Search Space—Restricts directory handler searches to users and contacts who are associated with a partition that is a member of the selected search space.
Inherit Search Space from Call—Restricts directory handler searches to users and contacts who are associated with a partition that is a member of the search space of the call. The search space of the call can be set by the call routing rules or by a call handler that receives the call before it reaches the directory handler.
Default setting: Entire Server.</td></tr>
<tr><td>Search Criteria Order</td><td>(Not applicable to voice-enabled directory handlers.) Select the method that callers use to spell a username:
First Name, Last Name—For example, callers press 535 (KEL) to reach Kelly Bader.
Last Name, First Name—For example, callers press 223 (BAD) to reach Kelly Bader.
Include instructions that reflect the Search By selection in the call handler greeting that routes callers to this directory handler.
Default setting: Last Name, First Name.</td></tr>
<tr><td>Route Automatically on a Unique Match</td><td>When this option is selected, Cisco Unity Connection SRSV routes a call to the extension assigned to the user without prompting the caller to verify the match.
Note For voice-enabled directory handlers, the Route Automatically on a Unique Match option is supported.</td></tr>
<tr><td>Always Request Caller Input</td><td>(Not applicable to voice-enabled directory handlers.) When this option is selected, Cisco Unity Connection SRSV prompts a caller to verify the match before sending the caller to the specified user extension.</td></tr>
<tr><td>Announce Matched Names Using Extension Format</td><td>(Not applicable to voice-enabled directory handlers.) When this option is selected, Cisco Unity Connection SRSV announces to callers the names and extensions of matching users. For example, “For Pat Amos, press 123. For Gerry Anderson, press 104.” Callers enter the extension number to choose a user.
This functionality is supported only when the Search Scope of the directory handler is set to Search Space or Inherit Search Space from Call.</td></tr>
</table>

Table 9-12 Edit Directory Handler Basics Page (continued)

Field	Description
Announce Matched Names Using Menu Format	*(Not applicable to voice-enabled directory handlers.)* When this option is selected, Cisco Unity Connection SRSV provides a menu of users to callers. For example, "For Pat Amos, press 1. For Gerry Anderson, press 2." Callers enter the menu number to choose a user. To provide callers with the user extension, also check the Announce Extension with Each Name check box. Then, Connection SRSV provides a menu of users that includes user extensions. For example, "For Pat Amos at extension 123, press 1. For Gerry Anderson at extension 104, press 2."
Announce Extension with Each Name	Check this check box to indicate that Cisco Unity Connection SRSV provides a menu of users that includes user extensions. Callers enter the menu number to select a user. For example, "For Pat Amos at extension 123, press 1. For Gerry Anderson at extension 104, press 2." Callers might take note of user extensions to skip using the corporate directory the next time they call. For directory handlers that are not voice enabled, this functionality is supported only when Announce Matched Names Using Menu Format is also selected. Default setting: Check box checked.
Maximum Number of Matches	*(Not applicable to voice-enabled directory handlers.)* Indicate the maximum number of matching names that are presented to a caller when more than one user matches the key presses entered by the caller. Default setting: 8 matches.

Directory Handler Caller Input

Table 9-13 Directory Handler Caller Input Page

Field	Description
Timeout If No Input	*(Not applicable to voice-enabled directory handlers.)* Enter the number of seconds that Cisco Unity Connection SRSV waits for caller input. When the caller does not press any key, Connection asks for confirmation that the caller is still there. If there is no response, Connection performs the action selected in the If Caller Exits fields. Default setting: Five seconds.
Timeout After Last Input	*(Not applicable to voice-enabled directory handlers.)* Enter the number of seconds that Cisco Unity Connection SRSV waits after caller input before performing the action indicated by the input. Default setting: Four seconds.
Times to Repeat Request for Name Entry	*(Not applicable to voice-enabled directory handlers.)* Enter the number of times that Cisco Unity Connection SRSV reprompts the caller for input. When the caller does not press any key after being reprompted, Connection SRSV asks for confirmation that the caller is still there. If there is no response, Connection SRSV performs the action selected in the If Caller Exits fields. Default setting: One time.
Allow Caller to Exit Using * Key	*(Not applicable to voice-enabled directory handlers.)* Check this check box to allow callers to press the * key on the phone to exit. Cisco Unity Connection SRSV immediately sends the caller to the destination you specify in the If Caller Exits field.

Table 9-13 Directory Handler Caller Input Page (continued)

Field	Description
If Caller Exits	Select the destination to which calls are sent when the caller does not respond to the name entry prompt: • Call Action—Select the applicable action from the list. When Hang Up is selected, Cisco Unity Connection SRSV immediately terminates the call when a caller presses the applicable key on the phone. • Call Handler—Sends the call to the system call handler that you specify. Specify whether the call should transfer to the call handler extension or go directly to the greeting of the handler. • Directory Handler—Sends the call to the directory handler that you specify. • Conversation—Sends the call to the conversation that you specify: – Sign-In—Sends the call to the user sign in conversation, which prompts the caller to enter an ID. – User System Transfer—Sends the call to a conversation that allows users to transfer to a number that they specify. Users are prompted to sign in and then can enter numbers that are not associated with Connection users—such as lobby and conference room phones, and even phone numbers outside the organization. Connection SRSV performs the transfer only when the user restriction table permits it.
If No Input	*(Not applicable to voice-enabled directory handlers.)* Select the destination to which calls are sent when the caller does not respond to the name entry prompt: • Call Action—Select the applicable action from the list. When Hang Up is selected, Cisco Unity Connection SRSV immediately terminates the call when a caller presses the applicable key on the phone. • Call Handler—Sends the call to the system call handler that you specify. Specify whether the call should transfer to the call handler extension or go directly to the greeting of the handler. • Directory Handler—Sends the call to the directory handler that you specify. • Conversation—Sends the call to the conversation that you specify: – Sign-In—Sends the call to the user sign in conversation, which prompts the caller to enter an ID. – User System Transfer—Sends the call to a conversation that allows users to transfer to a number that they specify. Users are prompted to sign in and then can enter numbers that are not associated with Connection users—such as lobby and conference room phones, and even phone numbers outside the organization. Connection SRSV performs the transfer only when the user restriction table permits it. Default setting: Goodbye Call Handler.

Table 9-13 Directory Handler Caller Input Page (continued)

Field	Description
If No Selection	*(Not applicable to voice-enabled directory handlers.)* Select the destination to which calls are sent when the caller does not respond to the name entry prompt: • Call Action—Select the applicable action from the list. When Hang Up is selected, Cisco Unity Connection SRSV immediately terminates the call when a caller presses the applicable key on the phone. • Call Handler—Sends the call to the system call handler that you specify. Specify whether the call should transfer to the call handler extension or go directly to the greeting of the handler. • Directory Handler—Sends the call to the directory handler that you specify. • Conversation—Sends the call to the conversation that you specify: – Sign-In—Sends the call to the user sign in conversation, which prompts the caller to enter an ID. – User System Transfer—Sends the call to a conversation that allows users to transfer to a number that they specify. Users are prompted to sign in and then can enter numbers that are not associated with Connection users—such as lobby and conference room phones, and even phone numbers outside the organization. Connection SRSV performs the transfer only when the user restriction table permits it. Default setting: Goodbye Call Handler.
If Caller Presses Zero	Select the destination to which calls are sent when the caller presses zero in response to the name entry prompt: • Call Action—Select the applicable action from the list. When Hang Up is selected, Cisco Unity Connection SRSV immediately terminates the call when a caller presses the applicable key on the phone. • Call Handler—Sends the call to the system call handler that you specify. Specify whether the call should transfer to the call handler extension or go directly to the greeting of the handler. • Directory Handler—Sends the call to the directory handler that you specify. • Conversation—Sends the call to the conversation that you specify: – Sign-In—Sends the call to the user sign in conversation, which prompts the caller to enter an ID. – User System Transfer—Sends the call to a conversation that allows users to transfer to a number that they specify. Users are prompted to sign in and then can enter numbers that are not associated with Connection users—such as lobby and conference room phones, and even phone numbers outside the organization. Connection SRSV performs the transfer only when the user restriction table permits it. Default setting: Operator Call Handler.

Directory Handler Greeting

Table 9-14 Directory Handler Greeting Page

Field	Description
Use Custom Greeting	Check this check box to have Cisco Unity Connection SRSV play the custom greeting that you record when callers reach the directory handler. Use the Recording fields to play or record the custom greeting. Uncheck this check box to have Connection SRSV play the system default greeting when callers reach the directory handler. Default setting: Check box not checked.
Recording	If more than one language is installed on Cisco Unity Connection SRSV, greetings can be recorded in multiple languages. The Recorded Languages field displays each language in which the greeting has been recorded. To play or record the greeting here, select the language for the greeting that you will be recording, then select the Play/Record button to open the Media Master. On the Options menu of the Media Master, select Open File to use a prerecorded WAV file as the recording. Note that when a greeting recording is available in multiple languages, the recording that plays to a caller depends on the language that is set for the call. You can set the language via the Language setting on the Edit Call Directory Basics page for the directory handler. When the Inherit Language from Caller option is selected for this setting, Connection SRSV determines the language to use on a per-call basis, depending on the language set by the call routing rule or handler that most recently processed the call. (If the language is set to inherited for every rule and handler that processes a call, when the call reaches the directory handler greeting, the greeting that corresponds to the system default language is played.)

CHAPTER 10

Managing Call Handlers in Cisco Unity Connection SRSV

See the following sections:

Overview of Default Call Handlers in Cisco Unity Connection SRSV

Cisco Unity Connection Survivable Remote Site Voicemail comes with the following predefined call handlers, which you can modify but not delete. Note that you should at least modify the greetings for these call handlers.

Opening Greeting	Acts as an automated attendant, playing the greeting that callers first hear when they call your organization, and performing the actions you specify. The Opening Greeting Call Routing rule transfers all incoming calls to the Opening Greeting call handler. By default, the Opening Greeting call handler allows callers to press * to reach the Sign-In conversation, or press # to reach the Operator call handler. Messages left in the Opening Greeting call handler are sent to the Undeliverable Messages distribution list.

Operator	Calls are routed to this call handler when callers press "0" or do not press any key (the default setting). You can configure the Operator call handler so that callers can leave a message or be transferred to a live operator. By default, the Operator call handler allows callers to press * to reach the Sign-In conversation, or press # to reach the Opening Greeting call handler. Messages left in the Operator call handler are sent to the mailbox for the Operator user.
Goodbye	Plays a brief goodbye message and then hangs up if there is no caller input. By default, the Goodbye call handler allows callers to press * to reach the Sign-In conversation, or press # to reach the Opening Greeting call handler. If you change the After Greeting action from Hang Up to Take Message, messages left in the Goodbye call handler are sent to the Undeliverable Messages distribution list.

Creating, Modifying, and Deleting Call Handler Templates in Cisco Unity Connection Survivable Remote Site Voicemail

Each call handler that you add in Cisco Unity Connection SRSV is based on a template. Settings from the template are applied as the call handler is created. Connection SRSV comes with one default call handler template, which has settings that are suitable for most call handlers. The administrator is allowed to create new templates.

Before you create call handlers, review the settings in the template that you plan to use and determine whether you need to make changes or create new templates. For each template, you should consider enabling the transfer, caller input, greetings, and message settings that will be needed for the call handlers that you plan to create. Note that if you change settings on a call handler template, the new settings are in effect only for new call handlers that are created by using that template. Changes to template settings do not affect existing call handlers.

Deleting a call handler template does not affect any call handlers that were based on that template when they were created. Note that you cannot delete the default template.

See the following procedures:

- To Create a Call Handler Template, page 10-2
- To Modify a Call Handler Template, page 10-3
- To Delete a Call Handler Template, page 10-4

To Create a Call Handler Template

Step 1 In Cisco Unity Connection SRSV Administration, expand **Templates**, then select **Call Handler Templates**.

Step 2 On the Search Call Handler Templates page, select **Add New**.

Step 3 On the New Call Handler Template page, enter basic settings, as applicable. (For field information, on the Help menu, select **This Page**.)

Note Fields marked with * (an asterisk) are required.

Step 4 Select **Save**.

Step 5 On the Edit Call Handler Template page, continue entering applicable settings.

Step 6 When you have finished entering settings on the Edit Call Handler Template page, select **Save**.

Step 7 On the Edit menu, select any (or all) of the following related pages, to continue adding applicable settings to the new call handler template:

- Transfer Rules (see the "Transferring Calls in Cisco Unity Connection SRSV" section on page 10-10 for details)
- Caller Input (see the "Managing Caller Input During Greetings in Cisco Unity Connection SRSV" section on page 10-7 for details)
- Greetings (see the "Overview of Call Handler Greetings in Cisco Unity Connection SRSV" section on page 10-6 for details)
- Message Settings (see the "Taking Messages in Cisco Unity Connection SRSV" section on page 10-10 for details)

Step 8 If you change any of the default settings on any of the pages listed in Step 7, select **Save** before leaving the page.

To Modify a Call Handler Template

Step 1 In Cisco Unity Connection SRSV Administration, expand **Templates**, then select **Call Handler Templates**.

Step 2 On the Search Call Handler Templates page, select the display name of the call handler template that you want to modify.

Note If the call handler template that you want to modify does not appear in the search results table, set the applicable parameters in the search fields at the top of the page, and select **Find**.

Step 3 On the Edit Call Handler Template, change settings, as applicable. (For field information, on the Help menu, select **This Page**.)

Step 4 When you have finished changing settings on the Edit Call Handler Template page, select **Save**.

Step 5 You may also want to change settings on any (or all) of the following related pages, as applicable:

- Transfer Rules (see the "Transferring Calls in Cisco Unity Connection SRSV" section on page 10-10 for details)
- Caller Input (see the "Managing Caller Input During Greetings in Cisco Unity Connection SRSV" section on page 10-7 for details)
- Greetings (see the "Overview of Call Handler Greetings in Cisco Unity Connection SRSV" section on page 10-6 for details)
- Message Settings (see the "Taking Messages in Cisco Unity Connection SRSV" section on page 10-10 for details)

Step 6 If you change any of the settings on a page listed in Step 5, select **Save** before leaving the page.

To Delete a Call Handler Template

Step 1 In Cisco Unity Connection Administration, expand **Templates**, then select **Call Handler Templates**.

Step 2 On the Search Call Handler Templates page, check the check box adjacent to the call handler template that you want to delete.

> **Note** If the call handler template that you want to delete does not appear in the search results table, set the applicable parameters in the search fields at the top of the page, and select **Find**.

Step 3 Select **Delete Selected**.

Step 4 Select **OK**.

Creating Call Handlers in Cisco Unity Connection SRSV

After you have created or updated the templates that you plan to use, you are ready to create call handlers.

To Create a Call Handler

Step 1 In Cisco Unity Connection Administration, expand **Call Management**, then select **System Call Handlers**.

Step 2 On the Search Call Handlers page, select **Add New**.

Step 3 On the New Call Handler page, enter basic settings, as applicable. (For field information, on the Help menu, select **This Page**.)

> **Note** Fields marked with * (an asterisk) are required.

Step 4 Select **Save**.

Step 5 On the Edit Call Handler page, continue entering settings for the call handler.

Step 6 When you have finished entering settings on the Edit Call Handler page, select **Save**.

Step 7 On the Edit menu, select any (or all) of the following related pages, to continue adding applicable settings to the new call handler:

- Transfer Rules (see the "Transferring Calls in Cisco Unity Connection SRSV" section on page 10-10 for details)
- Caller Input (see the "Managing Caller Input During Greetings in Cisco Unity Connection SRSV" section on page 10-7 for details)
- Greetings (see the "Overview of Call Handler Greetings in Cisco Unity Connection SRSV" section on page 10-6 for details)
- Message Settings (see the "Taking Messages in Cisco Unity Connection SRSV" section on page 10-10 for details)
- Call Handler Owners

Note Depending on how you set up the call handler template on which this new call handler is based, you may not need to change any settings on these additional pages. At a minimum, however, you should record a name and one or more greetings for the call handler.

Step 8 If you change any of the settings on a page listed in Step 7, select **Save** before leaving the page.

Modifying Call Handlers in Cisco Unity Connection SRSV

After a call handler has been created, you may need to adjust settings. The tools in Cisco Unity Connection SRSV Administration allow you to modify a single call handler at a time, or make changes to multiple call handlers at once. Do the applicable procedure:

To Modify a Single Call Handler

Step 1 In Cisco Unity Connection SRSV Administration, expand **Call Management**, then select **System Call Handlers**.

Step 2 On the Search Call Handlers page, select the display name of the call handler that you want to modify.

Note If the call handler that you want to modify does not appear in the search results table, set the applicable parameters in the search fields at the top of the page, and select **Find**.

Step 3 On the Edit Call Handler page, change settings as applicable. (For field information, on the Help menu, select **This Page**.)

Step 4 When you have finished changing settings on the Edit Call Handler page, select **Save**.

Step 5 You may also want to change settings on any (or all) of the following related pages, as applicable:

- Transfer Rules (see the "Transferring Calls in Cisco Unity Connection SRSV" section on page 10-10 for details)
- Caller Input (see the "Managing Caller Input During Greetings in Cisco Unity Connection SRSV" section on page 10-7 for details)
- Greetings (see the "Overview of Call Handler Greetings in Cisco Unity Connection SRSV" section on page 10-6 for details)
- Message Settings (see the "Taking Messages in Cisco Unity Connection SRSV" section on page 10-10 for details)
- Call Handler Owners

Step 6 If you change any of the settings on a page listed in Step 5, select **Save** before leaving the page.

Overview of Call Handler Greetings in Cisco Unity Connection SRSV

Each call handler can have up to seven greetings. The greeting settings specify which greetings are enabled, how long they are enabled, the greeting source, and the actions that Connection SRSV takes during and after each greeting. When a greeting is enabled, Connection SRSV plays the greeting in the applicable situation until the specified date and time arrives, and then the greeting is automatically disabled. A greeting can also be enabled to play indefinitely.

Note that Call Handler greetings can be recorded in multiple languages. See the "Changing Phone Language Settings in Cisco Unity Connection SRSV" section on page 10-10 for instructions.

You can customize how Connection SRSV handles calls to call handlers that have the alternate greeting enabled. For example, you can specify that as long as the alternate greeting is enabled, Connection SRSV:

- Transfers callers directly to the greeting without ringing the extension that is assigned to the call handler (as applicable) whenever calls are transferred from the automated attendant or a directory handler to the user extension. (The phone rings if an outside caller or another Connection SRSV user dials a user extension directly.)
- Prevents all callers from skipping the greeting.
- Prevents all callers from leaving messages (when the call handler is set up to take message).

Note that Connection SRSV plays the greetings that you enable for the applicable situation; however, some greetings override other greetings when they are enabled:

Standard	Plays at all times unless overridden by another greeting. You cannot disable the standard greeting.
Closed	Plays during the closed (nonbusiness) hours defined for the active schedule. A closed greeting overrides the standard greeting, and thus limits the standard greeting to the open hours defined for the active schedule.
Error	Plays if the caller enters invalid digits. This can happen if the digits do not match an extension, the extension is not found in the search scope, or the caller is otherwise restricted from dialing the digits. You cannot disable the error greeting. The system default error recording is, "I did not recognize that as a valid entry." By default, after the error greeting plays, Connection replays the greeting that was playing when the caller entered the invalid digits.

Call handler owners can select a different call handler greeting or record the call handler greetings from the System Call Handlers > Greetings page in Cisco Unity Connection SRSV Administration, or they can use the Cisco Unity Greetings Administrator to do so by phone. (For more information on recording greetings and using the Cisco Unity Greetings Administrator, see the "Managing Recorded Greetings and Recorded Names in Cisco Unity Connection 9.x" chapter of the *System Administration Guide for Cisco Unity Connection* available at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/administration/guide/9xcucsagx.html.)

Managing Call Handler Greetings in Cisco Unity Connection SRSV

You can modify call handlers greetings by using Cisco Unity Connection SRSV Administration, or by calling Connection SRSV by phone. When you use Connection SRSV Administration to modify greetings, you can do so for a single call handler, or you can modify the greetings for multiple call handlers at once.

To manage call handler greetings when you—or the call handler owners that you assign—cannot access Cisco Unity Connection SRSV Administration, you can use the Cisco Unity Greetings Administrator by phone. For more information, see the "Setting Up the 9.x Cisco Unity Greetings Administrator" section on page 18-4 and the "Using the 9.x Cisco Unity Greetings Administrator to Record or Rerecord Call Handler Greetings" section on page 18-2.

To Set Up Call Handler Greetings for a Single Call Handler

Step 1 In Cisco Unity Connection SRSV Administration, expand **Call Management**, then select **System Call Handlers**.

Step 2 On the Search Call Handlers page, select the display name of the applicable call handler.

> **Note** If the call handler does not appear in the search results table, set the applicable parameters in the search fields at the top of the page, and select **Find**.

Step 3 On the Edit menu, select **Greetings**.

Step 4 On the Greetings page, select the display name of the greeting that you want to set up.

Step 5 On the Edit Greeting page, enter settings as applicable.

Step 6 Select **Save**.

Step 7 To set up another greeting for the call handler, repeat Step 3 through Step 6.

Managing Caller Input During Greetings in Cisco Unity Connection SRSV

Caller input settings define actions that Cisco Unity Connection SRSV takes in response to phone keys pressed by callers during a call handler greeting. By using the settings on the Edit Greeting page for each individual greeting, you can specify on a per-greeting basis whether the greeting allows caller input and whether callers can perform transfers. Or, you can define caller input keys and options that apply to all of the call handler greetings by using the Caller Input page for the call handler.

See the following sections for details:

- Offering One-Key Dialing During Call Handler Greetings, page 10-8
- Offering System Transfers, page 10-9
- Abbreviated Extensions: Prepending Digits to Extensions That Callers Enter, page 10-9

Offering One-Key Dialing During Call Handler Greetings

One-key dialing enables you to designate a single digit to represent a user extension, alternate contact number, call handler, interview handler, or directory handler. Instead of entering the full extension, the caller presses a single key during a call handler greeting and Cisco Unity Connection SRSV responds accordingly. By specifying several different keys as caller input options, you can offer callers a menu of choices in the call handler greeting.

Configuring the transfer to alternate contact number action on one or more keys of a call handler allows you to quickly set up a simple audiotext tree that callers can use to transfer to specific non-user extensions on the phone system or to specific external numbers, without having to create separate call handlers for each number. When transferring a caller to an alternate contact number, Connection SRSV can either supervise the transfer or release the call to the phone system.

Callers can also bypass one-key dialing. You set the system to pause a certain number of seconds for additional key presses before routing the call according to the one-key dialing menu you have set up. These pauses allow callers to press full extension IDs to bypass one-key dialing menus, even during the handler greeting.

Further, you can lock certain keys to take the caller directly to the action programmed for that key without waiting for an additional key press. Note that you should not lock any key that matches the first digit of user extensions; otherwise, callers are not able to enter an extension to reach a user.

To Offer One-Key Dialing During a Call Handler Greeting

Step 1 In Cisco Unity Connection SRSV Administration, expand **Call Management**, then select **System Call Handlers**.

Step 2 On the Search Call Handler page, in the Search Results table, select the display name of the applicable call handler.

> **Note** If the call handler does not appear in the search results table, set the applicable parameters in the search fields at the top of the page, and select **Find**.

Step 3 On the Edit Call Handler page, on the Edit menu, select **Caller Input**.

Step 4 In the Single Key Settings table, select the applicable phone keypad key.

Step 5 On the Edit Caller Input page for the key that you selected, check the **Ignore Additional Input (Locked)** check box to instruct Connection SRSV to immediately process the key without waiting for the caller to enter additional digits.

Step 6 Under Action, select an option and change settings as applicable.

Step 7 Select **Save**.

Step 8 Optionally, you can rerecord the call handler greetings to mention the key that callers can press while listening to the greetings:

a. On the Edit menu, select **Greetings**.

b. On the Greetings page, select the display name of the greeting that you want to modify.

c. On the Edit Greeting page, select **Play/Record**, and record a greeting.

d. Select **Save**.

Offering System Transfers

System transfers allow callers to dial numbers that are not associated with a user, contact, call handler, or other entity. For example, users and outside callers may find it convenient to be able to call Cisco Unity Connection SRSV and transfer from a call handler to a lobby extension, conference room extension, or an extension that is assigned to someone in the organization who is not a Connection SRSV user, such as an employee who is visiting from another site and is using a guest office.

You can configure individual call handler greetings to allow callers to transfer to numbers that are not associated with Connection SRSV users or call handlers while the greeting is playing.

For more information see the "Setting Up System Transfers in Cisco Unity Connection 9.x" chapter.

Abbreviated Extensions: Prepending Digits to Extensions That Callers Enter

You can simulate abbreviated extensions by using prepended digits for call handlers and user mailboxes. When such digits are defined, they are prepended to any extension that a caller dials while listening to the greeting for the call handler or user mailbox.

Cisco Unity Connection SRSV first attempts to route the call to the prepended extension. If the prepended extension is not valid, Connection SRSV attempts to route the call to the dialed extension. In the following example, the call handler named Sales is configured with the prepended digits 123. When a caller dials 1000 while listening to the greeting for the Sales call handler, Connection SRSV attempts to route the call to extension 1231000; if the prepended extension is not valid, Connection SRSV attempts to route the call to extension 1000. (Note that if extension 1000 is not a valid extension and the greeting for the Sales call handler is configured to allow transfers to numbers not associated with users or call handlers, Connection SRSV performs a release transfer to 1231000.)

Abbreviated extensions can be used as a way for an organization to segment users into different groups. For example, suppose a company has two departments: Engineering and Marketing. The company uses six digit extensions, and all extensions in Engineering begin with 10 and all extensions in Marketing begin with 11. Call handlers could be created for Engineering and for Marketing, and each call handler could be configured to prepend a 10 or a 11, as applicable, to any extension dialed from that call handler. When set up this way, users would only have to enter the last four digits of a user extension.

To Configure Prepended Digits for Individual User or Call Handler Accounts

Step 1 In Cisco Unity Connection SRSV Administration, go to the Caller Input page for the applicable user, user template, call handler, or call handler template.

Step 2 In the Prepend Digits to Dialed Extensions section, check the **Enable** check box.

Step 3 In the Digits to Prepend field, enter the applicable digits.

Step 4 Select **Save**.

Changing Phone Language Settings in Cisco Unity Connection SRSV

Call handler greetings can be recorded in multiple languages when the language for the call handler is inherited from the caller. For example, if Cisco Unity Connection SRSV is configured to provide prompts in French and Spanish, it is possible to record the standard greeting in both languages so that Spanish- and French-speaking callers can hear the greeting in their own language.

To Change Phone Language Settings for a Call Handler

Step 1 In Cisco Unity Connection SRSV Administration, expand **Call Management**, then select **System Call Handlers**.

Step 2 On the Search Call Handler page, in the Search Results table, select the display name of the applicable call handler.

Note If the call handler does not appear in the search results table, set the applicable parameters in the search fields at the top of the page, and select **Find**.

Step 3 On the Edit Call Handler page, select **Use System Default Language** or **Inherit Language from Caller**, or select one of the languages in the Language list.

Step 4 Select **Save**.

Step 5 On the Edit menu, select **Greetings**.

Step 6 On the Greetings page, select the applicable greeting.

Step 7 On the Edit Greetings page, rerecord greetings in the new language.

Step 8 Select **Save**.

Taking Messages in Cisco Unity Connection SRSV

By using the settings for a particular call handler greeting, you can configure the call handler to take a message after playing the greeting. You can specify who receives messages for the call handler, whether messages are marked for dispatch delivery, the maximum recording length for messages from outside callers, what callers can do when leaving messages and whether their messages are automatically marked secure, and what action to take next on a call after a message is left.

Note that for some integrations, you can set up Cisco Unity Connection SRSV so that as a caller records a message, a warning prompt is played before the caller reaches the maximum allowable message length.

Transferring Calls in Cisco Unity Connection SRSV

The call transfer settings for a call handler specify how Cisco Unity Connection SRSV transfers calls that reach the call handler from the automated attendant. Each call handler has three transfer rules that you can customize: one for standard hours and one for closed (nonbusiness and holiday) hours of the active schedule, and an alternate transfer rule that, when enabled, overrides the standard and closed transfer rules and is in effect at all times. When a call is transferred to the call handler, Connection SRSV first checks the applicable transfer rule to determine where to transfer the call—either to the call handler greeting, or to an extension.

When transferring to the call handler greeting, Connection SRSV plays the applicable greeting (standard, closed, holiday, internal, busy, or alternate) based on the situation and which greetings are enabled. You configure a transfer rule to transfer to the greeting if you want to use the call handler to provide the caller with a prerecorded menu of options or an informational message.

To route callers to a specific user or to another call handler, you configure the transfer rule to transfer to the extension of the user or call handler. When transferring a call to a user extension, Connection SRSV can either release the call to the phone system, or it can supervise the transfer. When Connection SRSV is set to supervise transfers, it can provide call screening and call holding options on indirect calls:

- With call screening, Connection SRSV can ask for the name of the caller before connecting to a user. The user can then hear who is calling and, when a phone is shared by more than one user, who the call is for. The user can then accept or refuse the call.
- With call holding, when the phone is busy, Connection SRSV can ask callers to hold. Each caller on hold uses a Connection SRSV port and a phone system port, and therefore the total number of callers that can be holding in the queue at any one time is limited by the number of available ports.

  The default wait time in the call holding queue for the first caller in the queue is 25 seconds. If the caller is still on hold after this amount of time, Connection SRSV asks whether the caller wants to continue holding, leave a message, or try another extension. If the caller does not press a key on the phone keypad or say a voice command to indicate that he or she wants to continue holding, leave a message, or dial another extension, the caller is transferred back to the Opening Greeting. Subsequent callers in the holding queue are told how many other callers are in the queue ahead of them, in addition to these options. (See the "Call Holding Wait Time in Cisco Unity Connection 9.x" section on page 14-5 for more information on call holding.)

  If call holding is not selected, callers are sent to the user or handler greeting that is enabled: the standard, closed, holiday, busy, or alternate greeting.

Deleting Call Handlers in Cisco Unity Connection SRSV

If a call handler is referenced by other objects in Connection SRSV (for example, a routing rule or other call handler is set to route calls to the call handler), you are not allowed to delete the call handler until you have changed settings on the other objects to remove references to the call handler you want to delete. If you try to delete a call handler without first changing settings on objects that reference the call handler, the delete operation fails

If you delete call handlers that are referenced by other call handlers, be sure to rerecord the greetings so that callers hear the correct information about input options.

To Delete a Call Handler

Step 1 In Cisco Unity Connection SRSV Administration, expand **Call Management**, then select **System Call Handlers**.

Step 2 On the Search Call Handlers page, check the check box adjacent to the display name of the call handler that you want to delete.

Note If the call handler that you want to delete does not appear in the search results table, set the applicable parameters in the search fields at the top of the page, and select **Find**.

Step 3 On the Search Call Handlers page, check the check box adjacent to the display name of the call handler that you want to delete.

Step 4 Select **Delete Selected**.

Note If you are concerned that you might delete the wrong call handler from the search page, you can select the display name to navigate to the Edit Call Handler Basics page. Use the detailed data on that page to confirm that it is the call handler you want to delete.

Step 5 In the dialog box that asks you to confirm the deletion, select **OK**.

CHAPTER 11

Cisco Unity Connection SRSV Administration - Networking Settings Interface

See the following sections:

Central Server Configuration

Table 11-1 Central Server Configuration Page

Field	Description
Server Address	The Fully Qualified Domain Name (FQDN) address of the Cisco Unity Connection server, server host name or FQDN. To implement the Connection SRSV High Availability functionality, you must mention a common DNS A Record for both publisher and subscriber for central Connection server.
Test	Confirms the connectivity with the central Connection server. On the successful result, the status "Central Server Active" is displayed.

Configuring Central Server in Cisco Unity Connection SRSV

Do the following procedure.

To configure Conversation settings

Step 1 In Cisco Unity Connection SRSV Administration, expand **Networking**, then select **Central Server Configuration**.

Step 2 On the Central Server Configuration page, enter the applicable settings.

Step 3 Select **Save**.

Note A branch can be connected to only one central Connection.

CHAPTER 12

Cisco Unity Connection SRSV Administration - System Settings Interface

See the following sections:

Search Schedules

The Search Schedules page displays the status with the total number of schedules.

The search results, by default, return all schedules. By default, the administrator can view 25 records per page and can select rows per page from the dropdown list, the max limit being 250 search results returned on one page. The administrator can perform custom search on the schedule name field using the following options:

- Begins with
- Contains.
- Ends with
- Is Exactly
- Is Empty
- Is Not Empty

Table 12-1 Search Schedules Page

Field	Description
Delete Selected	Check the check box to the left of the display name, and select Delete Selected. You can delete multiple schedules at once.

Table 12-1 Search Schedules Page (continued)

Field	Description
Display Name	*(Display only)* The name of the schedule. Select the Display Name to edit the schedule.

New Schedule

Table 12-2 New Schedule Page

Field	Description
Display Name	Enter a descriptive name for the schedule.

Edit Schedule Basics

Table 12-3 Edit Schedule Basics Page

Field	Description
Display Name	Enter a descriptive name for the schedule.
Delete Selected	Check the check box to the left of the display name, and select Delete Selected. You can delete multiple schedules at once.
Add New	Click the Add New button. A new page opens, on which you enter data applicable to the new schedule.
Name	*(Display only)* The name of the schedule detail. Select the name to go to the specific page for the schedule detail.
Start Time	*(Display only)* The time at which the schedule becomes active based on this schedule detail.
End Time	*(Display only)* The time at which the schedule becomes inactive based on this schedule detail.
Days Active	*(Display only)* The days on which the schedule is active based on this schedule detail.

New Schedule Detail

Table 12-4 New Schedule Detail Page

Field	Description
Name	Enter a descriptive name that other administrators will recognize when they work with this schedule.
Start Time	From the lists, select the hour, minute, and a.m. or p.m. designation at which the schedule becomes active.
End Time	From the lists, select the hour, minute, and a.m. or p.m. designation at which time the schedule becomes inactive. **Note** The end time must be later than the start time. To specify the midnight end time (12:00 am), check the End of Day check box.
End of Day	Check this check box to specify that the schedule becomes inactive at midnight (the end of the day).

Table 12-4 New Schedule Detail Page (continued)

Field	Description
Active Every Day	Check this check box to make the schedule active every day of the week (including weekends) between the start time and end time that you specify for this schedule detail.
Active Weekdays	Check this check box to make the schedule active every week day (Monday through Friday, weekends excluded) between the start time and end time that you specify for this schedule detail.
Active Monday	Check this check box to make the schedule active each Monday between the start time and end time that you specify for this schedule detail.
Active Tuesday	Check this check box to make the schedule active each Tuesday between the start time and end time that you specify for this schedule detail.
Active Wednesday	Check this check box to make the schedule active each Wednesday between the start time and end time that you specify for this schedule detail.
Active Thursday	Check this check box to make the schedule active each Thursday between the start time and end time that you specify for this schedule detail.
Active Friday	Check this check box to make the schedule active each Friday between the start time and end time that you specify for this schedule detail.
Active Saturday	Check this check box to make the schedule active each Saturday between the start time and end time that you specify for this schedule detail.
Active Sunday	Check this check box to make the schedule active each Sunday between the start time and end time that you specify for this schedule detail.

Edit Schedule Detail

Table 12-5 Edit Schedule Detail Page

Field	Description
Name	Enter a descriptive name that other administrators will recognize when they work with this schedule.
Start Time	From the lists, select the hour, minute, and a.m. or p.m. designation at which the schedule becomes active.
End Time	From the lists, select the hour, minute, and a.m. or p.m. designation at which time the schedule becomes inactive. **Note** The end time must be later than the start time. To specify an end time of midnight (12:00 am), check the End of Day check box.
End of Day	Check this check box to specify that the schedule becomes inactive at midnight (the end of the day).
Active Every Day	Check this check box to make the schedule active every day of the week (including weekends) between the start time and end time that you specify for this schedule detail.
Active Weekdays	Check this check box to make the schedule active every week day (Monday through Friday, weekends excluded) between the start time and end time that you specify for this schedule detail.
Active Monday	Check this check box to make the schedule active each Monday between the start time and end time that you specify for this schedule detail.
Active Tuesday	Check this check box to make the schedule active each Tuesday between the start time and end time that you specify for this schedule detail.

Table 12-5 Edit Schedule Detail Page (continued)

Field	Description
Active Wednesday	Check this check box to make the schedule active each Wednesday between the start time and end time that you specify for this schedule detail.
Active Thursday	Check this check box to make the schedule active each Thursday between the start time and end time that you specify for this schedule detail.
Active Friday	Check this check box to make the schedule active each Friday between the start time and end time that you specify for this schedule detail.
Active Saturday	Check this check box to make the schedule active each Saturday between the start time and end time that you specify for this schedule detail.
Active Sunday	Check this check box to make the schedule active each Sunday between the start time and end time that you specify for this schedule detail.

Conversation Configuration

Table 12-6 Conversation Configuration Page

Field	Description
IP Addresses Allowed To Connect For Port Status Monitor Output (comma-separated)	Enter up to three IP addresses for the Remote Port Status Monitor clients that are allowed to connect to Cisco Unity Connection SRSV. You must separate the IP addresses with commas or semi-colons. Clients that do not have their IP address listed here, are refused access to Cisco Unity Connection SRSV.
Enable Remote Port Status Monitor Output	When this checkbox is checked, Cisco Unity Connection SRSV is enabled to send real-time port status information over port 5000 to the Remote Port Status Monitor clients. Default setting: Check box not checked.

Enterprise Parameters

Table 12-7 Enterprise Parameters Page

Field	Description
Parameter Name	*(Display only)* The name of the enterprise parameter.
Parameter Value	Enter or select the value for the parameter.
Suggested Value	*(Display only)* The suggested parameter value.
Set to Default	Select the Set to Default button to set all enterprise parameters to the default values.

Search Plugins

The Search Plugins page displays the status with the total number of plugins.

The search results, by default, return all plugins. By default, the administrator can view 25 records per page and can select rows per page from the dropdown list, the max limit being 250 search results returned on one page. The administrator can perform custom search on the plugins name field using the following options:

- Begins with
- Contains.
- Ends with
- Is Exactly
- Is Empty
- Is Not Empty

These fields do not apply to Cisco Unified Communications Manager or Cisco Unified Communications Manager Business Edition.

Table 12-8 Search Plugins Page

Field	Description
Find	Select the Find button to display the available plugins.
Download	Select Download and follow the on-screen instructions to download and install a plugin.
Plugin Name	*(Display only)* The name of the plugin that is available to download and install.
Description	*(Display only)* The description of the plugin.

CHAPTER 13

Managing System Settings in Cisco Unity Connection SRSV

See the following sections:

Managing Schedules in Cisco Unity Connection SRSV

See the following sections:

Creating Schedules in Cisco Unity Connection SRSV

To Create a New Schedule

Step 1 In Cisco Unity Connection SRSV Administration, expand **System Settings**, then select **Schedules**.

Step 2 On the Search Schedules page, select **Add New**.

Step 3 On the New Schedule page, enter a display name for this schedule.

Note Fields marked with * (an asterisk) are required.

Step 4 Select **Save**.

Step 5 To add time frames when the schedule is active, on the Edit Schedule Basics page, in the Schedule Details box, select **Add New**.

Step 6 On the New Schedule Detail page, enter settings as applicable. (For field information, on the Help menu, select **This Page**.)

Step 7 Select **Save**.

Step 8 To return to the Edit Schedule page, on the Edit menu, select **Schedule Basics**.

Modifying Schedules in Cisco Unity Connection SRSV

To Modify a Schedule

Step 1 In Cisco Unity Connection SRSV Administration, expand **System Settings**, then select **Schedules**.

Step 2 On the Search Schedules page, select the display name of the schedule that you want to modify.

> **Note** If the schedule that you want to modify does not appear in the search results table, set the applicable parameters in the search fields at the top of the page, and select **Find**.

Step 3 On the Edit Schedule Basics page, change the display name or holiday schedule settings, as applicable.

Step 4 When you have finished changing settings on the Edit Schedule page, select **Save**.

Step 5 To add time frames when the schedule is active, in the Schedule Details box, select **Add New**.

Step 6 If you change any settings on the New Schedule Detail page, select **Save**. To return to the Edit Schedule page, on the Edit menu, select **Edit Schedule**.

Step 7 To remove time frames, check the check box next to the schedule detail that you want to remove, and select **Delete Selected**.

> **Note** If you remove all schedule details from a schedule, the schedule is never active. Call handlers and users that use this schedule as per the default schedule, will always use the closed hours transfer settings, and the closed greeting always plays (if enabled) except when it is overridden by the holiday, internal, busy, or alternate greeting.

Deleting Schedules in Cisco Unity Connection SRSV

To Delete a Schedule

Step 1 In Cisco Unity Connection SRSV Administration, expand **System Settings**, then select **Schedules**.

Step 2 On the Search Schedules page, check the check box adjacent to the display name of the schedule that you want to delete.

> **Note** If the schedule that you want to delete does not appear in the search results table, set the applicable parameters in the search fields at the top of the page, and select **Find**.

Step 3 Select **Delete Selected**.

Note If the schedule that you are attempting to delete is referenced by a call routing table or call handler, you receive an error message and are not able to delete the schedule until you find and remove the reference.

Step 4 In the dialog box that asks you to confirm the deletion, select **OK**.

Configuring Conversations Settings in Cisco Unity Connection SRSV

To configure Conversation settings

Step 1 In Cisco Unity Connection SRSV Administration, expand **System Settings**, then select **Conversations**.

Step 2 On the Conversation Configuration page, enter the applicable settings.

Step 3 Select **Save**.

Configuring Enterprise Parameters in Cisco Unity Connection SRSV

Enterprise parameters for Cisco Unity Connection SRSV provide default settings that apply to all services in Cisco Unified Serviceability.

You cannot add or delete enterprise parameters, but you can use the procedure in this section to update the existing enterprise parameters.

Note Many of the enterprise parameters rarely require change. Do not change an enterprise parameter unless you fully understand the feature that you are changing or unless the Cisco Technical Assistance Center (Cisco TAC) specifies the change.

See the following sections:

- Configuring Enterprise Parameters for Cisco Unified Serviceability Services in Cisco Unity Connection SRSV, page 13-3
- Description of Enterprise Parameters in Cisco Unity Connection SRSV, page 13-4

Configuring Enterprise Parameters for Cisco Unified Serviceability Services in Cisco Unity Connection SRSV

Use the following procedure to configure enterprise parameters for Cisco Unified Serviceability services.

To Configure Enterprise Parameters for Cisco Unified Serviceability Services

Step 1 In Cisco Unity Connection SRSV Administration, expand **System Settings**, then select **Enterprise Parameters**.

Step 2 On the Enterprise Parameters page, enter the applicable settings. To set all service parameters to the default values, select **Set to Default**.

To view a list of enterprise parameters and their descriptions, select the **?** button on the right side of the page.

Step 3 Select **Save**.

Description of Enterprise Parameters in Cisco Unity Connection SRSV

Table 13-1 describes the enterprise parameters available in Cisco Unity Connection SRSV.

Table 13-1 Enterprise Parameter Descriptions

Enterprise Parameter	Description
Max Number of Device Level Trace	Specifies how many devices can be traced concurrently if device name-based trace is selected in Trace Configuration in Cisco Unified Serviceability. This is a required field. Default setting: 12 Minimum: 0 Maximum: 256
Localization Parameters	
Default Network Locale	Specifies the default network locale for tones and cadences. The chosen network locale applies to all gateways and phones that do not have the network locale set at the device or device pool level. This is a required field. **Note** Make sure that the chosen network locale is installed and supported for all gateways and phones. See the product documentation, if necessary. Reset all devices for the parameter change to take effect. Default setting: United States
Default User Locale	Specifies the default user locale for language selection. Not all locales are supported by all models. For models that do not support this setting, set their locale explicitly to something they support. This is a required field. **Note** Reset all devices for the parameter change to take effect. Default setting: English United States
Prepare Cluster for Rollback	

Table 13-1 Enterprise Parameter Descriptions (continued)

Enterprise Parameter	Description
Prepare Cluster for Rollback to Pre 8.0	If a Cisco Unity Connection cluster is configured and was upgraded, specifies whether the previous version of Connection was release 7.x. This is a required field. Default setting: False
Trace Parameters	
File Close Thread Flag	Enables the use of separate threads to close trace files. This may improve the performance of the system at the end of a trace file. This is a required field. Default setting: True
FileCloseThreadQueueWater Mark	Defines the high-water mark after which the separate thread used to close trace files stops accepting trace files to close; the trace file is then closed without the use of a separate thread. This is a required field. Default setting: 100 Minimum: 0 Maximum: 500
Clusterwide Domain Configuration Parameters	
Organization Top Level Domain	Defines the top level domain for the organization (for example, cisco.com). Maximum length: 255 Allowed values: Provide a valid domain (for example, cisco.com) with up to 255 of the following characters: any upper or lower case letter (a-z, A-Z), any number (0-9), the hyphen (-), or the dot (.) The dot serves as a domain label separator. Domain labels must not start with a hyphen. The last label (for example, .com) must not start with a number. Abc.1om is an example of an invalid domain.
Cluster Fully Qualified Domain Name	Defines one or more Fully Qualified Domain Names (FQDN) for this cluster. Multiple FQDNs must be separated by a space. Wildcards can be specified within an FQDN using an asterisk (*). Examples are cluster-1.rtp.cisco.com and *.cisco.com. Requests containing URLs (for example, SIP calls) whose host portion matches any of the FQDNs in this parameter are recognized as a request destined for this cluster and/or devices attached to it. Maximum length: 255 Allowed values: Provide one or more fully qualified domain names (FQDN), or partial FQDNs using the * wildcard (for example, cluster-1.cisco.com or *.cisco.com). Multiple FQDNs must be separated by a space. The following characters are allowed: any upper or lower case letter (a-z, A-Z), any number (0-9), hyphen (-), asterisk (*), or dot (.) The dot serves as a domain label separator. Domain labels must not start with a hyphen. The last label (for example, .com) must not start with a number. Abc.1om serves as an example of an invalid domain.

Table 13-1 Enterprise Parameter Descriptions (continued)

Enterprise Parameter	Description
Cisco Support Use	
Cisco Support Use 1	Is used by Cisco TAC only. Maximum length: 10
Cisco Support Use 2	Is used by Cisco Technical Support only. Maximum length: 10
Cisco Syslog Agent	
Remote Syslog Server Name 1 to Remote Syslog Server Name 5	Enter the name or IP address of the remote Syslog server that you want to use to accept Syslog messages. You can configure upto five Remote Syslog Servers to accept Syslog messages. If a server name is not specified, Cisco Unified Serviceability does not send the Syslog messages. Do not specify a Cisco Unified Communications Manager server as the destination because the Cisco Unified Communications Manager server does not accept Syslog messages from another server. Maximum length: 255 Allowed values: Provide a valid remote syslog server name with the following characters: A-Z, a-z, 0-9, ., -
Syslog Severity for Remote Syslog Messages	Select the desired Syslog messages severity for the remote syslog server. All the syslog messages with selected or higher severity level are sent to remote syslog. If a remote server name is not specified, Cisco Unified Serviceability does not send the Syslog messages. This is a required field. Default setting: Error
CUCReports Parameters	
Report Socket Connection Timeout	Specifies the maximum number of seconds used when attempting to establish a connection with another server. Increase this time if connection issues are experienced on a slow network. This is a required field. Default setting: 10 Minimum: 5 Maximum: 120
Report Socket Read Timeout	Specifies the maximum number of seconds used when reading data from another server. Increase this time if connection issues are experienced on a slow network. This is a required field. Default setting: 60 Minimum: 5 Maximum: 600

Installing Plugins in Cisco Unity Connection SRSV

Application plugins extend the functionality of Cisco Unity Connection SRSV. For example, the Real-Time Monitoring Tool (RTMT) allows you to monitor the health of the system remotely through tools such as performance-monitoring counters and the Port Monitor.

Do the following procedure.

Note Before you install any plugins, you must disable all intrusion detection or antivirus services that run on the server where you will install the plugin.

To Install a Plugin

Step 1 In Cisco Unity Connection SRSV Administration, expand **System Settings**, then select **Plugins**.

Step 2 On the Search Plugins page, select **Find**.

Step 3 For the plugin that you want to install, select **Download**.

Step 4 Follow the on-screen instructions for installing the plugin.

CHAPTER 14

Managing the Phone System Integrations in Cisco Unity Connection SRSV

Note You can manage the phone system integrations by adding or deleting phone systems, port groups, ports, and servers. You can also change the settings for existing phone systems, port groups, ports, phone, and servers.

See the following sections:

- Managing Phone Systems in Cisco Unity Connection SRSV, page 14-1
- Managing Port Groups in Cisco Unity Connection SRSV, page 14-3
- Managing Ports in Cisco Unity Connection SRSV, page 14-11
- Security in Cisco Unity Connection SRSV (Cisco Unified Communications Manager Integrations Only), page 14-15

Managing Phone Systems in Cisco Unity Connection SRSV

The phone system pages in Cisco Unity Connection SRSV Administration identify the phone systems that Cisco Unity Connection SRSV integrates with. In Connection SRSV Administration, a phone system has one or more port groups, which in turn have voice messaging ports. You can manage the phone systems to meet the changing needs of your system.

See the following sections:

- Adding a New Phone System Integration, page 14-1
- Deleting a Phone System Integration, page 14-2
- Changing Phone System Settings, page 14-2
- Changing Call Loop Detection Settings, page 14-3

Adding a New Phone System Integration

You can integrate multiple phone systems with Connection SRSV. For a matrix of supported combinations, see the *Multiple Integration Guide for Cisco Unity Connection* at http://www.cisco.com/en/US/products/ps6509/products_installation_and_configuration_guides_list.html.

To Add a New Phone System Integration

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Phone System**.

Step 2 On the Search Phone Systems page, under Phone System Search Results, select **Add New**. The New Phone System page appears.

Step 3 On the New Phone System page, in the Phone System Name field, enter a descriptive name for the phone system and select **Save**.

Step 4 On the Phone System Basics page, enter the applicable settings and select **Save**.

Deleting a Phone System Integration

You can delete a phone system when the phone system is no longer used by Connection SRSV. Before you delete a phone system, you must delete or reassign all of the following objects that are associated with the phone system that you want to delete:

- All users (including MWI devices and notification devices)
- All user templates
- All system call handlers
- All call handler templates

Note You can see a list of all users who are associated with the phone system on the Phone System Associations page. For instructions, see the "Changing Call Loop Detection Settings" section on page 14-3.

To Delete a Phone System Integration

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Phone System**.

Step 2 On the Search Phone Systems page, under Phone System Search Results, check the check box next to the name of the phone systems that you want to delete.

Step 3 Select **Delete Selected**.

Step 4 When prompted to confirm that you want to delete the phone systems, select **OK**.

Changing Phone System Settings

You can change the settings for a phone system after it is integrated with Connection SRSV. The phone system settings identify the phone system that Connection SRSV integrates with and regulate certain phone system features. (Integration configuration settings are located in the port groups that belong to the phone system.)

To Change Phone System Settings

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Phone System**.

Step 2 On the Search Phone Systems page, select the display name of the phone system for which you want to change the settings.

Step 3 On the Phone System Basics page, change the applicable settings and select **Save**.

Changing Call Loop Detection Settings

Calls that Cisco Unity Connection SRSV forwards (for example, to notify a user that a message has been received) are sometimes forwarded back to Connection SRSV. When call loop detection is enabled, Connection SRSV detects when a call loop has occurred and rejects the call.

You can change the call loop detection settings to enable or disable the types of calls that are checked, to set the fourth-column DTMF tone that Connection SRSV uses, and to set the guard time.

The call loop detection settings should not be changed without understanding the effect that they have on calls that Connection SRSV forwards.

To Change Call Loop Detection Settings

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Phone System**.

Step 2 On the Search Phone Systems page, select the display name of the phone system.

Step 3 On the Phone System Basics page, under Call Loop Detection by Using DTMF, enter applicable settings and select **Save**.

Managing Port Groups in Cisco Unity Connection SRSV

Port groups hold most of the integration configuration settings and some or all of the voice messaging ports for Connection SRSV.

Connection SRSV port groups provide flexibility for integration configuration settings that apply to different sets of ports.

See the following sections:

- Adding a Port Group, page 14-4
- Deleting a Port Group, page 14-4
- Changing Port Group Settings, page 14-5
- Changing the Audio Format That Cisco Unity Connection SRSV Uses for Calls, page 14-5
- Adding Secondary Cisco Unified Communications Manager Servers, page 14-6
- Deleting Cisco Unified Communications Manager Servers, page 14-6

Adding a Port Group

You can add multiple port groups, each with its own integration configuration settings and its own voice messaging ports.

Cisco Unified Communications Manager Business Edition (CMBE) only: Before you can add a port group, you must have existing voice messaging ports in Cisco Unified CM Administration that do not belong to a port group.

To Add a Port Group

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, under Port Group Search Results, select **Add New**.

Step 3 On the New Port Group page, enter the applicable settings and select **Save**.

Deleting a Port Group

When you delete a port group, any voice messaging ports that belong to it are deleted at the same time, but the phone system that the port group belongs to is not deleted.

To Delete a Port Group

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, under Port Group Search Results, check the check box next to the port group name of the port groups that you want to delete.

Step 3 Select **Delete Selected**.

Step 4 When prompted to confirm that you want to delete the port group, select **OK**.

Changing Port Group Settings

You can change the settings for a port group after it has been added. Changes to the settings affect only the voice messaging ports that belong to the port group.

To Change Port Group Settings

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, select the display name of the port group for which you want to change the settings.

Step 3 On the Port Group Basics page, change the applicable settings and select **Save**.

Changing the Audio Format That Cisco Unity Connection SRSV Uses for Calls

For calls, Cisco Unity Connection SRSV advertises the audio format (or codec) that is preferred for the media stream with the phone system. You should consider the following when setting the audio format:

- For the following reasons, Connection SRSV should use the same audio format for the media stream that the phone system uses:
  - To reduce the need for transcoding the media stream from one audio format to another.
  - To minimize the performance impact on the Connection SRSV server and on the phone system.
  - To preserve the audio quality of calls.
- When Connection SRSV advertises a different audio format than the one used by the phone system, the phone system transcodes the media stream.

To Change the Audio Format That Cisco Unity Connection Uses for Calls

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, select the first port group that belongs to the phone system integration for which you want to change the audio format of the media stream.

Step 3 On the Port Group Basics page, on the Edit menu, select **Codec Advertising**.

Step 4 On the Edit Codec Advertising page, select the **Up** and **Down** arrows to change the order of the codecs or to move codecs between the Advertised Codec box and the Unadvertised Codecs box.

If only one codec is in the Advertised Codecs box, Connection SRSV sends the media stream in that audio format. The phone system transcodes if it does not use this audio format.

If two or more codecs are in the Advertised Codecs box, Connection SRSV advertises its preference for the first codec in the list but sends the media stream in the audio format from the list that the phone system selects.

Step 5 Select **Save**.

Step 6 *(All integrations except SCCP)* If you want to change the packet size that is used by the advertised codecs, on the Port Group Basics page, under Advertised Codec Settings, select the applicable packet setting for each codec and select **Save**.

Step 7 Select **Next**.

Step 8 Repeat Step 3 through Step 7 for all remaining port groups that belong to the phone system integration for which you want to change the audio format of the media stream.

Adding Secondary Cisco Unified Communications Manager Servers

For Cisco Unified Communications Manager integrations, Related Links helps you create the integration only with one Cisco Unified CM server. The secondary Cisco Unified CM servers in the cluster must be added after the integration is created.

Note Cisco Unified Communications Manager Business Edition (CMBE) does not support secondary Cisco Unified CM servers.

To Add Secondary Cisco Unified Communications Manager Servers

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, select the display name of the port group for which you want to add secondary Cisco Unified CM servers.

Step 3 On the Port Group Basics page, on the Edit menu, select **Servers**.

Step 4 On the Edit Servers page, under Cisco Unified Communications Manager Servers, select **Add**.

Step 5 Enter the settings for the secondary Cisco Unified CM server and select **Save**.

Step 6 Repeat Step 4 and Step 5 for all remaining secondary Cisco Unified CM servers that you want to add.

Note You can select **Ping** to verify the IP address (or host name) of the Cisco Unified CM server.

Deleting Cisco Unified Communications Manager Servers

You can delete a Cisco Unified Communications Manager server when it is no longer used by the phone system integration.

If you want to move a Cisco Unified CM server to another port group, you must delete the Cisco Unified CM server from one port group and add it to the second port group.

Note Cisco Unified Communications Manager Business Edition (CMBE) does not support deleting Cisco Unified CM servers.

To Delete a Cisco Unified Communications Manager Server

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, select the display name of the port group for which you want to delete Cisco Unified CM servers.

Step 3 On the Port Group Basics page, on the Edit menu, select **Servers**.

Step 4 On the Edit Servers page, under Cisco Unified Communications Manager Servers, check the check box next to the Cisco Unified CM servers that you want to delete.

Step 5 Select **Delete Selected**.

Step 6 When prompted to confirm that you want to delete the Cisco Unified CM servers, select **OK**.

Changing Cisco Unified Communications Manager Server Settings

You can change the Cisco Unified CM server settings after the server has been added.

To Change Cisco Unified Communications Manager Server Settings

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, select the display name of the port group for which you want to change Cisco Unified CM server settings.

Step 3 On the Port Group Basics page, on the Edit menu, select **Servers**.

Step 4 On the Edit Servers page, under Cisco Unified Communications Manager Servers, change the applicable settings and select **Save**.

Note You can select **Ping** to verify the IP address (or host name) of the Cisco Unified CM server.

Adding a TFTP Server

For Cisco Unified Communications Manager integrations, TFTP servers are required only when the Cisco Unified CM cluster uses authentication and encryption for the Connection SRSV voice messaging ports.

If your system uses authentication and encryption for the Connection SRSV voice messaging ports, you must add a TFTP server after you create the Cisco Unified CM phone system integration.

To Add a TFTP Server

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, select the display name of the port group for which you want to add a TFTP server.

Step 3 On the Port Group Basics page, on the Edit menu, select **Servers**.

Step 4 On the Edit Servers page, under TFTP Servers, select **Add**.

Step 5 Enter the settings for the TFTP server and select **Save**.

Step 6 Repeat Step 4 and Step 5 for all remaining TFTP servers that you want to add.

Note You can select **Ping** to verify the IP address (or host name) of the TFTP server.

Deleting a TFTP Server

You can delete a TFTP server when it is no longer used by the port group.

For Cisco Unified Communications Manager integrations, TFTP servers are required only when the Cisco Unified CM cluster uses authentication and encryption for the Connection SRSV voice messaging ports.

To Delete a TFTP Server

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, select the display name of the port group for which you want to delete a TFTP server.

Step 3 On the Port Group Basics page, on the Edit menu, select **Servers**.

Step 4 On the Edit Servers page, under TFTP Servers, check the check box next to the TFTP server that you want to delete.

Step 5 Select **Delete Selected**.

Step 6 When prompted to confirm that you want to delete the TFTP server, select **OK**.

Changing TFTP Server Settings

You can change the TFTP server settings after the server has been added.

For Cisco Unified Communications Manager integrations, TFTP servers are required only when the Cisco Unified CM cluster uses authentication and encryption for the Connection SRSV voice messaging ports.

To Change TFTP Server Settings

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, select the display name of the port group for which you want to change TFTP server settings.

Step 3 On the Port Group Basics page, on the Edit menu, select **Servers**.

Step 4 On the Edit Servers page, under TFTP Servers, change the applicable settings and select **Save**.

Note You can select **Ping** to verify the IP address (or host name) of the TFTP server.

Adding a SIP Server

For a phone system integration with Cisco Unified Communications Manager through a SIP trunk or with another SIP server, you can add another SIP server after the phone system has been created.

Note Cisco Unified Communications Manager Business Edition (CMBE) does not support SIP servers.

To Add a SIP Server

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, select the display name of the port group for which you want to add SIP servers.

Step 3 On the Port Group Basics page, on the Edit menu, select **Servers**.

Step 4 On the Edit Servers page, under SIP Servers, select **Add**.

Step 5 Enter the settings for the SIP server and select **Save**.

Step 6 Repeat Step 4 and Step 5 for all remaining SIP servers that you want to add.

Note You can select **Ping** to verify the IP address (or host name) of the SIP server.

Deleting a SIP Server

For a phone system integration with Cisco Unified Communications Manager through a SIP trunk or with another SIP server, you can delete a SIP server when it is no longer used by the port group.

Note Cisco Unified Communications Manager Business Edition (CMBE) does not support SIP servers.

To Delete a SIP Server

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, select the display name of the port group for which you want to delete SIP servers.

Step 3 On the Port Group Basics page, on the Edit menu, select **Servers**.

Step 4 On the Edit Servers page, under SIP Servers, check the check box next to the SIP server that you want to delete.

Step 5 Select **Delete Selected**.

Step 6 When prompted to confirm that you want to delete the SIP server, select **OK**.

Changing SIP Server Settings

For a phone system integration with Cisco Unified Communications Manager through a SIP trunk or with another SIP server, you can change the SIP server settings after the server has been added.

Note Cisco Unified Communications Manager Business Edition (CMBE) does not support SIP servers.

To Change SIP Server Settings

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, select the display name of the port group for which you want to change SIP server settings.

Step 3 On the Port Group Basics page, on the Edit menu, select **Servers**.

Step 4 On the Edit Servers page, under SIP Servers, change the applicable settings and select **Save**.

Note You can select **Ping** to verify the IP address (or host name) of the SIP server.

Changing Port Group Advanced Settings

The port group advanced settings control infrequently used settings such as delays and MWI usage. We recommend that port group advanced settings be left at their default values.

To Change Port Group Advanced Settings

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, select the display name of the port group for which you want to change the advanced settings.

Step 3 On the Port Group Basics page, on the Edit menu, select **Advanced Settings**.

Step 4 On the Edit Advanced Settings page, under Port Group Advanced Settings, change the applicable settings and select **Save**.

Enabling or Disabling Normalization

Normalization controls automatic volume adjustments for recording messages. We recommend that you leave normalization enabled and that you not change the value of the Target Decibel Level for Recordings and Messages field on the System Settings > General Configuration page.

To Enable or Disable Normalization

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 2 On the Search Port Groups page, select the display name of the port group for which you want to change the advanced settings.

Step 3 On the Port Group Basics page, on the Edit menu, select **Advanced Settings**.

Step 4 On the Edit Advanced Settings page, under Audio Normalization for Recordings and Messages, change the applicable settings and select **Save**.

Managing Ports in Cisco Unity Connection SRSV

The voice messaging ports let Cisco Unity Connection SRSV receive calls (for example, to record a message) and let Connection SRSV make calls (for example to send message notifications or to set MWIs).

Each voice messaging port can belong to only one port group. Port groups, when there are several, each have their own voice messaging ports. The total voice messaging ports belonging to all port groups must not exceed the maximum number of voice messaging ports that are enabled by the Connection SRSV license files.

See the following sections:

- Adding a Port, page 14-12
- Deleting a Port, page 14-12
- Changing Port Settings, page 14-13
- Viewing the Port Certificate, page 14-14

Adding a Port

Voice messaging ports provide the connections for calls between Cisco Unity Connection SRSV and the phone system. You can add voice messaging ports after the phone system has been created. The number of voice messaging ports that you add cannot bring the total number of voice messaging ports for all port groups to more than the maximum number of voice messaging ports that are enabled by the Connection SRSV license files.

Cisco Unified Communications Manager Business Edition (CMBE) only: Before you can add ports, you must have existing voice messaging ports in Cisco Unified CM Administration that do not belong to a port group.

To Add a New Port

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port**.

Step 2 On the Search Ports page, under Port Search Results, select **Add New**.

Step 3 On the New Port page, enter the applicable settings and select **Save**.

Caution Verify that there are an appropriate number of ports set to answer calls and an appropriate number of ports set to dial out. Otherwise, the integration may not function correctly. See the "Planning How the Voice Messaging Ports Will Be Used by Cisco Unity Connection" section of the applicable Cisco Unity Connection integration guide at http://www.cisco.com/en/US/products/ps6509/products_installation_and_configuration_guides_list.html.

Step 4 In Connection SRSV Administration, in the Related Links list, select **Check Telephony Configuration** and select **Go** to confirm the phone system integration settings.

Step 5 If the test is not successful, the Task Execution Results list displays one or more messages with troubleshooting steps. After correcting the problems, check the configuration again.

Deleting a Port

Voice messaging ports provide the connections for calls between Connection SRSV and the phone system.

To Delete a Port

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port**.

Step 2 On the Search Ports page, under Port Search Results, check the check box next to the voice messaging ports that you want to delete.

Step 3 Select **Delete Selected**.

Step 4 For the remaining voice messaging ports in the port group, change the settings as necessary so that there are an appropriate number of voice messaging ports set to answer calls and an appropriate number of voice messaging ports set to dial out.

Changing Port Settings

Voice messaging ports provide the connections for calls between Connection SRSV and the phone system. You can change the voice messaging port settings after the phone system integration has been created.

To Change Port Settings

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port**.

Step 2 On the Search Ports page, select the display name of the voice messaging port for which you want to change the settings.

Step 3 On the Port Basics page, enter the applicable settings and select **Save**.

Depending on the phone system integration, some or all of the fields in Table 14-1 appear.

Table 14-1 Port Basics Page Settings

Field	Considerations
Enabled	Check this check box to enable the port. The port is enabled during normal operation. Uncheck this check box to disable the port. When the port is disabled, calls to the port get a ringing tone but are not answered. Typically, the port is disabled only by the installer during testing.
Server Name	*(For Connection SRSV redundancy only)* Select the name of the Connection SRSV server that you want to handle this port. Assign an equal number of answering and dial-out voice messaging ports to the Connection SRSV servers so that they equally share the voice messaging traffic.
Extension	Enter the extension for the port as assigned on the phone system.
Answer Calls	Check this check box to designate the port for answering calls. These calls can be incoming calls from outside callers or from users.
Perform Message Notification	Check this check box to designate the port for notifying users of messages. Assign Perform Message Notification to the least busy ports.
Send MWI Requests *(not used by serial integrations)*	Check this check box to designate the port for turning MWIs on and off. Assign Send MWI Requests to the least busy ports. For serial integrations, uncheck this check box. Otherwise, the integration may not function correctly.
Allow TRAP Connections	Check this check box so that users can use the port for recording and playback through the phone in Connection SRSV web applications. Assign Allow TRAP Connections to the least busy ports.
Outgoing Hunt Order *(not available for SIP integrations)*	Enter the priority order in which Connection SRSV uses the ports when dialing out (for example, if the Perform Message Notification, Send MWI Requests, or Allow TRAP Connections check box is checked). The highest numbers are used first. However, when multiple ports have the same Outgoing Hunt Order number, Connection SRSV uses the port that has been idle the longest.

Table 14-1 Port Basics Page Settings (continued)

Field	Considerations
Security Mode *(available for Cisco Unified CM SCCP integrations only)*	Select the applicable security mode: • **Non-secure**—The integrity and privacy of call-signaling messages are not ensured because call-signaling messages are sent as clear (unencrypted) text and are connected to Cisco Unified Communications Manager through a non-authenticated port rather than an authenticated TLS port. In addition, the media stream is not encrypted. • **Authenticated**—The integrity of call-signaling messages are ensured because they are connected to Cisco Unified CM through an authenticated TLS port. However, the privacy of call-signaling messages are not ensured because they are sent as clear (unencrypted) text. In addition, the media stream are not encrypted. • **Encrypted**—The integrity and privacy of call-signaling messages are ensured on this port because they are connected to Cisco Unified CM through an authenticated TLS port, and the call-signaling messages are encrypted. In addition, the media stream is encrypted.

Step 4 If there are no more voice messaging ports for which you want to change the settings, skip to Step 6. Otherwise, select **Next**.

Step 5 Repeat Step 3 and Step 4 for all remaining voice messaging ports for which you want to change the settings.

Step 6 On the Port menu, select **Search Ports**.

Step 7 On the Search Ports page, confirm that there are an appropriate number of voice messaging ports set to answer calls and an appropriate number of voice messaging ports set to dial out. If necessary, adjust the number of voice messaging ports set to answer calls and an appropriate number of voice messaging ports set to dial out.

Viewing the Port Certificate

Port certificates for voice messaging ports are used only by SCCP integrations with Cisco Unified Communications Manager 4.1 and later, and are required for authentication of the Connection SRSV voice messaging ports. You can view the port certificate to help in troubleshooting authentication and encryption problems.

To View the Port Certificate

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations**, then select **Port**.

Step 2 On the Search Ports page, select the display name of the voice messaging port for which you want to see the device certificate.

Step 3 On the Port Basics page, select **View Certificate**.

Step 4 In the View Port Certificate window, the information from the port device certificate is displayed.

Security in Cisco Unity Connection SRSV (Cisco Unified Communications Manager Integrations Only)

When Cisco Unified Communications Manager authentication and encryption is configured for Connection SRSV voice messaging ports, you can manage certifications and the security profile.

See the following sections:

- Viewing the Cisco Unity Connection SRSV Root Certificate, page 14-15
- Saving the Cisco Unity Connection SRSV Root Certificate as a File, page 14-15
- Adding a SIP Certificate (Cisco Unified Communications Manager SIP Trunk Integrations Only), page 14-16
- Deleting a SIP Certificate (Cisco Unified Communications Manager SIP Trunk Integrations Only), page 14-17
- Changing a SIP Certificate (Cisco Unified Communications Manager SIP Trunk Integrations Only), page 14-17
- Adding a SIP Security Profile (Cisco Unified Communications Manager SIP Trunk Integrations Only), page 14-17
- Deleting a SIP Security Profile (Cisco Unified Communications Manager SIP Trunk Integrations Only), page 14-18
- Changing a SIP Security Profile (Cisco Unified Communications Manager SIP Trunk Integrations Only), page 14-18

Viewing the Cisco Unity Connection SRSV Root Certificate

The root certificate is used by SCCP integrations with Cisco Unified Communications Manager 4.1 and later and SIP trunk integrations with Cisco Unified CM 7.0 and later, and is required for authentication of the Connection SRSV voice messaging ports. You can view the root certificate to help troubleshoot authentication and encryption problems.

To View the Cisco Unity Connection SRSV Root Certificate

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations > Security**, then select **Root Certificate**.

Step 2 On the View Root Certificate page, the information from the root certificate is displayed.

Saving the Cisco Unity Connection SRSV Root Certificate as a File

The root certificate is used by SCCP integrations with Cisco Unified CM 4.1 and later and SIP trunk integrations with Cisco Unified CM 7.0 and later, and is required for authentication of the Connection SRSV voice messaging ports.

To Save the Cisco Unity Connection SRSV Root Certificate as a File

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations > Security**, then select **Root Certificate**.

Step 2 On the View Root Certificate page, right-click the **Right-Click to Save the Certificate as a File** link, and select **Save Target As**.

Step 3 In the Save As dialog box, browse to the location where you want to save the Connection SRSV root certificate as a file.

Step 4 In the File Name field, confirm that the filename has the correct extension, depending on the version of Cisco Unified CM:

- For Cisco Unified CM 5.x or later, confirm that the extension is .pem (rather than .htm).
- For Cisco Unified CM 4.x, confirm that the extension is .0 (rather than .htm).

Caution The certificate must be saved as a file with the correct extension or Cisco Unified CM will not recognize the certificate.

Step 5 Select **Save**.

Step 6 In the Download Complete dialog box, select **Close**.

Step 7 The Connection SRSV root certificate file is ready to be copied to all Cisco Unified CM servers in this Cisco Unified CM phone system integration. For instructions, see the applicable Cisco Unified CM integration guide at http://www.cisco.com/en/US/products/ps6509/products_installation_and_configuration_guides_list.html.

Adding a SIP Certificate (Cisco Unified Communications Manager SIP Trunk Integrations Only)

The SIP certificate is used only by SIP trunk integrations with Cisco Unified CM 7.0 and later, and is required for authentication of the Connection SRSV voice messaging ports.

To Add a SIP Certificate (Cisco Unified Communications Manager SIP Trunk Integrations Only)

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations > Security**, then select **SIP Certificate**.

Step 2 On the Search SIP Certificates page, select **Add New**.

Step 3 On the New SIP Certificate page, in the Display Name field, enter a display name for the SIP certificate.

Step 4 In the Subject Name field, enter a subject name that matches the X.509 subject name of the SIP security profile for the SIP trunk in Cisco Unified CM Administration.

Caution This subject name must match the X.509 subject name of the SIP security profile used by Cisco Unified CM. Otherwise, Cisco Unified CM authentication and encryption fail.

Step 5 Select **Save**.

Deleting a SIP Certificate (Cisco Unified Communications Manager SIP Trunk Integrations Only)

You can delete a SIP certificate when the Cisco Unified CM server is no longer configured for authentication of the Cisco Unity Connection voice messaging ports.

To Delete a SIP Certificate (Cisco Unified Communications Manager SIP Trunk Integrations Only)

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations > Security**, then select **SIP Certificate**.

Step 2 On the Search SIP Certificates page, check the check box next to the display name of the SIP certificate that you want to delete.

Step 3 Select **Delete Selected**.

Step 4 When prompted to confirm that you want to delete the SIP certificate, select **OK**.

Changing a SIP Certificate (Cisco Unified Communications Manager SIP Trunk Integrations Only)

You can change a SIP certificate after it is created.

To Change a SIP Certificate (Cisco Unified Communications Manager SIP Trunk Integrations Only)

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations > Security**, then select **SIP Certificate**.

Step 2 On the Search SIP Certificates page, select the name of the SIP certificate that you want to change.

Step 3 On the Edit SIP Certificate page, enter the applicable settings and select **Save**.

Adding a SIP Security Profile (Cisco Unified Communications Manager SIP Trunk Integrations Only)

The SIP security profile is used only by SIP trunk integrations with Cisco Unified CM 7.0 and later, and is required for authentication of the Connection SRSV voice messaging ports.

To Add a SIP Security Profile (Cisco Unified Communications Manager SIP Trunk Integrations Only)

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations > Security**, then select **SIP Security Profile**.

Step 2 On the Search SIP Security Profiles page, select **Add New**.

Step 3 On the New SIP Security Profile page, in the Port field, enter the port number that the Cisco Unified CM server uses for SIP trunk authentication and encryption of the voice messaging ports.

Step 4 To encrypt the call signaling messages, check the **Do TLS** check box.

Step 5 Select **Save**.

Deleting a SIP Security Profile (Cisco Unified Communications Manager SIP Trunk Integrations Only)

You can delete a SIP security profile when the Cisco Unified CM server is no longer configured for authentication of the Connection SRSV voice messaging ports.

To Delete a SIP Security Profile (Cisco Unified Communications Manager SIP Trunk Integrations Only)

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations > Security**, then select **SIP Security Profile**.

Step 2 On the Search SIP Security Profiles page, check the check box next to the display name of the SIP security profile that you want to delete.

Step 3 Select **Delete Selected**.

Step 4 When prompted to confirm that you want to delete the SIP security profile, select **OK**.

Changing a SIP Security Profile (Cisco Unified Communications Manager SIP Trunk Integrations Only)

You can change a SIP security profile after it is created.

To Change a SIP Security Profile (Cisco Unified Communications Manager SIP Trunk Integrations Only)

Step 1 In Cisco Unity Connection SRSV Administration, expand **Telephony Integrations > Security**, then select **SIP Security Profile**.

Step 2 On the Search SIP Certificates page, select the name of the SIP security profile that you want to change.

Step 3 On the Edit SIP Security Profile page, enter the applicable settings and select **Save**.

CHAPTER 15

Cisco Unity Connection SRSV Administration - Telephony Integration Settings Interface

See the following sections:

Search Phone Systems

The Search Phone Systems page displays the status with the total number of phone systems.

The search results, by default, return all phone systems. By default, the administrator can view 25 records per page and can select rows per page from the dropdown list, the max limit being 250 search results returned on one page. The administrator can perform custom search on the phone system display name field using the following options:

- Begins with
- Contains.
- Ends with
- Is Exactly
- Is Empty
- Is Not Empty

Table 15-1 Search Phone Systems Page

Field	Description
Display Name	*(Display only)* The name of the phone system. Select the Display Name to open the detail of a phone system.
Port Count	*(Display only)* The number of voice messaging ports that have been created in Cisco Unity Connection SRSV for use by the phone system.

Phone System Basics

Table 15-2 Phone System Basics Page

Field	Description
Phone System Name	Enter a descriptive name for the phone system.
Enable for Supervised Transfers	Check this check box so that Cisco Unity Connection SRSV uses DTMF to detect and reject calls that have been transferred to another extension (by using supervised transfer) and that have been transferred back to Connection SRSV. If the call loop is not detected and rejected, Connection SRSV records a voice message that contains the prompt to leave a voice message. Default setting: Check box not checked.
Enable Outgoing Calls	When this option is selected, Cisco Unity Connection SRSV places outgoing calls (for example, setting MWIs) as needed through the phone system. Default setting: Option selected.
Disable All Outgoing Calls Immediately	When this option is selected, Cisco Unity Connection SRSV does not place any outgoing calls (for example, setting MWIs). This option is useful when the phone system cannot respond to outgoing calls because of maintenance. Default setting: Option not selected.
Disable All Outgoing Calls Between	When this option is selected, Cisco Unity Connection SRSV does not place any outgoing calls (for example, setting MWIs) between the times set in the Beginning Time field and the Ending Time field. This option is useful when the phone system cannot respond to outgoing calls because of scheduled maintenance. Default setting: Option not selected.

Search Port Groups

The Search Port Groups page displays the status with the total number of port groups.

The search results, by default, return all port groups. By default, the administrator can view 25 records per page and can select rows per page from the dropdown list, the max limit being 250 search results returned on one page. The administrator can perform custom search on the port group's display name field using the following options:

- Begins with
- Contains.
- Ends with
- Is Exactly
- Is Empty
- Is Not Empty

Table 15-3 Search Port Groups Page

Field	Description
Delete Selected	To delete a port group, check the check box to the left of the display name, and select Delete Selected. You can delete multiple port groups at once.
Add New	To add a port group, select the Add New button. A new page opens, on which you enter data applicable to the new port group.
Port Group Name	The descriptive name for the port group. Select this name to view and edit the phone system settings. Select the Port Group Name to open the detail of a particular port group.
Phone System Display Name	*(Display only)* The phone system that uses the port group.
Port Count	*(Display only)* The number of voice messaging ports that have been created in Cisco Unity Connection SRSV for use by the port group.
Integration Method	*(Display only)* The method of integration that is used to connect Cisco Unity Connection SRSV and the phone system.
Needs Reset	*(Display only)* Indicates whether the port group needs to be reset to assure all functions.

New Port Group

Table 15-4 New Port Group Page

Field	Description
Phone System	Select the phone system that uses the port group.

Table 15-4 New Port Group Page (continued)

Field	Description
Create From	Select one of the following: • Port Group Type —Connection SRSV creates the new port group based on the type that is selected from the list. The new port group has default settings as specified in the port group type. • Port Group—Connection SRSV creates the new port group from the existing port group that is selected from the list. The new port group has the current settings of the selected port group.
Display Name	Enter a descriptive name for the port group.
Device Name Prefix	*(Cisco Unified CM SCCP integrations only)* Enter the prefix that Cisco Unified Communications Manager adds to the device name for voice ports. This prefix must match the prefix used by Cisco Unified CM.
IPv4 Address or Host Name	Enter the IPv4 address (or host name) of the phone system, or SIP server that the port group connects to. You must enter an IP address or host name in this field, or an IP address or host name in the IPv6 Address or Host Name field (or, if applicable, enter information in both fields). You cannot leave both fields blank. **Note** If you will use Cisco Unified CM authentication and encryption with SCCP ports, enter an IP address or host name in this field. The CTL file used for encryption between Connection and Cisco Unified CM for SCCP ports requires an IPv4 address or host name, even if you are otherwise using IPv6 addressing.
IPv6 Address or Host Name	Enter the IPv6 address (or host name) of the Cisco Unified Communications Manager server that the port group connects to. You must enter an IP address or host name in this field, or an IP address or host name in the IPv4 Address or Host Name field (or, if applicable, enter information in both fields). You cannot leave both fields blank. **Note** This setting is applicable to Connection only. IPv6 is not supported in Cisco Unified Communications Manager Business Edition. **Note** This setting is applicable to Cisco Unified Communications Manager integrations only. IPv6 is not supported with other phone system integrations.
Port	*(Cisco Unified CM SCCP integrations only)* Enter the TCP port of the primary Cisco Unified Communications Manager server that Cisco Unity Connection connects to. We recommend that you use the default setting. Default setting: 2000.
TLS Port	*(Cisco Unified CM SCCP integrations only)* Enter the TLS port of the Cisco Unified Communications Manager server that you are integrating with Cisco Unity Connection. Default setting: 2443.

Note The SCCP and SIP port groups support both the IPv4 and IPv6 addresses. However, the IPv6 address works only when Connection platform is configured in Dual (IPv4/IPv6) mode. For more information on Configuring IPv6 settings, see Adding or Changing the IPv6 Addresses of Cisco Unity Connection chapter of *Reconfiguration and Upgrade Guide for Cisco Unity Connection* guide at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/upgrade/guide/9xcucrug051.html.

Port Group Basics

Table 15-5 Port Group Basics Page

Field	Description
Display Name	Enter a descriptive name for the port group.
Integration Method	*(Display only)* The method of integration that is used to connect Cisco Unity Connection SRSV and the phone system.
Device Name Prefix	*(Cisco Unified CM SCCP integrations only)* Enter the prefix that Cisco Unified Communications Manager adds to the device name for voice ports. This prefix must match the prefix used by Cisco Unified CM.
Reset Status	*(Display only)* Indicates whether the port group needs to be reset to assure all functions.

Note The SCCP and SIP port groups support both the IPv4 and IPv6 addresses. For more information on Configuring IPv6 settings, see Adding or Changing the IPv6 Addresses of Cisco Unity Connection chapter of *Reconfiguration and Upgrade Guide for Cisco Unity Connection* guide at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/upgrade/guide/9xcucrug051.html.

Edit Servers

Table 15-6 Edit Servers Page

Field	Description
Cisco Unified Communications Manager Servers	
Delete Selected	To delete a server, check the check box to the left of the display name, and select Delete Selected. You can delete multiple servers at once.
Add	To add a server, select the Add button. A new row appears, in which you enter data applicable to the new server.
Order	Enter the order of priority for the Cisco Unified Communications Manager server that the port group connects to. The lowest number is the primary Cisco Unified CM server, the higher numbers are the secondary servers.
IPv4 Address or Host Name	Enter the IPv4 address (or host name) of the Cisco Unified Communications Manager server that the port group connects to. You must enter an IP address or host name in this field, or an IP address or host name in the IPv6 Address or Host Name field (or, if applicable, enter information in both fields). You cannot leave both fields blank. **Note** If you will use Cisco Unified CM authentication and encryption with SCCP ports, enter an IP address or host name in this field. The CTL file used for encryption between Connection SRSV and Cisco Unified CM for SCCP ports requires an IPv4 address or host name, even if you are otherwise using IPv6 addressing.

Table 15-6 Edit Servers Page (continued)

Field	Description
IPv6 Address or Host Name	Enter the IPv6 address (or host name) of the Cisco Unified Communications Manager server that the port group connects to. You must enter an IP address or host name in this field, or an IP address or host name in the IPv4 Address or Host Name field (or, if applicable, enter information in both fields). You cannot leave both fields blank. **Note** This setting is applicable to Connection SRSV only. IPv6 is not supported in Cisco Unified Communications Manager Business Edition. **Note** This setting is applicable to Cisco Unified Communications Manager integrations only. IPv6 is not supported with other phone system integrations.
Port	Enter the TCP port of the Cisco Unified Communications Manager server that Cisco Unity Connection SRSV uses. We recommend that you use the default setting. Default setting: 2000.
TLS Port	Enter the TLS port of the Cisco Unified Communications Manager server. We recommend that you use the default setting. Default setting: 2443.
Server Type	Select the type of Cisco Unified Communications Manager server that Cisco Unity Connection is integrating with—Cisco Unified Communications Manager or Cisco Unified Communications Manager Express. Default setting: Cisco Unified Communications Manager.
Reconnect To a Higher-Order Cisco Unified Communications Manager When Available	Check this check box so that Cisco Unity Connection SRSV reregisters ports in the port group to a server listed as higher priority in the Cisco Unified Communications Manager Servers table as soon as possible after an outage involving the higher-priority server. The connection between Cisco Unified CM and Connection is maintained by using a keep-alive that is sent on an interval specified by Cisco Unified CM. When this check box is checked, Connection will reconnect with a higher-priority Cisco Unified CM server as soon as the keep-alive indicates that the server is available. Uncheck this check box so that Cisco Unity Connection SRSV continues to connect to a lower-priority Cisco Unified CM server after an outage involving a higher-priority server, even when keep-alives indicate that the higher-priority server has become available again.
TFTP Servers	
Delete Selected	To delete a TFTP server, check the check box to the left of the display name, and select Delete Selected. You can delete multiple TFTP servers at once.
Add	To add a server, select the Add button. A new row appears, in which you enter data applicable to the new server.
Order	Enter the order of priority for the TFTP server that the port group connects to. The lowest number is the primary TFTP server, the higher numbers are the secondary servers.

Table 15-6 Edit Servers Page (continued)

Field	Description
IPv4 Address or Host Name	Enter the IPv4 address (or host name) of the TFTP server that the port group connects to. You must enter an IP address or host name in this field, or an IP address or host name in the IPv6 Address or Host Name field (or, if applicable, enter information in both fields). You cannot leave both fields blank. **Note** If you will use Cisco Unified CM authentication and encryption with SCCP ports, enter an IP address or host name in this field. The CTL file used for encryption between Connection SRSV and Cisco Unified CM for SCCP ports requires an IPv4 address or host name, even if you are otherwise using IPv6 addressing.
IPv6 Addressing Mode	
Preference for Signaling	This setting determines the call control signaling preference when registering with Cisco Unified Communications Manager via SCCP and when initiating SIP requests. This setting is applicable only when the IP Addressing Mode option on the System Settings > General Configuration page is set to IPv4 and IPv6. Select the option from the list to control how Connection controls out-going traffic: • IPv4 Default Setting: IPv4 **Note** The settings on this page are applicable to Connection SRSV only. IPv6 is not supported in Cisco Unified Communications Manager Business Edition. **Note** This setting is applicable to Cisco Unified CM integrations only. IPv6 is not supported with other phone system integrations.

Note The SCCP and SIP port groups support both the IPv4 and IPv6 addresses. For more information on Configuring IPv6 settings, see Adding or Changing the IPv6 Addresses of Cisco Unity Connection chapter of *Reconfiguration and Upgrade Guide for Cisco Unity Connection* guide at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/upgrade/guide/9xcucrug051.html.

Edit Advanced Settings

Table 15-7 Edit Advanced Settings Page

Field	Description
Delay After Answer ____ Milliseconds	Enter the wait time, in milliseconds, after a call is connected to Connection SRSV and before Connection plays a greeting. Default setting: 0 milliseconds.
Outgoing Guard Time _____ Milliseconds	Enter the wait time, in milliseconds, that a voice messaging port must be inactive before Connection SRSV uses it for an outgoing call. Default setting: 1,000 milliseconds.

Table 15-7 Edit Advanced Settings Page (continued)

Field	Description
Outgoing Pre-Dial Delay ____ Milliseconds	Enter the wait time, in milliseconds, before Connection SRSV dials an outgoing call. Default setting: 0 milliseconds.
Outgoing Post-Dial Delay ____ Milliseconds	Enter the wait time, in milliseconds, after Connection SRSV dials an outgoing call. Default setting: 0 milliseconds.
DTMF Interdigit Delay ____ Milliseconds	Enter the wait time, in milliseconds, after a caller dials a digit and before Connection SRSV acts on the digits that it has received. Default setting: 300 milliseconds.
Recording DTMF Clip ____ Milliseconds	Enter the amount of time, in milliseconds, to truncate at the end of a recording when a message is terminated with a DTMF touchtone. Default setting: 170 milliseconds.
Recording Tone Extra Clip ____ Milliseconds	Enter the amount of time, in milliseconds, to truncate at the end of a recording when a message is terminated by the caller hanging up, which may cause the phone system to provide a tone (such as a reorder tone). Default setting: 250 milliseconds.
Enable Audio Normalization	Check this check box so that Connection SRSV automatically adjusts the recording volume of voice messages and user greetings to match the setting of the Target Decibel Level for Recordings and Messages field on the System Settings > General Configuration page. Default setting: Check box checked.
Enable Noise Reduction	Check this check box so that Connection SRSV enables a noise-reduction filter to improve audio quality and voice-recognition accuracy in noisy environments. The filter is applied to all voice utterances entered by users of the voice-recognition conversation and to all audio that is recorded by Connection SRSV on calls to voice messaging ports in the port group. Uncheck the check box to disable the filter for all utterances and recorded audio on calls to voice messaging ports in the port group. Default setting: Check box checked.

Edit Codec Advertising

Table 15-8 Edit Codec Advertising Page

Field	Description
Advertised Codecs	Move to this list the codecs (audio formats) that Cisco Unity Connection SRSV advertises that it can use when dialing out. The phone system must transcode if it uses different codecs.
Unadvertised Codecs	Move to this list the codecs (audio formats) that Cisco Unity Connection SRSV does not advertise when dialing out.

Search Ports

The Search Ports page displays the status with the total number of ports.

The search results, by default, return all ports. By default, the administrator can view 25 records per page and can select rows per page from the dropdown list, the max limit being 250 search results returned on one page. The administrator can perform custom search on the port's display name field using the following options:

- Begins with
- Contains.
- Ends with
- Is Exactly
- Is Empty
- Is Not Empty

Table 15-9 Search Ports Page

Field	Description
Delete Selected	To delete a port, check the check box to the left of the display name, and select Delete Selected. You can delete multiple ports at once.
Add New	To add a port, select the Add New button. A new page opens, on which you enter data applicable to the new port.
Display Name	*(Display only)* The descriptive name for the voice messaging port. The name is created from the port group display name followed by a hyphen and sequence number of the voice messaging port.
Phone System Display Name	*(Display only)* The phone system that uses the port. Select this name to view and edit the phone system settings.
Extension	*(Display only)* The extension of the voice messaging port, if applicable.
Server	*(Display only)* The Cisco Unity Connection SRSV server (when a Connection cluster is configured) that handles this port.
Enabled	*(Display only)* When the column has an X, the port is enabled during normal operation.
Answer Calls	*(Display only)* When the column has an X, the port is designated for answering calls.
Message Notification	*(Display only)* When the column has an X, the port is designated for notifying users of messages.
Dialout MWI	*(Display only)* When the column has an X, the port is designated for turning MWIs on and off.
TRAP Connection	*(Display only)* When the column has an X, the port enables users to use the phone as a recording and playback device in Cisco Unity Connection web applications and email clients. Typically, TRAP Connection is assigned to the least busy ports.
Security Mode	*(Display only)* Indicates whether Cisco Unified Communications Manager authentication or encryption is enabled.

New Port

Table 15-10 New Port Page

Field	Description
Enabled	Check this check box so that the port is enabled during normal operation. Uncheck this check box to disable the port. When the port is disabled, calls to the port get a ringing tone but are not answered. Typically, the port is disabled only by the installer during testing. Default setting: Check box checked.
Number of Ports	Enter the number of voice messaging ports that you want to add. Default setting: 1.
Phone System	Select the phone system that the voice messaging port uses.
Port Group	Select the port group that the voice messaging ports belong to.
Server	(Connection SRSV server that handles the voice messaging port.
Answer Calls	Check this check box so that the port is designated for answering calls. These calls can be incoming calls from unidentified callers or from users. Uncheck this check box so that the port does not answer calls. Default setting: Check box checked.
Allow TRAP Connections	Check this check box so that users can use the phone as a recording and playback device in Cisco Unity Connection web applications and email clients. Assign Allow TRAP Connections to the least busy ports. Default setting: Check box checked.
Security Mode	*(Cisco Unified CM SCCP integrations only)* Select the Cisco Unified Communications Manager security mode that you want to use for the voice messaging port. Default setting: Non-secure.

Port Basics

Table 15-11 Port Basics Page

Field	Description
Enabled	Check this check box so that the port is enabled during normal operation. Uncheck this check box to disable the port. When the port is disabled, calls to the port get a ringing tone but are not answered. Typically, the port is disabled only by the installer during testing. Default setting: Check box checked.
Port Name	*(Display only)* The descriptive name for the voice messaging port. The name is created from the port group display name followed by a hyphen and sequence number of the voice messaging port.

Table 15-11 Port Basics Page (continued)

Field	Description
Restart	Select this button to restart the voice messaging port. **Caution** Restarting a voice messaging port terminates any call that is in progress on that port. In Cisco Unity Connection SRSV Serviceability, you can stop a port from taking new incoming calls on the Tools > Cluster Management page.
Phone System	*(Display only)* The display name for the phone system that uses the voice messaging port.
Port Group	*(Display only)* The port group that the voice messaging ports belong to.
Extension	(Enter the extension that the phone system uses to connect to the port.
Server	*(Cisco Unified CM SCCP and SIP integrations only)* Select the name of the Connection SRSV server that handles the voice messaging port.
Answer Calls	Check this check box so that the port is designated for answering calls. These calls can be incoming calls from unidentified callers or from users. Uncheck this check box so that the port does not answer calls. Default setting: Check box checked.
Allow TRAP Connections	Check this check box so that users can use the phone as a recording and playback device in Cisco Unity Connection web applications and email clients. Assign Allow TRAP Connections to the least busy ports. Default setting: Check box checked.
Outgoing Hunt Order	Enter the order of priority that the port is used for outgoing calls, if applicable. When available ports have the same hunt order number, Cisco Unity Connection uses the port that has been idle the longest.
Security Mode	*(Cisco Unified CM SCCP integrations only)* Select the Cisco Unified Communications Manager security mode that you want to use for the voice messaging port. Default setting: Non-secure.
SCCP (Skinny) Device Name	*(Display only)* The device name that Cisco Unified Communications Manager assigned to the voice messaging port. This device name may be helpful for troubleshooting.
View Certificate	*(Cisco Unified CM SCCP integrations only)* Select this button to view the device certificate data for the voice messaging port.

View Port Certificate

Table 15-12 View Port Certificate Page

Field	Description
Subject	*(Display only)* The content of the Subject field for the port certificate.
Issuer	*(Display only)* The content of the Issuer field for the port certificate.
Valid From	*(Display only)* The date and time of beginning validity for the port certificate.
Valid Until	*(Display only)* The date and time of ending validity for the port certificate.

Table 15-12 View Port Certificate Page (continued)

Field	Description
Version	*(Display only)* The version of the port certificate.
Serial Number	*(Display only)* The serial number of the port certificate.
Certificate Text	*(Display only)* The text content of the port certificate.
Private Key	*(Display only)* The encrypted private key of the port certificate.
Generate New	Select this button to generate a new port certificates for all voice messaging ports.

View Root Certificate

Table 15-13 View Root Certificate Page

Field	Description
Subject	*(Display only)* The content of the Subject field for the root certificate.
Issuer	*(Display only)* The content of the Issuer field for the root certificate.
Valid From	*(Display only)* The date and time of beginning validity for the root certificate.
Valid Until	*(Display only)* The date and time of ending validity for the root certificate.
Version	*(Display only)* The version of the root certificate.
File Name	*(Display only)* The file name of the root certificate.
Serial Number	*(Display only)* The serial number of the root certificate.
Certificate Text	*(Display only)* The text content of the root certificate.
Private Key	*(Display only)* The encrypted private key of the root certificate.
Right-Click to Save the Certificate as a File	Right-click this link and select Save Target As so that you can save the root certificate as a file at the location that you indicate. Note that the file name must match the name indicated and that the extension must be 0 rather than htm.
Generate New	Select this button to generate a new root certificate and new port certificates for all voice messaging ports.

Search SIP Certificates

The Search SIP Certificates page displays the status with the total number of SIP certificates.

The search results, by default, return all SIP certificates. By default, the administrator can view 25 records per page and can select rows per page from the dropdown list, the max limit being 250 search results returned on one page. The administrator can perform custom search on the certificate display name field using the following options:

- Begins with
- Contains.
- Ends with
- Is Exactly

- Is Empty
- Is Not Empty

Table 15-14 Search SIP Certificates Page

Field	Description
Delete Selected	To delete a SIP certificate, check the check box to the left of the display name, and select Delete Selected. You can delete multiple SIP certificates at once.
Add New	To add an SIP certificate, select the Add New button. A new page opens, on which you enter data applicable to the new SIP certificate.
Display Name	*(Display only)* The name of the SIP certificate.
Subject Name	*(Display only)* The subject name that matches the subject name of the SIP certificate for the SIP trunk in Cisco Unified CM Administration.

New SIP Certificate

Table 15-15 New SIP Certificate Page

Field	Description
Display Name	Enter a descriptive name for the SIP certificate.
Subject Name	Enter a subject name that matches the subject name of the SIP certificate for the SIP trunk in Cisco Unified CM Administration. **Caution** This subject name must match the subject name of the SIP certificate used by Cisco Unified Communications Manager. Otherwise, Cisco Unified CM authentication and encryption fail.

Edit SIP Certificate

Table 15-16 Edit SIP Certificate Page

Field	Description
Display Name	Enter a descriptive name for the SIP certificate.
Subject Name	Enter a subject name that matches the subject name of the SIP certificate for the SIP trunk in Cisco Unified CM Administration. **Caution** This subject name must match the subject name of the SIP certificate used by Cisco Unified Communications Manager. Otherwise, Cisco Unified CM authentication and encryption fail.

Table 15-16 Edit SIP Certificate Page (continued)

Field	Description
Subject	*(Display only)* The content of the Subject field for the SIP certificate.
Issuer	*(Display only)* The content of the Issuer field for the SIP certificate.
Valid From	*(Display only)* The date and time of beginning validity for the SIP certificate.
Valid Until	*(Display only)* The date and time of ending validity for the SIP certificate.
Version	*(Display only)* The version of the SIP certificate.
Serial Number	*(Display only)* The serial number of the SIP certificate.
Certificate Text	*(Display only)* The text content of the SIP certificate.
Private Key	*(Display only)* The encrypted private key of the SIP certificate.
Generate New	Select this button to generate a new SIP certificate.

Search SIP Security Profiles

The Search Security Profiles page displays the status with the total number of security profiles.

The search results, by default, return all security profiles. By default, the administrator can view 25 records per page and can select rows per page from the dropdown list, the max limit being 250 search results returned on one page. The administrator can perform custom search on the security profile's display name field using the following options:

- Begins with
- Contains.
- Ends with
- Is Exactly
- Is Empty
- Is Not Empty

Table 15-17 Search SIP Security Profiles Page

Field	Description
Delete Selected	To delete a SIP security profile, check the check box to the left of the display name, and select Delete Selected. You can delete multiple SIP security profiles at once.
Add New	To add a SIP security profile, select the Add New button. A new page opens, on which you enter data applicable to the new SIP security profile.
Display Name	*(Display only)* The name of the SIP security profile.

New SIP Security Profile

Table 15-18 New SIP Security Profile Page

Field	Description
Port	The port that the Cisco Unified Communications Manager server uses for SIP trunk authentication and encryption of the voice messaging ports. **Note** You cannot use the same port for both TLS and non-TLS SIP security.
Do TLS	Check this check box so that call signaling messages are encrypted when sent through the SIP trunk between Cisco Unity Connection and the Cisco Unified Communications Manager server. Uncheck this check box so that call signaling messages are not encrypted when sent through the SIP trunk between Connection and the Cisco Unified CM server. **Caution** When this check box is checked, the Cisco Unified CM server must also enable TLS. Otherwise, SIP security does not function correctly.

Edit SIP Security Profile

Table 15-19 Edit SIP Security Profile Page

Field	Description
Port	The port that the Cisco Unified Communications Manager server uses for SIP trunk authentication and encryption of the voice messaging ports. **Note** You cannot use the same port for both TLS and non-TLS SIP security.
Do TLS	Check this check box so that call signaling messages are encrypted when sent through the SIP trunk between Cisco Unity Connection and the Cisco Unified Communications Manager server. Uncheck this check box so that call signaling messages are not encrypted when sent through the SIP trunk between Connection and the Cisco Unified CM server. **Caution** When this check box is checked, the Cisco Unified CM server must also enable TLS. Otherwise, SIP security does not function correctly.

CHAPTER 16

Cisco Unity Connection SRSV Tool Settings

See the following sections:

Search Custom Keypad Mappings

The Search Custom Keypad Mappings page displays the status with the total number of custom keypad mappings.

The search results, by default, return all keypad mappings. By default, the administrator can view 25 records per page and can select rows per page from the dropdown list, the max limit being 250 search results returned on one page. The administrator can perform custom search on the conversation name field using the following options:

- Begins with
- Contains.
- Ends with
- Is Exactly
- Is Empty
- Is Not Empty

Table 16-1 Search Custom Keypad Mappings Page

Field	Description
Conversation Name	Select the applicable custom keypad mapping conversation from the available conversations.

Edit Custom Keypad Mapping

Table 16-2 Edit Custom Keypad Mapping Page

Field	Description
Menu Tabs	Select the applicable menu tab to customize the conversation for that menu. There are eight conversation menus that can be customized: • Main menu • Message Header • Message Body • Message Footer • After Message menu
Option	The list of options that can be used for the selected menu. For a detailed description of each option, see the "Custom Keypad Mapping Tool in Cisco Unity Connection 9.x" chapter of the *System Administration Guide for Cisco Unity Connection Release 9.x*, available at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/administration/guide/9xcucsagx.html.
Key Assignment	The key or keys that are assigned to each menu option. Note the following guidelines: • The only characters allowed are: 0 – 9, *, # or blank. • A maximum of 3 digits is allowed for each menu option. • Duplicate key entries are not allowed for any unique menu. (For example, you cannot map the "1" key to both Hear New Messages and Send a Message in the Main menu. However, you can map the "1" key to Hear New Messages in the Main menu and also to Greetings in the Settings menu.) • Leaving a key assignment blank disables that option for the menu. • When changes are saved, all new calls that use that conversation follow the new key mapping settings. • When you leave a key assignment black, uncheck the Option Voiced in Menu check box.
Option Voiced in Menu	Check or uncheck the check box to indicate whether the option is voiced in the menu. You can use this setting to assign a key or keys to an option but not have it presented in the menu. The option would still be enabled and Cisco Unity Connection would respond appropriately if the assigned key is pressed, but the user would not hear the option in the menu. For example, your users may know that "0" is always mapped to Help and "*" is always mapped to Cancel, but in an effort to keep menus as short as possible, you may not want to have those options voiced in every menu.
Move To	Select a menu option and use the Move To button or the up and down arrows to change the order in which options are voiced in the menu.
Action Description	*(Display only)* Describes the action that is taken for the selected option.

Using the Custom Keypad Mapping Tool in Cisco Unity Connection SRSV

The Custom Keypad Mapping tool is divided into eight tabs that represent eight different conversation menus that can be customized. On each of these menu tabs you can:

- Customize which key or keys are assigned to each menu option. Leaving a key assignment blank disables that option for the menu.
- Configure whether the option is voiced in the menu. This allows you to assign a key or keys to an option but not have it presented verbally in the menu. The option would still be enabled for that menu and Connection would respond appropriately if the assigned key is pressed, but the user would not hear the option in the menu.
- Configure the order in which the menu items are offered to users. This is done by selecting the radio button of the row that you want to reorder and then using either the Up or Down arrows or the Move To button to arrange the menu items. The order in which the options appear in the tool is the order in which they are presented to the user by phone regardless of which keys are mapped to the options

To Use the Custom Keypad Mapping Tool to Make Changes to a Custom Keypad Map

Step 1 In Cisco Unity Connection SRSV Administration, expand **Tools**, then select **Custom Keypad Mapping**.

Step 2 On the Search Custom Keypad Mappings page, select the applicable custom keypad mapping conversation.

Step 3 On the Edit Custom Keypad Mapping page, select the applicable tab to select the menu for which you would like to change key assignments.

Step 4 Change key assignments as applicable. (For guidelines on allowed entries, see the Guidelines for Assigning Keys to Menu Options, page 16-3.)

Step 5 Select **Save**.

When changes are saved, all new calls that use this conversation follow the new key mapping settings.

Step 6 Repeat Step 3 through Step 5 for each menu that you want to customize.

Guidelines for Assigning Keys to Menu Options

- The only characters allowed are: 0 – 9, *, # or blank.
- A maximum of 3 digits is allowed for each menu option.
- Duplicate key entries are not allowed for any unique menu. (For example, you cannot map the “1” key to both Hear New Messages and Send a Message in the Main menu. However, you can map the “1” key to Hear New Messages in the Main menu and also to Greetings in the Settings menu.)
- Leaving a key assignment blank disables that option for the menu.
- When you leave a key assignment black, uncheck the Option Voiced in Menu check box.
- When changes are saved, all new calls that use the conversation follow the new key mapping settings.

Conversation Menus That Can Be Customized in Cisco Unity Connection SRSV

The Custom Keypad Mapping tool is divided into eight tabs that represent eight different conversation menus that can be customized. The Message Playback menu is represented on three tabs because messages contain three distinct parts: the message header, the message body, and the message footer. The options on these three tabs are identical, but you may want to map different options to different keys for certain parts.

The following menus can be customized:

- Main Menu Tab, page 16-4
- Message Playback Menu Tabs (Message Header Tab, Message Body Tab, and Message Footer Tab), page 16-4
- After Message Menu Tab, page 16-7

Main Menu Tab

The Main menu is what users hear immediately after they sign in and hear their message counts (if applicable).

See Table 16-3 for a list of options that can be mapped.

Table 16-3 Main Menu Tab

Option	Description
Play New Messages	Takes users to the new (unread) message stack.
Review Old Messages	Takes users to the saved message stack. If applicable, users are also offered an opportunity to review deleted messages.
Cancel or Back Up	Exits the user mailbox. By default, when users exit their mailboxes they are sent to the Opening Greeting call handler. However, you can customize the exit behavior by changing the When Exiting the Conversation setting on the Phone Menu page for each user.
Help	Plays the Main menu Help.
Repeat Menu Options	Plays the Main menu again.
Call a Number	Allows users to access the User System Transfer conversation and dial any number that is allowed by their transfer restriction table.

Message Playback Menu Tabs (Message Header Tab, Message Body Tab, and Message Footer Tab)

When a message is played in the Cisco Unity Connection user conversation, there are three separate parts: the header, the body, and the footer. By default, the message header contains the message number and the sender information. The message body is the actual recording of the message. The message footer is the time stamp.

The contents of the header and footer sections can be modified on the Playback Message Settings page. For example, the message number, the sender information, the sender extension, and the time stamp can be added or removed from the header. These settings are controlled by the check boxes under the "Before Playing Each Message, Play" section on the Playback Message Settings page. For the message footer, you have the option of playing the time stamp after the message; you can exclude it altogether or have it played as part of the header. This option is controlled with the check box under the "After Playing Each Message, Play" section on the Playback Message Settings page. If you choose not to play the time stamp after the message, the effect is to have no footer to the message. In Cisco Unity Connection, the "After Playing Each Message, Play" section now includes the sender information, extension or ANI, and the message number, in addition to the time the message was sent and message duration.

The Custom Keypad Mapping tool includes separate tabs for each part of the message. As a best practice, we recommend that you map the same keys to each option for all three parts. However, in some cases it may be useful to map the same key to different actions. For example, during the message header you might want to press the "1" key to skip to the start of the message body, and during the message body press the "1" key to skip to the message footer.

The same message playback key mappings are used when listening to new messages, saved messages, and deleted messages, rather than separate mappings for each message stack. Keep this in mind as you are deciding on key mapping preferences, particularly for options such as marking messages as new (unread) or saved (read).

Message playback options are not voiced in a menu format by phone, but they are listed if the user presses the key that is mapped to the Help option. The Custom Keypad Mapping tool allows you to configure which items are voiced in the Help.

See Table 16-4 for a list of options that can be mapped.

Table 16-4 Message Playback Menu Tabs

Option	Description
Repeat Message	Jumps to the beginning of the header portion of the message.
Save	Skips to the next message and marks the current message as saved.
Delete	Deletes the message that is currently being played. The user class of service determines whether the message is moved to the deleted items folder or is deleted permanently.
Slow Playback	Slows down the message that is currently being played. Pressing the mapped key slows the message playback by 50 percent. **Note** If the Save Speed and Volume Changes Made by User setting is enabled on the System Settings > Advanced > Conversation Configuration page, the last change made to playback speed is saved as the default playback speed for the user.
Change Volume	Cycles the volume of the message that is currently being played through three volume levels: normal, loud, and quiet.If the Save Speed and Volume Changes Made by User setting is enabled on the System Settings > Advanced > Conversation Configuration page, the last change made to playback volume is saved as the default playback volume for the user.

Table 16-4 Message Playback Menu Tabs (continued)

Option	Description
Fast Playback	Speeds up the message that is currently being played. Pressing the mapped key speeds the message playback by 50 percent. Pressing the key again speeds the message playback by 100 percent. **Note** If the Save Speed and Volume Changes Made by User setting is enabled on the System Settings > Advanced > Conversation Configuration page, the last change made to playback speed is saved as the default playback speed for the user.
Rewind	Jumps backward in the message that is currently being played. By default, the message rewinds five seconds. You can adjust the rewind time on the Playback Message Settings page.
Pause/Resume	Pauses playback of the message, or resumes playback when the message is already paused.
Fast-Forward	Jumps forward in the message that is currently being played. By default, the message fast-forwards five seconds. You can adjust the fast-forward time on the Playback Message Settings page.
Skip to After Message Menu	Jumps directly to the After Message menu.
Skip Message, Save As Is	Skips to the next message in the stack and leaves the message in the state it was in. When a new message is skipped, it is saved as unread; when a saved message is skipped, it remains saved; and when a deleted message is skipped, it remains deleted.
Play Message By Number	Asks the user to enter the number of a message in the current stack (new, saved, or deleted messages) and then takes the user directly to that message. For users who have large numbers of messages, this is a useful way to jump ahead or back in the stacks. This option is offered only when the Enable Go to Message setting is enabled on the System Settings > Advanced > Conversation page.
Go to Previous Message	Takes the user to the previous message in the stack.
Go to Next Message	Takes the user to the next message in the stack. The message the user was listening to is left in the state it was in (new, saved, or deleted). Go to Next Message functions the same as the Skip Message, Save As Is option.
Cancel or Back Up	Terminates message playback and goes up a menu level. Users who are listening to new or saved messages go to the Main menu. Users who are listening to deleted messages go to the Deleted Message Option menu.
Operator	Signs users out of their mailboxes and sends them to the Operator call handler. The message is left in the state that it was in.
Play Message Properties	Plays the properties of the message that is currently being played. This includes the sender information (including ANI if it is provided for outside callers) and the time that the message was sent.
Go to First Message	Jumps to the first message of the message stack. Connection plays the “First message” prompt as an audible cue to the user.
Help	Plays Help for all of the options that are mapped to a key, and for which the Option Voiced in Help check box is checked.

Table 16-4 Message Playback Menu Tabs (continued)

Option	Description
Go to Last Message	Jumps to the last message of the message stack. Connection plays the “Last message” prompt as an audible cue to the user.
List Message Recipients	Lists all recipients of the current message.
Reset Volume to Default	Resets the volume of the message that is currently being played to the default message playback volume setting for the user. **Note** If the Save Speed and Volume Changes Made by User setting is enabled on the System Settings > Advanced > Conversation Configuration page, the last change made to playback volume is saved as the default playback volume for the user.
Save as New	Skips to the next message in the stack and marks the message as new. When this option is selected, if a user skips messages when listening to saved or deleted messages, the messages are marked as unread and are moved to the new message stack.
Louder Playback	Increases the volume of the message that is currently being played. **Note** If the Save Speed and Volume Changes Made by User setting is enabled on the System Settings > Advanced > Conversation Configuration page, the last change made to playback volume is saved as the default playback volume for the user.
Reset Speed to Default	Resets the speed of the message that is currently being played to the default message playback speed setting for the user. **Note** If the Save Speed and Volume Changes Made by User setting is enabled on the System Settings > Advanced > Conversation Configuration page, the last change made to playback speed is saved as the default playback speed for the user.
Quieter Playback	Decreases the volume of the message that is currently being played. **Note** If the Save Speed and Volume Changes Made by User setting is enabled on the System Settings > Advanced > Conversation Configuration page, the last change made to playback volume is saved as the default playback volume for the user.
Skip to End	Jumps to the beginning of the message footer. When the After Playing Each Message, Play options are not enabled for the user on the Playback Message Settings page, these options effectively skips to the end of the message and goes directly to the After Message menu.
Forward Message	Allows the user to forward the message to another user or distribution list.
Replay Message	Jumps to the beginning of the message body, effectively repeating the message. If you assign a key to this option for the message header, it allows users to skip the header and jump right to the message.

After Message Menu Tab

The After Message menu plays after the user has listened to a message.

See Table 16-5 for a list of options that can be mapped.

Table 16-5 After Message Menu Tab

Option	Description
Repeat Message	Plays the message again, starting with the header.
Save	Marks the message as saved (read) and moves to the next message in the stack. When the user is listening to a deleted message, this option moves the message to the saved message stack.
Delete	Deletes the message that is currently being played. The user class of service determines whether the message is moved to the deleted items folder or is deleted permanently.
Reply	Replies to the sender of the message. Only the sender receives the reply; other recipients of the original message do not receive the reply. This option is available only when the message is from another user; users cannot reply to outside caller messages.
Save as New	Marks the message as new (unread) and moves to the next message in the stack. When the user is listening to a saved or deleted message, this option moves the message to the new message stack.
Rewind	Jumps backward into the message. By default, the message rewinds five seconds. You can adjust the rewind time on the Playback Message Settings page.
Play Message Properties	Plays the properties of the current message. This includes the sender information (including ANI if it is provided for outside callers) and the time that the message was sent.
Cancel or Back Up	Exits the After Message menu and goes up a menu level. Users who are listening to new or saved messages go to the Main menu. Users who are listening to deleted messages go to the Deleted Message Option menu.
Help	Plays the After Message menu Help.
Play Message By Number	Asks the user to enter the number of a message in the current stack (new, saved, or deleted messages) and then takes the user directly to that message. For users who have large numbers of messages, this is a useful way to jump ahead or back in the stacks. This option is available only when the Enable Go to Message setting is enabled on the System Settings > Advanced > Conversations page.
List Message Recipients	Lists all recipients of the current message.
Go to Previous Message	Takes the user to the previous message in the stack.
Go to Next Message	Takes the user to the next message in the stack. The message the user was listening to is left in the state that it was in (new, saved, or deleted). Go to Next Message functions the same as the Skip Message, Save As Is option.
Save As Is	Goes to the next message in the stack and leaves the message in the state that it was in. New messages are saved as unread; saved messages remain saved; and deleted messages remain deleted.
Go to First Message	Jumps to the first message of the message stack. Connection plays the “First message” prompt as an audible cue to the user.

Table 16-5 After Message Menu Tab (continued)

Option	Description
Go to Last Message	Jumps to the last message of the message stack. Connection plays the “Last message” prompt as an audible cue to the user.
Operator	Signs users out of their mailboxes and sends them to the operator call handler. The message is left in the state it was in.
Skip Message, Save As Is	Skips to the next message in the stack and leaves the message in the state it was in. When a new message is skipped, it is saved as unread; when a saved message is skipped, it remains saved; and when a deleted message is skipped, it remains deleted.

CHAPTER 17

Securing Connections in Cisco Unity Connection Survivable Remote Site Voicemail 9.1(1)

This chapter contains information on how to secure communication between central Connection and Connection SRSV. In addition, it explains about securing communication between Cisco Unity Connection SRSV Administration and Connection SRSV.

See the following sections:

Using Self-Signed Certificate Based Access

You can use the self-signed certificate based access for communication between central Connection server and Connection SRSV. By default, central Connection server and Connection SRSV does not accept self-signed certificates. To accepts self-signed certificates on the central Connection server and Connection SRSV, you need to perform the following steps on command prompt using administrator credentials:

Step 1 Run the following command:

```
run cuc dbquery unitydirdb EXECUTE PROCEDURE
csp_ConfigurationModify(pFullName='System.SRSV.AcceptSrsvSelfSignedCertificates',
pValue='1')
```

Step 2 Run the following command to confirm that the value of “System.SRSV.AcceptSrsvSelfSignedCertificates” field is set to 1:

```
run cuc dbquery unitydirdb select objectid,fullname,value from vw_configuration
where fullname like '%SRSV%'
```

After changing the value of System.SRSV.AcceptSrsvSelfSignedCertificates to 1, you need to restart the Connection Branch Sync Service and Tomcat Service to reflect the changes and allow the self-signed certificate access.

To Restart the Tomcat service please follow following steps:

Step 1 Sign in to the Connection server by using an SSH application.

Step 2 Run the following CLI command to restart the Tomcat service:

```
utils service restart Cisco Tomcat
```

Securing Connections between Central Connection server and Connection SRSV

Connection SRSV uses both Secure Sockets Layer (SSL) certificate and shared secrets to secure communication between the central Connection and the branch.

1. **Installing SSL Certificate**

When you install Cisco Unity Connection SRSV, a local certificate is automatically created and installed to secure communication between Connection SRSV and Connection. This means that all network traffic (including usernames, passwords, other text data, and voice messages) between Connection SRSV and Connection is automatically encrypted. For more information on installing SSL certificate, refer to the "Using SSL to Secure Client/Server Connections in Cisco Unity Connection 9.x" chapter of the *Security Guide for Cisco Unity Connection Release 9.x* at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/security/guide/9xcucsecx.html.

2. **Using Shared Secrets**

Connection SRSV uses shared secrets to authenticate Connection access. For more information on shared secrets, refer to the "Cisco Unity Connection SRSV Passwords and Shared Secrets" section of the "Passwords, PINs, and Authentication Rule Management in Cisco Unity Connection 9.x" chapter of the *Security Guide for Cisco Unity Connection Release 9.x* at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/security/guide/9xcucsecx.html.

Securing Connections between Connection SRSV Administration and Connection SRSV

Do the following tasks to create and install an SSL server certificate to secure Connection SRSV Administration access to Connection SRSV:

1. If you are using Microsoft Certificate Services to issue certificates, install Microsoft Certificate Services. For information on installing Microsoft Certificate Services on a server running Windows Server 2003, see the "Installing Microsoft Certificate Services (Windows Server 2003 Only)" section on page 17-5. For information on installing Microsoft Certificate Services on a server running a later version of Windows Server, refer to Microsoft documentation.

   If you are using another application to issue certificates, install the application. See the manufacturer documentation for installation instructions. Then skip to Task 2.

   If you are using an external certification authority to issue certificates, skip to Task 2.

Note If you already have installed Microsoft Certificate Services or another application that can create certificate signing requests, skip to Task 2.

2. Create a certificate signing request. Then download the certificate signing request to the server on which you installed Microsoft Certificate Services or another application that issues certificates, or download the request to a server that you can use to send the certificate signing request to an external CA. Do the "To Create and Download a Certificate Signing Request" procedure on page 17-3.
3. If you are using Microsoft Certificate Services to export the root certificate and to issue the server certificate, do the procedure in the "Exporting the Root Certificate and Issuing the Server Certificate (Microsoft Certificate Services Only)" section on page 17-6.

   If you are using another application to issue the certificate, see the documentation for the application for information on issuing certificates.

   If you are using an external CA to issue the certificate, send the certificate signing request to the external CA. When the external CA returns the certificate, continue with Task 4.

   Only PEM-formatted (also known as Base-64 encoded DER) certificates can be uploaded to Connection SRSV. The certificate must have a .pem filename extension. If the certificate is not in this format, you can usually convert what you have to PEM format by using freely available utilities like OpenSSL.
4. Upload the root certificate and the server certificate to the Connection SRSV server. Do the "To Upload the Root and Server Certificates to the Cisco Unity Connection SRSV Server" procedure on page 17-4.
5. To prevent users from seeing a security alert whenever they access Connection SRSV by using the Connection SRSV Administration or Connection, do the following tasks on all computers from which users will access Connection SRSV:
   - Import the server certificate that you uploaded to the Connection SRSV server in Task 4. into the certificate store. The procedure differs based on the browser. For more information, see the documentation for the browser.
   - Import the server certificate that you uploaded to the Connection SRSV server in Task 4. into the Java store. The procedure differs based on the operating system running on the client computer. For more information, see the operating system documentation and the Java Runtime Environment documentation.

To Create and Download a Certificate Signing Request

Step 1 On the Cisco Unity Connection SRSV server, sign in to Cisco Unified Operating System Administration.

Step 2 On the Security menu, select **Certificate Management**.

Step 3 On the Certificate List page, select **Generate CSR**.

Step 4 On the Generate Certificate Signing Request page, in the **Certificate Name** list, select **tomcat**.

Step 5 Select **Generate CSR**.

Step 6 When the Status area displays a message that the CSR was successfully generated, select **Close**.

Step 7 On the Certificate List page, select **Download CSR**.

Step 8 On the Download Certificate Signing Request page, in the **Certificate Name** list, select **tomcat**.

Step 9 Select **Download CSR**.

Step 10 In the File Download dialog box, select **Save**.

Step 11 In the Save As dialog box, in the **Save As Type** list, select **All Files**.

Step 12 Save the file **tomcat.csr** to a location on the server on which you installed Microsoft Certificate Services or on a server that you can use to send the CSR to an external certification authority.

Step 13 On the Download Certificate Signing Request page, select **Close**.

To Upload the Root and Server Certificates to the Cisco Unity Connection SRSV Server

Step 1 On the Cisco Unity Connection SRSV server on which you created the certificate signing request, sign in to Cisco Unified Operating System Administration.

Step 2 On the Security menu, select **Certificate Management**.

> **Note** If you select **Find** and display a list of the certificates currently installed on the server, you will see an existing, automatically generated, self-signed certificate for Tomcat. That certificate is unrelated to the Tomcat certificates that you upload in this procedure.

Step 3 Upload the root certificate:

a. On the Certificate List page, select **Upload Certificate**.

b. On the Upload Certificate page, in the Certificate Name list, select **tomcat-trust**.

c. Leave the Root Certificate field blank.

d. Select **Browse**, and browse to the location of the root CA certificate.

   If you used Microsoft Certificate Services to issue the certificate, this is the location of the root certificate that you exported in the "To Export the Root Certificate and to Issue the Server Certificate" procedure on page 17-6.

   If you used an external certification authority to issue the certificate, this is the location of the root CA certificate that you received from the external certification authority.

e. Select the name of the file.

f. Select **Open**.

g. On the Upload Certificate page, select **Upload File**.

h. When the Status area reports that the upload succeeded, select **Close**.

Step 4 Upload the server certificate:

a. On the Certificate List page, select **Upload Certificate**.

b. On the Upload Certificate page, in the Certificate Name list, select **tomcat**.

c. In the Root Certificate field, enter the filename of the root certificate that you uploaded in Step 3.

d. Select **Browse**, and browse to the location of the server certificate.

   If you used Microsoft Certificate Services to issue the certificate, this is the location of the server certificate that you issued in the "To Export the Root Certificate and to Issue the Server Certificate" procedure on page 17-6.

   If you used an external certification authority to issue the certificate, this is the location of the server certificate that you received from the external certification authority.

e. Select the name of the file.

f. Select **Open**.

g. On the Upload Certificate page, select **Upload File**.

h. When the Status area reports that the upload succeeded, select **Close**.

Step 5 Restart the Tomcat service (the service cannot be restarted from Cisco Unified Serviceability):

a. Sign in to the Cisco Unity Connection SRSV server by using an SSH application.

b. Run the following CLI command to restart the Tomcat service:

```
utils service restart Cisco Tomcat
```

To Restart the Connection Branch Sync Service

Step 1 Sign in to Cisco Unity Connection Serviceability.

Step 2 On the Tools menu, select **Service Management**.

Step 3 In the Optional Services section, for the Connection Branch Sync Service, select **Stop**.

Step 4 When the Status area displays a message that the Connection IMAP Server service was successfully stopped, select **Start** for the service.

Installing Microsoft Certificate Services (Windows Server 2003 Only)

If you want to use a third-party certificate authority to issue SSL certificates, or if Microsoft Certificate Services is already installed, skip this section.

Do the procedure in this section if you want to use Microsoft Certificate Services to issue your own certificate and if you want to install the application on a server running Windows Server 2003.

If you want to install a root certification authority (the generic term for Microsoft Certificate Services) on a Windows Server 2008 server, refer to the Windows Server 2008 online help.

To Install the Microsoft Certificate Services Component

Step 1 On any server whose DNS name (FQDN) or IP address can be resolved by all client computers that will access Connection SRSV voice messages, sign in to Windows by using an account that is a member of the local Administrators group.

Step 2 On the Windows Start menu, select **Settings > Control Panel > Add or Remove Programs**.

Step 3 In the left pane of the Add or Remove Programs control panel, select **Add/Remove Windows Components**.

Step 4 In the Windows Components dialog box, check the **Certificate Services** check box. Do not change any other items.

Step 5 When the warning appears about not being able to rename the computer or to change domain membership, select **Yes**.

Step 6 Select **Next**.

Step 7 On the CA Type page, select **Stand-alone Root CA**, and select **Next**. (A stand-alone certification authority (CA) is a CA that does not require Active Directory.)

Step 8 On the CA Identifying Information page, in the Common Name for This CA field, enter a name for the certification authority.

Step 9 Accept the default value in the Distinguished Name Suffix field.

Step 10 For Validity Period, accept the default value of **5 Years**.

Step 11 Select **Next**.

Step 12 On the Certificate Database Settings page, select **Next** to accept the default values.

If a message appears indicating that Internet Information Services is running on the computer and must be stopped before proceeding, select **Yes** to stop the services.

Step 13 If you are prompted to insert the Windows Server 2003 disc into the drive, do so.

Step 14 In the Completing the Windows Components Wizard dialog box, select **Finish**.

Step 15 Close the Add or Remove Programs dialog box.

Exporting the Root Certificate and Issuing the Server Certificate (Microsoft Certificate Services Only)

Do the following procedure only when you are using Microsoft Certificate Services to issue the certificate.

To Export the Root Certificate and to Issue the Server Certificate

Step 1 On the server on which you installed Microsoft Certificate Services, sign in to Windows by using an account that is a member of the Domain Admins group.

Step 2 On the Windows Start menu, select **Programs > Administrative Tools > Certification Authority**.

Step 3 In the left pane, expand **Certification Authority (Local) > <Certification authority name>**, where <Certification authority name> is the name that you gave to the certification authority when you installed Microsoft Certificate Services in the "To Install the Microsoft Certificate Services Component" procedure on page 17-5.

Step 4 Export the root certificate:

a. Right-click the name of the certification authority, and select **Properties**.

b. On the General tab, select **View Certificate**.

c. Select the **Details** tab.

d. Select **Copy to File**.

e. On the Welcome to the Certificate Export Wizard page, select **Next**.

f. On the Export File Format page, select **Next** to accept the default value of **DER Encoded Binary X.509 (.CER)**.

g. On the File to Export page, enter a path and filename for the .cer file. Select a network location that you can access from the Connection server.

Write down the path and filename. You will need it in a later procedure.

h. Follow the onscreen prompts until the wizard has finished the export.

i. Select **OK** to close the Certificate dialog box, and select **OK** again to close the Properties dialog box.

Step 5 Issue the server certificate:

a. Right-click the name of the certification authority, and select **All Tasks > Submit New Request**.

b. Browse to the location of the certificate signing request file that you created in the "To Create and Download a Certificate Signing Request" procedure on page 17-3, and double-click the file.

c. In the left pane of Certification Authority, select **Pending Requests**.

d. Right-click the pending request that you submitted in b., and select **All Tasks > Issue**.

e. In the left pane of Certification Authority, select **Issued Certificates**.

f. Right-click the new certificate, and select **All Tasks > Export Binary Data**.

g. In the Export Binary Data dialog box, in the Columns that Contain Binary Data list, select **Binary Certificate**.

h. Select **Save Binary Data to a File**.

i. Select **OK**.

j. In the Save Binary Data dialog box, enter a path and filename. Select a network location that you can access from the Connection SRSV server.

Write down the path and filename. You will need it in a later procedure.

k. Select **OK**.

Step 6 Close Certification Authority.

CHAPTER 18

Securing PINs and Passwords in Cisco Unity Connection SRSV

See the following sections:

Cisco Unity Connection SRSV Passwords and Shared Secrets

All the requests initiated from the central Connection server to the Connection SRSV server use administrator credentials of Connection SRSV for communication whereas the requests from Connection SRSV to Connection use secret tokens for authentication.

The central Connection server uses the administrator username and password of Connection SRSV to authenticate access to the server. The username and password of the Connection SRSV get stored in the Connection database as you create a new branch on the central Connection server.

During each provisioning cycle with Connection SRSV, the central Connection server generates a secret token and shares the token with Connection SRSV. After the provisioning is completed from the Connection SRSV site, it notifies Connection using the same token. Then this token is removed from both Connection and Connection SRSV servers as soon as the provisioning cycle is completed. This concept of runtime token keys is known as shared secrets.

For more information on Connection SRSV, refer to the *Cisco Unity Connection Survivable Remote Site Voicemail (SRSV) Guide for Cisco Unity Connection Release 9.x* at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/quick_start/guide/9xcucqsgsrsv.html.

Changing the Cisco Unity Connection SRSV User PIN

If you want to change the PIN of a Connection SRSV user, you can do it through the Cisco Unity Connection Administration interface. After changing the PIN of the selected user, you need to provision the associated branch to update the user information in the Connection SRSV database.

Note You cannot change the PIN of an SRSV user through Cisco Unity Connection SRSV Administration interface.

CHAPTER 19

Managing Cisco Unity Connection SRSV Services

This chapter provides information on managing services in Cisco Unity Connection Serviceability for Connection SRSVand contains the following sections:

Cisco Unity Connection SRSV Services

Cisco Unity Connection has the services described in Table 19-1.

Table 19-1 Cisco Unity Connection SRSV Services

Service	Description
Status Only Services	
Connection DB	This service enables the Connection database and can be deactivated only by using the command-line interface (CLI).
Connection License Manager Server	This service manages license status of the Connection server.
Connection Server Role Manager	This service enables the server status when a Connection cluster is configured and can be deactivated only by using the command-line interface (CLI).
Connection Serviceability	This service enables the Cisco Unity Connection Serviceability Administration interface and can be deactivated only by using the command-line interface (CLI).
Critical Services	
Connection Conversation Manager	This service enables Connection to handle calls. Disabling this service will degrade the ability of Connection to function.
Connection Message Transfer Agent	This service enables the delivery of messages to the message store. Disabling this service will degrade the ability of Connection to function.

Table 19-1 Cisco Unity Connection SRSV Services (continued)

Service	Description
Connection Mixer	This service enables the audio (media stream) for calls, recorded messages, and Text to Speech (TTS). Disabling this service will degrade the ability of Connection to function.
Base Services	
Connection Administration	This service enables Cisco Unity Connection Administration and the settings that are saved in the interface.
Connection DB Event Publisher	This service enables notifying Connection components of changes to the Connection database.
Connection Exchange Notification Web Service	This service allows single inbox to receive message change notifications from Exchange Web Services-based external services.
Connection License Server	This service enables the Connection licensing by reading the installed license files, tracking the number of seats that are in use, and enabling licensed features.
Connection SNMP Agent	This service enables the Simple Network Management Protocol (SNMP), which uses the Cisco-Unity-MIB.
Connection SRSV Administration	This service enables Cisco Unity Connection SRSV Administration and the settings that are saved in the interface.
Optional Services	
Connection Branch Sync Service	This service enables Survivable Remote Site Voicemail (SRSV) feature.
Connection CM Database Event Listener	This service enables the detection of changes in the Cisco Unified Communications Manager database.
Connection Database Proxy	This service allows tools that are not installed on the Connection server (COBRAS, User Data Dump, Distribution List Builder, and so on) to gain direct access to the Connection database via ODBC from a Windows client on the network. The service is off by default. To use any of these tools, you must enable the service, configure the time out for the service, and create a user that has the remote admin role. For more information, see the help file for the applicable tool.
Connection Diagnostic Portal Service	This service enables access to data on Connection SRSV by the Diagnostic Portal in the Real-Time Monitoring Tool (RTMT).
Connection Directory Feeder	For Intersite Networking, this service checks the local site change-tracking database for directory changes and responds to poll requests from the remote site gateway Reader task.
Connection Realtime Monitoring APIs	This service enables access to data on Connection SRSV by Real-Time Monitoring Tool (RTMT).

Table 19-1 Cisco Unity Connection SRSV Services (continued)

Service	Description
Connection Reports Data Harvester	This service enables conversion of data in log files to entries in the reports database, which is used to generate reports.
Connection REST Service	This service enables Representational State Transfer (REST) API clients.
Connection SMTP Server	This service enables access to data on Connection by an SMTP server.
Connection System Agent	This service enables schedules system tasks (such as re-synchronizing MWIs) that the administrator can enter in Cisco Unity Connection Administration.

Managing Services in Control Center

Control Center in Cisco Unity Connection Serviceability lets you do the following tasks:

- Activate and deactivate Connection SRSV services in the Optional Services section.
- Start and stop all Connection SRSV services except the services in the Status Only Services section.

  Stopping Connection SRSV services in the Critical Services section may cause calls in progress to be dropped and degrades the normal function of the Connection SRSV.
- View the status the status of Connection SRSV services.
- Refresh the status of Connection SRSV services.

Tip You may need to manage services in both Cisco Unity Connection Serviceability and Cisco Unified Serviceability to troubleshoot a problem.

The Cisco Unified Serviceability services are described in the *Cisco Unified Serviceability Administration Guide*.

This section contains five procedures; do the applicable procedure to activate, deactivate, start, or stop Connection SRSV services, or to refresh the status of services. You can activate, deactivate, start, and stop only one service at a time.

To Activate a Service in Control Center

Step 1 In Cisco Unity Connection Serviceability, select **Tools > Service Management**.

Step 2 From the Server drop-down box, select the applicable Connection SRSV and select **Go**.

Step 3 Under Optional Services, locate the service that you want to activate.

Step 4 In the Change Activate Status column, select **Activate**.

To Deactivate a Service in Control Center

Step 1 In Cisco Unity Connection Serviceability, select **Tools > Service Management**.

Step 2 From the Server drop-down box, select the applicable Connection SRSV and select **Go**.

Step 3 Under Optional Services, locate the service that you want to deactivate.

Step 4 In the Change Activate Status column, select **Deactivate**.

To Start a Service in Control Center

Step 1 In Cisco Unity Connection Serviceability, select **Tools > Service Management**.

Step 2 From the Server drop-down box, select the applicable Connection or Cisco Unified CMBE server, and select **Go**.

Step 3 Locate the service that you want to start.

> **Note** Services that are deactivated must be activated before they can be started.

Step 4 In the Change Service Status column, select **Start**.

To Stop a Service in Control Center

Step 1 In Cisco Unity Connection Serviceability, select **Tools > Service Management**.

Step 2 From the Server drop-down box, select the applicable Connection SRSV and select **Go**.

Step 3 Locate the service that you want to stop.

> **Note** Services in the Status Only Services section cannot be started or stopped in Cisco Unity Connection Serviceability. You must use the command line interface (CLI) to start or stop these services.

Step 4 In the Change Service Status column, select **Stop**.

> **Note** Stopping Connection SRSV services in the Critical Services section may cause calls in progress to be dropped and degrades the normal function of the Connection SRSV.
>
> A service that is not activated cannot be started or stopped.

To Refresh Service Status in Control Center

Step 1 In Cisco Unity Connection Serviceability, select **Tools > Service Management**.

Step 2 From the Server drop-down box, select the applicable Connection SRSV and select **Go**.

Step 3 Select **Refresh**.

The status information is updated to reflect the current status.

Accessing Cisco Unity Connection SRSV by Phone

This document provides information about the platforms supported for Cisco Unity Connection SRSV, including those shipped by Cisco and those provided by customers.

The Connection installation application prevents installation on servers that do not meet the exact specifications or models listed in this document.

Contents

About the Connection SRSV Conversation

When you access Cisco Unity Connection SRSV by phone, you hear the Connection SRSV conversation. Its recorded instructions and prompts guide you to receive messages. You can use any phone to access Connection SRSV.

You can use Connection SRSV by phone using:

Phone Keypad	Press the keys on the phone keypad

Americas Headquarters:
Cisco Systems, Inc., 170 West Tasman Drive, San Jose, CA 95134-1706 USA

Using the Phone Keypad with the Connection SRSV Conversation

There are several versions of the Connection SRSV conversation, each providing different keypad mappings for the Connection SRSV menu options. (For example, you might press 3 to delete a message in one version but press 7 to delete a message in another version.)

Your Connection SRSV administrator determines the conversation version that you hear. Typically, an administrator will select a conversation that has a keypad mapping that is familiar to you. Ask your Connection SRSV administrator which conversation you are set up to use.

Voicemail Basics

Calling Cisco Unity Connection SRSV

You can call Cisco Unity Connection SRSV from your desk phone, from another phone within your organization.

Procedure

Step 1 Dial the applicable number to call Connection SRSV.

Step 2 If you are calling from another phone within your organization, press * (star key) when Connection SRSV answers.

Step 3 If prompted, enter your ID and press # (pound key).

Step 4 Enter your Connection SRSV PIN and press #.

Sending Voice Messages

You can send voice messages to other Cisco Unity Connection SRSV users without dialing their extensions. This can

Tip Connection SRSV plays a list of matches that you can navigate quickly. Press # to select a recipient from a list; press 7 to skip to the previous name and 9 to skip to the next name; and press 77 to skip to the beginning of a list and 99 to skip to the end of a list.

Managing Receipts

As you work with Cisco Unity Connection SRSV, you may manage the following types of receipts:

NondeliveryReciept message that informs you when your message could not be delivered to the intended recipient.

When you check messages, Connection SRSV plays receipts along with your other messages. You play and delete receipts in the same way as other messages; you cannot reply to or forward them.

For read receipts, Connection SRSV plays a list of the recipients who played the message you sent. For nondelivery receipts (NDRs), Connection SRSV identifies recipients whose mailboxes did not accept the message.

After Connection SRSV plays an NDR, you can hear the original message and resend it to the recipient(s) who failed to receive it. You can record an introduction, modify the recipient list, and change delivery options when resending a message. Once you resend the message, Connection SRSV automatically deletes the NDR.

- Managing Receipts by Using the Phone Keypad
- Managing Receipts by Using the Phone Keypad

Procedure

Step 1 Call and sign in to Connection SRSV.

Step 2 At the Main menu, select the option Play New Messages, then Receipts.

Step 3 Follow the prompts to manage your receipt.

Finding Messages by Using the Go to Message Option

As you listen to your messages, you can use the Go to Message option to find a particular message by entering the number of the message.

- Finding Messages with Go to Message by Using the Phone Keypad
- Finding Messages with Go to Message by Using the Phone Keypad

Procedure

Step 1 Calland sign in to Connection SRSV.

Step 2 At the Main menu, select the applicable option, Play New Messages or Review Old Messages.

Step 3 Press the Go to Message shortcut keys.

Tip Ask your Connection SRSV administrator for the shortcut keys that you use to hear the prompt for entering the message number.

Step 4 When prompted, enter the message number followed by #.

Step 5 Follow the prompts to manage the message after you have listened to it.

Managing Deleted Messages

About Deleted Messages

Cisco Unity Connection SRSV saves your deleted messages; you can play, restore, or permanently delete them.

Permanently Deleting Deleted Messages

Deleting messages can be an important way to reduce the size of your mailbox, especially when Cisco Unity Connection SRSV is not set up to automatically delete messages once they reach a certain age.

Ask your Connection SRSV administrator if the system is set up to enforce a message-retention policy. Connection SRSV does not indicate when a message-retention policy is enforced, nor does it warn you before messages are permanently deleted as a result of such a policy. If Connection SRSV is not set up to do so, make sure that you permanently delete messages periodically.

Permanently Deleting Messages by Using the Phone Keypad

Procedure

Step 1 Call and sign in to Connection SRSV.

Step 2 At the Main menu, select the option Review Old Messages, then Deleted Messages.

Step 3 Follow the prompts to review your deleted messages and delete them individually, or to delete all messages at once.

Checking Deleted Messages

You can play your deleted messages, just as you can play new and saved messages. You can also restore a deleted message as a new or saved message.

By default, the most recent messages are played first. Note that you cannot enable the Message Type menu or specify a playback order by message type for deleted messages.

Checking Deleted Messages by Using the Phone Keypad

Procedure

Step 1 Call and sign in to Connection SRSV.

Step 2 At the Main menu, select the option Review Old Messages, then Deleted Messages.

Step 3 Follow the prompts to manage a deleted message after you have listened to it.

Changing Your Alternate Contact Numbers

To specify an alternate contact number outside your organization, begin with any access code needed to make an external call (for example, 9). For long-distance numbers, include the applicable dialing codes (for example, 1 and the area code).

About Playback Settings

Playback settings allow you to change the playback volume and the playback speed of:

- An individual message as you are listening to it.
- The conversation for your current phone session at any point while Connection SRSV is playing a prompt.

Changes for individual message playback do not affect playback for other messages you hear during the same phone session. Changes for conversation playback last until you hang up the phone; the next time you call Connection SRSV, playback settings are reset to the defaults.

Note To adjust the conversation speed or volume, you use voice commands; you cannot use the phone keypad.

Changing Playback Volume for Individual Messages

As you listen to a message by phone, you can adjust the volume for that message. Changes do not affect the playback volume of other messages you hear during the same phone session.

Changing Playback Volume for an Individual Message by Using the Phone Keypad

Procedure

The key that you press to adjust playback volume will depend on your conversation. Ask your system administrator which key is assigned to change playback volume. While listening to a message, toggle among these volume settings:

Option	Description
Press key once	Increases the volume

Option	Description
Press key again	Decreases the volume
Press key again	Returns the volume to normal

Changing Playback Speed for Individual Messages

As you listen to a message by phone, you can adjust the playback speed for that message. Changes do not affect the playback speed of other messages you hear during the same phone session.

- Changing Playback Speed for an Individual Message by Using the Phone Keypad
- Changing Playback Speed for an Individual Message by Using the Phone Keypad

Procedure

The key that you press to adjust playback speed will depend on your conversation. Ask your system administrator which keys are assigned to increase and decrease playback speed. While listening to a message, use the following speed settings:

Option	Description
Press decrease key	Slow message playback
Press increase key once	Fast message playback
Press increase key again	Faster message playback

Changing Playback Volume for the Connection Conversation

You can use voice commands to change the volume of the Cisco Unity Connection SRSV conversation at any point while Connection SRSV is playing prompts. (You cannot use the phone keypad to adjust the conversation volume.)

Changes last until you hang up the phone; the next time you call Connection SRSV, the volume is reset to the default setting.

Changing Playback Speed for the Connection Conversation

You can use voice commands to change the speed of the Cisco Unity Connection SRSV conversation at any point while Connection SRSV is playing prompts. (You cannot use the phone keypad to adjust the conversation speed.)

Changes last until you hang up the phone; the next time you call Connection SRSV, the speed is reset to the default setting.

Cisco Unity Connection Phone Menus

Phone Menus for the Classic Conversation

Main Menu and Shortcuts (Classic Conversation)

While listening to the Main menu, press:

Action	Key (s)
Hear new messages	1
Review saved messages	3, 1
Review deleted messages (Not available on some systems)	3, 2

During Message Menu and Shortcuts (Classic Conversation)

While listening to a message, press:

Action	Key (s)
Restart message	1
Play message by number	1 2
Play previous message	1 4
Play next message	6
Save	2
Delete	3
Slow playback	4
Change volume (Not available on some systems)	5
Fast playback	6
Rewind message	7

Action	Key (s)
Pause or resume	8
Fast-forward	9
Fast-forward to end	#
Restore as saved (Not available on some systems)	# 2
Save or restore as new (Not available on some systems)	# 6
Play message properties	# 9
Skip message, save as is	# #
Cancel or back up	*
Help	0

After Message Menu and Shortcuts (Classic Conversation)

After listening to a message, press:

Action	Key (s)
Replay message	1
Play message by number	1 2
Play previous message	1 4
Save or restore as saved (Not available on some systems)	2
Delete	3
ForwardMessage	5
Save or Restore as new	6
Rewind	7
PlayMessage Properties	9
Save as i	#
Cancel or back up	*
Help	0

Recording Menu (Classic Conversation)

Use the following keys while you record messages:

Action	Key (s)
Pause or resume	8
End a recording	#

After Message Menu and Shortcuts (Alternate Keypad Mapping N)

After listening to a message, press:

Action	Key (s)
Rewind	4
Save As is	6
Call the sender (Not available on some systems)	9
Play message properties	7 0
Reply	7 1
Replay Messages	7 2
Forward Message	7 3
Reply to All	7 4
Delete	7 6
Save or restore as saved (Not available on some systems)	7 7
Save or restore as new (Not available on some systems)	7 8
Cancel or back up	*
Operator	0

Recording Menu (Alternate Keypad Mapping N)

Use the following keys while you record messages, names, and greetings:

Action	Key (s)
Pause or resume	8
End a recording	#

CHAPTER 21

Troubleshooting Cisco Unity Connection SRSV in Connection 9.1(1)

Cisco Unity Connection Surviable Remote Site Voicemail is a backup voicemail system that allows you to receive voice messages during WAN outages. See the following sections for information on troubleshooting problems with the Connection SRSV:

Error Message Appears When You Test the Connectivity of Connection with the branch

You may receive the following error messages on the **Edit Branch** page of Cisco Unity Connection Administration when you test the connectivity of Connection with the branch:

- **“Authentication failed. Incorrect Username and Password.”**:If you receive the “Authentication failed. Incorrect Username and Password.” error message on the **Edit Branch** page when you test the connectivity of the central Connection server with the branch, make sure that the username and password of the branch entered on the Edit Branch page are correct.

- **“Branch is unreachable”**: If you receive the “Branch is unreachable” error message on the **Edit Branch** page when you test the connectivity of the central Connection server with the branch, make sure that the PAT port number specified on the **Edit Branch** page is correct.
- **“Server Address is Invalid”**: If you receive the “Server Address is Invalid” error on the **Edit Branch** page, make sure that the FQDN/IP address of the branch entered on the **Edit Branch** page is correct. In case DNS is configured, make sure that the IP address of the branch is added to it.

Certificate Mismatch Error Appears on the Central Connection Server

If you are getting the “**Unable to start provisioning of the branch:: Message=Certificate Mismatch**” error on the **Edit Branch** page of Connection Administration page, make sure that the hostname of the branch mentioned in the certificate installed on the central Connection Server is correct.

Unable to login to the Cisco Unity Connection SRSV Administration

Connection SRSV Administration gets locked if you enter incorrect administrator username and password of the branch three times on the **Edit Branch** page of Connection Administration. To unlock the Connection SRSV Administration interface, you need to reset the administrator credentials for the branch using the **utilsreset_application_ui_administrator_password** CLI command. For more information on this command refer to the "Utils commands” chapter of the *Command Line Interface Reference Guide for Cisco Unified Communications Solutions*, Release 9.0(1) at http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/cli_ref/9_0_1/CUCM_BK_C3A58B83_00_cucm-cli-reference-guide-90_chapter_01001.html.

Branch User is Unable to Login through Telephony User Interface (TUI)

If a branch user is unable to login through the TUI, check the following:

- Make sure that the PIN entered on the central Connection server is synchronized with the branch through provisioning.
- If the branch user is logging in for the first time through the TUI, make sure that the user has set the PIN at the central Connection server and provisioning is done successfully.

Status of Provisioning Remains In Progress for a long time

If the status of provisioning on Cisco Unity Connection Administration remains “In Progress” for a long time, consider the following:

- Check the network connectivity of the central Cisco Unity Connection server with the branch.
- Check whether the central Connection server details are entered correctly on the branch.

- Check whether the **Connection Branch Sync Service** is active on both central Connection server and branch. For more information on the services required for Connection SRSV, refer to the "Managing Cisco Unity Connection Services in Version 9.x" chapter of the *Cisco Unified Serviceability Administration Guide* Release 9.x at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/serv_administration/guide/9xcucservagx.html.
- Check whether the REST services are active on the central Connection server and the branch.

Provisioning from the Central Connection Server to the Branch Is Not Working

If the provisioning of the users from the central Connection Server to the branch does not work, make sure that the license status at the central Connection server is not "Expire". If the license status at the central Connection server is "Expire", you need to install the required licenses for the central Connection server to make the license status as "Compliance" and start provisioning. For more information on licensing requirements, refer to the "Managing Licenses in Cisco Unity Connection 9.x" chapter of the *System Administration Guide for Cisco Unity Connection* Release 9.x at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/administration/guide/9xcucsagx.html.

Status of Provisioning is Partial Success

If the status of the provisioning on the **Branch Sync Results** page is "Partial Success", check the following:

- Make sure that the name of an administrator on the branch is not same as the name of a subscriber on the central Connection server associated with the branch.
- Make sure that the extension of a call handler at the branch is not used as the extension of a branch user at the central Connection server.
- Make sure that the deleted user on the central Connection server is not used on the branch server. For example, if a branch user on Connection is used as operator on Connection SRSV, make sure to change the operator at branch before deleting the user at central Connection server.
- Make sure that the deleted distribution list on the central Connection server is not used on the branch. For example, if a distribution list is used in a call handler on the branch, make sure to change the distribution list in the call handler before deleting the distribution list.

Provisioning/Voicemail Upload Remains in Scheduled state for a long time

If the provisioning of the users or voicemail upload remains in the Scheduled state for a long time, make sure that the **Connection Branch Sync Service** is active on the central Connection server.

Unable to Reach a Branch User through Telephony User Interface (TUI)

If you are not able to reach a branch user through TUI, make sure that the associated partition is added in the **Search Space** of the central Connection server.

Unable to Send a Voice Message to a Branch User During WAN Outage

If you are unable to send a voice message to a branch user during WAN outage, make sure that Visual VoiceMail (VVM) is not installed on your phone. VVM is not supported in the Cisco Unified SRST mode. For more information, contact your phone service provider.

Error Messages Appear on the Branch Sync Results Page

If the username and password of the branch is not entered correctly on the **Edit Branch** page, the provisioning of the users and the voicemail upload does not work and you will receive the following error messages or status in the **Description** field of the **Branch Sync Results** page of Cisco Unity Connection Administration:

- Unable to start Provisioning of the branch:: Message = Authentication failed.
- Unable to fetch voice mail summary of the branch:: Message = Authentication failed.

If you receive the **"Unable to start provisioning of the branch:: Message=Central Server is not Configured on CUCE"** error message on the **Branch Sync Results** page of Cisco Unity Connection Administration when you start provisioning of the branch, enter the correct FQDN/ IP address of the central Connection server on Cisco Unity Connection SRSV Administration to resolve the problem.

Logs are Not Created or SRSV feature is Not Working Properly

If the logs for the branch are not generated or the SRSV feature is not working properly, you may restart the **Connection Branch Sync Service** and the REST APIs on both the branch and Connection sites to correct this issue.

Unable to Perform Backup/Restore Operation on the Branch

If you are unable to perform the backup/restore operation on the branch, make sure that the backup server is configured correctly on the branch.

Central Connection Server Moves to Violation State

If the central Connection server moves to the Violation state, make sure that the number of licenses for the Connection features, such as SpeechView and Connection SRSV, does not exceed its maximum limit. For more information on licensing, refer to the "Managing Licenses in Cisco Unity Connection 9.x" chapter of the *System Administration Guide for Cisco Unity Connection* Release 9.x at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/administration/guide/9xcucsagx.html.

Non-Delivery Receipts (NDR) on the Central Connection Server

If you are getting NDR on the central Connection server but the same email is delivered on the branch, check the NDR code and take action accordingly. For example, if user A sends email to user B from the branch, the email gets successfully delivered to user B on the branch. However, on the central Connection server, user A receives "4.2.2" NDR code stating that the mailbox quota of user B has exceeded its maximum limit. In this case, user B needs to take appropriate action, such as delete existing emails or get the mailbax quota increased to receive further emails. For more information on NDR codes, refer to the "Troubleshooting Non-Delivery Receipts in Cisco Unity Connection 9.x" chapter of this guide.

CHAPTER 22

Cisco Unity Connection SRSV Supported Platforms List

This chapter contains the following sections:

- Hardware Supported by Cisco Unity Connection SRSV, page 22-1
- Specifications for Virtual Platform Hardware Supported by Cisco Unity Connection SRSV 9.1(1), page 22-2

Hardware Supported by Cisco Unity Connection SRSV

This section lists the supported hardware platforms and the minimum Cisco IOS software release required to support the hardware platformCisco Unity Connection SRSV.

Table 1 Specifications for Cisco Unity Connection SRSV

Cisco Platform	Connection SRSV on SM-SRE-900-K9	Connection SRSV on SM-SRE-910-K9
Cisco 2911 Cisco 2921 Cisco 2951	15.1(4)M	15.1(4)M
Cisco 3925 Cisco 3945	15.1(4)M	15.1(4)M
Cisco 3925E Cisco 3945E	15.1(4)M	15.1(4)M

Note A different Cisco IOS software release may be required depending on the version of Cisco Unified Communications Manager Express or Cisco Unified Survivable Remote Site Telephony (SRST) being used. For more information refer to Cisco Unified Communications Manager Express documentation available at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/index.html.

Specifications for Virtual Platform Hardware Supported by Cisco Unity Connection SRSV 9.1(1)

This section lists the supported hardware platforms and the minimum Cisco IOS software release required to support the hardware platform Cisco Unity Connection SRSV.

Table 2 Specifications for Cisco Unity Connection SRSV

	SM-SRE-900-K9	SM-SRE-910-K9	UCS E 140S	UCS E140D, E140DP, E160D, and E160 DP
Minimum Connection SRSV Version	9.1(1)	9.1(1)	9.1(1)	9.1(1)
vCPU (Number of Virtual Processors) Cores and Speed per Core	*Connection 9.1(1) and later only:* 1@a minimum of 2.0 GHz	*Connection 9.1(1) and later only:* 1@a minimum of 2.0 GHz	*Connection 9.1(1) and later only:* 1@a minimum of 2.0 GHz	*Connection 9.1(1) and later only:* 1@a minimum of 2.0 GHz
vRAM (Amount of Virtual RAM)	4 GB	4 GB	4 GB	4 GB
vDisk (Size of Virtual Hard Disks)	1x160GB	1x160GB	1x160GB	1x160GB
Total number of available voice ports[1]	8	8	8	8
Total number of users with mailboxes	200	200	200	200
Approximate message storage, G-711 codec, minutes	72,944	72,944	72,944	72,944
Number of public distribution lists	500	500	500	500
Number of Call Handlers	200	200	500	500
Number of languages supported	2	2	2	2

1. 8 ports G.711 or G.729a (combined TUI or TTS) and 2 ports iLBC or G.722

CHAPTER 23

Alarm Category: EVENT

This chapter gives detail of the following alarms:

Alarm Name: EvtBranchNotReachable

Severity: ERROR_ALARM

Description: Branch[name= %1, address= %2] is not reachable.

Route To: Event Log, Alert Log

Explanation: There is an issue with the connectivity between the central Connection server and the specified branch.

Recommended Action: If there is no connectivity between the central Connection server and the branch office, refer to the *Troubleshooting Guide for Cisco Unity Connection* available at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/troubleshooting/guide/9xcuctsgx.html. If the problem is not resolved then please contact Cisco TAC.

Alarm Name: EvtBranchProvisioned

Severity: INFORMATIONAL_ALARM

Description: The branch[name= %1, address= %2] has been successfully provisioned.

Route To: Event Log, Alert Log

Explanation: The branch has been successfully associated with the central Connection server.

Recommended Action: NONE

Alarm Name: EvtBranchProvisioningFailed

Severity: WARNING_ALARM

Description: Provisioning for branch[name= %1, address= %2] has failed.

Route To: Event Log, Alert Log

Explanation: The provisioning of branch has been failed.

Recommended Action: If there is no connectivity between the central Connection server and the branch office, refer to the *Troubleshooting Guide for Cisco Unity Connection* available at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/troubleshooting/guide/9xcuctsgx.html. If the problem is not resolved then please contact Cisco TAC.

Alarm Name: EvtBranchProvisioningFailedMaxRetries

Severity: ERROR_ALARM

Description: Provisioning for branch[name= %1, address= %2] has failed after maximum %3 retries.

Route To: Event Log, Alert Log

Explanation: Provisioning for a branch has failed in all the retries.

Recommended Action: If there is no connectivity between the central Connection server and the branch office, refer to the *Troubleshooting Guide for Cisco Unity Connection* available at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/troubleshooting/guide/9xcuctsgx.html. If the problem is not resolved then please contact Cisco TAC.

Alarm Name: EvtBranchProvisioningFailedMaxWait

Severity: ERROR_ALARM

Description: A provisioning completion notification was not received for branch[name= %1, address= %2] within the maximum wait time of %3 minutes.

Route To: Event Log, Alert Log

Explanation: Provisioning for a branch has failed because the branch did not return the provisioning completion status within the defined timeframe.

Recommended Action: If there is no connectivity between the central Connection server and the branch office, refer to the *Troubleshooting Guide for Cisco Unity Connection* available at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/troubleshooting/guide/9xcuctsgx.html. If the problem is not resolved then please contact Cisco TAC.

Alarm Name: EvtBranchVoiceMailUpload

Severity: INFORMATIONAL_ALARM

Description: Voice mail upload for branch[name= %1, address= %2] completed successfully. %3 messages were uploaded.

Route To: Event Log

Explanation: Voicemails from branch are uploaded on the central Connection server.

Recommended Action: NONE

Alarm Name: EvtBranchVoiceMailUploadFailed

Severity: ERROR_ALARM

Description: Voice mail upload for branch[name= %1, address= %2] has failed.

Route To: Event Log

Explanation: No voicemail could be uploaded from the branch to the central Connection server.

Recommended Action: If there is no connectivity between the central Connection server and the branch office, refer to the *Troubleshooting Guide for Cisco Unity Connection* available at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/troubleshooting/guide/9xcuctsgx.html. If the problem is not resolved then please contact Cisco TAC.

Alarm Name: EvtBranchVoiceMailUploadPartial

Severity: WARNING_ALARM

Description: Voice mail upload for branch[name= %1, address= %2] partially completed. %3 messages out of %4 were uploaded.

Route To: Event Log

Explanation: All the voicemails could not be uploaded from branch to the central Connection server.

Recommended Action: If there is no connectivity between the central Connection server and the branch office, refer to the *Troubleshooting Guide for Cisco Unity Connection* available at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/troubleshooting/guide/9xcuctsgx.html. If the problem is not resolved then please contact Cisco TAC.

Alarm Name: EvtCentralNotReachable

Severity: ERROR_ALARM

Description: Cenral connection[address= %1] is not reachable.

Route To: Event Log, Alert Log

Explanation: There is an issue with the connectivity between the central Connection server and the specified branch.

Recommended Action: If there is no connectivity between the central Connection server and the branch office, refer to the *Troubleshooting Guide for Cisco Unity Connection* available at http://www.cisco.com/en/US/docs/voice_ip_comm/connection/9x/troubleshooting/guide/9xcuctsgx.html. If the problem is not resolved then please contact Cisco TAC.

CHAPTER 24

Cisco Survivable Remote Site Voicemail (SRSV) APIs

This chapter contains the following sections:

Listing the Branches

The following is an example of the ***GET*** request that lists the branches present in the Connection server:

GET https://<connection-server>/vmrest/branches

The following is an example of response from the above *GET* request and the actual result will depend upon the information that has been provided by you:

Response Code: 200


```
<Branches total="2">
 <Branch>
  <URI>/vmrest/branches/1e0ed69d-028d-4156-9d68-f14a90438448</URI>
  <ObjectId>1e0ed69d-028d-4156-9d68-f14a90438448</ObjectId>
  <IsAlive>true</IsAlive>
  <IsDisabled>false</IsDisabled>
  <OperatorObjectId>159bb671-cbba-4964-b06b-871f990e1de8</OperatorObjectId>
  <Port>443</Port>
  <ProvisionState>0</ProvisionState>
  <ServerAddress>mysrsv.cisco.com</ServerAddress>
  <SyncGreetings>false</SyncGreetings>
  <SyncVoiceName>false</SyncVoiceName>
```

```
 <UserName>admin</UserName>
 <VmUploadState>0</VmUploadState>
 <DisplayName>Branch1</DisplayName>
 <PartitionObjectId>d6ac04c5-fb36-4e21-9e60-d15e0f9c6971</PartitionObjectId>
 <PartitionURI>/vmrest/partitions/d6ac04c5-fb36-4e21-9e60-d15e0f9c6971</PartitionURI>
 <SmtpDomain>mysrsv.cisco.com</SmtpDomain>
 </Branch>
 <Branch>
 <URI>/vmrest/branches/c3816faf-8dc6-48f3-9c6a-b8e93bba1c42</URI>
 <ObjectId>c3816faf-8dc6-48f3-9c6a-b8e93bba1c42</ObjectId>
 <IsAlive>true</IsAlive>
 <IsDisabled>false</IsDisabled>
 <OperatorObjectId>159bb671-cbba-4964-b06b-871f990e1de8</OperatorObjectId>
 <Port>443</Port>
 <ProvisionState>0</ProvisionState>
 <ServerAddress>mysrsv1.cisco.com</ServerAddress>
 <SyncGreetings>false</SyncGreetings>
 <SyncVoiceName>false</SyncVoiceName>
 <UserName>admin</UserName>
 <VmUploadState>0</VmUploadState>
 <DisplayName>Branch2</DisplayName>
 <PartitionObjectId>765cd618-0cff-43a4-b781-efdba282dba4</PartitionObjectId>
 <PartitionURI>/vmrest/partitions/765cd618-0cff-43a4-b781-efdba282dba4</PartitionURI>
 <SmtpDomain>mysrsv1.cisco.com</SmtpDomain>
 </Branch>
</Branches>
```

The following chart lists the data fields:

Table 24-1 Explanation of Data Fields - Listing the Branches

Field Name	Read/Write	Possible Values	Description
ObjectId	Read/Write	objectid	The object id of the branch at central Connection server.
IsAlive	Read	true/false	Connectivity status between the central Connection server and the branch Connection SRSV server.
IsDisabled	Read/Write	true/false	Disabled status of branch on the central Connection server.
OperatorObjectId	Read/Write	objectid	The object id of the user which is assigned as the operator user for the branch on the central Connection server.
Port	Read/Write	Port number	PAT port number for the branch server.
ProvisionState	Read/Write	0 – Idle, 1 – Scheduled, 2 – In-progress	Current provisioning status of branch on the central Connection server.
ServerAddress	Read/Write	FQDN, IP Address	The address of the branch server.
SyncGreetings	Read/Write	true/false	Option to enable/disable syncing of greetings for users.
SyncVoiceName	Read/Write	true/false	Option to enable/disable syncing of voice names for users.

Table 24-1 Explanation of Data Fields - Listing the Branches

Field Name	Read/Write	Possible Values	Description
UserName	Read/Write	String	User name to be used for REST communication between the central Connection and the branch Connection SRSV server.
VmUploadState	Read/Write	0 – Idle, 1 – Scheduled, 2 – In-progress	Displays the current voicemail upload status of branch on the central Connection server.
DisplayName	Read/Write	String	Display name of the branch Connection SRSV server on the central Connection server.
PartitionObjectId	Read/Write	ObjectId	Partition object ID associated with the branch on the central Connection server.
SmtpDomain	Read/Write	Domain name	Smtp domain of the branch Connection SRSV server.

Viewing Data for an Individual Branch

The following is an example of the ***GET*** request that lists the properties for an individual branch present in the central Connection server:

GET https://<connection-server>/vmrest/branches/<objectid>

The following is an example of response from the above *GET* request and the actual result will depend upon the information that has been provided by you:

Response Code: 200


```
<Branch>
 <URI>/vmrest/branches/c3816faf-8dc6-48f3-9c6a-b8e93bba1c42</URI>
 <ObjectId>c3816faf-8dc6-48f3-9c6a-b8e93bba1c42</ObjectId>
 <IsAlive>true</IsAlive>
 <IsDisabled>false</IsDisabled>
 <OperatorObjectId>159bb671-cbba-4964-b06b-871f990e1de8</OperatorObjectId>
 <Port>443</Port>
 <ProvisionState>0</ProvisionState>
 <ServerAddress>mysrsv.cisco.com</ServerAddress>
 <SyncGreetings>false</SyncGreetings>
 <SyncVoiceName>false</SyncVoiceName>
 <UserName>admin</UserName>
 <VmUploadState>0</VmUploadState>
 <DisplayName>branch16</DisplayName>
 <PartitionObjectId>765cd618-0cff-43a4-b781-efdba282dba4</PartitionObjectId>
 <PartitionURI>/vmrest/partitions/765cd618-0cff-43a4-b781-efdba282dba4</PartitionURI>
 <SmtpDomain>mysrsv.cisco.com</SmtpDomain>
</Branch>
```


The following chart lists the data fields.

Table 24-2 Explanation of Data Fields - Viewing Data for Individual Branch

Field Name	Read/Write	Possible Values	Description
URI	Read	URL to access the branch.	Server address of a particular branch.
ObjectId	Read/Write	object ID	The object id of the branch at the central Connection server.
IsAlive	Read/Write	true/false	Connectivity status between the central and branch Connection server.
IsDisabled	Read/Write	true/false	Disabled status of branch on the central Connection server.
OperatorObjectId	Read/Write	Objectid	The object id of the user which is assigned as the operator user for the branch on the central Connection server.
Port	Read/Write	Port number	PAT port number for the branch server.
Provision	Read/Write	State 0 – Idle, 1 – Scheduled, 2 – In-progress	Current provisioning status of branch on the central Connection server.
ServerAddress	Read/Write	FQDN, IP Address	The address of the branch Connection server.
SyncGreetings	Read/Write	true/false	Option to enable/disable syncing of greetings for users.
SyncVoiceName	Read/Write	true/false	Option to enable/disable syncing of voice names for users.
UserName	Read/Write	String	User name of the administrator of a particular branch.
VmUploadState	Read/Write	0 – Idle, 1 – Scheduled, 2 – In-progress	Current voicemail upload status of branch on the central Connection server.
DisplayName	Read/Write	String	Display name of the branch server on the central Connection server.
PartitionObjectId	Read/Write	ObjectId	Partition object ID associated with the branch on the central Connection server.
PartitionURI	Read/Write	URL Partition	URL associated with the branch on the central Connection server.
SmtpDomain	Read/Write	Domain name	Smtp domain of the branch server.

Creating a Branch

The following is an example of the ***POST*** request that is used for creating a branch on the central Connection server:

POST https://<connection-server>/vmrest/branches

```
<Branch>
    <IsDisabled>false</IsDisabled>
    <OperatorObjectId>159bb671-cbba-4964-b06b-871f990e1de8</OperatorObjectId>
    <Port>443</Port>
    <ServerAddress>mysrsv.cisco.com</ServerAddress>
    <SyncGreetings>false</SyncGreetings>
    <SyncVoiceName>false</SyncVoiceName>
    <UserName>admin</UserName>
    <Password>test</Password>
    <DisplayName>branch16</DisplayName>
    <PartitionObjectId>765cd618-0cff-43a4-b781-efdba282dba4</PartitionObjectId>
    <SmtpDomain>mysrsv.cisco.com</SmtpDomain>
</Branch>
```

The mandatory properties are ServerAddress, UserName, Password, DisplayName, PartitionObjectId, and SmtpDomain.

The successful response code returned for this API is 201. The error response code and data will depend upon the information provided by you:

Response Code: 201

```
/vmrest/branches/c3816faf-8dc6-48f3-9c6a-b8e93bba1c42
```

The following chart lists the data fields:

Table 24-3 Explanation of Data Fields - Creating a Branch

Field Name	Read/Write	Possible Values	Description
IsDisabled	Read/Write	true/false	Enables or activates the branch.
OperatorObjectId	Read/Write	Object ID of the operator.	The operator or the user that must be used to synchronize the messages received by the branch Connection server.
Port	Read/Write	Port number	A port number that the branch uses to communicate with Cisco Unity Connection.
ServerAddress	Read/Write	FQDN, IP Address	The IP address or the Fully Qualified Domain Name (FQDN) of the branch Connection SRSV server.
SyncGreetings	Read/Write	true/false	Synchronize the greetings for the users on the branch Connection SRSV server.
SyncVoiceName	Read/Write	true/false	Synchronize the recorded voice name of the user on the branch Connection SRSV server.
UserName	Read/Write	String	The user name of the administrator of the branch Connection server.
Password	Read/Write	String	The password of the administrator of the branch Connection server.

Table 24-3 Explanation of Data Fields - Creating a Branch

DisplayName	Read/Write	String	Display name of the branch server on the central Connection server.
PartitionObjectId	Read/Write	ObjectId	Partition object ID associated with the branch on the central Connection server.
SmtpDomain	Read/Write	Domain name	Smtp domain of the branch server.

Updating a Branch

The following is an example of the *PUT* request that is used for updating a branch on the central Connection server:

PUT https://<connection-server>/vmrest/branches/c3816faf-8dc6-48f3-9c6a-b8e93bba1c42

```
<Branch>
    <IsDisabled>false</IsDisabled>
    <OperatorObjectId>159bb671-cbba-4964-b06b-871f990e1de8</OperatorObjectId>
    <Port>443</Port>
    <ServerAddress>mysrsv.cisco.com</ServerAddress>
    <SyncGreetings>false</SyncGreetings>
    <SyncVoiceName>false</SyncVoiceName>
    <UserName>admin</UserName>
    <Password>test</Password>
    <DisplayName>branch16</DisplayName>
    <PartitionObjectId>765cd618-0cff-43a4-b781-efdba282dba4</PartitionObjectId>
    <SmtpDomain>mysrsv.cisco.com</SmtpDomain>
    <ProvisionState>1</ProvisionState>
    <VmUploadState>0</VmUploadState>
</Branch>
```

This ***PUT*** request is also used for scheduling a branch for provisioning and voicemail upload. Only the properties mentioned in above XML are writable at the time of modifying a branch. The properties, ProvisionState and VmUploadState, can not be put in the request XML at the same time as a branch can be scheduled either for provisioning or voicemail upload, at a given point of time. The value of those fields can only be 1.

The successful response code returned for this API is 201. The error response code and data will depend upon the information provided by you:

Response Code: 201

```
/vmrest/branches/c3816faf-8dc6-48f3-9c6a-b8e93bba1c42
```

The following chart lists the data fields:

Table 24-4 Explanation of Data Fields - Updating a Branch

Field Name	Read/Write	Possible Values	Description
IsDisabled	Read/Write	true/false	Enables or activates the branch.

Table 24-4 Explanation of Data Fields - Updating a Branch

OperatorObjectId	Read/Write	Object ID of the operator.	The operator or the user that must be used to synchronize the messages received by the branch Connection server.
Port	Read/Write	Port number	A port number that the branch uses to communicate with Cisco Unity Connection.
ServerAddress	Read/Write	FQDN, IP Address	The IP address or the Fully Qualified Domain Name (FQDN) of the branch Connection server.
SyncGreetings	Read/Write	true/false	Synchronize the greetings for the users on the branch.
SyncVoiceName	Read/Write	true/false	Synchronize the recorded voice name of the user on the branch.
UserName	Read/Write	String	The user name of the administrator of the branch Connection SRSV server.
Password	Read/Write	String	The password of the administrator of the branch Connection SRSV server.
DisplayName	Read/Write	String	Display name of the branch server on the central Connection server.
PartitionObjectId	Read/Write	ObjectId	Partition object ID associated with the branch on the central Connection server.
SmtpDomain	Read/Write	Domain name	Smtp domain of the branch server.
ProvisionState	Read/Write	0 – Idle, 1 – Scheduled, 2 – In-progress	Current provisioning status of branch on central Connection server.
VmUploadState	Read/Write	0 – Idle, 1 – Scheduled, 2 – In-progress	Current voicemail upload status of branch on central Connection server.

Deleting a Branch

The following is an example of the Delete request that is used for deleting a branch on the central Connection server:

```
DELETE  /vmrest/branches/c3816faf-8dc6-48f3-9c6a-b8e93bba1c42
```

A branch using this API can only be deleted, if the branch is not in the in-progress state neither for provisioning nor voicemail upload.

The successful response code returned for this API is 201. The error response code and data will depend upon the information provided by you:

Response Code: 201 Data: NA

Assigning a User to Branch

The following is an example of the Put request that is used for assigning a branch to a user by allocating the branch partition to it:

```
PUT /vmrest/users/<userObjectId>


<User>
    <PartitionObjectId>partitionObjectIdMappedToBranch</PartitionObjectId>
</User>
```

To fetch the partition information of a branch, you can use the API to view the details of a branch. Refer to the Viewing Data for an Individual Branch, page 24-3 section for more information. The PartitionObjectId element given in the response XML of this section denotes the partition mapped with the branch.

Response Code: 204

Removing a User from a Branch

The following is an example of the Put request that is used for removing a user from a branch by modifying its partition to some other partition that is not mapped to that branch:

```
PUT /vmrest/users/<userObjectId>


<User>
    <PartitionObjectId>partitionObjectIdNotMappedToBranch</PartitionObjectId>
</User>
```

To fetch the partition information of a branch, you can use the API to view the details of a branch. Refer to the Viewing Data for an Individual Branch, page 24-3 section for more information. The PartitionObjectId element given in the response XML of this section denotes the partition mapped with the branch.

Response Code: 204

Listing All Users Those Are Part of a Particular Branch

The following is an example of the Get request that is used to list the users those are part of a particular branch by searching with the partition object ID of the branch:

```
GET  /vmrest/users?query=(PartitionObjectId is partitionObjectIdMappedToBranch)
```

Response Code: 200

```
<Users total="10">
    <User>
        <URI>/vmrest/users/cb13e6a9-7322-45fa-91cd-7a0b1e21b754</URI>
        <ObjectId>cb13e6a9-7322-45fa-91cd-7a0b1e21b754</ObjectId>
    </User>
</Users>
```

Table 24-5 Explanation of Data Fields - Listing All Users Those are Part of a Particular Branch

Field Name	Read/Write	Possible Values	Description
URI	Read	URL to access the branch.	Server address of a particular branch.
ObjectId	Read/Write	object ID	The object id of the branch at the central Connection server.

Creating a Call Handler for a Branch

The following is an example of the Put request that is used to create a call handler:

```
POST  /vmrest/handlers/callhandlers?templateObjectId=<callhandlerTemplateObjectId>


<Callhandler>
    <DisplayName>Test</DisplayName>
</Callhandler>
```

This is an existing API for creating a call handler that can be used at the branch as well.

Response Code: 201

```
/vmrest/handlers/callhandlers/<callhandlerObjectId>
```

CHAPTER 25

Cisco Survivable Remote Site Voicemail (SRSV) Limitations and Restrictions

This chapter contains the following sections:

Voicemail Limitations and Restrictions

- The following features are not supported with Cisco Unity Connection SRSV:
  - Fax support
  - Addressing contacts
  - Dispatch messages
  - Scheduled base services, such as alternate greetings and notifications
  - Advanced telephony features, such as call screening.
  - Updating spoken name, distribution lists, or PINs through the touchtone conversation users functionality
  - Touchtone conversation users administration interfaces, such as broadcast or greeting administration.
  - Private distribution lists.
  - Text-to-speech or voice recognition features.
  - Customizing the voicemail flow for touchtone conversation users (TUI) on a Connection SRSV.
  - VPIM
  - IMAP
  - Single Inbox
  - Cisco Personal Communication Assistant (CPCA) and Web Inbox

- The Compose, Forward, and Reply to voice messages functionalities are not supported with Connection SRSV. Only the Ring No Answer/Call forward Busy functionalities are supported.
- The voicemail synchronization is supported only via central Connection server. The voice messages received on central Connection server are not replicated to the Connection SRSV.
- The Message Waiting Indicator (MWI) for Connection SRSV is not supported.
- The voice messages upload is not synchronized with phone re-home to Cisco Unified Communications Manager.
- Few class of service (COS) features of central Connection server, such as distribution list access and message deletion behavior, are provisioned for all Connection SRSV users.
- The subscribers cannot log in to Cisco Unity Connection SRSV Administration until they set up their voicemail preferences at central Connection server.
- The Live Record and Live Reply functionalities are not supported.

Auto-Attendant Limitations

The auto-attendant configuration is done at branch site only. There is no syncronization required from central Connection server.

The following auto-attendant features are supported:

- Local user only lookup

There is no support given for the following auto-attendant features:

- Partitions or search spaces
- Advanced calling features, such as call screening
- Interview handlers
- Dispatch messages

Network Address Translation (NAT) Restrictions

- NAT is only supported at branch locations and not at the central Connection server.
- Only one Connection SRSV can be provisioned at each NAT site.
- Only static NAT and Port Address Translation (PAT) are supported. Dynamic NAT is not supported.

Backup and Restore Limitations

To avoid creating duplicate email messages, we do not recommend taking back up of data on Connection SRSV.

Distribution Lists

- The voice messages sent to distribution lists in survivable mode are sent to members only after WAN gets recovered.

- The system does not provision distribution lists with the spoken name.
- The system does not provision recorded names for distribution lists.
- Only public distribution lists are supported.