



# Cisco Unified CallManager 5.0 SIP Trunk Integration Guide for Cisco Unity Connection 1.1

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*Revised December 8, 2006*

This document provides instructions for integrating the Cisco Unified CallManager phone system with Cisco Unity Connection by a SIP trunk.

Cisco Unity Connection supports a SIP trunk integration with the Cisco Unified CallManager phone system when the Cisco Unified CallManager phone system has only SIP phones.



**Note**

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If you are configuring MWI relay across trunks in a distributed phone system, you must refer to the Cisco Unified CallManager documentation for requirements and instructions. Configuring MWI relay across trunks does not involve Cisco Unity Connection settings.

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## Integration Tasks

Before doing the following tasks to integrate Cisco Unity Connection with the Cisco Unified CallManager phone system, confirm that the Cisco Unity Connection server is ready for the integration by completing the applicable tasks in the *Cisco Unity Connection Installation Guide*.

The following task lists describe the process for creating, changing, and deleting integrations.

## Task List to Create the Integration by a SIP Trunk

Use the following task list to set up a new integration with the Cisco Unified CallManager phone system. If you are installing a new Cisco Unity Connection server by using the *Cisco Unity Connection Installation Guide*, you may have already completed some of the following tasks.

1. Review the system and equipment requirements to confirm that all phone system and Cisco Unity Connection server requirements have been met. See the [“Requirements” section on page 3](#).
2. Plan how the voice messaging ports will be used by Cisco Unity Connection. See the [“Planning How the Voice Messaging Ports Will Be Used by Cisco Unity Connection” section on page 5](#).



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3. Program Cisco Unified CallManager. See the “[Programming the Cisco Unified CallManager Phone System](#)” section on page 7.
4. Create the integration. See the “[Creating a New Integration with the Cisco Unified CallManager Phone System](#)” section on page 12.

**Note**

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An additional Cisco Unified CallManager cluster can be added by creating a new phone system integration through the Phone System Integration Wizard. Each Cisco Unified CallManager cluster is a separate phone system integration.

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5. Test the integration. See the “[Testing the Integration](#)” section on page 15.
6. If this integration is a second or subsequent integration, add the applicable new user templates for the new phone system. See the [\(Multiple Integrations Only\) Adding New User Templates](#), page 19.

## Task List to Change the Number of Voice Messaging Ports

Use the following task list to change the number of voice messaging ports for an integration after it has been created.

1. Change the number of voice messaging ports in Cisco Unified CallManager Administration and in Cisco Unity Connection Administration. See the “[Changing the Number of Voice Messaging Ports](#)” section on page 19.

## Task List to Add a Cisco Unified CallManager Express Server to a Cisco Unified CallManager Cluster

Use the following task list to add a Cisco Unified CallManager Express server to a Cisco Unified CallManager cluster.

1. Confirm that the Cisco Unified CallManager Express server meets the requirements for integrating with Cisco Unity Connection. Refer to the applicable Cisco Unified CallManager Express integration guide at [http://www.cisco.com/en/US/products/ps6509/products\\_installation\\_and\\_configuration\\_guides\\_list.html](http://www.cisco.com/en/US/products/ps6509/products_installation_and_configuration_guides_list.html).
2. Add the Cisco Unified CallManager Express server to the port group for the Cisco Unified CallManager phone system integration. See “[Adding a Cisco Unified CallManager Express Server to a Cisco Unified CallManager Phone System Integration](#)” section on page 21.
3. If needed, add voice messaging ports. See the “[Changing the Number of Voice Messaging Ports](#)” section on page 19.

# Requirements

The Cisco Unified CallManager integration supports configurations of the following components:

## Phone System

- A Cisco IP telephony applications server consisting of Cisco Unified CallManager 5.0(x), running on a Cisco Media Convergence Server (MCS) or customer-provided server meeting approved Cisco configuration standards.
- For the Cisco Unified CallManager extensions, SIP phones that support DTMF relay as described in RFC-2833.
- A LAN connection in each location where you will plug a SIP phone into the network.
- For multiple Cisco Unified CallManager clusters, the capability for users to dial an extension on another Cisco Unified CallManager cluster without having to dial a trunk access code or prefix.

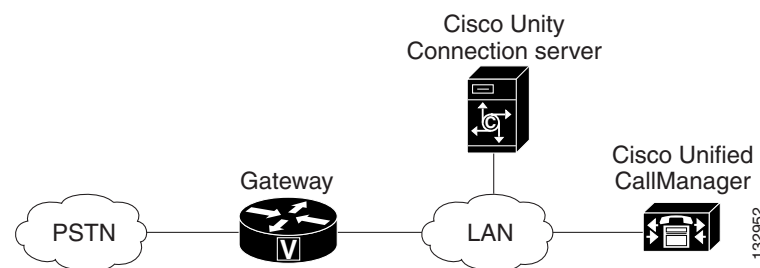
## Cisco Unity Connection Server

- The applicable version of Cisco Unity Connection. For details on compatible versions of Cisco Unity Connection and Cisco Unified CallManager, refer to the *SIP Trunk Compatibility Matrix: Cisco Unity Connection and Cisco Unified CallManager* at [http://www.cisco.com/en/US/products/ps6509/products\\_device\\_support\\_table09186a0080624b9f.html](http://www.cisco.com/en/US/products/ps6509/products_device_support_table09186a0080624b9f.html).
- Cisco Unity Connection installed and ready for the integration, as described in the *Cisco Unity Connection Installation Guide* at [http://www.cisco.com/en/US/products/ps6509/prod\\_installation\\_guides\\_list.html](http://www.cisco.com/en/US/products/ps6509/prod_installation_guides_list.html).
- A license that enables the appropriate number of voice messaging ports.

# Integration Description

The Cisco Unified CallManager integration uses the LAN to connect Cisco Unity Connection and the phone system. The gateway provides connections to the PSTN. [Figure 1](#) shows the connections.

**Figure 1** Connections Between the Phone System and Cisco Unity Connection



# Call Information

The phone system sends the following information with forwarded calls:

- The extension of the called party

- The extension of the calling party (for internal calls) or the phone number of the calling party (if it is an external call and the system uses caller ID)
- The reason for the forward (the extension is busy, does not answer, or is set to forward all calls)

Cisco Unity Connection uses this information to answer the call appropriately. For example, a call forwarded to Cisco Unity Connection is answered with the personal greeting of the user. If the phone system routes the call to Cisco Unity Connection without this information, Cisco Unity Connection answers with the opening greeting.

## Integration Functionality

The Cisco Unified CallManager integration with Cisco Unity Connection provides the following features:

- Call forward to personal greeting
- Call forward to busy greeting
- Caller ID
- Easy message access (a user can retrieve messages without entering an ID; Cisco Unity Connection identifies a user based on the extension from which the call originated; a password may be required)
- Identified subscriber messaging (Cisco Unity Connection automatically identifies a user who leaves a message during a forwarded internal call, based on the extension from which the call originated)
- Message waiting indication (MWI)

The functionality of this integration may be affected by the issues described below.

### Use of Cisco Unified Survivable Remote Site Telephony (SRST) Router

When a Cisco Unified Survivable Remote Site Telephony (SRST) router is part of the network and the Cisco Unified SRST router takes over call processing functions from Cisco Unified CallManager (for example, because the WAN link is down), phones at a branch office can continue to function. In this situation, however, the integration features have the following limitations:

- **Call forward to busy greeting**—When the Cisco Unified SRST router uses FXO/FXS connections to the PSTN and a call is forwarded from a branch office to Cisco Unity Connection, the busy greeting cannot play.
- **Call forward to internal greeting**—When the Cisco Unified SRST router uses FXO/FXS connections to the PSTN and a call is forwarded from a branch office to Cisco Unity Connection, the internal greeting cannot play. Because the PSTN provides the calling number of the FXO line, the caller is not identified as a user.
- **Call transfers**—Because an access code is needed to reach the PSTN, call transfers from Cisco Unity Connection to a branch office will fail.
- **Identified user messaging**—When the Cisco Unified SRST router uses FXO/FXS connections to the PSTN and a user at a branch office leaves a message or forwards a call, the user is not identified. The caller appears as an unidentified caller.
- **Message waiting indication**—MWIs are not updated on branch office phones, so MWIs will not correctly reflect when new messages arrive or when all messages have been listened to. We recommend resynchronizing MWIs after the WAN link is reestablished.
- **Message notification**—Because an access code is needed to reach the PSTN, message notifications from Cisco Unity Connection to a branch office will fail.

- **Routing rules**—When the Cisco Unified SRST router uses FXO/FXS connections to the PSTN and a call arrives from a branch office to Cisco Unity Connection (either a direct or forwarded call), routing rules will fail.

When the Cisco Unified SRST router uses PRI/BRI connections, the caller ID for calls from a branch office to Cisco Unity Connection may be the full number (exchange plus extension) provided by the PSTN and therefore may not match the extension of the Cisco Unity Connection user. If this is the case, you can let Cisco Unity Connection recognize the caller ID by using alternate extensions (for instructions, see the “Appendix: Using Alternate Extensions and MWIs” section).

Redirected Dialed Number Information Service (RDNIS) needs to be supported when using SRST.

For information on setting up Cisco Unified SRST routers, refer to the “Integrating Voice Mail with Cisco Unified SRST” section of the *Cisco Unified SRST System Administrator Guide* at <http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122limit/122z/122zj15/index.htm>.

#### **Impact of Non-Delivery of RDNIS on Voice Mail Calls Routed via AAR**

RDNIS needs to be supported when using Automated Alternate Routing (AAR).

AAR can route calls over the PSTN when the WAN is oversubscribed. However, when calls are rerouted over the PSTN, RDNIS can be affected. Incorrect RDNIS information can affect voice mail calls that are rerouted over the PSTN by AAR when Cisco Unity Connection is remote from its messaging clients. If the RDNIS information is not correct, the call will not reach the voice mail box of the dialed user but will instead receive the automated attendant prompt, and the caller might be asked to reenter the extension number of the party they wish to reach. This behavior is primarily an issue when the telephone carrier is unable to ensure RDNIS across the network. There are numerous reasons why the carrier might not be able to ensure that RDNIS is properly sent. Check with your carrier to determine whether it provides guaranteed RDNIS delivery end-to-end for your circuits. The alternative to using AAR for oversubscribed WANs is simply to let callers hear reorder tone in an oversubscribed condition.

## **Integrations with Multiple Phone Systems**

Cisco Unity Connection can be integrated with multiple phone systems at one time. For information on and instructions for integrating Cisco Unity Connection with multiple phone systems, refer to the *Multiple Phone System Integration Guide* at [http://www.cisco.com/en/US/products/ps6509/products\\_installation\\_and\\_configuration\\_guides\\_list.html](http://www.cisco.com/en/US/products/ps6509/products_installation_and_configuration_guides_list.html).


## **Planning How the Voice Messaging Ports Will Be Used by Cisco Unity Connection**

Before programming the phone system, you need to plan how the voice messaging ports will be used by Cisco Unity Connection. The following considerations will affect the programming for the phone system (for example, setting up the hunt group or call forwarding for the voice messaging ports):

- The number of voice messaging ports installed.
- The number of voice messaging ports that will answer calls.
- The number of voice messaging ports that will only dial out, for example, to send message notification, to set message waiting indicators (MWIs), and to make telephone record and playback (TRAP) connections.

The following table describes the voice messaging port settings in Cisco Unity Connection that can be set on Telephony Integrations > Port of Cisco Unity Connection Administration.

**Table 1**      **Settings for the Voice Ports**

Field	Considerations
Enabled	Check this check box.
Answer Calls	Check this check box.   <b>Caution</b> All voice messaging ports connecting to the Cisco Unified CallManager server must have the Answer Calls box checked. Otherwise, calls to Cisco Unity Connection may not be answered.
Perform Message Notification	Check this check box to designate the port for notifying users of messages.
Send MWI Requests	Check this check box to designate the port for turning MWIs on and off.
Allow TRAP Connections	Check this check box so that users can use the port for recording and playback through the phone in Cisco Unity Connection web applications.

#### The Number of Voice Messaging Ports to Install

The number of voice messaging ports to install depends on numerous factors, including:

- The number of calls Cisco Unity Connection will answer when call traffic is at its peak.
- The expected length of each message that callers will record and that users will listen to.
- The number of users.
- The number of ports that will be set to dial out only.
- The number of calls made for message notification.
- The number of MWIs that will be activated when call traffic is at its peak.
- The number of TRAP connections needed when call traffic is at its peak. (TRAP connections are used by Cisco Unity Connection web applications to play back and record over the phone.)
- The number of calls that will use the automated attendant and call handlers when call traffic is at its peak.

It is best to install only the number of voice messaging ports that are needed so that system resources are not allocated to unused ports.

#### The Number of Voice Messaging Ports That Will Answer Calls

The calls that the voice messaging ports answer can be incoming calls from unidentified callers or from users. Typically, the voice messaging ports that answer calls are the busiest.

You can set voice messaging ports to both answer calls and to dial out (for example, to send message notifications). However, when the voice messaging ports perform more than one function and are very active (for example, answering many calls), the other functions may be delayed until the voice messaging port is free (for example, message notifications cannot be sent until there are fewer calls to answer). For best performance, dedicate certain voice messaging ports for only answering incoming calls, and dedicate other ports for only dialing out. Separating these port functions eliminates the possibility of a collision, in which an incoming call arrives on a port at the same time that Cisco Unity Connection takes the port off-hook to dial out.

**The Number of Voice Messaging Ports That Will Only Dial Out, and Not Answer Calls**

Ports that will only dial out and will not answer calls can do one or more of the following:

- Notify users by phone, pager, or e-mail of messages that have arrived.
- Turn MWIs on and off for user extensions.
- Make a TRAP connection so that users can use the phone as a recording and playback device in Cisco Unity Connection web applications.

Typically, these voice messaging ports are the least busy ports.

**Caution**

In programming the phone system, do not send calls to voice messaging ports in Cisco Unity Connection that cannot answer calls (voice messaging ports that are not set to Answer Calls). For example, if a voice messaging port is set only to Perform Message Notifications, do not send calls to it.

**Preparing for Programming the Phone System**

Record your decisions about the voice messaging ports to guide you in programming the phone system.

# Programming the Cisco Unified CallManager Phone System

Do the following procedures in the order given.

**Note**

There must be a calling search space that is used by all user phones (directory numbers). Otherwise, the integration will not function correctly. For instructions on setting up a calling search space and assigning user phones to it, refer to the Cisco Unified CallManager Help.

**To Create the SIP Trunk Security Profile**

- Step 1** In Cisco Unified CallManager Administration, on the System menu, click **Security Profile > SIP Trunk Security Profile**.
- Step 2** On the Find and List SIP Trunk Security Profiles page, click **Add New**.
- Step 3** On the SIP Trunk Security Profile Configuration page, under SIP Trunk Security Profile Information, enter the following settings.

**Table 2** Settings for the SIP Trunk Security Profile Configuration Page

Field	Setting
Name	Enter <b>Connection SIP Trunk Security Profile</b> or another name.
Description	Enter <b>SIP trunk security profile for Cisco Unity Connection</b> or another description.
Device Security Mode	Accept the default of <b>Non-secure</b> .
Accept Out-of-Dialog REFER	Check this check box.

**Table 2** Settings for the SIP Trunk Security Profile Configuration Page (continued)

Field	Setting
Accept Unsolicited Notification	Check this check box.
Accept Header Replacement	Check this check box.

**Step 4** Click **Save**.

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#### To Create the SIP Profile

- Step 1** On the Device menu, click **Device Settings > SIP Profile**.
- Step 2** On the Find and List SIP Profiles page, click **Add New**.
- Step 3** On the SIP Profile Configuration page, enter the following settings.

**Table 3** Settings for the SIP Profile Configuration Page

Field	Setting
Name	Enter <b>Connection SIP Profile</b> or another name.
Description	Enter <b>SIP profile for Cisco Unity Connection</b> or another description.

**Step 4** Click **Save**.

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#### To Create the SIP Trunk

- Step 1** On the Device menu, click **Trunk**.
- Step 2** On the Find and List Trunks page, click **Add New**.
- Step 3** On the Trunk Configuration page, in the Trunk Type field, click **SIP Trunk**.
- Step 4** In the Device Protocol field, click **SIP** and click **Next**.
- Step 5** Under Device Information, enter the following settings.

**Table 4** Settings for Device Information on the Trunk Configuration Page

Field	Setting
Device Name	Enter <b>Connection_SIP_Trunk</b> or another name.
Description	Enter <b>SIP trunk for Cisco Unity Connection</b> or another description.

**Step 6** Under Outbound Calls, check the **Redirecting Diversion Header Delivery - Outbound** check box.

**Step 7** Under SIP Information, enter the following settings.

**Table 5** Settings for SIP Information on the Trunk Configuration Page

Field	Setting
Destination Address	Enter the IP address of the Cisco Unity Connection SIP port to which Cisco Unified CallManager will connect.
Destination Port	We recommend that you accept the default of <b>5060</b> .
SIP Trunk Security Profile	Click the name of the SIP trunk security profile that you created in the <a href="#">“To Create the SIP Trunk Security Profile” procedure on page 7</a> . For example, click “Cisco Unity Connection SIP Trunk Security Profile.”
Rerouting Calling Search Space	Click the name of the calling search space that is used by user phones.
Out-of-Dialog Refer Calling Search Space	Click the name of the calling search space that is used by user phones.
SIP Profile	Click the name of the SIP profile that you created in the <a href="#">“To Create the SIP Profile” procedure on page 8</a> . For example, click “Cisco Unity Connection SIP Profile.”

**Step 8** Adjust any other settings that are needed for your site.

**Step 9** Click **Save**.

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### To Create a Route Pattern

**Step 1** On the Call Routing menu, click **Route/Hunt > Route Pattern**.

**Step 2** On the File and List Route Patterns page, click **Add New**.

**Step 3** On the Route Pattern Configuration page, enter the following settings.

**Table 6** Settings for the Route Pattern Configuration Page

Field	Setting
Route Pattern	Enter the voice mail pilot number for Cisco Unity Connection.
Gateway/Route List	Click the name of the SIP trunk that you created in the <a href="#">“To Create the SIP Trunk” procedure on page 8</a> . For example, click “Connection_SIP_Trunk.”

**Step 4** Click **Save**.

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### To Create the Voice Mail Pilot

**Step 1** On the Voice Mail menu, click **Voice Mail Pilot**.

**Step 2** On the Find and List Voice Mail Pilots page, click **Add New**.

**Step 3** On the Voice Mail Pilot Configuration page, enter the following voice mail pilot number settings.

**Table 7** Settings for the Voice Mail Pilot Configuration Page

Field	Setting
Voice Mail Pilot Number	Enter the voice mail pilot number that users will dial to listen to their voice messages. This number must match the route pattern that you entered in the <a href="#">“To Create a Route Pattern” procedure on page 9</a> .
Calling Search Space	Click the calling search space that includes partitions containing the user phones and the partition that you set up for the voice mail pilot number.
Description	Enter <b>Connection Pilot</b> or another description.
Make This the Default Voice Mail Pilot for the System	Check this check box. When this check box is checked, this voice mail pilot number replaces the current default pilot number.

**Step 4** Click **Save**.

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#### To Create the Voice Mail Profile

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**Step 1** On the Voice Mail menu, click **Voice Mail Profile**.

**Step 2** On the Find and List Voice Mail Profiles page, click **Add New**.

**Step 3** On the Voice Mail Profile Configuration page, enter the following voice mail profile settings.

**Table 8** Settings for the Voice Mail Profile Configuration Page

Field	Setting
Voice Mail Profile Name	Enter <b>Connection Profile</b> or another name to identify the voice mail profile.
Description	Enter <b>Profile for Cisco Unity Connection</b> or another description.
Voice Mail Pilot	Click the voice mail pilot number that you defined in the <a href="#">“To Create the Voice Mail Pilot” procedure on page 9</a> .
Voice Mail Box Mask	When multitenant services are not enabled on Cisco Unified CallManager, leave this field blank.  When multitenant services are enabled, each tenant uses its own voice mail profile and must create a mask to identify the extensions (directory numbers) in each partition that is shared with other tenants. For example, one tenant can use a mask 972813XXXX, while another tenant can use the mask 214333XXXX. It is also necessary to set up translation patterns for MWIs.
Make This the Default Voice Mail Profile for the System	Check this check box to make this voice mail profile the default.  When this check box is checked, this voice mail profile replaces the current default voice mail profile.

**Step 4** Click **Save**.

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Do the following two procedures only if you want to set up SIP Digest authentication.

If you do not want to set up SIP digest authentication, continue to the [“Creating a New Integration with the Cisco Unified CallManager Phone System” procedure on page 12.](#)

#### (Optional) To Set Up SIP Digest Authentication

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- Step 1** On the System menu, click **Security Profile > SIP Trunk Security Profile**.
- Step 2** On the Find and List SIP Trunk Security Profiles page, click the SIP trunk security profile that you created in the [“To Create the SIP Trunk Security Profile” procedure on page 7.](#)
- Step 3** On the SIP Trunk Security Profile Configuration page, check the **Enable Digest Authentication** check box.
- Step 4** Click **Save**.
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#### (Optional) To Create the Application User

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- Step 1** On the User Management menu, click **Application User**.
- Step 2** On the Find and List Application Users page, click **Add New**.
- Step 3** On the Application User Configuration page, enter the following settings.

**Table 9** Settings for the Application User Configuration Page

Field	Setting
User ID	Enter the application user identification name. Cisco Unified CallManager does not permit modifying the user ID after it is created. You may use the following special characters: =, +, <, >, #, ;, \, , “”, and blank spaces.
Password	Enter the same password that you use for the digest credentials.
Confirm Password	Enter the password again.
Digest Credentials	Enter the name of the digest credentials.
Presence Group	Used with the Presence feature, the application user (for example, IPMASysUser) serves as the watcher because it requests status about the presence entity.  If you want the application user to receive the status of the presence entity, make sure that the Application User Presence group is allowed to view the status of the Presence group that is applied to the directory number, as indicated in the Presence Group Configuration window.
Accept Presence Subscription	Leave this check box unchecked.
Accept Out-of-Dialog REFER	Check this check box.
Accept Unsolicited Notification	Check this check box.
Accept Header Replacement	Leave this check box unchecked.

**Table 9** Settings for the Application User Configuration Page (continued)

Field	Setting
Available Devices	<p>This list box displays the devices that are available for association with this application user.</p> <p>To associate a device with this application user, select the device and click the Down arrow below this list box.</p> <p>If the device that you want to associate with this application user does not appear in this pane, click one of these buttons to search for other devices:</p> <ul style="list-style-type: none"> <li>• <b>Find More Phones</b>—Click this button to find more phones to associate with this application user. The Find and List Phones window appears to enable a phone search.</li> <li>• <b>Find More Route Points</b>—Click this button to find more phones to associate with this application user. The Find and List CTI Route Points window displays to enable a CTI route point search.</li> </ul>
Associated CAPF Profiles	In the Associated CAPF Profile pane, the Instance ID for the Application User CAPF Profile displays if you configured an Application User CAPF Profile for the user. To edit the profile, click the Instance ID; then, click Edit Profile. The Application User CAPF Profile Configuration window appears.
Groups	This list box appears after an application user has been added. The list box displays the groups to which the application user belongs.
Roles	This list box appears after an application user has been added. The list box displays the roles that are assigned to the application user.

**Step 4** Click **Save**.

## Creating a New Integration with the Cisco Unified CallManager Phone System


After ensuring that the Cisco Unified CallManager phone system and Cisco Unity Connection are ready for the integration, do the following procedures to set up the integration and to enter the port settings.

### To Create an Integration

- 
- Step 1** Log on to Cisco Unity Connection Administration.
- Step 2** In Cisco Unity Connection Administration, expand **Telephony Integrations**, then click **Phone System**.
- Step 3** On the Search Phone Systems page, on the Phone System menu, click **New Phone System**. The Phone System Integration Wizard appears.
- Step 4** On the Select Phone System Manufacturer page, in the Manufacturer field, click **Cisco Systems** and click **Next**.

- Step 5** On the Select Phone System Model page, in the Model field, click **CallManager** and click **Next**.
- Step 6** On the Set Up Phone System page, in the Phone System Name field, accept the default name or enter the descriptive name that you want, and click **Next**.
- Step 7** On the Select Port Group Template page, in the Port Group Template field, click **SIP - Session Initiation Protocol** and click **Next**.
- Step 8** On the Set Up Port Group page, enter the following settings and click **Next**.

**Table 10** Settings for the Set Up Port Group Page

Field	Setting
Port Group Name	<a descriptive name for the port group; accept the default name or enter the name that you want>
Contact Line Name	<the voice messaging line name (or pilot number) that users will use to contact Cisco Unity Connection and that Cisco Unity Connection will use to register with the Cisco Unified CallManager server>
Authenticate with SIP Proxy Server	<your indication whether Cisco Unity Connection will authenticate with the Cisco Unified CallManager server>
Authentication User Name	<the name that Cisco Unity Connection will use to authenticate with the Cisco Unified CallManager server>
Authentication Password	<the password that Cisco Unity Connection will use to authenticate with the Cisco Unified CallManager server>
Number of Ports	<the number of voice messaging ports that you want to create in this port group>
IP Address or Host Name	<the IP address (or host name) of the primary Cisco Unified CallManager server that you are integrating with Cisco Unity Connection>
Test Address	Click this button to test the IP address that you entered. The results of the test appear in the field to the right of the button.   <b>Note</b> Even though the integration is successful, the test may fail on networks where the “ping” command is disabled or ignored, or when the Cisco Unified CallManager server is not running.
Port	<the IP port of the primary Cisco Unified CallManager server that you are integrating with Cisco Unity Connection; we recommend that you use the default setting>

- Step 9** On the Confirm Phone System Settings page, confirm the settings that you have entered and click **Finish**.
- Step 10** On the Phone System Creation Summary page, click **Close**.
- Step 11** In Cisco Unity Connection Administration, expand **Telephony Integrations**, then click **Port Group**.
- Step 12** On the Search Port Groups page, click the display name of the port group that you created for the Cisco Unified CallManager integration.



**Note** By default, the display name for a port group is composed of the phone system display name followed by an incrementing number.

- Step 13** On the Port Group Basics page, on the Edit menu, click **Servers**.
- Step 14** On the Edit Servers page, do the following substeps if there are secondary Cisco Unified CallManager servers. Otherwise, continue to [Step 15](#).
- Under SIP Proxy Servers, click **Add**.
  - Enter the following settings for the secondary Cisco Unified CallManager server and click **Save**.

**Table 11** Settings for the Cisco Unified CallManager Server

Field	Setting
Order	<the order of priority for the Cisco Unified CallManager server; the lowest number is the primary Cisco Unified CallManager server, the higher numbers are the secondary servers>
IP Address or Host Name	<the IP address (or host name) of the secondary Cisco Unified CallManager server>
Port	<the IP port of the secondary Cisco Unified CallManagery server that you are integrating with Cisco Unity Connection; we recommend that you use the default setting>



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

- Repeat [Step 14a](#). and [Step 14b](#). for all remaining secondary Cisco Unified CallManager servers in the cluster.


- Step 15** In Cisco Unity Connection Administration, expand **Telephony Integrations**, then click **Port**.
- Step 16** On the Search Ports page, click the display name of the first voice messaging port that you created for this phone system integration.



**Note** By default, the display names for the voice messaging ports are composed of the port group display name followed by incrementing numbers.

- Step 17** On the Port Basics page, enter the following settings. The fields in the following table are the ones that you can change.

**Table 12** Settings for the Voice Ports

Field	Considerations
Enabled	Check this check box.
Answer Calls	Check this check box.   <b>Caution</b> All voice messaging ports connecting to the Cisco Unified CallManager server must have the Answer Calls box checked. Otherwise, calls to Cisco Unity Connection may not be answered.
Perform Message Notification	Check this check box to designate the port for notifying users of messages.

**Table 12** Settings for the Voice Ports (continued)

Field	Considerations
Send MWI Requests	Check this check box to designate the port for turning MWIs on and off.
Allow TRAP Connections	Check this check box so that users can use the port for recording and playback through the phone in Cisco Unity Connection web applications.

**Step 18** Click **Save**.

**Step 19** Click **Next**.

**Step 20** Repeat [Step 17](#) through [Step 19](#) for all remaining voice messaging ports for the phone system.

**Step 21** If another phone system integration exists, in Cisco Unity Connection Administration, expand **Telephony Integrations**, then click **Trunk**. Otherwise, skip to [Step 25](#).

**Step 22** On the Search Phone System Trunks page, on the Phone System Trunk menu, click **New Phone System Trunk**.

**Step 23** On the New Phone System Trunk page, enter the following settings for the phone system trunk and click **Save**.

**Table 13** Settings for the Phone System Trunk

Field	Setting
From Phone System	<the display name of the phone system that you are creating a trunk for>
To Phone System	<the display name of the previously existing phone system that the trunk will connect to>
Trunk Access Code	<the extra digits that Cisco Unity Connection must dial to transfer calls through the gateway to extensions on the previously existing phone system>

**Step 24** Repeat [Step 22](#) and [Step 23](#) for all remaining phone system trunks that you want to create.

**Step 25** If prompted to restart Cisco Unity Connection, in the Windows task bar, right-click the **Cisco Unity Connection** icon and click **Restart > Voice Processing Server Role**.

**Step 26** When prompted to confirm stopping the Voice Processing server role, click **Yes**.

**Step 27** In Cisco Unity Connection Administration, in the Related Links drop-down list, click **Check Telephony Configuration** and click **Go** to confirm the phone system integration settings.

If the test is not successful, the Task Execution Results displays one or more messages with troubleshooting steps. After correcting the problems, test the connection again.

**Step 28** In the Task Execution Results window, click **Close**.

**Step 29** Log off Cisco Unity Connection Administration.

## Testing the Integration

To test whether Cisco Unity Connection and the phone system are integrated correctly, do the following procedures in the order listed.

If any of the steps indicate a failure, refer to the following documentation as applicable:

- The installation guide for the phone system.
- The setup information earlier in this guide.

### To Set Up the Test Configuration

**Step 1** Set up two test extensions (Phone 1 and Phone 2) on the same phone system that Cisco Unity Connection is connected to.

**Step 2** Set Phone 1 to forward calls to the Cisco Unity Connection pilot number when calls are not answered.



**Caution** The phone system must forward calls to the Cisco Unity Connection pilot number in no fewer than four rings. Otherwise, the test may fail.

**Step 3** To create a test user for testing, in Cisco Unity Connection Administration, expand **Users**, then click **Users**.

**Step 4** On the Search Users page, on the User menu, click **New User**.

**Step 5** On the New User page, enter the following settings.

**Table 14** Settings for the New User Page

Field	Setting
User Type	<b>User with Voice Mailbox</b>
Based on Template	<the applicable user template>
Alias	<b>testuser</b>
First Name	<b>Test</b>
Last Name	<b>User</b>
Display Name	<b>Test User</b>
Extension	<the extension of Phone 1>

**Step 6** Click **Save**.

**Step 7** On the Edit User Basics page, in the Voice Name field, record a voice name for the test user.

**Step 8** In the Phone System field, confirm that the phone system selected is the phone system that Phone 1 is connected to.

**Step 9** Uncheck the **Set for Self-enrollment at Next Login** check box.

**Step 10** Click **Save**.

**Step 11** On the Edit menu, click **Message Waiting Indicators**.

**Step 12** On the Message Waiting Indicators page, click the message waiting indicator. If no message waiting indication is in the table, click **Add New**.

**Step 13** On the Edit Message Waiting Indicator page, enter the following settings.

**Table 15**      **Settings for the Edit MWI Page**

Field	Setting
Enabled	Check this check box to enable MWIs for the test user.
Display Name	Accept the default or enter a different name.
Inherit User's Extension	Check this check box to enable MWIs on Phone 1.

- Step 14** Click **Save**.
- Step 15** On the Edit menu, click **Transfer Options**.
- Step 16** On the Transfer Options page, click the active option.
- Step 17** On the Edit Transfer Option page, under Transfer Action, click the **Extension** option and enter the extension of Phone 1.
- Step 18** In the Transfer Type field, click **Release to Switch**.
- Step 19** Click **Save**.
- Step 20** Minimize the Cisco Unity Connection Administration window.  
Do not close the Cisco Unity Connection Administration window because you will use it again in a later procedure.
- Step 21** On the Cisco Unity Connection desktop, double-click the **Tools Depot** icon.
- Step 22** In the left pane of the Tools Depot window, expand **Switch Integration Tools**, then double-click **Port Status Monitor**. The Port Status Monitor window appears.
- Step 23** On the Ports menu, click **Start All**, and arrange the port monitors so that you can notice which port will handle the calls that you will make.

---

#### To Test an External Call with Release Transfer

- Step 1** From Phone 2, enter the access code necessary to get an outside line, then enter the number outside callers use to dial directly to Cisco Unity Connection.
- Step 2** In the Port Status Monitor, note which port handles this call.
- Step 3** When you hear the opening greeting, enter the extension for Phone 1. Hearing the opening greeting means that the port is configured correctly.
- Step 4** Confirm that Phone 1 rings and that you hear a ringback tone on Phone 2. Hearing a ringback tone means that Cisco Unity Connection correctly released the call and transferred it to Phone 1.
- Step 5** Leaving Phone 1 unanswered, confirm that the state of the port handling the call changes to "Idle." This state means that release transfer is successful.
- Step 6** Confirm that, after the number of rings that the phone system is set to wait, the call is forwarded to Cisco Unity Connection and that you hear the greeting for the test user. Hearing the greeting means that the phone system forwarded the unanswered call and the call-forward information to Cisco Unity Connection, which correctly interpreted the information.
- Step 7** On the Port Status Monitor, note which port handles this call.
- Step 8** Leave a message for the test user and hang up Phone 2.

- Step 9** In the Port Status Monitor, confirm that the state of the port handling the call changes to “Idle.” This state means that the port was successfully released when the call ended.
- Step 10** Confirm that the MWI on Phone 1 is activated. The activated MWI means that the phone system and Cisco Unity Connection are successfully integrated for turning on MWIs.
- 

#### To Test Listening to Messages

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- Step 1** From Phone 1, enter the internal pilot number for Cisco Unity Connection.
- Step 2** When asked for your password, enter the password for the test user. Hearing the request for your password means that the phone system sent the necessary call information to Cisco Unity Connection, which correctly interpreted the information.
- Step 3** Confirm that you hear the recorded voice name for the test user (if you did not record a voice name for the test user, you will hear the extension number for Phone 1). Hearing the voice name means that Cisco Unity Connection correctly identified the user by the extension.
- Step 4** Listen to the message.
- Step 5** After listening to the message, delete the message.
- Step 6** Confirm that the MWI on Phone 1 is deactivated. The deactivated MWI means that the phone system and Cisco Unity Connection are successfully integrated for turning off MWIs.
- Step 7** Hang up Phone 1.
- Step 8** On the Port Status Monitor, confirm that the state of the port handling the call changes to “Idle.” This state means that the port was successfully released when the call ended.
- 

#### To Set Up Supervised Transfer on Cisco Unity Connection

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- Step 1** In Cisco Unity Connection Administration, on the Edit Transfer Option page for the test user, in the Transfer Type field, click **Supervise Transfer**.
- Step 2** In the Rings to Wait For field, enter **3**.
- Step 3** Click **Save**.
- Step 4** Minimize the Cisco Unity Connection Administration window.
- Do not close the Cisco Unity Connection Administration window because you will use it again in a later procedure.
- 

#### To Test Supervised Transfer

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- Step 1** From Phone 2, enter the access code necessary to get an outside line, then enter the number outside callers use to dial directly to Cisco Unity Connection.
- Step 2** On the Port Status Monitor, note which port handles this call.
- Step 3** When you hear the opening greeting, enter the extension for Phone 1. Hearing the opening greeting means that the port is configured correctly.

- Step 4** Confirm that Phone 1 rings and that you do not hear a ringback tone on Phone 2. Instead, you should hear the indication your phone system uses to mean that the call is on hold (for example, music).
- Step 5** Leaving Phone 1 unanswered, confirm that the state of the port handling the call remains “Busy.” This state and hearing an indication that you are on hold mean that Cisco Unity Connection is supervising the transfer.
- Step 6** Confirm that, after three rings, you hear the greeting for the test user. Hearing the greeting means that Cisco Unity Connection successfully recalled the supervised-transfer call.
- Step 7** During the greeting, hang up Phone 2.
- Step 8** On the Port Status Monitor, confirm that the state of the port handling the call changes to “Idle.” This state means that the port was successfully released when the call ended.
- Step 9** Exit the Port Status Monitor.
- 

#### To Delete the Test User

---

- Step 1** In Cisco Unity Connection Administration, expand **Users**, then click **Users**.
- Step 2** On the Search Users page, check the check box to the left of the test user.
- Step 3** Click **Delete Selected**.
- 

## (Multiple Integrations Only) Adding New User Templates

When you create the first phone system integration, this phone system is automatically selected in the default user template. The users that you add after creating this phone system integration will be assigned to this phone system by default.

However, for each additional phone system integration that you create, you must add the applicable new user templates that will assign users to the new phone system. You must add the new templates before you add new users who will be assigned to the new phone system.

For details on adding new user templates, refer to the “Adding, Changing, or Deleting an Account Template” chapter in the *Cisco Unity Connection User Moves, Adds, and Changes Guide* at [http://www.cisco.com/en/US/products/ps6509/prod\\_maintenance\\_guides\\_list.html](http://www.cisco.com/en/US/products/ps6509/prod_maintenance_guides_list.html).

For details on selecting a user template when adding a new user, refer to the applicable chapter for adding user accounts in the *Cisco Unity Connection User Moves, Adds, and Changes Guide* at [http://www.cisco.com/en/US/products/ps6509/prod\\_maintenance\\_guides\\_list.html](http://www.cisco.com/en/US/products/ps6509/prod_maintenance_guides_list.html).

## Changing the Number of Voice Messaging Ports

If you are adding voice messaging ports, do the “[To Add Voice Messaging Ports in Cisco Unity Connection Administration](#)” procedure on page 20.

If you are deleting voice messaging ports, do the “[To Delete Voice Messaging Ports in Cisco Unity Connection Administration](#)” procedure on page 20.

### To Add Voice Messaging Ports in Cisco Unity Connection Administration

- 
- Step 1** If the Cisco Unity Connection license does not enable the additional voice messaging ports you added, see your sales representative to request the applicable license.
- Step 2** When you have the license, log on to Cisco Unity Connection Administration.
- Step 3** In Cisco Unity Connection Administration, expand **System Settings**, then click **Licenses**.
- Step 4** On the License page, on the License menu, click **Add New License**.
- Step 5** On the Add New License page, click **Browse**.
- Step 6** In the Choose File dialog box, browse to license file and click **Open**.
- Step 7** On the Add New License page, click **Add**.
- Step 8** On the Licenses page, check the check box for the license file that you added in [Step 7](#) and click **Install Selected**.
- Step 9** In Cisco Unity Connection Administration, expand **Telephony Integrations**, then click **Port**.
- Step 10** On the Search Ports page, under Port Search Results, click **Add New**.
- Step 11** On the New Port page, enter the applicable settings and click **Save**.




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**Caution** Make sure 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 will not function correctly. For details, see to the “Planning How the Voice Messaging Ports Will be Used by Cisco Unity Connection” section.

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- Step 12** If prompted to restart Cisco Unity Connection, in the Windows task bar, right-click the **Cisco Unity Connection** icon and click **Restart > Voice Processing Server Role**.
- Step 13** When prompted to confirm stopping the Voice Processing server role, click **Yes**.
- Step 14** Log off Cisco Unity Connection Administration.
- 

### To Delete Voice Messaging Ports in Cisco Unity Connection Administration

- 
- Step 1** Log on to the Cisco Unity Connection Administration.
- Step 2** Go to the **Telephony Integrations > Port** page.
- Step 3** Under Port Search Results, check the check boxes next to the voice messaging ports that you want to delete.
- Step 4** Click **Delete Selected**.
- Step 5** 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.
- Step 6** In the Windows task bar, right-click the **Cisco Unity Connection** icon and click **Restart > Voice Processing Server Role**.
- Step 7** When prompted to confirm stopping the Voice Processing server role, click **Yes**.
- Step 8** In Cisco Unity Connection Administration, in the Related Links drop-down list, click **Check Telephony Configuration** and click **Go** to confirm the phone system integration settings.

If the test is not successful, the Task Execution Results displays one or more messages with troubleshooting steps. After correcting the problems, test the connection again.

**Step 9** In the Task Execution Results window, click **Close**.

**Step 10** Log off the Cisco Unity Connection Administration.

## Adding a Cisco Unified CallManager Express Server to a Cisco Unified CallManager Phone System Integration

Cisco Unity Connection can integrate a Cisco Unified CallManager phone system integration that has a port group of Cisco Unified CallManager servers and a port group of a Cisco Unified CallManager Express server. This configuration is typically used to ensure call processing functionality at a branch office when the WAN link is down.

There are, however, the following considerations:

- The version of Cisco Unified CallManager Express and the version of the Cisco Unity-CM TSP must be a supported combination in the *Compatibility Matrix: Cisco Unity Connection, the Cisco Unity-CM TSP, Cisco Unified CallManager, and Cisco Unified CallManager Express* at [http://www.cisco.com/univercd/cc/td/doc/product/voice/c\\_unity/cmptblty/tspmtrx.htm](http://www.cisco.com/univercd/cc/td/doc/product/voice/c_unity/cmptblty/tspmtrx.htm).
- The Cisco Unified CallManager phone system integration is typically already created before adding the Cisco Unified CallManager Express server.
- The Cisco Unified CallManager Express server is in its own port group, which is separate from the port group for the Cisco Unified CallManager servers.
- The Cisco Unified CallManager Express port group has its own voice messaging ports that connect only to the Cisco Unified CallManager Express server.

To add a Cisco Unified CallManager Express server to a Cisco Unified CallManager phone system integration, do the following procedure.

### To Add a Cisco Unified CallManager Express Server to a Cisco Unified CallManager Phone System Integration

**Step 1** Log on to Cisco Unity Connection Administration.

**Step 2** In Cisco Unity Connection Administration, expand **Telephony Integrations**, then click **Port Group**.

**Step 3** On the Search Port Groups page, under Port Group Search Results, click **Add New**.

**Step 4** On the New Port Group page, enter the following settings.

**Table 16** Settings for the Cisco Unified CallManager Express Server

Field	Setting
Display Name	Enter the display name for the port group. Accept the default name, which is composed of the phone system display name followed by an incrementing number, or enter another descriptive name.
Phone System	Click the name of the Cisco Unified CallManager phone system.
Create From	Click <b>Port Group Template</b> and, in the drop-down box, click <b>SCCP - Skinny Client Control Protocol</b> .

**Table 16** Settings for the Cisco Unified CallManager Express Server (continued)

Field	Setting
Device Name Prefix	Enter the prefix that was added in Cisco Unified CallManager Express to the device name for voice messaging ports; this prefix must match the prefix used by Cisco Unified CallManager Express.
MWI On Extension	Enter the extension that you specified in Cisco Unified CallManager Express for turning MWIs on.
MWI Off Extension	Enter the extension that you specified in Cisco Unified CallManager Express for turning MWIs off.
IP Address or Host Name	Enter the IP address (or host name) of the Cisco Unified CallManager Express server that you are adding to the Cisco Unified CallManager phone system integration.
Port	Enter the TCP port of the Cisco Unified CallManager Express server that you are adding to the Cisco Unified CallManager phone system integration. We recommend that you use the default setting.
TLS Port	Enter the TLS port of the Cisco Unified CallManager Express server that you are adding to the Cisco Unified CallManager phone system integration. We recommend that you use the default setting.

**Step 5** Click **Save**.

**Step 6** On the Port Group Basics page, on the Edit menu, click **Servers**.

**Step 7** On the Edit Servers page, in the Server Type field, click **Cisco Unified CallManager Express**.



**Note** You can click **Ping** to verify the IP address of the Cisco Unified CallManager Express server.

**Step 8** Click **Save**.

**Step 9** On the Edit menu, click **Advanced Settings**.

**Step 10** On the Edit Advanced Settings page, in the Delay Before Opening Greeting field, enter **1000** and click **Save**.

**Step 11** In Cisco Unity Connection Administration, expand **Telephony Integrations**, then click **Port**.

**Step 12** On the Search Ports page, under Port Search Results, click **Add New**.

**Step 13** On the New Port page, enter the following settings.

**Table 17** Settings for the New Port Page

Field	Setting
Number of Ports	Enter the number of voice messaging ports that you want to create on Cisco Unity Connection for connecting to the Cisco Unified CallManager Express server.
Phone System	Click the display name of the Cisco Unified CallManager phone system integration.
Port Group	Click the display name of the port group that you created for the Cisco Unified CallManager Express server in <a href="#">Step 5</a> .

**Step 14** Click **Save**.

**Step 15** On the Search Ports page, click the display name of the first voice messaging port that you created for the Cisco Unified CallManager Express port group.



**Note** By default, the display names for the voice messaging ports are composed of the port group display name followed by incrementing numbers.

**Step 16** On the Port Basics page, set the voice messaging port settings as applicable. The fields in the following table are the ones that you can change.

**Table 18** *Settings for the Voice Ports*

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.
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 unidentified 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	Check this check box to designate the port for turning MWIs on and off. Assign Send MWI Requests to the least busy ports.
Allow TRAP Connections	Check this check box so that users can use the port for recording and playback through the phone in Cisco Unity Connection web applications. Assign Allow TRAP Connections to the least busy ports.
Outgoing Hunt Order	Enter the priority order in which Cisco Unity Connection will use 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, Cisco Unity Connection will use the port that has been idle the longest.
Security Mode	Click <b>Non-secure</b> . (Cisco Unified CallManager authentication and encryption are not available.)

**Step 17** Click **Save**.

**Step 18** Click **Next**.

**Step 19** Repeat [Step 16](#) through [Step 18](#) for all remaining voice messaging ports in the Cisco Unified CallManager Express port group.

**Step 20** In the Windows task bar, right-click the **Cisco Unity Connection** icon and click **Restart > Voice Processing Server Role**.

**Step 21** When prompted to confirm stopping the Voice Processing server role, click **Yes**.

**Step 22** In Cisco Unity Connection Administration, in the Related Links drop-down list, click **Test Port Group** and click **Go** to confirm the Cisco Unified CallManager port group settings.

**Step 23** When prompted that the test will terminate call in progress, click **OK**.

If the test is not successful, the Task Execution Results displays one or more messages with troubleshooting steps. After correcting the problems, test the connection again.

**Step 24** In the Task Execution Results window, click **Close**.

**Step 25** Log off Cisco Unity Connection Administration.

## Appendix: Documentation and Technical Assistance

### Conventions

The *Cisco Unified CallManager 5.0 SIP Trunk Integration Guide for Cisco Unity Connection 1.1* uses the following conventions.

**Table 19** *Cisco Unified CallManager 5.0 SIP Trunk Integration Guide for Cisco Unity Connection 1.1 Conventions*

Convention	Description
boldfaced text	Boldfaced text is used for: <ul style="list-style-type: none"> <li>Key and button names. (Example: Click <b>OK</b>.)</li> <li>Information that you enter. (Example: Enter <b>Administrator</b> in the User Name box.)</li> </ul>
< > (angle brackets)	Angle brackets are used around parameters for which you supply a value. (Example: In the Command Prompt window, enter <b>ping &lt;IP address&gt;</b> .)
- (hyphen)	Hyphens separate keys that must be pressed simultaneously. (Example: Press <b>Ctrl-Alt-Delete</b> .)
> (right angle bracket)	A right angle bracket is used to separate selections that you make: <ul style="list-style-type: none"> <li>On menus. (Example: On the Windows Start menu, click <b>Settings &gt; Control Panel &gt; Phone and Modem Options</b>.)</li> <li>In the navigation bar of Cisco Unity Connection Administration. (Example: In Cisco Unity Connection Administration, expand <b>System Settings &gt; Advanced</b>.)</li> </ul>
[x] (square brackets)	Square brackets enclose an optional element (keyword or argument). (Example: [reg-e164])
[x   y] (vertical line)	Square brackets enclosing keywords or arguments separated by a vertical line indicate an optional choice. (Example: [transport tcp   transport udp])
{ x   y } (braces)	Braces enclosing keywords or arguments separated by a vertical line indicate a required choice. (Example: {tcp   udp})

The *Cisco Unified CallManager 5.0 SIP Trunk Integration Guide for Cisco Unity Connection 1.1* also uses the following conventions:

**Note**

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

**Caution**

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

For descriptions and URLs of Cisco Unity Connection documentation on Cisco.com, see the *About Cisco Unity Connection Documentation*. The document is shipped with Cisco Unity Connection and is available at [http://www.cisco.com/en/US/products/ps6509/products\\_documentation\\_roadmaps\\_list.html](http://www.cisco.com/en/US/products/ps6509/products_documentation_roadmaps_list.html).

## Obtaining Documentation

Cisco documentation and additional literature are available on Cisco.com. This section explains the product documentation resources that Cisco offers.

### Cisco.com

You can access the most current Cisco documentation at this URL:

<http://www.cisco.com/techsupport>

You can access the Cisco website at this URL:

<http://www.cisco.com>

You can access international Cisco websites at this URL:

[http://www.cisco.com/public/countries\\_languages.shtml](http://www.cisco.com/public/countries_languages.shtml)

### Product Documentation DVD

The Product Documentation DVD is a library of technical product documentation on a portable medium. The DVD enables you to access installation, configuration, and command guides for Cisco hardware and software products. With the DVD, you have access to the HTML documentation and some of the PDF files found on the Cisco website at this URL:

<http://www.cisco.com/univercd/home/home.htm>

The Product Documentation DVD is created and released regularly. DVDs are available singly or by subscription. Registered Cisco.com users can order a Product Documentation DVD (product number DOC-DOCDVD= or DOC-DOCDVD=SUB) from Cisco Marketplace at the Product Documentation Store at this URL:

<http://www.cisco.com/go/marketplace/docstore>

## Ordering Documentation

You must be a registered Cisco.com user to access Cisco Marketplace. Registered users may order Cisco documentation at the Product Documentation Store at this URL:

<http://www.cisco.com/go/marketplace/docstore>

If you do not have a user ID or password, you can register at this URL:

<http://tools.cisco.com/RPF/register/register.do>

## Documentation Feedback

You can provide feedback about Cisco technical documentation on the Cisco Support site area by entering your comments in the feedback form available in every online document.

## Cisco Product Security Overview

Cisco provides a free online Security Vulnerability Policy portal at this URL:

[http://www.cisco.com/en/US/products/products\\_security\\_vulnerability\\_policy.html](http://www.cisco.com/en/US/products/products_security_vulnerability_policy.html)

From this site, you will find information about how to do the following:

- Report security vulnerabilities in Cisco products
- Obtain assistance with security incidents that involve Cisco products
- Register to receive security information from Cisco

A current list of security advisories, security notices, and security responses for Cisco products is available at this URL:

<http://www.cisco.com/go/psirt>

To see security advisories, security notices, and security responses as they are updated in real time, you can subscribe to the Product Security Incident Response Team Really Simple Syndication (PSIRT RSS) feed. Information about how to subscribe to the PSIRT RSS feed is found at this URL:

[http://www.cisco.com/en/US/products/products\\_psirt\\_rss\\_feed.html](http://www.cisco.com/en/US/products/products_psirt_rss_feed.html)

## Reporting Security Problems in Cisco Products

Cisco is committed to delivering secure products. We test our products internally before we release them, and we strive to correct all vulnerabilities quickly. If you think that you have identified a vulnerability in a Cisco product, contact PSIRT:

- For emergencies only — [security-alert@cisco.com](mailto:security-alert@cisco.com)

An emergency is either a condition in which a system is under active attack or a condition for which a severe and urgent security vulnerability should be reported. All other conditions are considered nonemergencies.

- For nonemergencies — [psirt@cisco.com](mailto:psirt@cisco.com)

In an emergency, you can also reach PSIRT by telephone:

- 1 877 228-7302

- 1 408 525-6532

**Tip**

We encourage you to use Pretty Good Privacy (PGP) or a compatible product (for example, GnuPG) to encrypt any sensitive information that you send to Cisco. PSIRT can work with information that has been encrypted with PGP versions 2.x through 9.x.

Never use a revoked encryption key or an expired encryption key. The correct public key to use in your correspondence with PSIRT is the one linked in the Contact Summary section of the Security Vulnerability Policy page at this URL:

[http://www.cisco.com/en/US/products/products\\_security\\_vulnerability\\_policy.html](http://www.cisco.com/en/US/products/products_security_vulnerability_policy.html)

The link on this page has the current PGP key ID in use.

If you do not have or use PGP, contact PSIRT to find other means of encrypting the data before sending any sensitive material.

## Product Alerts and Field Notices

Modifications to or updates about Cisco products are announced in Cisco Product Alerts and Cisco Field Notices. You can receive these announcements by using the Product Alert Tool on Cisco.com. This tool enables you to create a profile and choose those products for which you want to receive information.

To access the Product Alert Tool, you must be a registered Cisco.com user. Registered users can access the tool at this URL:

<http://tools.cisco.com/Support/PAT/do/ViewMyProfiles.do?local=en>

To register as a Cisco.com user, go to this URL:

<http://tools.cisco.com/RPF/register/register.do>

## Obtaining Technical Assistance

Cisco Technical Support provides 24-hour-a-day award-winning technical assistance. The Cisco Support website on Cisco.com features extensive online support resources. In addition, if you have a valid Cisco service contract, Cisco Technical Assistance Center (TAC) engineers provide telephone support. If you do not have a valid Cisco service contract, contact your reseller.

## Cisco Support Website

The Cisco Support website provides online documents and tools for troubleshooting and resolving technical issues with Cisco products and technologies. The website is available 24 hours a day at this URL:

<http://www.cisco.com/en/US/support/index.html>

Access to all tools on the Cisco Support website requires a Cisco.com user ID and password. If you have a valid service contract but do not have a user ID or password, you can register at this URL:

<http://tools.cisco.com/RPF/register/register.do>

**Note**

Before you submit a request for service online or by phone, use the **Cisco Product Identification Tool** to locate your product serial number. You can access this tool from the Cisco Support website by clicking the **Get Tools & Resources** link, clicking the **All Tools (A-Z)** tab, and then choosing **Cisco Product Identification Tool** from the alphabetical list. This tool offers three search options: by product ID or model name; by tree view; or, for certain products, by copying and pasting **show** command output. Search results show an illustration of your product with the serial number label location highlighted. Locate the serial number label on your product and record the information before placing a service call.

**Tip****Displaying and Searching on Cisco.com**

If you suspect that the browser is not refreshing a web page, force the browser to update the web page by holding down the Ctrl key while pressing **F5**.

To find technical information, narrow your search to look in technical documentation, not the entire Cisco.com website. After using the Search box on the Cisco.com home page, click the **Advanced Search** link next to the Search box on the resulting page and then click the **Technical Support & Documentation** radio button.

To provide feedback about the Cisco.com website or a particular technical document, click **Contacts & Feedback** at the top of any Cisco.com web page.

## Submitting a Service Request

Using the online TAC Service Request Tool is the fastest way to open S3 and S4 service requests. (S3 and S4 service requests are those in which your network is minimally impaired or for which you require product information.) After you describe your situation, the TAC Service Request Tool provides recommended solutions. If your issue is not resolved using the recommended resources, your service request is assigned to a Cisco engineer. The TAC Service Request Tool is located at this URL:

<http://www.cisco.com/techsupport/servicerequest>

For S1 or S2 service requests, or if you do not have Internet access, contact the Cisco TAC by telephone. (S1 or S2 service requests are those in which your production network is down or severely degraded.) Cisco engineers are assigned immediately to S1 and S2 service requests to help keep your business operations running smoothly.

To open a service request by telephone, use one of the following numbers:

Asia-Pacific: +61 2 8446 7411

Australia: 1 800 805 227

EMEA: +32 2 704 55 55

USA: 1 800 553 2447

For a complete list of Cisco TAC contacts, go to this URL:

<http://www.cisco.com/techsupport/contacts>

## Definitions of Service Request Severity

To ensure that all service requests are reported in a standard format, Cisco has established severity definitions.

Severity 1 (S1)—An existing network is “down” or there is a critical impact to your business operations. You and Cisco will commit all necessary resources around the clock to resolve the situation.

Severity 2 (S2)—Operation of an existing network is severely degraded, or significant aspects of your business operations are negatively affected by inadequate performance of Cisco products. You and Cisco will commit full-time resources during normal business hours to resolve the situation.

Severity 3 (S3)—Operational performance of the network is impaired while most business operations remain functional. You and Cisco will commit resources during normal business hours to restore service to satisfactory levels.

Severity 4 (S4)—You require information or assistance with Cisco product capabilities, installation, or configuration. There is little or no effect on your business operations.

## Obtaining Additional Publications and Information

Information about Cisco products, technologies, and network solutions is available from various online and printed sources.

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- The *Cisco Product Quick Reference Guide* is a handy, compact reference tool that includes brief product overviews, key features, sample part numbers, and abbreviated technical specifications for many Cisco products that are sold through channel partners. It is updated twice a year and includes the latest Cisco channel product offerings. To order and find out more about the *Cisco Product Quick Reference Guide*, go to this URL:  
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- *Internet Protocol Journal* is a quarterly journal published by Cisco for engineering professionals involved in designing, developing, and operating public and private internets and intranets. You can access the *Internet Protocol Journal* at this URL:  
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