Cisco Unified Communications Manager 6.x SIP Trunk Integration Guide for Cisco Unity Connection 1.2

Revised November 21, 2007

This document provides instructions for integrating Cisco Unified Communications Manager (CM) (formerly known as Cisco Unified CallManager) with Cisco Unity Connection by a SIP trunk.

Cisco Unity Connection supports a SIP trunk integration when the Cisco Unified CM phone system has only SIP phones (best practice) or has both SCCP and SIP phones.

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

Integration Tasks

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

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 CM phone system.

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 2.
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.
3. Program Cisco Unified CM. See the "Programming the Cisco Unified Communications Manager Phone System" section on page 6.

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4. Create the integration. See the "Creating a New Integration with the Cisco Unified Communications Manager Phone System" section on page 12.

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

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.

Requirements

The Cisco Unified Communications Manager integration supports configurations of the following components:

Phone System

- A Cisco IP telephony applications server consisting of Cisco Unified CM 6.x, running on a Cisco Media Convergence Server (MCS) or customer-provided server meeting approved Cisco configuration standards.

  For details on compatible versions of Cisco Unified CM, refer to the *SIP Trunk Compatibility Matrix: Cisco Unity Connection, Cisco Unified Communications Manager, and Cisco Unified Communications Manager Express* at http://www.cisco.com/en/US/products/ps6509/products_device_support_tables_list.html.
- For the Cisco Unified CM extensions, one of the following configurations:
  - *(Best practice)* Only SIP phones that support DTMF relay as described in RFC-2833.
  - Both SCCP and SIP phones.

    Note that older SCCP phone models may require a Media Termination Point (MTP) to function correctly.
- A LAN connection in each location where you will plug the applicable phone into the network.
- For multiple Cisco Unified CM clusters, the capability for users to dial an extension on another Cisco Unified CM 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 CM, refer to the *SIP Trunk Compatibility Matrix: Cisco Unity Connection, Cisco Unified Communications Manager, and Cisco Unified Communications Manager Express* at http://www.cisco.com/en/US/products/ps6509/products_device_support_tables_list.html.
- Cisco Unity Connection installed and ready for the integration, as described in the *Installation Guide for Cisco Unity Connection* at http://www.cisco.com/en/US/products/ps6509/prod_installation_guides_list.html.
- A license that enables the applicable number of voice messaging ports.

Integration Description

The Cisco Unified Communications Manager integration uses a LAN or WAN to connect Cisco Unity Connection and the phone system. A gateway provides connections to the PSTN.

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 Communications Manager 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 the user based on the extension from which the call originated; a password may be required)
- Identified user messaging (Cisco Unity Connection identifies the 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 CM (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.
- **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.

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/en/US/products/sw/voicesw/ps2169/products_installation_and_configuration_guides_list.html.

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.

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. **Caution** All voice messaging ports connecting to the Cisco Unified CM 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 Communications Manager 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 CM Help.

To Create the SIP Trunk Security Profile

Step 1 In Cisco Unified CM 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 **Connection SIP Trunk Security Profile** or another name.
Description	Enter **SIP trunk security profile for Cisco Unity Connection** or another description.
Device Security Mode	Accept the default of **Non Secure**.
Accept Out-of-Dialog REFER	Check this check box.
Accept Unsolicited Notification	Check this check box.
Accept Header Replacement	Check this check box.

Step 4 Click **Save**.

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 **Connection SIP Profile** or another name.
Description	Enter **SIP profile for Cisco Unity Connection** or another description.

Step 4 Click **Save**.

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 **Connection_SIP_Trunk** or another name.
Description	Enter **SIP trunk for Cisco Unity Connection** or another description.

Step 6 If user phones are contained in a calling search space, under Inbound Calls, enter the following settings. Otherwise, continue to Step 7.

Table 5 Settings for Inbound Calls on the Trunk Configuration Page

Field	Setting
Calling Search Space	Click the name of the calling search space that contains the user phones.
Redirecting Diversion Header Delivery - Inbound	Check this check box.

Step 7 Under Outbound Calls, check the **Redirecting Diversion Header Delivery - Outbound** check box.

Step 8 Under SIP Information, enter the following settings.

Table 6 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 CM will connect.
Destination Port	We recommend that you accept the default of **5060**.
SIP Trunk Security Profile	Click the name of the SIP trunk security profile that you created in the "To Create the SIP Trunk Security Profile" procedure on page 6. 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 "To Create the SIP Profile" procedure on page 7. For example, click "Cisco Unity Connection SIP Profile."

Step 9 Adjust any other settings that are needed for your site.

Step 10 Click **Save**.

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 7 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 "To Create the SIP Trunk" procedure on page 7. For example, click "Connection_SIP_Trunk."

Step 4 Click **Save**.

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 8 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 "To Create a Route Pattern" procedure on page 8.
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 **Connection Pilot** 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**.

To Create the Voice Mail Profile

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 9 Settings for the Voice Mail Profile Configuration Page

Field	Setting
Voice Mail Profile Name	Enter **Connection Profile** or another name to identify the voice mail profile.
Description	Enter **Profile for Cisco Unity Connection** or another description.
Voice Mail Pilot	Click the voice mail pilot number that you defined in the "To Create the Voice Mail Pilot" procedure on page 9.
Voice Mail Box Mask	When multitenant services are not enabled on Cisco Unified CM, 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. Each tenant also uses its own 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**.

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 Communications Manager Phone System" procedure on page 12.

(Optional) To Set Up SIP Digest Authentication

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 6.

Step 3 On the SIP Trunk Security Profile Configuration page, check the **Enable Digest Authentication** check box.

Step 4 Click **Save**.

(Optional) To Create the Application User

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 10 Settings for the Application User Configuration Page

Field	Setting
User ID	Enter the application user identification name. Cisco Unified CM 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.
Available Devices	This list box displays the devices that are available for association with this application user. To associate a device with this application user, select the device and click the Down arrow below this list box. 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: • **Find More Phones**—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. • **Find More Route Points**—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.
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 Communications Manager Phone System

After ensuring that the Cisco Unified CM phone system and Cisco Unity Connection are ready for the integration, do the following procedure 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 **Cisco Unified 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 11 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 CM server>
Authenticate with SIP Proxy Server	<your indication whether Cisco Unity Connection will authenticate with the Cisco Unified CM server>
Authentication User Name	<the name that Cisco Unity Connection will use to authenticate with the Cisco Unified CM server>
Authentication Password	<the password that Cisco Unity Connection will use to authenticate with the Cisco Unified CM 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 CM server that you are integrating with Cisco Unity Connection>

Table 11 Settings for the Set Up Port Group Page (continued)

Field	**Setting**
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. **Note** 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 CM server is not running.
Port	<the IP port of the primary Cisco Unified CM 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 CM 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 CM servers. Otherwise, continue to Step 15.

a. Under SIP Proxy Servers, click **Add**.

b. Enter the following settings for the secondary Cisco Unified CM server and click **Save**.

Table 12 Settings for the Cisco Unified CM Server

Field	**Setting**
Order	<the order of priority for the Cisco Unified CM server; the lowest number is the primary Cisco Unified CM server, the higher numbers are the secondary servers>
IP Address or Host Name	<the IP address (or host name) of the secondary Cisco Unified CM server>
Port	<the IP port of the secondary Cisco Unified CMy 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 CM server.

c. Repeat Step 14a. and Step 14b. for all remaining secondary Cisco Unified CM 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 13 Settings for the Voice Ports

Field	Considerations
Enabled	Check this check box.
Answer Calls	Check this check box. **Caution** All voice messaging ports connecting to the Cisco Unified CM 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.

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 14 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 15 Settings for the New User Page

Field	Setting
User Type	**User with Voice Mailbox**
Based on Template	<the applicable user template>
Alias	**testuser**
First Name	**Test**
Last Name	**User**

Table 15 Settings for the New User Page (continued)

Field	Setting
Display Name	**Test User**
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 16 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 Log on to Real-Time Monitoring Tool (RTMT).

Step 22 On the Unity Connection menu, click **Port Monitor**. The Port Monitor tool appears in the right pane.

Step 23 In the right pane, click **Start Polling**. the Port Monitor will display which port is handling 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 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 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 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

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 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

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

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 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 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 Click **Stop Polling**.

Step 10 Exit RTMT.

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 *User Moves, Adds, and Changes Guide for Cisco Unity Connection* at 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 *User Moves, Adds, and Changes Guide for Cisco Unity Connection* at http://www.cisco.com/en/US/products/ps6509/prod_maintenance_guides_list.html.

Appendix: Documentation and Technical Assistance

Documentation Conventions

The *Cisco Unified Communications Manager 6.x SIP Trunk Integration Guide for Cisco Unity Connection 1.2* uses the following conventions.

Table 17 Cisco Unified Communications Manager 6.x SIP Trunk Integration Guide for Cisco Unity Connection 1.2 Conventions

Convention	Description
boldfaced text	Boldfaced text is used for: • Key and button names. (Example: Click **OK**.) • Information that you enter. (Example: Enter **Administrator** in the User Name box.)
< > (angle brackets)	Angle brackets are used around parameters for which you supply a value. (Example: In the Command Prompt window, enter **ping <IP address>**.)

Table 17 Cisco Unified Communications Manager 6.x SIP Trunk Integration Guide for Cisco Unity Connection 1.2 Conventions (continued)

Convention	Description
- (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 on menus. (Example: On the Windows Start menu, click **Programs > Cisco Unified Serviceability > Real-Time Monitoring Tool**.) In the navigation bar of the Cisco Unity Connection Administration. (Example: In the Cisco Unity Connection Administration, expand **System Settings > Advanced**.)

The *Cisco Unified Communications Manager 6.x SIP Trunk Integration Guide for Cisco Unity Connection 1.2* 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 *Documentation Guide for Cisco Unity Connection*. 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.

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.

Any Internet Protocol (IP) addresses used in this document are not intended to be actual addresses. Any examples, command display output, and figures included in the document are shown for illustrative purposes only. Any use of actual IP addresses in illustrative content is unintentional and coincidental.