



## **Cisco Unified Communications Manager Express SIP Trunk Integration Guide for Cisco Unity**

Release 5.0

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*Cisco Unified Communications Manager Express SIP Trunk Integration Guide for Cisco Unity Release 5.0*

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## Preface

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This Preface contains the following sections:

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- [Documentation Conventions](#), page v
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## Audience and Use

This document provides instructions for setting up an integration between Cisco Unity and supported versions of Cisco Unified Communications Manager (CM) (formerly known as Cisco Unified CallManager). For a list of supported versions of Cisco Unified CM that are qualified to integrate with Cisco Unity by Skinny Call Control Protocol (SCCP), see the *SCCP Compatibility Matrix: Cisco Unity, the Cisco Unity-CM TSP, Cisco Unified Communications Manager, and Cisco Unified Communications Manager Express* at

[http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products\\_device\\_support\\_tables\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products_device_support_tables_list.html).

## Documentation Conventions

The *Cisco Unified Communications Manager Express SIP Trunk Integration Guide for Cisco Unity Release 5.0* uses the following conventions.

**Table 1** *Cisco Unified Communications Manager Express SIP Trunk Integration Guide for Cisco Unity Release 5.0 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> .)

**Table 1** *Cisco Unified Communications Manager Express SIP Trunk Integration Guide for Cisco Unity Release 5.0 Conventions (continued)*

Convention	Description
- (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 on menus. (Example: On the Windows Start menu, click <b>Programs &gt; Cisco Unified Serviceability &gt; Real-Time Monitoring Tool</b> .)  In the navigation bar of the Cisco Unity Administrator. (Example: In the Cisco Unity Administrator, expand <b>System Settings &gt; Advanced</b> .)

The *Cisco Unified Communications Manager Express SIP Trunk Integration Guide for Cisco Unity Release 5.0* 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.

## Cisco Unity Documentation

For descriptions and URLs of Cisco Unity documentation on Cisco.com, see the *Documentation Guide for Cisco Unity*. The document is shipped with Cisco Unity and is available at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products\\_documentation\\_roadmaps\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products_documentation_roadmaps_list.html).

## Obtaining Documentation, Obtaining Support, and Security Guidelines

For information on obtaining documentation, obtaining support, providing documentation feedback, security guidelines, and also recommended aliases and general Cisco documents, 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>



# CHAPTER 1

## Introduction

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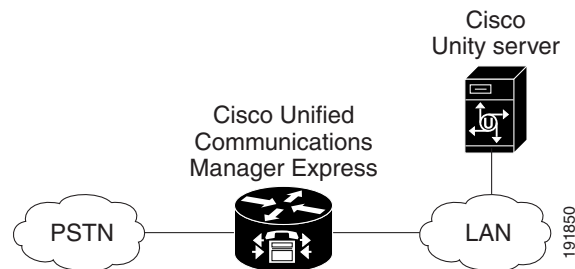
See the following sections in this chapter:

- [Integration Description, page 1-1](#)
- [Call Information, page 1-1](#)
- [Integration Functionality, page 1-2](#)
- [Integrations with Multiple Phone Systems, page 1-2](#)

## Integration Description

The Cisco Unified Communications Manager (CM) Express (formerly known as Cisco Unified CallManager Express) SIP trunk integration uses a LAN to connect Cisco Unity and the phone system. The Cisco Unified Communications Manager Express also provides connections to the PSTN. [Figure 1-1](#) shows the connections for a system with a single Cisco Unified CM Express router.

**Figure 1-1** *Connections Between the Cisco Unified Communications Manager Express Router and Cisco Unity*



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

When forwarding calls to greetings, Cisco Unity uses the original redirected number, not the last redirected number. For example, when A calls B and forwards the call to C whose phone forwards to voice mail, the call will go to the voice mailbox for B.

## Integration Functionality

The Cisco Unified CM Express SIP trunk integration with Cisco Unity provides the following features:

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

## Integrations with Multiple Phone Systems

Cisco Unity can be integrated with two or more phone systems at one time. For information on the maximum supported combinations and instructions for integrating Cisco Unity with multiple phone systems, see the *Multiple Phone System Integration Guide for Cisco Unity 5.0* at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products\\_installation\\_and\\_configuration\\_guides\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products_installation_and_configuration_guides_list.html).



## CHAPTER 2

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

See the following sections in this chapter:

- [Introduction: Issues to Consider When Planning Port Setup, page 2-1](#)
- [Determining How Many Voice Messaging Ports to Install, page 2-2](#)
- [Determining How Many Voice Messaging Ports Will Answer Calls, page 2-2](#)
- [Determining How Many Voice Messaging Ports Will Only Dial Out, and Not Answer Calls, page 2-2](#)

## Introduction: Issues to Consider When Planning Port Setup

Before programming the phone system, you need to plan how the voice messaging ports will be used by Cisco Unity. 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 and to make telephone record and playback (TRAP) connections.

The following table describes the voice messaging port settings in Cisco Unity that can be set in UTIM, and that are displayed as read-only text on the System > Ports page of Cisco Unity Administrator.

**Table 2-1**      **Settings for the Voice Messaging 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 Communications Manager Express server must have the Answer Calls box checked. Otherwise, calls to Cisco Unity may not be answered.
Message Notification	Check this check box to designate the port for notifying subscribers of messages.

Table 2-1 Settings for the Voice Messaging Ports (continued)

Field	Considerations
TRAP Connection	Check this check box so that subscribers can use the phone as a recording and playback device in Cisco Unity web applications and e-mail clients.

## Determining How Many Voice Messaging Ports to Install

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

- The number of calls Cisco Unity will answer when call traffic is at its peak.
- The expected length of each message that callers will record and that subscribers will listen to.
- The number of subscribers.
- The number of ports that will be set to dial out only.
- The number of calls made for message notification.
- The number of TRAP connections needed when call traffic is at its peak. (TRAP connections are used by Cisco Unity 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.

## Determining How Many Voice Messaging Ports Will Answer Calls

The calls that the voice messaging ports answer can be incoming calls from unidentified callers or from subscribers. 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 takes the port off-hook to dial out.

## Determining How Many Voice Messaging Ports 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 subscribers by phone, pager, or e-mail of messages that have arrived.

- Make a TRAP connection so that subscribers can use the phone as a recording and playback device in Cisco Unity web applications.

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

**Caution**

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In programming the phone system, do not send calls to voice messaging ports in Cisco Unity 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 Dialout MWI, do not send calls to it.

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## CHAPTER 3

# Setting Up a Cisco Unified Communications Manager Express SIP Trunk Integration with Cisco Unity

For detailed instructions for setting up a Cisco Unified Communications Manager Express SIP trunk integration with Cisco Unity, see the following sections in this chapter.

- [Task List to Create the Integration Through a SIP Trunk, page 3-1](#)
- [Requirements, page 3-2](#)
- [Programming the Cisco Unified Communications Manager Express Phone System for Integrating with Cisco Unity, page 3-3](#)
- [Creating a New Integration with Cisco Unified Communications Manager Express, page 3-3](#)



### Note

Cisco Unity failover is not available with the Cisco Unified CM Express integration.

AMIS Networking and call loop detection will not function when Cisco Unity is integrated with Cisco Unified CM Express.

The G.729a codec is not supported.

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

Before doing the following tasks to integrate Cisco Unity with Cisco Unified CM Express through a SIP trunk, confirm that the Cisco Unity server is ready for the integration by completing the applicable tasks in the applicable Cisco Unity installation guide. If you are installing a new Cisco Unity server by using the applicable Cisco Unity 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 server requirements have been met. See the [“Requirements” section on page 3-2](#).
2. Plan how the voice messaging ports will be used by Cisco Unity. See [Chapter 2, “Planning How the Voice Messaging Ports Will Be Used by Cisco Unity.”](#)
3. Program Cisco Unified CM Express. See the [“Programming the Cisco Unified Communications Manager Express Phone System for Integrating with Cisco Unity” section on page 3-3](#).

4. Create the integration. See the “[Creating a New Integration with Cisco Unified Communications Manager Express](#)” section on page 3-3.
5. If want to disable transcoding into the G.729a audio format, remove the G.729a codec from the Cisco Unity server. See [Chapter 4, “Disabling Transcoding into the G.729a Audio Format.”](#)
6. Test the integration. See [Chapter 5, “Testing the Integration.”](#)

## Requirements

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

### Phone System

- Cisco Unified CM Express.  
For details on compatible versions of Cisco Unified CM Express, refer to the *SIP Trunk Compatibility Matrix: Cisco Unity, Cisco Unified CM, and Cisco Unified CM Express* at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products\\_device\\_support\\_tables\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products_device_support_tables_list.html).
- A compatible Cisco IOS software version. Refer to the *Cisco Unified Communications Manager Express and Cisco IOS Software Version Compatibility Matrix* at [http://www.cisco.com/en/US/products/sw/voicesw/ps4625/prod\\_installation\\_guides\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps4625/prod_installation_guides_list.html).
- Cisco Unified CM Express feature license.
- Cisco IP phone feature licenses, and Cisco licenses for other H.323-compliant devices or software (such as Cisco VirtualPhone and Microsoft NetMeeting clients) that will be connected to the network, as well as one license for each Cisco Unity port.
- For the Cisco Unified CM Express extensions, SIP phones that support DTMF relay as described in RFC-2833. For a list of supported Cisco IP phone models, refer to the applicable *Cisco Unified Communications Manager Express Supported Firmware, Platforms, Memory, and Voice Products* at [http://www.cisco.com/en/US/products/sw/voicesw/ps4625/prod\\_installation\\_guides\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps4625/prod_installation_guides_list.html).
- For the Cisco Unified CM Express extensions, one of the following configurations:
  - (Best practice) Only SIP phones that support DTMF relay as described in RFC-2833.
  - Both SCCP phones 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 an IP phone into the network.

### Cisco Unity Server

- The applicable version of Cisco Unity. For details on compatible versions of Cisco Unity, refer to the *SIP Trunk Compatibility Matrix: Cisco Unity, Cisco Unified CM, and Cisco Unified CM Express* at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products\\_device\\_support\\_tables\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products_device_support_tables_list.html).
- Cisco Unity installed and ready for the integration, as described in the applicable Cisco Unity installation guide at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod\\_installation\\_guides\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod_installation_guides_list.html).
- A license that enables the appropriate number of voice messaging ports.

**Centralized Voice Messaging**

Cisco Unity supports centralized voice messaging by supporting various inter-phone system networking protocols including, for example, proprietary protocols such as Avaya DCS, Nortel MCDN, or Siemens CorNet, and standards-based protocols such as QSIG or DPNSS. For details, see the “Centralized Voice Messaging” section in the “Integrating Cisco Unity with the Phone System” chapter of the *Design Guide for Cisco Unity Release 5.x* at

[http://www.cisco.com/en/US/docs/voice\\_ip\\_comm/unity/5x/design/guide/5xcudgx.html](http://www.cisco.com/en/US/docs/voice_ip_comm/unity/5x/design/guide/5xcudgx.html).

## Programming the Cisco Unified Communications Manager Express Phone System for Integrating with Cisco Unity

For details on programming the Cisco Unified CM Express router for the integration with Cisco Unity, refer to the “Integrating Voice Mail” chapter of the *Cisco Unified Communications Manager Express System Administrator Guide* at

[http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products\\_installation\\_and\\_configuration\\_guides\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_installation_and_configuration_guides_list.html).

## Creating a New Integration with Cisco Unified Communications Manager Express

After ensuring that Cisco Unified Communications Manager Express and Cisco Unity 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** If UTIM is not already open, on the Windows Start menu of the Cisco Unity server, click **Programs > Cisco Unity > Manage Integrations**. UTIM appears.
- Step 2** In the left pane of the UTIM window, click **Cisco Unity Server**.
- Step 3** On the Integration menu of the UTIM window, click **New**. The Telephony Integration Setup Wizard appears.
- Step 4** On the Welcome page, click **SIP (including CUCM/CCM)** and click **Next**.
- Step 5** On the Name This SIP Integration and Cluster page, enter the following settings, then click **Next**.

**Table 3-1 Settings for the Name This SIP Integration and Cluster Page**

Field	Setting
Integration Name	Enter the name you will use to identify this SIP integration. Accept the default name or enter another name.
Cluster Name	Enter the name you will use to identify this SIP server cluster. Accept the default name or enter another name.

- Step 6** On the Enter Primary and Secondary SIP Server page, enter the following settings, then click **Next**.

**Table 3-2 Settings for the Enter Primary and Secondary SIP Server Page**

Field	Setting
Primary: IP Address/Name	Enter the IP address of the primary SIP server that you are connecting to Cisco Unity.
Primary: Port	Enter the IP port of the primary SIP server that you are connecting to Cisco Unity.
Secondary: IP Address/Name	(Optional) Enter the IP address of the secondary SIP server that you are connecting to Cisco Unity.
Secondary: Port	(Optional) Enter the IP port of the secondary SIP server that you are connecting to Cisco Unity.


You can click **Ping Server** to confirm that the IP address is correct.

- Step 7** On the Set Number of Voice Messaging Ports page, enter the number of voice messaging ports on Cisco Unity that you want to connect to the SIP server, then click **Next**.

This number must not be more than the number of ports set up on the SIP server.

- Step 8** On the Configure Cisco Unity SIP Settings page, enter the following settings, then click **Next**.

**Table 3-3 Settings for the Configure Cisco Unity SIP Settings Page**

Field	Setting
Contact Line Name	Enter the voice messaging line name that subscribers will use to contact Cisco Unity and that Cisco Unity uses to register with the SIP server.
Cisco Unity SIP Port	Enter the SIP port on Cisco Unity that callers and the SIP server use to connect to voice mail. In most circumstances, we recommend using the default setting. However, for multiple phone system integrations, do not use 5061 as the setting for this integration.   <b>Caution</b> Do not use 5061 as the Cisco Unity SIP port for this integration. Otherwise, MWIs and dialouts may not function correctly.
Preferred Codec	Click the codec Cisco Unity will first attempt to use on outgoing calls.
Preferred Transport Protocol	Click <b>UDP</b> .

- Step 9** On the Enter SIP Server Authentication page, enter the following settings, then click **Next**.

**Table 3-4 Settings for the Enter SIP Server Authentication Page**

Field	Setting
Authenticate with the SIP Server	Check this check box if you want Cisco Unity to authenticate with the SIP server.
Name	Enter the name that the SIP server will use for authentication.
Password	Enter the password that the SIP server will use for authentication.

**Step 10** If other integrations already exist, the Enter Trunk Access Code page appears. Enter the extra digits that Cisco Unity must use to transfer calls through the gateway to extensions on the other phone systems with which it is integrated. Then click **Next**.

**Step 11** On the Reassign Subscribers page, any subscribers whose phone system integration has been deleted and who are not currently assigned to a phone system integration will appear in the list.

If no subscribers appear in the list, click **Next** and continue to [Step 12](#).

Otherwise, select the subscribers that you want to assign to this phone system integration and click **Next**. You can use the following selection controls for selecting subscribers.

**Table 3-5 Selection Controls for the Reassign Subscribers Page**

Selection Control	Effect
Check All	Checks the check boxes for all subscribers in the list.
Uncheck All	Unchecks the check boxes for all subscribers in the list.
Toggle Selected	For the subscribers highlighted in the list, toggles between checking and unchecking the check boxes.  If some highlighted subscriber check boxes are checked and others are unchecked, clicking this button will check all the check boxes. Clicking again will uncheck all the check boxes.

**Step 12** On the Reassign Call Handlers page, any call handlers whose phone system integration has been deleted and that are not currently assigned to a phone system integration will appear in the list.

If no call handlers appear in the list, click **Next** and continue to [Step 13](#).

Otherwise, select the call handlers that you want to assign to this phone system integration and click **Next**. You can use the following selection controls for selecting call handlers.

**Table 3-6 Selection Controls for the Reassign Call Handlers Page**

Selection Control	Effect
Check All	Checks the check boxes for all call handlers in the list.
Uncheck All	Unchecks the check boxes for all call handlers in the list.
Toggle Selected	For the call handlers highlighted in the list, toggles between checking and unchecking the check boxes.  If some highlighted call handler check boxes are checked and others are unchecked, clicking this button will check all the check boxes. Clicking again will uncheck all the check boxes.

**Step 13** On the Completing page, verify the settings you entered, then click **Finish**.

**Step 14** At the prompt to restart the Cisco Unity services, click **Yes**. The Cisco Unity services restart.


Alternatively, you can restart the Cisco Unity services in UTIM on the Tools menu by clicking **Restart Cisco Unity**.

### To Enter the Voice Messaging Port Settings for the Integration

- Step 1** After the Cisco Unity services restart, on the View menu, click **Refresh**.
- Step 2** In the left pane of the UTIM window, expand the phone system integration that you are creating.
- Step 3** In the left pane, click the name of the cluster.
- Step 4** In the right pane, on the Servers tab, confirm that the CUCM/CCM Security Mode field is set to Non-Secure.
- Step 5** Click the **Ports** tab.
- Step 6** Enter the settings shown in [Table 3-7](#) for the voice messaging ports.

To get the best performance, use the first voice messaging ports for incoming calls and the last ports to dial out. This arrangement helps minimize the possibility of a collision, in which an incoming call arrives on a port at the same time that Cisco Unity takes the port off-hook to dial out.

**Table 3-7** Settings for the Voice Messaging 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 CM Express server must have the Answer Calls box checked. Otherwise, calls to Cisco Unity may not be answered.
Message Notification	Check this check box to designate the port for notifying subscribers of messages.
Dialout MWI	Check this check box to designate the port for turning MWIs on and off. Assign Dialout MWI to the least busy ports.
TRAP Connection	Check this check box so that subscribers can use the phone as a recording and playback device in Cisco Unity web applications and e-mail clients.

- Step 7** Click **Save**.

### To Adjust the Advanced Settings for 72 or More Voice Messaging Ports

- Step 1** On the Cisco Unity server, on the Windows Start menu, click **Programs > Cisco Unity > Cisco Unity Tools Depot**.
- Step 2** In the Tools Depot window, in the left pane, expand **Administration Tools** and double-click **Advanced Settings Tool**.
- Step 3** If the Cisco Unity server is running Windows Server 2003, do the following substeps. Otherwise, continue to [Step 4](#).
- In the Cisco Unity Advanced Settings window, in the left pane, click **Messaging - 72 or More Voice Ports - Enable Low-Fragmentation Heap**.
  - In the New Value drop-down box, click **1** and click **Set**.
  - When prompted that the value has been set, click **OK**.

- Step 4** If Exchange is the message store, do the following substeps. Otherwise, continue to [Step 5](#).
- In the Cisco Unity Advanced Settings window, in the left pane, click **Messaging - 72 or More Voice Ports - Number of MAPI Sessions Per Exchange Server**.
  - In the New Value drop-down box, click the applicable setting, then click **Set**.

Number of Voice Messaging Ports	Setting
72 to 83	Click <b>2</b> .
84 to 95	Click <b>3</b> .
96 to 119	Click <b>4</b> .
120 to 143	Click <b>5</b> .
144	Click <b>6</b> .

- When prompted that the value has been set, click **OK**.

**Step 5** Close the Tools Depot window.

**Step 6** Restart the Cisco Unity server.

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## CHAPTER 4

# Disabling Transcoding into the G.729a Audio Format

If you want to disable transcoding into the G.729a audio format, do the following procedure. Otherwise, continue to [Chapter 5, “Testing the Integration.”](#)



### Caution

Disabling transcoding into the G.729a audio format will block the audio stream for phones that use this audio format when connected to Cisco Unity. For the phones that use the G.729a audio format to receive the audio stream from Cisco Unity, you must set up a Cisco Unified CM transcoder to transcode the audio stream into the G.729a audio format.

When Cisco Unity has multiple integrations, disabling transcoding into the G.729a audio format will block G.729 audio streams to the Cisco Unity server for other integrations that use the G.729a audio format (for example, Cisco Unified CM SCCP integrations or integrations through PIMG units).

### To Disable Transcoding into the G.729a Audio Format

- Step 1** On the Windows Start menu, click **Settings > Control Panel > Sounds and Multimedia**.
- Step 2** In the Sounds and Multimedia dialog box, click the **Hardware** tab.
- Step 3** Under Devices, click **Audio Codecs** and click **Properties**.
- Step 4** In the Audio Codecs Properties dialog box, click the **Properties** tab.
- Step 5** Under Audio Compression Codecs, click **Sipro Labs G.729A** and click **Remove**.
- Step 6** When prompted to confirm removing the codec, click **Yes**.
- Step 7** If prompted to restart the system, click **Restart Later**.
- Step 8** In the Audio Codecs Properties dialog box, click **OK**.
- Step 9** In the Sounds and Multimedia dialog box, click **OK**.
- Step 10** Browse to **Windows\System32**.
- Step 11** Rename the file **Sl\_g729a.acm** to be **Sl\_g729a.old**.
- Step 12** On the Windows Start menu, click **Programs > Cisco Unity > Manage Integrations**.
- Step 13** In the left pane of the UTIM window, expand the Cisco Unified CM SIP trunk integration and click the first cluster.
- Step 14** In the right pane, click the **SIP Info** tab.

**Step 15** In the Preferred Codec field, confirm that the setting is **G.711 (mu-law)**. If the field has this setting, continue to [Step 16](#).

If this field has a different setting, do the following substeps.

- a. Click **G.711 (mu-law)**.
- b. Click **Save**.
- c. When prompted to restart the Cisco Unity services, click **No**.

**Step 16** Restart the Cisco Unity server.

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## CHAPTER 5

# Testing the Integration

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

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

- The installation guide for the phone system.
- *Troubleshooting Guide for Cisco Unity*, available at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod\\_troubleshooting\\_guides\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod_troubleshooting_guides_list.html).
- The setup information earlier in this guide.

### To Set Up the Test Configuration

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

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



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

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**Step 3** In Cisco Unity Administrator, create a test subscriber to use for testing by doing the applicable substeps below.

If your message store is Microsoft Exchange, do the following:

- a. In Cisco Unity Administrator, go to the **Subscribers > Subscribers > Profile** page.
- b. Click the **Add** icon.
- c. In the New Subscriber field, click **Exchange**.
- d. On the Add Subscriber page, enter the applicable information.
- e. Click **Add**.

If your message store is IBM Lotus Domino, do the following:

- a. In Cisco Unity Administrator, go to the **Subscribers > Subscribers > Profile** page.
- b. Click the **Add** icon.
- c. In the New Subscriber field, click **Notes**.
- d. In the Address Book list, confirm that the address book listed is the one that contains the user data that you want to import.

If the address book that you want to use is not listed, go to the **System > Configuration > Subscriber Address Books** page and add a different address book.

- e. In the Find Domino Person By list, indicate whether to search by short name, first name, or last name.
- f. Enter the applicable short name or name. You also can enter \* to display a list of all users, or enter one or more characters followed by \* to narrow your search.
- g. Click **Find**.
- h. On the list of matches, click the name of the user to import.
- i. On the Add Subscriber page, enter the applicable information.
- j. Click **Add**.

**Step 4** In the Extension field, enter the extension of Phone 1.

**Step 5** In the Active Schedule field, click **All Hours - All Days**.

**Step 6** Click the **Save** icon.

**Step 7** In the navigation bar, click **Call Transfer** to go to the Subscribers > Subscribers > Call Transfer page for the test subscriber.

For more information on transfer settings, see the “Subscriber Template Call Transfer Settings” section in Cisco Unity Administrator Help.

**Step 8** In the Transfer Rule Applies To field, click **Standard**.

**Step 9** Under Transfer Incoming Calls, click **Yes, Ring Subscriber’s Extension**, and confirm that the extension number is for Phone 1.

**Step 10** Under Transfer Type, click **Release to Switch**.

**Step 11** Click the **Save** icon.

**Step 12** In the navigation bar, click **Messages** to go to the Subscribers > Subscribers > Messages page for the test subscriber.

**Step 13** Under Message Waiting Indicators (MWIs), check the **Use MWI for Message Notification** check box.

**Step 14** In the Extension field, enter **x**.

**Step 15** Click the **Save** icon.

**Step 16** Open the Status Monitor by doing one of the following:

- In Internet Explorer, go to **http://<Cisco Unity server name>/web/sm**.
- Double-click the desktop shortcut to the Status Monitor.
- In the status bar next to the clock, right-click the Cisco Unity tray icon and click **Status Monitor**.

### 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.

**Step 2** On the 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 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 and that you hear the greeting for the test subscriber. Hearing the greeting means that the phone system forwarded the unanswered call and the call-forward information to Cisco Unity, which correctly interpreted the information.
- Step 7** On the Status Monitor, note which port handles this call.
- Step 8** Leave a message for the test subscriber and hang up Phone 2.
- Step 9** On the 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 are successfully integrated for turning on MWIs.
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#### To Test Listening to Messages

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- Step 1** From Phone 1, enter the internal pilot number for Cisco Unity.
- Step 2** When asked for your password, enter the default password. Hearing the request for your password means that the phone system sent the necessary call information to Cisco Unity, which correctly interpreted the information.
- Step 3** Confirm that you hear the recorded voice name for the test subscriber (if you did not record a voice name for the test subscriber, you will hear the extension number for Phone 1). Hearing the voice name means that Cisco Unity correctly identified the subscriber by the extension.
- Step 4** When asked whether you want to listen to your message, press **1**.
- Step 5** After listening to the message, press **3** to 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 are successfully integrated for turning off MWIs.
- Step 7** Hang up Phone 1.
- Step 8** On the 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.
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#### To Set Up Supervised Transfer on Cisco Unity

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- Step 1** In Cisco Unity Administrator, go to the **Subscribers > Subscribers > Call Transfer** page.  
If the name of the test subscriber is not displayed, click the **Find** icon (the magnifying glass) in the title bar, then click **Find**, and select the name of the test subscriber in the list that appears.  
For more information on transfer settings, see the “Subscriber Template Call Transfer Settings” section in Cisco Unity Administrator Help.
- Step 2** Under Transfer Type, click **Supervise Transfer**.
- Step 3** Set the Rings to Wait For field to **3**.

**Step 4** Click the **Save** icon.

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#### 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.
- Step 2** On the 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 or beeps).
- 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 is supervising the transfer.
- Step 6** Confirm that, after three rings, you hear the greeting for the test subscriber. Hearing the greeting means that Cisco Unity successfully recalled the supervised-transfer call.
- Step 7** During the greeting, hang up Phone 2.
- Step 8** On the 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.
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#### To Delete the Test Subscriber Account

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- Step 1** In Cisco Unity Administrator, go to the **Subscribers > Subscribers > Profile** page.  
If the name of the test subscriber is not displayed, click the **Find** icon (the magnifying glass) in the title bar, then click **Find**, and select the name of the test subscriber in the list that appears.
- Step 2** In the title bar, click the **Delete Subscriber** icon (the X).
- Step 3** Click **Delete**.
- Step 4** When prompted to confirm deleting the subscriber, click **OK**.
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