



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

Release 7.x  
December 11, 2008

**Americas Headquarters**  
Cisco Systems, Inc.  
170 West Tasman Drive  
San Jose, CA 95134-1706  
USA  
<http://www.cisco.com>  
Tel: 408 526-4000  
800 553-NETS (6387)  
Fax: 408 527-0883

Customer Order Number:  
Text Part Number: OL-15443-01

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS.

THE SOFTWARE LICENSE AND LIMITED WARRANTY FOR THE ACCOMPANYING PRODUCT ARE SET FORTH IN THE INFORMATION PACKET THAT SHIPPED WITH THE PRODUCT AND ARE INCORPORATED HEREIN BY THIS REFERENCE. IF YOU ARE UNABLE TO LOCATE THE SOFTWARE LICENSE OR LIMITED WARRANTY, CONTACT YOUR CISCO REPRESENTATIVE FOR A COPY.

The Cisco implementation of TCP header compression is an adaptation of a program developed by the University of California, Berkeley (UCB) as part of UCB's public domain version of the UNIX operating system. All rights reserved. Copyright © 1981, Regents of the University of California.

NOTWITHSTANDING ANY OTHER WARRANTY HEREIN, ALL DOCUMENT FILES AND SOFTWARE OF THESE SUPPLIERS ARE PROVIDED "AS IS" WITH ALL FAULTS. CISCO AND THE ABOVE-NAMED SUPPLIERS DISCLAIM ALL WARRANTIES, EXPRESSED OR IMPLIED, INCLUDING, WITHOUT LIMITATION, THOSE OF MERCHANTABILITY, FITNESS FOR A PARTICULAR PURPOSE AND NONINFRINGEMENT OR ARISING FROM A COURSE OF DEALING, USAGE, OR TRADE PRACTICE.

IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.

CCDE, CCENT, Cisco Eos, Cisco Lumin, Cisco Nexus, Cisco StadiumVision, Cisco TelePresence, Cisco WebEx, the Cisco logo, DCE, and Welcome to the Human Network are trademarks; Changing the Way We Work, Live, Play, and Learn and Cisco Store are service marks; and Access Registrar, Aironet, AsyncOS, Bringing the Meeting To You, Catalyst, CCDA, CCDP, CCIE, CCIP, CCNA, CCNP, CCSP, CCVP, Cisco, the Cisco Certified Internetwork Expert logo, Cisco IOS, Cisco Press, Cisco Systems, Cisco Systems Capital, the Cisco Systems logo, Cisco Unity, Collaboration Without Limitation, EtherFast, EtherSwitch, Event Center, Fast Step, Follow Me Browsing, FormShare, GigaDrive, HomeLink, Internet Quotient, IOS, iPhone, iQuick Study, IronPort, the IronPort logo, LightStream, Linksys, MediaTone, MeetingPlace, MeetingPlace Chime Sound, MGX, Networkers, Networking Academy, Network Registrar, PCNow, PIX, PowerPanels, ProConnect, ScriptShare, SenderBase, SMARTnet, Spectrum Expert, StackWise, The Fastest Way to Increase Your Internet Quotient, TransPath, WebEx, and the WebEx logo are registered trademarks of Cisco Systems, Inc. and/or its affiliates in the United States and certain other countries.

All other trademarks mentioned in this document or website are the property of their respective owners. The use of the word partner does not imply a partnership relationship between Cisco and any other company. (0809R)

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.

*Cisco Unified Communications Manager Express SIP Trunk Integration Guide for Cisco Unity Connection Release 7.x*  
© 2008 Cisco Systems, Inc. All rights reserved.



# CONTENTS

## **Preface** v

- Audience and Use v
- Documentation Conventions v
- Cisco Unity Connection Documentation vi
- Obtaining Documentation and Submitting a Service Request vi

---

## **CHAPTER 1**

### **Introduction** 1-1

- Integration Description 1-1
- Call Information 1-2
- Integration Functionality 1-2
- Integrations with Multiple Phone Systems 1-2

---

## **CHAPTER 2**

### **Planning How the Voice Messaging Ports Will Be Used by Cisco Unity Connection** 2-1

- Introduction: Issues to Consider When Planning Port Setup 2-1
- Determining How Many Voice Messaging Ports to Install 2-2
- Determining How Many Voice Messaging Ports Will Answer Calls 2-3
- Determining How Many Voice Messaging Ports Will Only Dial Out, and Not Answer Calls 2-3
- Considerations for a Cisco Unity Connection Cluster 2-3
  - When Both Cisco Unity Connection Servers Are Functioning Normally 2-3
  - When Only One Cisco Unity Connection Server Is Functioning 2-4

---

## **CHAPTER 3**

### **Setting Up a Cisco Unified Communications Manager Express SIP Trunk Integration with Cisco Unity Connection** 3-1

- Integration Tasks 3-1
- Requirements 3-2
- Programming the Cisco Unified Communications Manager Express Phone System for Integrating with Cisco Unity Connection 3-3
- Creating a New Integration with Cisco Unified Communications Manager Express 3-6

---

**CHAPTER 4**     **Testing the Integration**     4-1

---

**CHAPTER 5**     **Adding New User Template for Multiple Integrations**     5-1

---

**INDEX**



## Preface

---

This Preface contains the following sections:

- [Audience and Use, page v](#)
- [Documentation Conventions, page v](#)
- [Cisco Unity Connection Documentation, page vi](#)
- [Obtaining Documentation and Submitting a Service Request, page vi](#)

## Audience and Use

This document provides instructions for setting up an integration between Cisco Unity Connection 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 Connection through a SIP trunk, see 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](http://www.cisco.com/en/US/products/ps6509/products_device_support_tables_list.html).

## Documentation Conventions

The *Cisco Unified Communications Manager Express SIP Trunk Integration Guide for Cisco Unity Connection Release 7.x* uses the following conventions.

**Table 1** *Cisco Unified Communications Manager Express SIP Trunk Integration Guide for Cisco Unity Connection Release 7.x 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 Connection Release 7.x 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 Connection Administration. (Example: In the Cisco Unity Connection Administration, expand <b>System Settings &gt; Advanced</b> .)

The *Cisco Unified Communications Manager Express SIP Trunk Integration Guide for Cisco Unity Connection Release 7.x* 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 Connection Documentation

For descriptions and URLs of Cisco Unity Connection documentation on Cisco.com, see the *Documentation Guide for Cisco Unity Connection*. The document is shipped with 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 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.



# CHAPTER 1

## Introduction

---

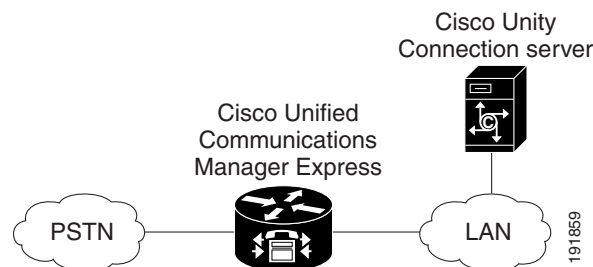
See the following sections in this chapter:

- [Integration Description, page 1-1](#)
- [Call Information, page 1-2](#)
- [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 Connection and the phone system. The Cisco Unified Communications Manager Express router also provides connections to the PSTN. [Figure 1-1](#) shows the connections.

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



For a list of supported versions of Cisco Unified CM Express that are qualified to integrate with Cisco Unity Connection through a SIP trunk, see 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](http://www.cisco.com/en/US/products/ps6509/products_device_support_tables_list.html).

This document does not apply to the configuration in which Cisco Unity Connection is installed as Cisco Unified Communications Manager Business Edition (CMBE)—on the same server with Cisco Unified CM.

## 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 CM Express SIP trunk 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 user 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)

## Integrations with Multiple Phone Systems

When Cisco Unity Connection is installed as Cisco Unified Communications Manager Business Edition (CMBE)—on the same server with Cisco Unified Communications Manager—Cisco Unity Connection cannot be integrated with multiple phone systems at one time.

When Cisco Unity Connection is not installed as Cisco Unified CMBE, Cisco Unity Connection can be integrated with two or more 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 for Cisco Unity Connection* 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).



## CHAPTER 2

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

---

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 2-3](#)
- [Determining How Many Voice Messaging Ports Will Only Dial Out, and Not Answer Calls, page 2-3](#)
- [Considerations for a Cisco Unity Connection Cluster, page 2-3](#)


## 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 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.  
For a Cisco Unity Connection cluster, each Cisco Unity Connection server must have enough ports to handle all voice messaging traffic in case the other server stops functioning.
- 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 2-1 Settings for the Voice Messaging Ports**

Field	Considerations
Enabled	Check this check box.
Server Name	<p>(When a Cisco Unity Connection cluster is configured) Click the name of the Cisco Unity Connection server that you want to handle this port.</p> <p>Assign an equal number of answering and dial-out voice messaging ports to the Cisco Unity Connection servers so that they equally share the voice messaging traffic.</p>
Answer Calls	<p>Check this check box.</p> <p> <b>Caution</b> 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.</p>
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 phone as a recording and playback device in Cisco Unity Connection web applications.

## 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 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.
- Whether a Cisco Unity Connection cluster is configured. For considerations, see the [“Considerations for a Cisco Unity Connection Cluster”](#) section on page 2-3.

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 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.

If your system is configured for a Cisco Unity Connection cluster, see the [“Considerations for a Cisco Unity Connection Cluster”](#) section on page 2-3.

## 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 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.

If your system is configured for a Cisco Unity Connection cluster, see the [“Considerations for a Cisco Unity Connection Cluster”](#) section on page 2-3.



### 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 Send MWI Requests, do not send calls to it.

## Considerations for a Cisco Unity Connection Cluster

If your system is configured for a Cisco Unity Connection cluster, consider how the voice messaging ports will be used in different scenarios.

## When Both Cisco Unity Connection Servers Are Functioning Normally

- A hunt group is configured to send incoming calls first to the subscriber server, then to the publisher server if no answering ports are available on the subscriber server.

- Both Cisco Unity Connection servers are active and handle voice messaging traffic for the system.
- In Cisco Unity Connection Administration, the voice messaging ports are configured so that an equal number of voice messaging ports are assigned to each Cisco Unity Connection server. This guide directs you to assign the voice messaging ports to their specific server at the applicable time.
- The number of voice messaging ports that are assigned to one Cisco Unity Connection server must be sufficient to handle all of the voice messaging traffic for the system (answering calls and dialing out) when the other Cisco Unity Connection server stops functioning.

If both Cisco Unity Connection servers must be functioning to handle the voice messaging traffic, the system will not have sufficient capacity when one of the servers stops functioning.

- Each Cisco Unity Connection server is assigned half the total number of voice messaging ports.  
If all the voice messaging ports are assigned to one Cisco Unity Connection server, the other Cisco Unity Connection server will not be able to answer calls or to dial out.
- Each Cisco Unity Connection server must have voice messaging ports that will answer calls and that can dial out (for example, to set MWIs).

## When Only One Cisco Unity Connection Server Is Functioning

- The hunt group on the phone system sends all calls to the functioning Cisco Unity Connection server.
- The functioning Cisco Unity Connection server receives all voice messaging traffic for the system.
- The number of voice messaging ports that are assigned to the functioning Cisco Unity Connection server must be sufficient to handle all of the voice messaging traffic for the system (answering calls and dialing out).
- The functioning Cisco Unity Connection server must have voice messaging ports that will answer calls and that can dial out (for example, to set MWIs).

If the functioning Cisco Unity Connection server does not have voice messaging ports for answering calls, the system will not be able to answer incoming calls. Similarly, if the functioning Cisco Unity Connection server does not have voice messaging ports for dialing out, the system will not be able to dial out (for example, to set MWIs).



## CHAPTER 3

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

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

- [Integration Tasks, page 3-1](#)
- [Requirements, page 3-2](#)
- [Programming the Cisco Unified Communications Manager Express Phone System for Integrating with Cisco Unity Connection, page 3-3](#)
- [Creating a New Integration with Cisco Unified Communications Manager Express, page 3-6](#)

This document does not apply to the configuration in which Cisco Unity Connection is installed as Cisco Unified Communications Manager Business Edition (CMBE)—on the same server with Cisco Unified Communications Manager.



### Note

Cisco Unified Communications Manager Express does not support Keypad Markup Language (KPML) for sending DTMF keystrokes in a SIP message (out-of-band).

## Integration Tasks

Before doing the following tasks to integrate Cisco Unity Connection with Cisco Unified CM Express through a SIP trunk, 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*.

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](#).
2. Plan how the voice messaging ports will be used by Cisco Unity Connection. See [Chapter 2, “Planning How the Voice Messaging Ports Will Be Used by Cisco Unity Connection.”](#)
3. Program Cisco Unified CM Express. See the [“Programming the Cisco Unified Communications Manager Express Phone System for Integrating with Cisco Unity Connection” section on page 3-3](#).
4. Create the integration. See the [“Creating a New Integration with Cisco Unified Communications Manager Express” section on page 3-6](#).
5. Test the integration. See [Chapter 4, “Testing the Integration.”](#)

6. If this integration is a second or subsequent integration, add the applicable new user templates for the new phone system. See [Chapter 5, “Adding New User Template for Multiple Integrations.”](#)

## Requirements

The Cisco Unified CM Express SIP integration supports configurations of the following components:

### Phone System

- A compatible version of Cisco Unified CM Express.  
For details on compatible versions of Cisco Unified CM Express, 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](http://www.cisco.com/en/US/products/ps6509/products_device_support_tables_list.html).
- A compatible Cisco IOS software version. Refer to the *Cisco Unified CME and Cisco IOS Software Version Compatibility Matrix* at [http://www.cisco.com/en/US/products/sw/voicesw/ps4625/prod\\_installation\\_guide09186a00805acf50.html](http://www.cisco.com/en/US/products/sw/voicesw/ps4625/prod_installation_guide09186a00805acf50.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 Connection 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 *Supported Firmware, Platforms, Memory, and Voice Products* document 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:
  - Only SIP phones.
  - 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 the applicable phone into the network.

### Cisco Unity Connection Server

- The applicable version of Cisco Unity Connection. For details on compatible versions of Cisco Unity Connection, 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](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](http://www.cisco.com/en/US/products/ps6509/prod_installation_guides_list.html).
- A license that enables the applicable number of voice messaging ports.

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

**Revised December 11, 2008**

For details on programming the Cisco Unified Communications Manager Express router for the integration with Cisco Unity Connection, 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).

After you have configured the Cisco Unified CM Express router for the integration, do the applicable following procedures:

- For Cisco Unified CM Express 4.1 or later, if calls can be received from a Cisco Unified Communications Manager SIP trunk, do the “[To Configure the Cisco Unified Communications Manager Express Router When It is Connected to a Cisco Unified Communications Manager SIP Trunk \(Cisco Unified Communications Manager Express 4.1 or Later\)](#)” procedure on page 3-3.
- For a Cisco Unity Connection cluster, do the “[To Configure the Cisco Unified Communications Manager Express Router for a Cisco Unity Connection Cluster](#)” procedure on page 3-3.

## To Configure the Cisco Unified Communications Manager Express Router When It is Connected to a Cisco Unified Communications Manager SIP Trunk (Cisco Unified Communications Manager Express 4.1 or Later)

---

- Step 1** On the Cisco Unified CM Express router, go into the global configuration mode by entering the following command:
- configure terminal**
- Step 2** To enter the voice service configuration mode, enter the following command:
- voice service voip**
- Step 3** To disable the 302 “Moved Temporarily” SIP message, enter the following command:
- no supplementary-service sip moved-temporarily**
- Step 4** To exit the global configuration mode, enter the following command:
- exit**
- 

## To Configure the Cisco Unified Communications Manager Express Router for a Cisco Unity Connection Cluster

---

- Step 1** On the Cisco Unified CM Express router, go into the global configuration mode by entering the following command:
- configure terminal**
- Step 2** To enter dial-peer configuration mode for port group for the publisher server, enter the following command:
- dial-peer voice <number> voip**
- Step 3** To set the description for the dial-peer, enter the following command:
- description <name of publisher server>**

- Step 4** To set the Cisco Unity Connection pilot number for the dial-peer, enter the following command:  
**destination-pattern <pilot number>**
- Step 5** To configure the dial-peer to use Session Initiation Protocol (SIP) for calls, enter the following command:  
**session protocol sipv2**
- Step 6** To specify the IP address (or DNS name) of the publisher server, enter the following command:  
**session target {ipv4:<IP address>|dns:<host name>}**
- Step 7** To enable DTMF relay, enter the following command:  
**dtmf-relay rtp-nte**
- Step 8** To set the codec for calls, enter the following command:  
**codec <codec used for calls>**
- Step 9** To equalize the number of calls sent to each Cisco Unity Connection server in the Connection cluster, enter the following command:  
**max-conn <number of ports handled by publisher server>**
- Step 10** To set the dial-peer preference for port group so that the calls will be routed first to the subscriber server, then to the publisher server if no ports are available on the subscriber server, enter the following command:  
**preference 2**
- Step 11** To enable huntstop, enter the following command:  
**huntstop**
- Step 12** To exit the global configuration mode, enter the following command:  
**exit**
- Step 13** To set a dial-peer that will prevent Cisco Unified CM Express from hunting beyond the dial-peers for the Cisco Unity Connection port groups, entering the following command:  
**configure terminal**
- Step 14** To enter dial-peer configuration mode, enter the following command:  
**dial-peer voice <number> voip**
- Step 15** To set the description for the dial-peer, enter the following command:  
**description <name of subscriber server>**
- Step 16** To set the Cisco Unity Connection pilot number for the dial-peer, enter the following command:  
**destination-pattern <pilot number>**
- Step 17** To configure the dial-peer to use Session Initiation Protocol (SIP) for calls, enter the following command:  
**session protocol sipv2**
- Step 18** To specify the IP address (or DNS name) of the subscriber server, enter the following command:  
**session target {ipv4:<IP address>|dns:<host name>}**
- Step 19** To enable DTMF relay, enter the following command:  
**dtmf-relay rtp-nte**
- Step 20** To set the codec for calls, enter the following command:

**codec <codec used for calls>**

- Step 21** To equalize the number of calls sent to each Cisco Unity Connection server in the Connection cluster, enter the following command:

**max-conn <number of ports handled by subscriber server>**

- Step 22** To set the dial-peer preference so that the calls will be routed first to the subscriber server, then to the publisher server if no ports are available on the subscriber server, enter the following command:

**preference 1**

- Step 23** To disable huntstop so that the calls will use the next available voice messaging port on the subscriber server, then use voice messaging ports on the publisher server if no ports are available on the subscriber server, enter the following command:

**no huntstop**

- Step 24** To exit the global configuration mode, enter the following command:

**exit**

---

The following is an example of the configuration without a Cisco Unity Connection cluster:

```
!  
dial-peer voice 2 voip  
  description CUC1  
  destination-pattern 8000  
  session protocol sipv2  
  session target ipv4:<IP address>  
  dtmf-relay rtp-nte  
  codec g711ulaw  
  max-conn 4  
  preference 1  
  huntstop  
!
```

The following is an example of the configuration with a Cisco Unity Connection cluster configured:

```
!  
dial-peer voice 2 voip  
  description CUC1  
  destination-pattern <pilot number>  
  session protocol sipv2  
  session target ipv4:<IP address of publisher server>  
  dtmf-relay rtp-nte  
  codec g711ulaw  
  max-conn 4  
  preference 2  
  huntstop  
!  
dial-peer voice 4 voip  
  description CUC2  
  destination-pattern <pilot number>  
  session protocol sipv2  
  session target ipv4:<IP address of subscriber server>  
  dtmf-relay rtp-nte  
  codec g711ulaw  
  max-conn 4  
  preference 1  
  no huntstop  
!
```

# Creating a New Integration with Cisco Unified Communications Manager Express

After ensuring that Cisco Unified Communications Manager Express 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, under Display Name, click the name of the default phone system.
- Step 4** On the Phone System Basics page, in the Phone System Name field, enter the descriptive name that you want for the phone system.
- Step 5** If you want to use this phone system for TRaP connections (when users record and playback through the phone in Cisco Unity Connection web applications), check the Default TRAP Switch check box. If you want to use another phone system for TRaP connections, uncheck this check box.
- Step 6** Click **Save**.
- Step 7** On the Phone System Basics page, in the Related Links drop-down box, click **Add Port Group** and click **Go**.
- Step 8** On the New Port Group page, enter the applicable settings and click **Save**.

**Table 3-1 Settings for the New Port Group Page**

Field	Setting
Phone System	Click the name of the phone system that you entered in <a href="#">Step 4</a> .
Create From	Click <b>Port Group Template</b> and click <b>SIP</b> in the drop-down box.
Display Name	Enter a descriptive name for the port group. You can accept the default name or enter the name that you want.
Authenticate with SIP Server	Check this check box if you want Cisco Unity Connection to authenticate with the Cisco Unified CM Express router.
Authentication User Name	Enter the name that Cisco Unity Connection will use to authenticate with the Cisco Unified CM Express router.
Authentication Password	Enter the password that Cisco Unity Connection will use to authenticate with the Cisco Unified CM Express router.
Contact Line Name	Enter 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 Express router.
SIP Security Profile	Click the SIP security profile that Cisco Unity Connection will use. <b>Note</b> Cisco Unified CM Express does not support Cisco Unified CM authentication and encryption.
SIP Transport Protocol	Click the SIP transport protocol that Cisco Unity Connection will use.

**Table 3-1** Settings for the New Port Group Page (continued)

Field	Setting
IP Address or Host Name	Enter the IP address (or host name) of the Cisco Unified CM Express router that you are integrating with Cisco Unity Connection.
Port	Enter the IP port of the Cisco Unified CM Express router that you are integrating with Cisco Unity Connection. We recommend that you use the default setting.

**Step 9** On the Port Group Basics page, in the Related Links drop-down box, click **Add Ports** and click **Go**.

**Step 10** On the New Port page, enter the following settings and click **Save**.

**Table 3-2** Settings for the New Port Page

Field	Setting
Enabled	Check this check box.
Number of Ports	Enter the number of voice messaging ports that you want to create in this port group. <b>Note</b> For a Cisco Unity Connection cluster, you must enter the total number of voice messaging ports that will be used by all Cisco Unity Connection servers. Each port will later be assigned to a specific Cisco Unity Connection server.
Phone System	Click the name of the phone system that you entered in <a href="#">Step 4</a> .
Port Group	Click the name of the port group that you added in <a href="#">Step 8</a> .
Server Name	Click the name Cisco Unity Connection server.

**Step 11** 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 12** 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 3-3** Settings for the Voice Messaging 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.
Server Name	<i>(For Cisco Unity Connection clusters only)</i> Click the name of the Cisco Unity Connection server that you want to handle this port.  Assign an equal number of answering and dial-out voice messaging ports to the Cisco Unity Connection servers so that they equally share the voice messaging traffic.

**Table 3-3** Settings for the Voice Messaging Ports (continued)

Field	Considerations
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.

**Step 13** Click **Save**.

**Step 14** Click **Next**.

**Step 15** Repeat [Step 12](#) through [Step 14](#) for all remaining voice messaging ports for the phone system.

**Step 16** If another phone system integration exists, in Cisco Unity Connection Administration, expand **Telephony Integrations**, then click **Trunk**. Otherwise, skip to [Step 20](#).

**Step 17** On the Search Phone System Trunks page, on the Phone System Trunk menu, click **New Phone System Trunk**.

**Step 18** On the New Phone System Trunk page, enter the following settings for the phone system trunk and click **Save**.

**Table 3-4** Settings for the Phone System Trunk

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

**Step 19** Repeat [Step 17](#) and [Step 18](#) for all remaining phone system trunks that you want to create.

**Step 20** 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 21** In the Task Execution Results window, click **Close**.



# CHAPTER 4

## 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, see 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 **Add New**.
- Step 5** On the New User page, enter the following settings.

**Table 4-1** Settings for the New User Page

Field	Setting
User Type	Click <b>User with Voice Mailbox</b> .
Based on Template	Click the applicable user template.
Alias	Enter <b>testuser</b> .
First Name	Enter <b>Test</b> .
Last Name	Enter <b>User</b> .
Display Name	Enter <b>Test User</b> .
Extension	Enter 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 4-2 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 the 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**.
-



## CHAPTER 5

# Adding New User Template for Multiple Integrations

---

When you create the first phone system integration, this first 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, Modifying, or Deleting a User 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](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 *User Moves, Adds, and Changes Guide for Cisco Unity Connection* 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).





## INDEX

---

### A

- Allow TRAP Connections (port setting) [2-2](#)
- Answer Calls (port setting) [2-2](#)

---

### C

- call information [1-2](#)
- Cisco Unified CM Express
  - creating a new integration [3-6](#)
  - phone system programming instructions [3-3](#)
  - requirements [3-2](#)
  - task list, creating a SIP trunk integration [3-1](#)
- Cisco Unified CM Express, supported versions for SIP trunk integrations [1-1](#)

---

### E

- Enabled (port setting) [2-2](#)

---

### F

- features provided by integration [1-2](#)
- forwarded calls, information sent by phone system [1-2](#)

---

### I

- integration functionality [1-2](#)

---

### M

- multiple integrations
  - about [1-2](#)
  - adding new user template [5-1](#)

---

### P

- Perform Message Notification (port setting) [2-2](#)
- phone systems, supported for SIP trunk integrations [1-1](#)
- ports
  - considerations for Cisco Unity Connection cluster [2-3](#)
  - planning how many will answer calls [2-3](#)
  - planning how many will dial out [2-3](#)
  - planning number to install [2-2](#)
  - planning setup [2-1](#)
  - settings [2-2](#)

---

### S

- Send MWI Requests (port setting) [2-2](#)
- Server Name (port setting) [2-2](#)
- supported phone systems [1-1](#)

---

### T

- template, adding a new user template for multiple integrations [5-1](#)
- testing
  - deleting the test user [4-4](#)
  - setting up supervised transfer [4-3](#)
  - setting up test configuration [4-1](#)
  - testing an external call with release transfer [4-2](#)

testing supervised transfer [4-4](#)

testing the functionality for listening to messages [4-3](#)

---

## U

user template, adding new for multiple integrations [5-1](#)

---

## V

voice messaging ports, settings [2-2](#)