



Cisco Unified Communications Manager Express SIP Trunk Integration Guide for Cisco Unity Connection Release 14

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Cisco Unified Communications Manager Express SIP Trunk Integration Guide for Cisco Unity Connection Release 14

Integration Tasks

Before doing the following tasks to integrate Unity Connection with Cisco Unified Communications Manager Express through a SIP trunk, confirm that the Unity Connection server is ready for the integration by completing the applicable tasks in the “[Installing Cisco Unity Connection](#)” chapter of the *Install, Upgrade, and Maintenance Guide for Cisco Unity Connection* Release 14, available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/14/install_upgrade/guide/b_14cuciumg.html.

The following task list describes the process for creating the integration.

Task List to Create the Integration

Use the following task list to set up a Cisco Unified CM Express SIP trunk integration.

1. Review the system and equipment requirements to confirm that all phone system and Unity Connection server requirements have been met. See the “Requirements” section on page 2.
2. Plan how the voice messaging ports are used by Unity Connection. See the “Planning the Usage of Voice Messaging in Unity Connection” section on page 4.
3. Program Cisco Unified Communications Manager Express. See the “Programming the Cisco Unified Communications Manager Express Phone System” section on page 7.
4. Create the integration. See the “Creating a New Integration with Cisco Unified CM Express” section on page 10.
5. Test the integration. See the “Testing the Integration” section on page 13.
6. If this integration is a second or subsequent integration, add the applicable new user templates for the new phone system. See the “Adding New User Templates for Multiple Integrations” section on page 16.



Note While integrating the Cisco Unity Connection with Cisco Unified Call Manager through a SIP trunk uncheck the Synchronize guest time to host option for Unified Communications product line in Virtualized environment. This enables the Unified Communications to synchronize with their clock to external NTP servers.

Requirements

The Cisco Unified CM Express SIP trunk 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, see the *Compatibility Matrix for Cisco Unity Connection* at <http://www.cisco.com/c/en/us/support/unified-communications/unity-connection/products-device-support-tables-list.html>.
- A compatible Cisco IOS software version. See the Cisco Unified CME and Cisco IOS Software Version Compatibility Matrix at <http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-express/products-device-support-tables-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) connected to the network, as well as one license for each 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, see the applicable compatibility information document at <http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-express/products-device-support-tables-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 Unity Connection in each location where you plug the applicable phone into the network.

Unity Connection Server

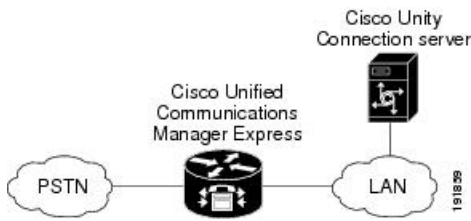
- The applicable version of Unity Connection. For details on compatible versions of Cisco Unity Connection, see the *Compatibility Matrix for Cisco Unity Connection*, at <http://www.cisco.com/c/en/us/support/unified-communications/unity-connection/products-device-support-tables-list.html>.
- Unity Connection installed and ready for the integration, as described in the “Installing Cisco Unity Connection” chapter of the *Install, Upgrade, and Maintenance Guide for Cisco Unity Connection Release 14*, available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/14/install_upgrade/guide/b_14cuciumg.html.
- A license that enables the applicable number of voice messaging ports.

Centralized Voice Messaging

Unity Connection supports centralized voice messaging through the phone system, which supports various inter-phone system networking protocols including proprietary protocols such as Avaya DCS, Nortel MCDN, or Siemens CorNet, and standards-based protocols such as QSIG or DPNSS. Note that centralized voice messaging is a function of the phone system and its inter-phone system networking, not voicemail. Unity Connection supports centralized voice messaging as long as the phone system and its inter-phone system networking are properly configured. For details, see the “[Centralized Voice Messaging](#)” section in the “Integrating Cisco Unity Connection with the Phone System” chapter of the *Design Guide for Cisco Unity Connection*, Release 14 at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/14/design/guide/b_14cucdg.html.

Integration Description

The Cisco Unified Communications Manager (CM) Express SIP trunk integration uses a LAN to connect Unity Connection and the phone system. The Cisco Unified Communications Manager Express router also provides connections to the PSTN.



For a list of supported versions of Cisco Unified CM Express that are qualified to integrate with Unity Connection through a SIP trunk, see the Compatibility Matrix for Cisco Unity Connection at <http://www.cisco.com/c/en/us/support/unified-communications/unity-connection/products-device-support-tables-list.html>.

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)

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

Integration Functionality

The Cisco Unified CM Express SIP trunk integration with 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; Unity Connection identifies a user based on the extension from which the call originated; a password may be required)
- Identified user messaging (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 Unity Connection is installed as Cisco Business Edition—on the same server with Cisco Unified Communications Manager—Unity Connection cannot be integrated with multiple phone systems at one time.

When Unity Connection is not installed as Cisco Business Edition, Unity Connection can be integrated with two or more phone systems at one time. For information on and instructions for integrating Unity Connection with multiple phone systems, see the *Multiple Phone System Integration Guide for Cisco Unity Connection*, Release 14 at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/14/integration/multiple/b_cuc14intmultiple.html.

Planning the Usage of Voice Messaging in Unity Connection

Before programming the phone system, you need to plan how the voice messaging ports are used by Unity Connection. The following considerations 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 Unity Connection cluster, each 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 answer calls.
- The number of voice messaging ports that 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 Unity Connection that can be set on Telephony Integrations > Port of Cisco Unity Connection Administration.

Table 1: Settings for the Voice Messaging Ports

| Field | Considerations |
|------------------------------|---|
| Enabled | Check this check box. |
| Server | <i>(When a Unity Connection cluster is configured)</i> Select the name of the Unity Connection server that you want to handle this port. Assign an equal number of answering and dial-out voice messaging ports to the Unity Connection servers so that they equally share the voice messaging traffic. |
| Answer Calls | Check this check box. Caution All voice messaging ports connecting to the Cisco Unified CM server must have the Answer Calls check box checked. Otherwise, calls to 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 phone as a recording and playback device in Unity Connection web applications. |

Determining the Number Voice of Messaging Ports to Install

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

- The number of calls Unity Connection answers when call traffic is at its peak.
- The expected length of each message that callers record and that users listen to.
- The number of users.
- The number of calls made for message notification.
- The number of MWIs 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 Unity Connection web applications to play back and record over the phone.)
- The number of calls that use the automated attendant and call handlers when call traffic is at its peak.
- Whether a Unity Connection cluster is configured. For considerations, see the [Considerations for a Unity Connection Cluster, on page 6](#) section.

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 the Number of Voice Messaging Ports to Answers 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.

If your system is configured for a Unity Connection cluster, see the [Considerations for a Unity Connection Cluster, on page 6](#) section.

Determining the Number of Voice Messaging Ports to Dials Out

Ports that only dial out can do one or more of the following:

- Notify users by phone, pager, or email 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 Unity Connection web applications.

If your system is configured for a Unity Connection cluster, see the [Considerations for a Unity Connection Cluster, on page 6](#) section.



Caution In programming the phone system, do not send calls to voice messaging ports in 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 Unity Connection Cluster

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

When Both Unity Connection Servers are Functioning

- 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.

- In Cisco Unity Connection Administration, the voice messaging ports are configured so that an equal number of voice messaging ports are assigned to each 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 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 Unity Connection server stops functioning. If both Unity Connection servers must be functioning to handle the voice messaging traffic, the system do not have sufficient capacity when one of the servers stops functioning.
- Each Unity Connection server is assigned half the total number of voice messaging ports. If all the voice messaging ports are assigned to one Unity Connection server, the other Unity Connection server cannot answer calls or to dial out.
- Each Unity Connection server must have voice messaging ports that answer calls and can dial out (for example, to set MWIs).

When Only One Unity Connection Server is Functioning

- The hunt group on the phone system sends all calls to the functioning Unity Connection server.
 - The functioning Unity Connection server receives all voice messaging traffic for the system.
 - The number of voice messaging ports that are assigned to the functioning Unity Connection server must be sufficient to handle all of the voice messaging traffic for the system (answering calls and dialing out).
 - The functioning Unity Connection server must have voice messaging ports that answer calls and can dial out (for example, to set MWIs).
- If the functioning Unity Connection server does not have voice messaging ports for answering calls, the system cannot answer incoming calls. Similarly, if the functioning Unity Connection server does not have voice messaging ports for dialing out, the system cannot dial out (for example, to set MWIs).

Programming the Cisco Unified Communications Manager Express Phone System

For details on programming the Cisco Unified Communications Manager Express router for the integration with Unity Connection, see the “Integrating Voice Mail” chapter of the Cisco Unified Communications Manager Express System Administrator Guide at <http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-express/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 4.1 and Later, on page 8](#)
- For a Unity Connection cluster, do the [To Configure the Cisco Unified Communications Manager Express Router for a Unity Connection Cluster, on page 8](#)

To Configure the Cisco Unified Communications Manager Express Router 4.1 and Later

Procedure

- Step 1** On the Cisco Unified CM Express router connected to a Cisco Unified Communications Manager SIP trunk, 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 Unity Connection Cluster

Procedure

- 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 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 Unity Connection server in the 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 are 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 prevents Cisco Unified CM Express from hunting beyond the dial-peers for the 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 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 Unity Connection server in the 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 are 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 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
-

Example

The following is an example of the configuration without a 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 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 CM Express

After ensuring that Cisco Unified Communications Manager Express and Unity Connection are ready for the integration, do the following procedure to set up the integration and to enter the port settings.

Procedure

- Step 1** In Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Phone System**.
- Step 2** On the Search Phone Systems page, under **Display Name**, select the name of the default phone system.
- Step 3** On the Phone System Basics page, in the **Phone System Name** field, enter the descriptive name that you want for the phone system.

Step 4 If you want to use this phone system as the default for TRaP connections so that administrators and users without voicemail boxes can record and playback through the phone in the Unity Connection web applications, check the **Default TRAP Switch check box**. If you want to use another phone system as the default for TRaP connections, uncheck this check box.

Step 5 Select **Save**.

Step 6 On the Phone System Basics page, in the Related Links drop-down box, select **Add Port Group** and select **Go**.

Step 7 On the New Port Group page, enter the applicable settings and select **Save**.

Table 2: Settings for the New Port Group Page

| Field | Value |
|------------------------------|--|
| Phone System | Select the name of the phone system that you entered in Step 3. |
| Create From | Select Port Group Template and select SIP 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 Unity Connection to authenticate with the Cisco Unified CM Express router. |
| Authentication User Name | Enter the name that Unity Connection uses to authenticate with the Cisco Unified CM Express router. |
| Authentication Password | Enter the password that Unity Connection uses to authenticate with the Cisco Unified CM Express router. |
| Contact Line Name | Enter the voice messaging line name (or pilot number) that users use to contact Unity Connection and registers with the Cisco Unified CM Express router. |
| SIP Security Profile | Select the SIP security profile that Unity Connection uses. Note Cisco Unified CM Express does not support Cisco Unified CM authentication and encryption. |
| SIP Transport Protocol | Select the SIP transport protocol that Unity Connection uses. |
| IPv4 Address or Host Name | Enter the IP address (or host name) of the Cisco Unified CM Express router that you are integrating with Unity Connection. |
| IPv6 Address or Host Name | Do not enter a value in this field. IPv6 is not supported for Cisco Unified CM Express integrations. |
| IP Address or Host Name | Enter the IP address (or host name) of the Cisco Unified CM Express router that you are integrating with Unity Connection. |

| | |
|------|---|
| Port | Enter the IP port of the Cisco Unified CM Express router that you are integrating with Unity Connection. We recommend that you use the default setting. |
|------|---|

Step 8 On the **Port Group Basics** page, in the Related Links drop-down box, select **Add Ports** and select **Go**.

Step 9 On the New Port page, enter the following settings and select **Save**.

Table 3: 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. Note For a Unity Connection cluster, you must enter the total number of voice messaging ports that are used by all Unity Connection servers. Each port is later assigned to a specific Unity Connection server. |
| Phone System | Select the name of the phone system that you entered in Step 3. |
| Port Group | Select the name of the port group that you added in Step 7. |
| Server | Select the name Unity Connection server. |

Step 10 On the Search Ports page, select the display name of the first voice messaging port that you created for this phone 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 11 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 4: Settings for the Voice Messaging Ports

| Field | Setting |
|---------|--|
| 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 | (<i>For Unity Connection clusters only</i>) Select the name of the Unity Connection server that you want to handle this port. Assign an equal number of answering and dial-out voice messaging ports to the Unity Connection servers so that they equally share the voice messaging traffic. |
| 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 Unity Connection web applications. Assign Allow TRAP Connections to the least busy ports. |

Step 12 Select **Save**.

Step 13 Select **Next**.

Step 14 Repeat Step 11 through Step 13 for all remaining voice messaging ports for the phone system.

Step 15 If another phone system integration exists, in Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Trunk**. Otherwise, skip to Step 19.

Step 16 On the Search Phone System Trunks page, on the Phone System Trunk menu, select New Phone System **Trunk**.

Step 17 On the **New Phone System Trunk** page, enter the following settings for the phone system trunk and select **Save**.

Table 5: Settings for the Phone System Trunk

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

Step 18 Repeat Step 16 and Step 17 for all remaining phone system trunks that you want to create.

Step 19 In the Related Links drop-down list, select **Check Telephony Configuration** and select **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 Unity Connection again.

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

Testing the Integration

To test whether 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 “[Installing Cisco Unity Connection](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/14/install_upgrade/guide/b_14cuciumg.html)” chapter of the *Install, Upgrade, and Maintenance Guide for Cisco Unity Connection* Release 14, available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/14/install_upgrade/guide/b_14cuciumg.html.
- *Troubleshooting Guide for Cisco Unity Connection*, Release 14 at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/14/troubleshooting/guide/b_14cuctsg.html.

Setting Up the Test Configuration

Procedure

- Step 1** Set up two test extensions (Phone 1 and Phone 2) on the same phone system that Unity Connection is connected to.
- Step 2** Set Phone 1 to forward calls to the Unity Connection pilot number when calls are not answered.
- Caution** The phone system must forward calls to the Unity Connection pilot number in no fewer than four rings. Otherwise, the test may fail.
- Step 3** In Cisco Unity Connection Administration, expand **Users**, then select **Users**.
- Step 4** On the Search Users page, select the display name of a user to use for testing. The extension for this user must be the extension for Phone 1.
- Step 5** On the Edit User Basics page, uncheck the **Set for Self-enrollment at Next Login** check box.
- Step 6** In the **Voice Name** field, record a recorded name for the test user.
- Step 7** Select **Save**.
- Step 8** On the **Edit** menu, select **Message Waiting Indicators**.
- Step 9** On the Message Waiting Indicators page, select the message waiting indicator. If no message waiting indication is in the table, select **Add New**.
- Step 10** On the Edit Message Waiting Indicator page, enter the following settings.

Table 6: Table 15 Settings for the Edit MWI Page

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

- Step 11** Select **Save**.
- Step 12** On the **Edit** menu, select **Transfer Rules**.
- Step 13** On the Transfer Rules page, select the active transfer rule.
- Step 14** On the Edit Transfer Rule page, under **Transfer Action**, select **Extension** and enter the extension of Phone 1.
- Step 15** In the **Transfer Type** field, select **Release to Switch**.
- Step 16** Select **Save**.
- Step 17** Minimize the Cisco Unity Connection Administration window.
Do not close the Cisco Unity Connection Administration window because you use it again in a later procedure.
- Step 18** Sign in to the Real-Time Monitoring Tool (RTMT).
- Step 19** On the **Unity Connection** menu, select **Port Monitor**. The Port Monitor tool appears in the right pane.
- Step 20** In the right pane, select **Start Polling**. The Port Monitor displays which port is handling the calls that you make.
-

Testing an External Call with Release Transfer

Procedure

- 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 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 ring back tone means that 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 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 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 Unity Connection are successfully integrated for turning on MWIs.
-

Testing Listening to Messages

Procedure

- Step 1** From Phone 1, enter the internal pilot number for 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 Unity Connection, which correctly interpreted the information.
- Step 3** Confirm that you hear the recorded name for the test user (if you did not record a name for the test user, you hear the extension number for Phone 1). Hearing the recorded name means that 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 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.
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Setting Up Supervised Transfer on Cisco Unity Connection

Procedure

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

Adding New User Templates 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 is 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 assign users to the new phone system. You must add the new templates before you add new users who are assigned to the new phone system. For details on adding new user templates, or on selecting a user template when adding a new user, see the User Templates section of the “User Attributes” chapter of the *System Administration Guide for Cisco Unity Connection, Release 14*, available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/14/administration/guide/b_14cucsag.html.



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