QSIG-Enabled Phone System with Cisco ISR Voice Gateway Integration Guide for Cisco Unity Connection Release 15

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This document provides instructions for integrating a QSIG-enabled phone system with Cisco Unity Connection through a Cisco ISR voice gateway.

Integration Tasks

Confirm that Unity Connection is installed as per the steps mentioned in the "Installing Cisco Unity Connection" chapter of the *Install, Upgrade, and Maintenance Guide for Cisco Unity Connection Release 15,* available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/install_upgrade/guide/b_15cuciumg.html.

After completing the installation of Unity Connection, follow the tasks in the "Task List to Create the Integration" to complete the integration of Unity Connection with a QSIG-enabled phone system through a Cisco ISR voice gateway.

Task List to Create the Integration

Do the following steps in the given task list to integrate Unity Connection with a QSIG-enabled phone system through a Cisco ISR voice gateway.

1. Review the system and equipment requirements to confirm that all phone system and Unity Connection requirements have been met. See the Prerequisites.

2. Plan how the voice messaging ports are used by Unity Connection. See the Planning the Usage of Voice Messaging Ports.

3. Program the QSIG-enabled phone system. See the Programming the QSIG-Enabled Phone Systems.

4. Configure the Cisco ISR voice gateway. See the Configuring the Cisco ISR Voice Gateway.

5. Create the integration. See the "Creating a New Integration with QSIG-enabled Phone System" section.

6. Test the integration. See the Testing the Integration.

7. 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.

> **Note** While integrating the Cisco Unity Connection with Cisco Unified Call Manager through a QSIG- Enabled Phone system with Cisco ISR Voice gateway 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.

Prerequisites

The QSIG-enabled integration supports configurations of the following components:

Phone System

- A QSIG-enabled phone system.

- The phone system is ready for the integration.

Cisco ISR Voice Gateway

- Cisco IOS version 12.4(11)T or later.

- The QSIG-enabled phone system connected to the Cisco ISR voice gateway.

Unity Connection Server

- 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 15,* available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/install_upgrade/guide/b_15cuciumg.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 voice mail. 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 15* at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/design/guide/b_15cucdg.html.

Integration Description

This integration uses a Cisco ISR voice gateway and a LAN or WAN to connect Unity Connection and a QSIG-enabled phone system. The Cisco ISR voice gateway converts the QSIG communications to SIP. Figure 1: Connections between the Phone System and Unity Connection shows the required connections.

Figure 1: Connections between the Phone System and Unity Connection
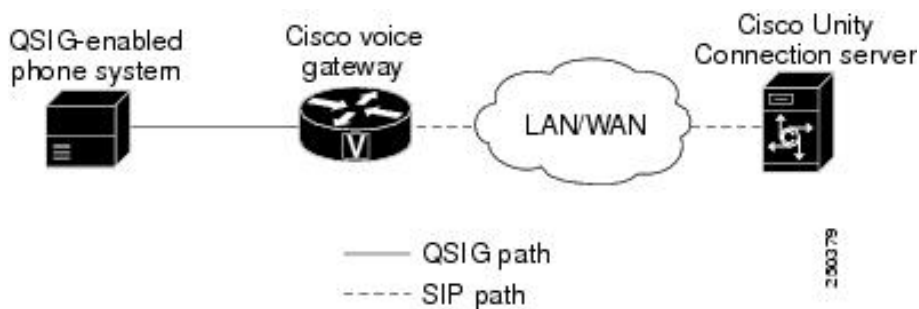
Call Information

The QSIG-enabled phone system integration 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 QSIG-enabled phone system 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 identifies the user who leaves a message during a forwarded internal call, based on the extension from which the call originated)

- Message waiting indication (MWI)

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 multiple 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 15* at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/integration/multiple/b_cuc15intmultiple.html.

> **Note** In Unity Connection integration with multiple phone systems is not supported for use with Cisco Business Edition 5000 and is supported only with Cisco Business Edition 6000/7000.

Planning the Usage of Voice Messaging Ports

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 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.

> ✎
>
> **Note**   The Cisco ISR voice gateway performs transfers by hairpinning two independent calls across two b-channels on the QSIG trunk. Hairpinned calls use more QSIG channels in comparison to the number of Cisco Unity Connection voice messaging ports that are available to answer calls.
>
> Release (blind) transfers that are forwarded back to Unity Connection uses three b-channels for the remainder of the call. However, supervised transfers pull back the consulting call when the target is unavailable so that only one b-channel is used for the remainder of the call.

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 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	*(When a Unity Connection cluster is configured)* 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.
	All voice messaging ports connecting to the Cisco ISR voice gateway 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. 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.

Field	Considerations
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.

Determining the Number of Voice Messaging Ports

The following tasks describe the process for determining the number of voice messaging ports for Cisco Unity Connection to install, answer call and dial out calls:

- For determining the number of voice messaging ports to Install, see Voice Messaging Ports to Install.

- For determining the number of voice messaging ports to Answer Calls, see Voice Messaging Ports to Answer Calls.

- For determining the number of voice messaging ports to Dial Out, see Voice Messaging Ports to Dial Out.

Voice Messaging Ports to Install

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

- The number of calls Unity Connection answer 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 that are 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.

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.

Voice Messaging Ports to Answer Calls

The calls that the voice messaging ports answer can be incoming calls from unidentified callers or from users. Assign all of the voice messaging ports to answer calls.

You can set voice messaging ports to both answer calls and to dial out (for example, to send message notifications).

Note The Cisco ISR voice gateway performs transfers by hairpinning two independent calls across two b-channels on the QSIG trunk. Hairpinned calls use more QSIG channels in comparison to the number of Unity Connection voice messaging ports that are available to answer calls. If your system uses Unity Connection auto-attendant transfers, you must provision a larger number of QSIG b-channels than the number of Unity Connection voice messaging ports that answer calls.

If your system is configured for a Unity Connection cluster, see the Considerations for a Unity Connection Cluster.

Voice Messaging Ports to Dial Out

Ports that only dial out 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 Unity 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.

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 Normally

- A hunt group is configured on the phone system to distribute calls equally to both Unity Connection servers.

- The network 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 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 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 do not be able to answer calls or to dial out.

- Each Unity Connection server must have voice messaging ports that answer calls and that 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 that 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 is not able to answer incoming calls. Similarly, if the functioning Unity Connection server does not have voice messaging ports for dialing out, the system is not able to dial out (for example, to set MWIs).

Programming the QSIG-Enabled Phone Systems

Note that you must program each extension to forward calls to the pilot number assigned to the voice messaging ports, based on one of the call transfer types shown in Table 2: Call Transfer Types.

Table 2: Call Transfer Types

Transfer Types	Usage
Release transfer (blind transfer)	Program the phone to forward calls to the pilot number when: • The extension is busy • The call is not answered
Supervised transfer	Program the phone to forward calls to the pilot number only when the call is not answered (on the phone system, the number of rings before forwarding must be more than the number of rings to supervise the call). Confirm that call forwarding is disabled when the extension is busy.

Configuring the Cisco ISR Voice Gateway

> **Note** If the Unity Connection SIP Port setting cannot be 5060 (for example, it is set to 5061), you must change the SIP port that the Cisco ISR voice gateway uses. You can use a command similar to the following: `dial-peer voice 1 voip  session protocol sipv2  session target ipv4:10.00.00.00:5061`
>
> For a Unity Connection cluster, identify the Unity Connection servers with a fully qualified domain name (FQDN), and configure a DNS server to resolve the FQDN to the IP addresses and SIP ports of the Unity Connection server.

Creating an Integration

Procedure

Step 1 In Cisco Unity Connection Administration, expand **Telephony Integrations** > **and** 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 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 3: Settings for the New Port Group Page

Field	Setting
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	Confirm that this check box is unchecked.
Authentication User Name	Leave this field blank.
Authentication Password	Leave this field blank.
Contact Line Name	Enter the voice messaging pilot number that matches the dial plan configuration of the gateway.
SIP Security Profile	Select **5060**.
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 primary gateway that you are connecting to Unity Connection.
IPv6 Address or Host Name	Do not enter a value in this field. IPv6 is not supported for this type of integration.
IP Address or Host Name	Enter the IP address (or host name) of the primary gateway that you are connecting to Unity Connection.
Port	Enter the IP port of the primary gateway that you are connecting to Unity Connection. We recommend that you use the default setting. This setting must match the port setting of the gateway. Otherwise the integration does not function correctly.

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 4: Settings for the New Ports Page

Field	Considerations
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 of the 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 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 11 On the Port Basics page, enter the following settings. The fields in the following table are the ones that you can change.

Table 5: 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	*(For Unity Connection clusters only)* 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 6: 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 installation guide for the phone system.

- *Troubleshooting Guide for Cisco Unity Connection Release 15*, available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/troubleshooting/guide/b_15cuctsg.html.

- The setup information earlier in this guide.

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** > **and** 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 voice name for the test user and select **Save**.

Step 7   On the Edit menu, select **Message Waiting Indicators**.

Step 8   On the Message Waiting Indicators page, select the message waiting indicator. If no message waiting indication is in the table, select **Add New**.

Step 9   On the Edit Message Waiting Indicator page, enter the following settings.

Table 7: 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 10   Select **Save**.

Step 11   On the Edit menu, select **Transfer Rules**.

Step 12   On the Transfer Rules page, select the active option.

Step 13   On the Edit Transfer Rule page, under Transfer Action, select the **Extension** option and enter the extension of Phone 1.

Step 14   In the Transfer Type field, select **Release to Switch**.

Step 15   Select **Save**.

Step 16   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 17   Sign in to the Real-Time Monitoring Tool (RTMT).

Step 18   On the Unity Connection menu, select **Port Monitor**. The Port Monitor tool appears in the right pane.

Step 19   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 ringback 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 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 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 the 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 voice name for the test user (if you did not record a voice name for the test user, you hear the extension number for Phone 1). Hearing the voice 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.

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.

Testing Supervised 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 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 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 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 Select **Stop Polling**.

Step 10 Sign out of RTMT.

Adding New User Templates for Multiple Integrations

When you create the first phone system integration, this phone system is automatically selected in the default user template. The users that you add after creating this phone system integration are 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 in "User Attributes" chapter of the *System Administration Guide for Cisco Unity Connection Release 15*. The guide is available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/administration/guide/b_15cucsag.html.

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