



# QSIG/DPNSS Phone System with Cisco EGW 2200 Integration Guide for Cisco Unity 5.0

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**Revised April 7, 2009**

This document provides instructions for integrating a QSIG/DPNSS phone system with Cisco Unity through a Cisco EGW 2200.

## Integration Tasks

Before doing the following tasks to integrate Cisco Unity with a QSIG/DPNSS phone system through a Cisco EGW 2200, confirm that the Cisco Unity server is ready for the integration by completing the applicable tasks in the applicable Cisco Unity installation guide.

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 new integration with a QSIG/DPNSS phone system. If you are installing a new Cisco Unity server by using the applicable Cisco Unity installation guide, you may have already completed some of the following tasks.

1. Review the system and equipment requirements to confirm that all phone system and Cisco Unity server requirements have been met. See the [“Requirements” section on page 2](#).
2. Plan how the voice messaging ports will be used by Cisco Unity. See the [“Planning How the Voice Messaging Ports Will Be Used by Cisco Unity” section on page 4](#).
3. Program the QSIG/DPNSS phone system and extensions. See the [“Programming the QSIG/DPNSS Phone System” section on page 6](#).
4. Set up the Cisco EGW 2200. See the [Setting Up the Cisco EGW 2200, page 6](#).
5. Create the integration. See the [Creating an Integration with the Phone System, page 6](#).
6. Test the integration. See the [Testing the Integration, page 9](#).
7. If you have a secondary server for Cisco Unity failover, integrate the secondary server. See the [“Integrating a Secondary Server for Cisco Unity Failover” section on page 12](#).



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# Requirements

The QSIG/DPNSS integration supports configurations of the following components:

## Phone System

- A QSIG/DPNSS phone system.
- The phone system is ready for the integration as described in the documentation for the phone system and for the Cisco EGW 2200.
- A VoIP gateway configured and connected to the QSIG/DPNSS phone system.

## Cisco Unity Server

- Cisco Unity installed and ready for the integration, as described in the applicable Cisco Unity installation guide at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod\\_installation\\_guides\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod_installation_guides_list.html).
- A license that enables the applicable number of voice messaging ports.
- A Cisco EGW 2200 ready for the integration as described in the documentation for the Cisco EGW 2200.
- The Cisco EGW 2200 connected to the LAN and configured for a QSIG/DPNSS backhaul signaling stream from the VoIP gateway.

## Centralized Voice Messaging

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

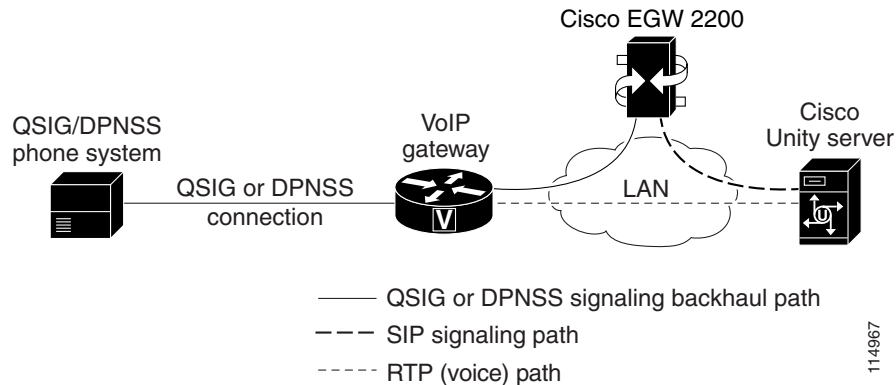
[http://www.cisco.com/en/US/docs/voice\\_ip\\_comm/unity/5x/design/guide/5xcudgx.html](http://www.cisco.com/en/US/docs/voice_ip_comm/unity/5x/design/guide/5xcudgx.html).

# Integration Description

The QSIG/DPNSS integration uses a Cisco EGW 2200 (which translates QSIG or DPNSS call signaling into SIP), a VoIP gateway, and the LAN to connect Cisco Unity and a QSIG/DPNSS phone system.

[Figure 1](#) shows the required connections.

**Figure 1** Connections Between a QSIG/DPNSS Phone System and Cisco Unity



## Call Information

The QSIG/DPNSS 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)

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

## Integration Functionality

The QSIG/DPNSS integration with Cisco Unity provides the following integration features:

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

# Integrations with Multiple Phone Systems

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


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

Before programming the phone system, you need to plan how the voice messaging ports will be used by Cisco Unity. The following considerations will affect the programming for the phone system (for example, setting up the hunt group or call forwarding for the voice messaging ports):

- The number of voice messaging ports installed.
- The number of voice messaging ports that will answer calls.
- The number of voice messaging ports that will only dial out, for example, to send message notification, to make AMIS deliveries, and to make telephone record and playback (TRAP) connections.

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

**Table 1** Settings for the Voice Messaging Ports

Field	Considerations
Enabled	Check this check box.
Answer Calls	Check this check box to designate the port for answering calls.   <b>Caution</b> All voice messaging ports connecting to the Cisco EGW 2200 must have the Answer Calls box checked. Otherwise, calls to Cisco Unity may not be answered.
Message Notification	Check this check box to designate the port for notifying subscribers of messages.
AMIS Delivery <i>(available with the AMIS licensed feature only)</i>	Check this check box to designate the port for making outbound AMIS calls to deliver voice messages from Cisco Unity subscribers to users on another voice messaging system. Cisco Unity supports the Audio Messaging Interchange Specification (AMIS) protocol, which provides an analog mechanism for transferring voice messages between different voice messaging systems.  This setting affects outbound AMIS calls only. All ports are used for incoming AMIS calls.  Because the transmission of outgoing AMIS messages may tie up voice ports for long periods of time, you may want to adjust the schedule on the Network > AMIS > Schedule page so that outgoing AMIS calls are placed during closed hours or at times when Cisco Unity is not processing many calls.
TRAP Connection	Check this check box so that subscribers can use the phone as a recording and playback device in Cisco Unity web applications and e-mail clients. Assign TRAP Connection to the least busy ports.

### The Number of Voice Messaging Ports to Install

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

- The number of calls Cisco Unity will answer when call traffic is at its peak.
- The expected length of each message that callers will record and that subscribers will listen to.
- The number of subscribers.
- The number of ports that will be set to dial out only.
- The number of calls made for message notification.
- The number of AMIS delivery calls.
- The number of TRAP connections needed when call traffic is at its peak. (TRAP connections are used by Cisco Unity web applications to play back and record over the phone.)
- The number of calls that will use the automated attendant and call handlers when call traffic is at its peak.

It is best to install only the number of voice messaging ports that are needed so that system resources are not allocated to unused ports.

### The Number of Voice Messaging Ports That Will Answer Calls

The calls that the voice messaging ports answer can be incoming calls from unidentified callers or from subscribers. Typically, the voice messaging ports that answer calls are the busiest.

You can set voice messaging ports to both answer calls and to dial out (for example, to send message notifications). However, when the voice messaging ports perform more than one function and are very active (for example, answering many calls), the other functions may be delayed until the voice messaging port is free (for example, message notifications cannot be sent until there are fewer calls to answer). For best performance, dedicate certain voice messaging ports for only answering incoming calls, and dedicate other ports for only dialing out. Separating these port functions eliminates the possibility of a collision, in which an incoming call arrives on a port at the same time that Cisco Unity takes the port off-hook to dial out.

### The Number of Voice Messaging Ports That Will Dial Out

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

- Notify subscribers by phone, pager, or e-mail of messages that have arrived.
- Make outbound AMIS calls to deliver voice messages from Cisco Unity subscribers to users on another voice messaging system. (This action is available only with the AMIS licensed feature.)
- Make a TRAP connection so that subscribers can use the phone as a recording and playback device in Cisco Unity web applications.

### Preparing for Programming the Phone System

Record your decisions about the voice messaging ports to guide you in programming the phone system.

# Programming the QSIG/DPNSS Phone System

For information on provisioning a QSIG or DPNSS phone system to integrate with Cisco Unity, see the documentation for the Cisco EGW 2200.

## Setting Up the Cisco EGW 2200

For information on setting up the Cisco EGW 2200, see the documentation for the Cisco EGW 2200.

## Creating an Integration with the Phone System

After ensuring that the QSIG/DPNSS phone system and the Cisco Unity server are ready for the integration, do the following procedures to set up the integration and to enter the port settings.

### To Create an Integration

- Step 1** If UTIM is not already open, on the Windows Start menu of the Cisco Unity server, click **Programs > Cisco Unity > Manage Integrations**. UTIM appears.
- Step 2** In the left pane of the UTIM window, click **Cisco Unity Server**.
- Step 3** On the Integration menu of the UTIM window, click **New**. The Telephony Integration Setup Wizard appears.
- Step 4** On the Welcome page, click **SIP (including CUCM/CCM)** and click **Next**.
- Step 5** On the Name This SIP Integration and Clusters page, enter the following settings, then click **Next**.

**Table 2** Settings for the Name This SIP Integration and Cluster Page

Field	Setting
Integration Name	Enter the name you will use to identify this QSIG/DPNSS integration. Accept the default name or enter another name.
Cluster Name	Enter the name you will use to identify this Cisco EGW 2200 or SIP proxy server cluster. Accept the default name or enter another name.

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

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

Field	Setting
Primary: IP Address/Name	Enter the first virtual IP address (or name alias) of the primary Cisco EGW 2200 or SIP proxy server that you are connecting to Cisco Unity.
Primary: Port	Enter the IP port of the primary Cisco EGW 2200 or SIP proxy server that you are connecting to Cisco Unity.

**Table 3 Settings for the Enter Primary and Secondary SIP Server Page (continued)**

Field	Setting
Secondary: IP Address/Name	(Optional) Enter the second virtual IP address (or name alias) of the secondary Cisco EGW 2200 or SIP proxy server that you are connecting to Cisco Unity.
Secondary: Port	(Optional) Enter the IP port of the secondary Cisco EGW 2200 or SIP proxy server that you are connecting to Cisco Unity.

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

**Step 7** On the Set Number of Cisco Unity Voice Messaging Ports page, enter the number of voice messaging ports on Cisco Unity that you want to connect to the Cisco EGW 2200 or the SIP proxy server, then click **Next**.

This number must not be more than the number of ports set up on the Cisco EGW 2200 or the SIP proxy server.

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

**Table 4 Settings for the Configure Cisco Unity SIP Settings Page**

Field	Setting
Contact Line Name	Enter the voice messaging pilot number that matches the Cisco EGW 2200 dial plan configuration.
Cisco Unity SIP Port	Enter the IP port on Cisco Unity that callers and the Cisco EGW or the SIP proxy server use to connect to voice mail. We recommend using the default setting.
Preferred Codec	Click the codec that Cisco Unity will first attempt to use on outgoing calls.
Preferred Transport Protocol	Click <b>UDP</b> .

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

**Table 5 Settings for the Enter SIP Server Authentication Page**

Field	Setting
Authenticate with the SIP Server	If you do not use a SIP proxy server, uncheck the check box. If you use a SIP proxy server, indicate whether Cisco Unity will authenticate with the SIP proxy server.
Name	Enter the name the SIP proxy server will use for authentication.
Password	Enter the password the SIP proxy server will use for authentication.

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

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

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

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

**Table 6 Selection Controls for the Reassign Subscribers Page**

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

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

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

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

**Table 7 Selection Controls for the Reassign Call Handlers Page**

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

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

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

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

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

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**Step 1** After the Cisco Unity services restart, on the View menu, click **Refresh**.


**Step 2** In the left pane of the UTIM window, expand the phone system integration that you are creating.

**Step 3** In the left pane, click the name of the phone system.

**Step 4** In the right pane, click the **Ports** tab.

**Step 5** Enter the settings shown in [Table 8](#) for the voice messaging ports.

**Table 8**      **Settings for the Voice Messaging Ports**

Field	Considerations
Extension	Check this check box.
Enabled	Check this check box.   <b>Caution</b> All voice messaging ports connecting to the Cisco EGW 2200 must have the Answer Calls box checked. Otherwise, calls to Cisco Unity may not be answered.
Answer Calls	Check this check box to designate the port for answering calls. These calls can be incoming calls from unidentified callers or from subscribers.
Message Notification	Check this check box to designate the port for notifying subscribers of messages.
AMIS Delivery <i>(available with the AMIS licensed feature only)</i>	Check this check box to designate the port for making outbound AMIS calls to deliver voice messages from Cisco Unity subscribers to users on another voice messaging system. Cisco Unity supports the Audio Messaging Interchange Specification (AMIS) protocol, which provides an analog mechanism for transferring voice messages between different voice messaging systems.  This setting affects outbound AMIS calls only. All ports are used for incoming AMIS calls.  Because the transmission of outgoing AMIS messages may tie up voice ports for long periods of time, you may want to adjust the schedule on the Network > AMIS > Schedule page so that outgoing AMIS calls are placed during closed hours or at times when Cisco Unity is not processing many calls.
TRAP Connection	Check this check box so that subscribers can use the phone as a recording and playback device in Cisco Unity web applications and e-mail clients.

**Step 6**      Click **Save**.

**Step 7**      Exit UTIM.

## Testing the Integration

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

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

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

### To Set Up the Test Configuration

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

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

**Caution**

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

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

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

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

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

- a. In the Cisco Unity Administrator, go to the **Subscribers > Subscribers > Profile** page.
- b. Click the **Add** icon.
- c. In the New Subscriber field, click **Notes**.
- d. In the Address Book list, confirm that the address book listed is the one that contains the user data that you want to import.  
If the address book that you want to use is not listed, go to the **System > Configuration > Subscriber Address Books** page and add a different address book.
- e. In the Find Domino Person By list, indicate whether to search by short name, first name, or last name.
- f. Enter the applicable short name or name. You also can enter \* to display a list of all users, or enter one or more characters followed by \* to narrow your search.
- g. Click **Find**.
- h. On the list of matches, click the name of the user to import.
- i. On the Add Subscriber page, enter the applicable information.
- j. Click **Add**.

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

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

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

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

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

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

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

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

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

- Step 12** In the navigation bar, click **Messages** to go to the Subscribers > Subscribers > Messages page for the test subscriber.
- Step 13** Under Message Waiting Indicators (MWIs), check **Use MWI for Message Notification**.
- Step 14** In the Extension field, enter **x**.
- Step 15** Click the **Save** icon.
- Step 16** Open the Status Monitor by doing one of the following:
  - In Internet Explorer, go to **http://<Cisco Unity server name>/web/sm**.
  - Double-click the desktop shortcut to the Status Monitor.
  - In the status bar next to the clock, right-click the Cisco Unity tray icon and click **Status Monitor**.

**To Test an External Call with Release Transfer**

- Step 1** From Phone 2, enter the access code necessary to get an outside line, then enter the number outside callers use to dial directly to Cisco Unity.
- Step 2** On the Status Monitor, note which port handles this call.
- Step 3** When you hear the opening greeting, enter the extension for Phone 1. Hearing the opening greeting means that the port is configured correctly.
- Step 4** Confirm that Phone 1 rings and that you hear a ringback tone on Phone 2. Hearing a ringback tone means that Cisco Unity correctly released the call and transferred it to Phone 1.
- Step 5** Leaving Phone 1 unanswered, confirm that the state of the port handling the call changes to “Idle.” This state means that release transfer is successful.
- Step 6** Confirm that, after the number of rings that the phone system is set to wait, the call is forwarded to Cisco Unity and that you hear the greeting for the test subscriber. Hearing the greeting means that the phone system forwarded the unanswered call and the call-forward information to Cisco Unity, which correctly interpreted the information.
- Step 7** On the Status Monitor, note which port handles this call.
- Step 8** Leave a message for the test subscriber and hang up Phone 2.
- Step 9** On the Status Monitor, confirm that the state of the port handling the call changes to “Idle.” This state means that the port was successfully released when the call ended.
- Step 10** Confirm that the MWI on Phone 1 is activated. The activated MWI means that the phone system and Cisco Unity are successfully integrated for turning on MWIs.

**To Test Listening to Messages**

- Step 1** From Phone 1, enter the internal pilot number for Cisco Unity.
- Step 2** When asked for your password, enter the default password. Hearing the request for your password means that the phone system sent the necessary call information to Cisco Unity, which correctly interpreted the information.
- Step 3** Confirm that you hear the recorded voice name for the test subscriber (if you did not record a voice name for the test subscriber, you will hear the extension number for Phone 1). Hearing the voice name means that Cisco Unity correctly identified the subscriber by the extension.

- Step 4** When asked whether you want to listen to your message, press **1**.
  - Step 5** After listening to the message, press **3** to delete the message.
  - Step 6** Confirm that the MWI on Phone 1 is deactivated. The deactivated MWI means that the phone system and Cisco Unity are successfully integrated for turning off MWIs.
  - Step 7** Hang up Phone 1.
  - Step 8** On the Status Monitor, confirm that the state of the port handling the call changes to “Idle.” This state means that the port was successfully released when the call ended.
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#### To Delete the Test Subscriber

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

The Cisco Unity failover feature enables a secondary server to provide voice messaging services when the primary server becomes inactive. For information on installing a secondary server for failover, refer to the applicable Cisco Unity installation guide, available at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod\\_installation\\_guides\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod_installation_guides_list.html).

For information on failover, refer to the *Cisco Unity Failover Configuration and Administration Guide* at [http://cisco.com/en/US/products/sw/voicesw/ps2237/products\\_feature\\_guides\\_list.html](http://cisco.com/en/US/products/sw/voicesw/ps2237/products_feature_guides_list.html).

## Requirements

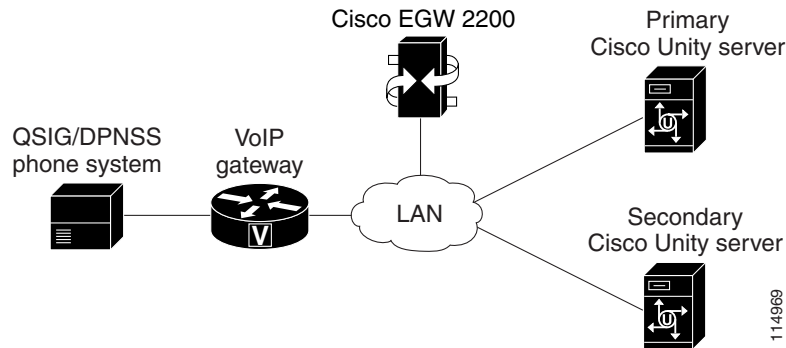
The following components are required to integrate a secondary server:

- One secondary server for each primary server installed and ready for the integration, as described in the applicable Cisco Unity installation guide and earlier in this integration guide.
- A license that enables failover.

## Integration Description

The QSIG/DPNSS phone system communicates with both the primary and secondary servers through the LAN or WAN. [Figure 2](#) shows the required connections.

**Figure 2** Connections Between a QSIG/DPNSS Phone System and the Cisco Unity Servers



The primary and secondary servers act in the following manner:

- When the primary server is operating normally, the secondary server is inactive.
- When the primary server becomes inactive, the secondary server becomes active.
- When the primary server becomes active again, the secondary server becomes inactive.

## Setting Up the Secondary Server for Failover

Do the following procedure to integrate the secondary server.

### To Set Up the Secondary Server for Failover

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- Step 1** Install a secondary server with the same configuration as the primary server. For installation instructions, refer to the applicable Cisco Unity installation guide.
  - Step 2** On the Windows Start menu of the secondary server, click **Programs > Cisco Unity > Manage Integrations**. The UTIM window appears.
  - Step 3** On the Integration menu of the UTIM window, click **New**. The Telephony Integration Setup Wizard appears.
  - Step 4** Enter the settings to match the integration settings on the primary server.



**Note** We recommend not reassigning any unassigned subscribers and call handlers to the new integration, if you are asked by the wizard. Failover replication will automatically assign the correct integration.

- Step 5** At the prompt to restart the Cisco Unity services, click **Yes**.



**Note** When restarting the Cisco Unity services, use the UTIM prompt instead of the Cisco Unity icon in the Windows taskbar. The taskbar icon does not restart all of the Cisco Unity services.

- Step 6** After Cisco Unity restarts, on the Windows Start menu of the Cisco Unity server, click **Programs > Cisco Unity > Manage Integrations**. UTIM appears.
- Step 7** In the left pane of the UTIM window, click the phone system integration that you created in [Step 3](#).
- Step 8** In the right pane, click **Properties**.

- Step 9** On the Integration tab, compare the setting of the Integration ID field for the secondary server to the setting of the Integration ID field for the primary server.
- Step 10** If the integration IDs of the phone system on the primary and secondary server are the same, continue to [Step 16](#).
- If the integration IDs of the phone system on the primary and secondary servers are different, on the secondary server, click **Modify Integration ID**.
- Step 11** When cautioned that subscribers associated with the current Integration ID setting will not be automatically associated with the new Integration ID setting, click **OK**.
- Step 12** In the Modify Integration ID dialog box, in the Enter New Integration ID field, enter the Integration ID setting for the phone system on the primary server and click **OK**.
- Step 13** Click **Save**.
- Step 14** At the prompt to restart the Cisco Unity services, click **Yes**.
- Step 15** In the left pane, click the phone system integration that you created in [Step 3](#).
- Step 16** In the right pane, click the **Ports** tab.
- Step 17** Enter the port settings to match the port settings on the primary server.
- Step 18** Click **Save**.
- Step 19** Exit UTIM.

## Appendix: Documentation and Technical Assistance

### Conventions

The *QSIG/DPNSS Phone System with Cisco EGW 2200 Integration Guide for Cisco Unity 5.0* uses the following conventions.

**Table 9** *QSIG/DPNSS Phone System with Cisco EGW 2200 Integration Guide for Cisco Unity 5.0 Conventions*

Convention	Description
boldfaced text	Boldfaced text is used for: <ul style="list-style-type: none"> <li>Key and button names. (Example: Click <b>OK</b>.)</li> <li>Information that you enter. (Example: Enter <b>Administrator</b> in the User Name box.)</li> </ul>
< > (angle brackets)	Angle brackets are used around parameters for which you supply a value. (Example: In the Command Prompt window, enter <b>ping &lt;IP address&gt;</b> .)

**Table 9** *QSIG/DPNSS Phone System with Cisco EGW 2200 Integration Guide for Cisco Unity 5.0 Conventions (continued)*

Convention	Description
- (hyphen)	Hyphens separate keys that must be pressed simultaneously. (Example: Press <b>Ctrl-Alt-Delete</b> .)
> (right angle bracket)	A right angle bracket is used to separate selections that you make: <ul style="list-style-type: none"> <li>On menus. (Example: On the Windows Start menu, click <b>Settings &gt; Control Panel &gt; Phone and Modem Options</b>.)</li> <li>In the navigation bar of the Cisco Unity Administrator. (Example: Go to the <b>System &gt; Configuration &gt; Settings</b> page.)</li> </ul>
[x] (square brackets)	Square brackets enclose an optional element (keyword or argument). (Example: [reg-e164])
[x   y] (vertical line)	Square brackets enclosing keywords or arguments separated by a vertical line indicate an optional choice. (Example: [transport tcp   transport udp])
{x   y} (braces)	Braces enclosing keywords or arguments separated by a vertical line indicate a required choice. (Example: {tcp   udp})

The *QSIG/DPNSS Phone System with Cisco EGW 2200 Integration Guide for Cisco Unity 5.0* also uses the following conventions:



**Note**

Means *reader take note*. Notes contain helpful suggestions or references to material not covered in the document.



**Caution**

Means *reader be careful*. In this situation, you might do something that could result in equipment damage or loss of data.

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

## Obtaining Documentation, Obtaining Support, and Security Guidelines

For information on obtaining documentation, obtaining support, providing documentation feedback, security guidelines, and also recommended aliases and general Cisco documents, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

<http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html>

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