Session Initiation Protocol (SIP) Integration Guide for Cisco Unity 5.0

Revised November 26, 2007

This document provides instructions for integrating the phone system with Cisco Unity.

Integration Tasks

Before doing the following tasks to integrate Cisco Unity with the SIP phone system, 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 the SIP 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 SIP proxy server and other call processing components. See the "Programming the SIP Phone System" section on page 5.
4. Set up the SIP gateway that services Cisco Unity. See the "Configure the SIP Gateway Servicing Cisco Unity for the SIP Integration" section on page 6.
5. Create the integration. See the "Creating a New Integration with the SIP Phone System" section on page 7.
6. Test the integration. See the "Testing the Integration" section on page 12.

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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 16.

Requirements

The SIP integration supports configurations of the following components:

Phone System

- SIP proxy server (Cisco SIP Proxy Server).
- SIP-enabled phones (for example, SIP-enabled Cisco IP Phone 7960 or Pingtel xpressa).

  The SIP phones must use the REFER method for call transfers.
- SIP-enabled gateways (for example, Cisco AS5300 Access Server, Cisco 2600 series router, or Cisco 3600 series router) for access to the PSTN.

For details on compatibility of the phone system components with the integration, see the "Appendix: Compatibility of Phone System Components" section on page 18.

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.
- A license that enables the applicable number of voice messaging ports.

Network Configuration

- Cisco Unity server, SIP proxy server, SIP-enabled phones, and SIP-enabled gateways installed on the same subnet (ensures adequate bandwidth and avoids latency issues affecting integration behavior).

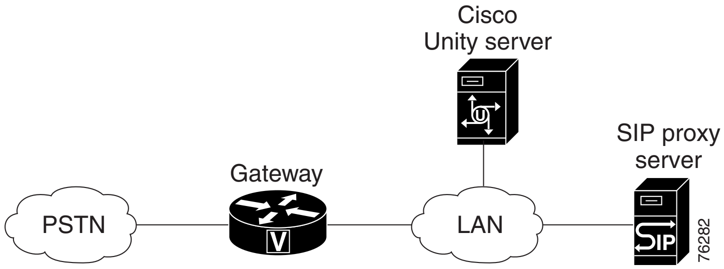
Integration Description

The SIP integration uses the SIP proxy server to set up communications between the voice messaging ports on the Cisco Unity server and the applicable end point (for example, a SIP-enabled phone). The communications occur through:

- An IP network (LAN, WAN, or Internet) to all SIP-enabled devices connected to it.
- A SIP-enabled gateway to the PSTN and all phones connected to it.

Figure 1 shows the connections.

Figure 1 Connections Between the SIP Phone System and Cisco Unity



Call Information

The proxy server sends the following information in the SIP message with the calls forwarded:

- In the Diversion header, the extension of the called party
- In the Diversion header, the reason for the forward (the extension is busy, does not answer, or is set to forward all calls)
- In the From header, the extension of the calling party (for internal calls) or the SIP URL of the calling party (if it is an external call and the system uses caller ID)

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 SIP 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 applicable *Multiple Phone System Integration Guide* at http://www.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.

Unlike other integrations, the hunt group mechanism for SIP integrations is implemented on the Cisco Unity server. Within an integration cluster, each incoming call hunts for an available voice messaging port among all the ports in a round-robin (or circular) fashion. If a voice messaging port in the cluster is set not to answer calls or is not enabled, a call reaching that port may receive a busy signal.

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. **Caution** All voice messaging ports connecting to the SIP proxy server must have the Answer Calls box checked. Otherwise, calls to Cisco Unity may not be answered.
Message Notification	Check this check box to designate the port for notifying subscribers of messages.
AMIS Delivery *(available with the AMIS licensed feature only)*	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.

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 calls made for message notification.
- The number of MWIs that will be activated when call traffic is at its peak.
- 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. 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 set MWIs).

The Number of Voice Messaging Ports That Will Dial Out

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

- Notify subscribers by phone, pager, or e-mail of messages that have arrived.
- Turn MWIs on and off for subscriber extensions.
- 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 SIP Phone System

If you use programming options other than those supplied in the following procedure, the performance of the integration may be affected.

Do the following procedure.

To Program the SIP Phone System

Step 1 Install and set up the SIP proxy server as described in the server documentation.

Step 2 Program each phone to forward calls to <the contact line name>@<SIP proxy server>, the voice messaging line name that subscribers will use to contact Cisco Unity.

Step 3 If Cisco Unity will authenticate with the SIP proxy server, enter a subscriber record for the contact line name that Cisco Unity will use.

Configure the SIP Gateway Servicing Cisco Unity for the SIP Integration

To configure the SIP gateway for the SIP integration with Cisco Unity, do the following three procedures.

To Configure Application Session on the Sip Gateway

Step 1 On the VoIP dial-peer servicing Cisco Unity, use the following command:

```
application session
```

Step 2 Create a destination pattern that matches the voice messaging port numbers. For example, if the system has voice messaging ports 1001 through 1016, enter the dial-peer destination pattern **10xx**.

Step 3 Repeat Step 1 and Step 2 for all remaining VoIP dial-peers servicing Cisco Unity.

To Disable the SIP Media Inactivity Timer

Step 1 On the gateway, go into the gateway configuration mode by entering the following command:

```
Router(config)# gateway
```

Step 2 Disable the RTCP timer by entering the following command:

```
Router(config-gateway)# no timer receive-rtcp
```

Step 3 Exit the gateway configuration mode by entering the following command:

```
Router(config-gateway)# exit
```

To Enable DTMF Relay for SIP Calls by Using Named Telephony Events

Step 1 On the gateway, go into dial-peer configuration mode and define the VoIP dial peer by entering the following command:

```
Router(config)# dial-peer voice <dial peer number> voip
```

Step 2 Configure the SIP protocol on the gateway by entering the following command:

```
Router(config-dial-peer)# session protocol sipv2
```

Step 3 Enable DTMF relay using NTE RTP packets by entering the following command:

```
Router(config-dial-peer)# dtmf-relay rtp-nte
```

Step 4 Configure the type of payload in the NTE packet by entering the following command:

```
Router(config-dial-peer)# rtp payload-type nte <NTE packet payload type>
```

Creating a New Integration with the SIP Phone System

After ensuring that the SIP 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 Cluster page, enter the following settings, then click **Next**.

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

Field	Setting
Integration Name	<the name you will use to identify this SIP integration; accept the default name or enter another name>
Cluster Name	<the name you will use to identify this SIP server cluster; accept the default name or enter another name>

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

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

Field	Setting
Primary: IP Address/Name	<the IP address of the primary SIP server that you are connecting to Cisco Unity>
Primary: Port	<the IP port of the primary SIP server that you are connecting to Cisco Unity>
Secondary: IP Address/Name	<optional; the IP address of the secondary SIP server that you are connecting to Cisco Unity>
Secondary: Port	<optional; the IP port of the secondary SIP 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 Voice Messaging Ports page, enter the number of voice messaging ports on Cisco Unity that you want to connect to the SIP server, then click **Next**.

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

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

Table 4 Settings for the Configure Cisco Unity SIP Settings Page

Field	Setting
Contact Line Name	<the voice messaging line name that subscribers will use to contact Cisco Unity and that Cisco Unity uses to register with the SIP server>
Cisco Unity SIP Port	<the IP port on Cisco Unity that callers and the SIP server use to connect to voice mail; we recommend using the default setting>
Preferred Codec	<the codec Cisco Unity will first attempt to use on outgoing calls>
Preferred Transport Protocol	Click **UDP**.

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	<your indication whether Cisco Unity will authenticate with the SIP server>
Name	<the name the SIP server will use for authentication>
Password	<the password the SIP server will use for authentication>

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

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

If no subscribers appear in the list, click **Next** and continue to Step 12.

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

Table 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**.

Unlike other integrations, the hunt group mechanism for SIP integrations is implemented on the Cisco Unity server. Within an integration cluster, each incoming call hunts for an available voice messaging port among all the ports in a round-robin fashion. If a voice messaging port in the cluster is set not to answer calls or is not enabled, a call reaching that port may receive a busy signal.

To Enter the Voice Messaging Port Settings for the Integration

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

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

Step 3 In the left pane, click the name of the 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
Enabled	Check this check box.
Answer Calls	Check this check box. **Caution** All voice messaging ports connecting to the SIP proxy server must have the Answer Calls box checked. Otherwise, calls to Cisco Unity may not be answered.
Message Notification	Check this check box to designate the port for notifying subscribers of messages.

Table 8 Settings for the Voice Messaging Ports (continued)

Field	Considerations
AMIS Delivery *(available with the AMIS licensed feature only)*	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.

Step 8 When integrated with a SIP phone system, Cisco Unity can use a non-20 ms packet size. See the following table to determine whether you must enable a non-20 ms packet size for your system.

Cisco Unity uses only the 20 ms packet size	Non-20 ms packet sizes are not needed. Skip to the "Testing the Integration" section on page 12.
Cisco Unity uses a non-20 ms packet size	If your system meets one of the following conditions, you must enable a non-20 ms packet size on Cisco Unity: • The SIP endpoints do not support ptime, and the SIP phone system uses a non-20 ms packet size. • You want Cisco Unity to initiate calls with a non-20ms packet size. Continue to the procedure "To Enable a Non-20 ms Packet Size on Cisco Unity" procedure on page 10. **Note** Cisco Unity does not support different packet-size intervals for sending and receiving.

To Enable a Non-20 ms Packet Size on Cisco Unity

Step 1 On the Windows Start menu, click **Run**.

Step 2 Enter **regedit** and click **OK**. The Registry Editor window appears.

Caution Changing the wrong registry key or entering an incorrect value can cause the server to malfunction. Before you edit the registry, confirm that you know how to restore it if a problem occurs. (Refer to the "Restoring" topics in Registry Editor Help.) Note that for Cisco Unity failover, registry changes on one Cisco Unity server must be made manually on the other Cisco Unity server, because registry changes are not replicated. If you have any questions about changing registry key settings, contact Cisco TAC.

Step 3 If you do not have a current backup of the registry, click **Registry > Export Registry File**, and save the registry settings to a file.

Step 4 Select the key HKEY_LOCAL_MACHINE\Software\ActiveVoice\MIU\1.0\Initialization\Integrations\Integration<n> where <n> is the number of the SIP integration.

Step 5 On the Edit menu, click New > **DWORD Value**.

Step 6 In the right pane, for the name of the new value, enter **G711 Packetization** and press **Enter**.

Step 7 On the Edit menu, click New > **DWORD Value**.

Step 8 In the right pane, for the name of the new value, enter **G729 Packetization** and press **Enter**.

Step 9 Double-click the **G711 Packetization** value.

Step 10 In the Edit DWORD Value dialog box, Under Base, click **Decimal**.

Step 11 In the Value Data field, enter one of the following values that you want to use for the packet size (in ms) for the G.711 codec:

- 10
- 20
- 30

Caution If you enter a setting other than one that appears in the list above, Cisco Unity will use the 20 ms default packet size.

Step 12 Click **OK**.

Step 13 Double-click the **G729 Packetization** value.

Step 14 In the Edit DWORD Value dialog box, Under Base, click **Decimal**.

Step 15 In the Value Data field, enter one of the following values that you want to use for packet size (in ms) for the G.729 codec:

- 10
- 20
- 30
- 40
- 50
- 60

Caution If you enter a setting other than one that appears in the list above, Cisco Unity will use the 20 ms default packet size.

Step 16 Click **OK**.

Step 17 Close the Registry Editor window.

Step 18 Restart the Cisco Unity server for these changes to take effect.

Step 19 If Cisco Unity is configured for failover, repeat this procedure on the secondary server.

When a non-20 ms packet size is enabled and depending on the situation, Cisco Unity will use the following packet sizes:

- When the initial SDP offer does not contain a ptime attribute, Cisco Unity will use the enabled packet size.
- When the initial SDP offer contains a ptime attribute, Cisco Unity will use the requested packet size.
- When Cisco Unity initiates the initial SDP offer, Cisco Unity will use the enabled packet size.

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.
- 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.

To Set Up Supervised Transfer on Cisco Unity

Step 1 In the Cisco Unity Administrator, go to the **Subscribers > Subscribers > Call Transfer** page.

If the name of the test subscriber is not displayed, click the **Find** icon (the magnifying glass) in the title bar, then click **Find**, and select the name of the test subscriber in the list that appears.

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

Step 2 Under Transfer Type, click **Supervise Transfer**.

Step 3 Set the Rings to Wait For field to **3**.

Step 4 Click the **Save** icon.

To Test Supervised Transfer

Step 1 From Phone 2, enter the access code necessary to get an outside line, then enter the number outside callers use to dial directly to Cisco Unity.

Step 2 On the Status Monitor, note which port handles this call.

Step 3 When you hear the opening greeting, enter the extension for Phone 1. Hearing the opening greeting means that the port is configured correctly.

Step 4 Confirm that Phone 1 rings and that you do not hear a ringback tone on Phone 2. Instead, you should hear the indication your phone system uses to mean that the call is on hold (for example, music or beeps).

Step 5 Leaving Phone 1 unanswered, confirm that the state of the port handling the call remains "Busy." This state and hearing an indication that you are on hold mean that Cisco Unity is supervising the transfer.

Step 6 Confirm that, after three rings, you hear the greeting for the test subscriber. Hearing the greeting means that Cisco Unity successfully recalled the supervised-transfer call.

Step 7 During the greeting, hang up Phone 2.

Step 8 On the Status Monitor, confirm that the state of the port handling the call changes to "Idle." This state means that the port was successfully released when the call ended.

To Delete the Test Subscriber

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**.

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.

For information on failover, refer to the applicable *Cisco Unity Failover Guide* at
http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products_feature_guides_list.html.

Requirements

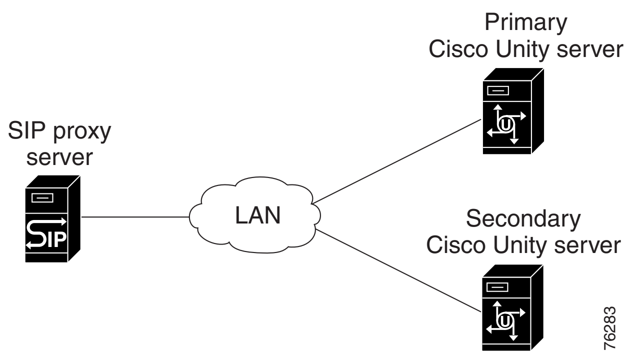
The following components are required to integrate a secondary server:

- 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 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 the SIP Proxy Server 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

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.

The Contact Line Name must be the same for both the primary and secondary servers.

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 of the UTIM window, click the **Ports** tab.

Step 17 Enter the port settings to match the port settings on the primary server.

Caution In programming the SIP proxy server, do not send calls to voice messaging ports in Cisco Unity that cannot answer calls (voice messaging ports that are not set to Answer Calls). For example, if a voice messaging port is set only to Dialout MWI, do not send calls to it.

Step 18 Click **Save**.

Step 19 Exit UTIM.

Appendix: Compatibility of Phone System Components

Testing has shown compatibility of the following phone system components with Cisco Unity in a SIP integration.

Table 9 Cisco SIP Proxy Server Compatibility with the Integration

Version	Comments
1.3	If Cisco Unity authenticates with the SIP proxy server, the authentication name entered in Cisco Unity must be the same as the contact line name in the SIP proxy server.
2.0	If Cisco Unity authenticates with the SIP proxy server, the authentication name entered in Cisco Unity must be the same as the contact line name in the SIP proxy server.

Table 10 Cisco 7960 IP Phone Compatibility with the Integration

Version	Comments
7960 P0S3-03-1-00	
7960 P0S3-03-2-00	
7960 P0S3-04-0-00	When the phone initiates a call, release transfer of the call is not available to Cisco Unity.
7960 P0S3-04-1-00	When the phone initiates a call, release transfer of the call is not available to Cisco Unity.
7960 P0S3-04-2-00	

Table 11 Pingtel xpressa Compatibility with the Integration

Version	Comments
1.2.6	To get the call forwarding to busy greeting integration feature, forwarding must be programmed on the SIP proxy server rather than configured on the Pingtel xpressa phones.
2.0.1 2.0.2	Not compatible. Silence is inserted into the audio stream every few seconds. To get the call forwarding to busy greeting integration feature, forwarding must be programmed on the SIP proxy server rather than configured on the Pingtel xpressa phones.

Table 12 Gateway IOS Compatibility with the Integration

Version	Comments
12.2(2)XB4	
12.2(2)XB6	

Other compatibility issues are:

- The Pingtel xpressa cannot connect to a backup SIP proxy server.
- To enable call forwarding when Cisco Unity is configured for failover, set the forwarding destinations in MySQL to be <contact line name>@proxy instead of <contact line name>@Unity.
- Caveat: CSCdx74350.
- Caveat: CSCdx66707.
- Caveat: CSCdx71968.

Appendix: Documentation and Technical Assistance

Conventions

The *Session Initiation Protocol (SIP) Integration Guide for Cisco Unity 5.0* uses the following conventions.

Table 13 Session Initiation Protocol (SIP) Integration Guide for Cisco Unity 5.0 Conventions

Convention	Description
boldfaced text	Boldfaced text is used for: • Key and button names. (Example: Click **OK.**) • Information that you enter. (Example: Enter **Administrator** in the User Name box.)
< > (angle brackets)	Angle brackets are used around parameters for which you supply a value. (Example: In the Command Prompt window, enter **ping <IP address>**.)
- (hyphen)	Hyphens separate keys that must be pressed simultaneously. (Example: Press **Ctrl-Alt-Delete.)**
> (right angle bracket)	A right angle bracket is used to separate selections that you make: • On menus. (Example: On the Windows Start menu, click **Settings > Control Panel > Phone and Modem Options**.) • In the navigation bar of the Cisco Unity Administrator. (Example: Go to the **System > Configuration > Settings** page.)
[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 *Session Initiation Protocol (SIP) 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 *About Cisco Unity Documentation*. 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.

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