Siemens Hicom 300 E (European) PIMG Integration Guide for Cisco Unity Connection 1.2

Revised October 12, 2007

This document provides instructions for integrating the Siemens Hicom 300 E (European) phone system with Cisco Unity Connection by using the Intel NetStructure PBX-IP Media Gateway (PIMG).

Integration Tasks

Before doing the following tasks to integrate Cisco Unity Connection with the Siemens Hicom 300 E (European) phone system by using the Intel NetStructure PBX-IP Media Gateway (PIMG), confirm that the Cisco Unity Connection server is ready for the integration by completing the applicable tasks in the *Cisco Unity Connection Installation Guide*.

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

Task List to Create the Integration

Use the following task list to set up a new integration with the Siemens Hicom 300 E (European) phone system. If you are installing a new Cisco Unity Connection server by using the *Cisco Unity Connection 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 Connection 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 Connection. See the “Planning How the Voice Messaging Ports Will Be Used by Cisco Unity Connection” section on page 4.
3. Program the Siemens Hicom 300 E (European) phone system and extensions. See the “Programming the Siemens Hicom 300 E (European) Phone System” section on page 6.
4. Set up the PIMG units. See the “Setting Up the PIMG Units” section on page 8.
5. Create the integration. See the “Creating a New Integration with the Siemens Hicom 300 E (European) Phone System” section on page 16.
6. Test the integration. See the “Testing the Integration” section on page 19.

Americas Headquarters:
Cisco Systems, Inc., 170 West Tasman Drive, San Jose, CA 95134-1706 USA

7. If this integration is a second or subsequent integration, add the applicable new user templates for the new phone system. See the (Multiple Integrations Only) Adding New User Templates, page 22.

Requirements

The Siemens Hicom 300 E (European) integration supports configurations of the following components:

Phone System

- Siemens Hicom 300 E (European) phone system.
- Software version 3.0, release 0.17 or later.
- One or more of the applicable PIMG units. For details, refer to the "Supported Circuit-Switched Phone System Integrations" section of *Cisco Unity Connection System Requirements, and Supported Hardware and Software* at http://www.cisco.com/en/US/products/ps6509/prod_installation_guides_list.html.
- The voice messaging ports in the phone system connected by digital lines to the ports on the PIMG units.

  We recommend that you connect the voice messaging ports on the phone system to the ports on the PIMG units in a planned manner to simplify troubleshooting. For example, the first phone system voice messaging port connects to the first port on the first PIMG unit, the second phone system voice messaging port connects to the second port on the first PIMG unit, and so on.
- The PIMG units connected to the same LAN or WAN that Cisco Unity Connection is connected to.
- If the PIMG units connect to a WAN, the requirements for the WAN network connections are:
  - For G.729a codec formatting, a minimum of 32.76 Kbps guaranteed bandwidth for each voice messaging port.
  - For G.711 codec formatting, a minimum of 91.56 Kbps guaranteed bandwidth for each voice messaging port.
  - No network devices that implement network address translation (NAT).
  - A maximum 200 ms network latency.
- The phone system ready for the integration, as described in the documentation for the phone system.

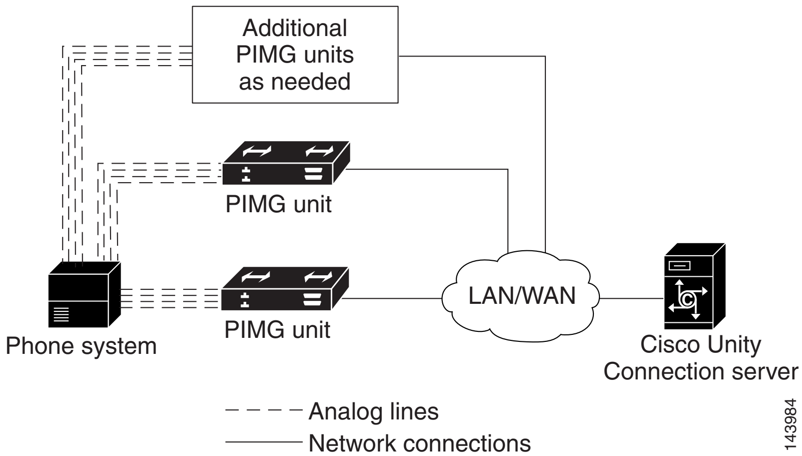
Cisco Unity Connection Server

- Cisco Unity Connection installed and ready for the integration, as described in the *Cisco Unity Connection Installation Guide* at http://www.cisco.com/en/US/products/ps6509/prod_installation_guides_list.html.
- A license that enables the applicable number of voice messaging ports.

Integration Description

The Siemens Hicom 300 E (European) integration sends call information and voice connections through the analog lines, which connect the phone system to the PIMG units. The PIMG units communicate with the Cisco Unity Connection server through the LAN or WAN by using Session Initiation Protocol (SIP). Figure 1 shows the required connections.

Figure 1 Connections Between the Phone System and Cisco Unity Connection



Call Information

The phone system sends the following information with forwarded calls:

- The extension of the called party
- The extension of the calling party (for internal calls) or the phone number of the calling party (if it is an external call and the system uses caller ID)
- The reason for the forward (the extension is busy, does not answer, or is set to forward all calls)

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

Integration Functionality

The Siemens Hicom 300 E (European) integration with Cisco Unity Connection provides the following features:

- Call forward to personal greeting
- Call forward to busy greeting
- Caller ID

- Easy message access (a subscriber can retrieve messages without entering an ID because Cisco Unity Connection identifies the subscriber based on the extension from which the call originated; a password may be required)
- Identified subscriber messaging (Cisco Unity Connection 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 Connection can be integrated with multiple phone systems at one time. For information on and instructions for integrating Cisco Unity Connection with multiple phone systems, refer to the *Multiple Phone System Integration Guide* at http://www.cisco.com/en/US/products/ps6509/products_installation_and_configuration_guides_list.html.

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

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

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

The following table describes the voice messaging port settings in Cisco Unity Connection that can be set on Telephony Integrations > Port of Cisco Unity Connection Administration.

Table 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.
Extension	Enter the extension for the port as assigned on the phone system.
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.

Table 1 Settings for the Voice Messaging Ports (continued)

Field	Considerations
Allow TRAP Connections	Check this check box so that users can use the port for recording and playback through the phone in Cisco Unity Connection web applications. Assign Allow TRAP Connections to the least busy ports.
Outgoing Hunt Order	Enter the priority order in which Cisco Unity Connection will use the ports when dialing out (for example, if the Perform Message Notification, Send MWI Requests, or Allow TRAP Connections check box is checked). The highest numbers are used first. However, when multiple ports have the same Outgoing Hunt Order number, Cisco Unity Connection will use the port that has been idle the longest.

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

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 users. Typically, the voice messaging ports that answer calls are the busiest.

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

The Number of Voice Messaging Ports That Will Only Dial Out, and Not Answer Calls

Ports that will only dial out and will not answer calls can do one or more of the following:

- Notify users by phone, pager, or e-mail of messages that have arrived.
- Turn MWIs on and off for user extensions.
- Make a TRAP connection so that users can use the phone as a recording and playback device in Cisco Unity Connection web applications.

Typically, these voice messaging ports are the least busy ports.

Caution In programming the phone system, do not send calls to voice messaging ports in Cisco Unity Connection that cannot answer calls (voice messaging ports that are not set to Answer Calls). For example, if a voice messaging port is set only to Send MWI Requests, do not send calls to it.

Preparing for Programming the Phone System

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

Programming the Siemens Hicom 300 E (European) Phone System

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

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

Do the following procedure.

To Program the Siemens Hicom 300 E (European) Phone System by Using the Command Line Interface

Step 1 Use the CHA-TAPRO command to create a button table to use on the voice messaging port extensions. The button table must have the following parameters while the remaining parameters keep their default values.

Table 2 Button Table Parameter Settings

English		German	
Parameter	**Setting**	**Parameter**	**Setting**
STNO	Leave this field blank.	STNO	Leave this field blank.
STD	**4**	SNU	**4**
DIGTYP	**OPTIT12**	DIGTYP	**OPTIT12**
KY03	**MB**	KY03	**BK**
KY04	**RLS**	KY04	**TR**
KY07	**NAME**	KY07	**NA**
KY10	**CONF**	KY10	**KF**
KY11	**CONS**	KY11	**RF**
KY12	**TRANSFER**	KY12	**UEG**

Step 2 If the Call Route mode on the PIMG unit will not be set to Pooled, continue to Step 3. Otherwise, use the ADD-SA command to create a hunt group for the voice messaging ports that are connected to the PIMG units. The hunt group must have the following parameters set while the remaining parameters keep their default values.

Table 3 ADD-SA Parameter Settings for Voice Messaging Ports on the Analog PIMG Units

English		German	
Parameter	**Setting**	**Parameter**	**Setting**
SERVICE	**VCE**	DIENST	**SPR**
CD	Enter the pilot number for the hunt group.	RNR	Enter the pilot number for the hunt group.
DPLN	**0**	WABE	**0**
NAME	Enter name for the hunt group. For example, enter "UNITY TELEPHONSVA."	NAME	Enter name for the hunt group. For example, enter "UNITY TELEPHONSVA."
CALL PROG. STATE	Leave this field blank.	VERKEHRSSITUATION	Leave this field blank.
STNO	Enter the extensions for the ports that will answer calls. Do not include the MWI-only ports.	TLNNU	Enter the extensions for the ports that will answer calls. Do not include the MWI-only ports.
TYP	**LIN**	ART	**LIN**
CQMAX	**6**	ANOKAP	**6**
OVERFLOW	**-**	UEBERLAUF	**-**

Step 3 Use the CHA-COSSU command to enable the mailbox feature in the class of service (COS) for the voice messaging ports. The mailbox feature must have the following parameters while the remaining parameters keep their default values.

Table 4 HA-COSSU Parameter Settings for the Mailbox Feature

English		German	
Parameter	**Setting**	**Parameter**	**Setting**
TYPE	**COS**	ART	**COS**
COS	Enter the number of the class of service for which the mailbox feature will be enabled. For example, enter "47".	COS	Enter the number of the class of service for which the mailbox feature will be enabled. For example, enter "47".
VOICE	**TA TNOTCR MB COSXCD VCE NOANSA FWDNWK ANSYN FWDECA**	SPRACHE	**FBKW QVKW BRK BUC SSM RWS AULEXT ANSYN AULERU**

Table 4 HA-COSSU Parameter Settings for the Mailbox Feature (continued)

English		German	
Parameter	**Setting**	**Parameter**	**Setting**
FAX	**NOCO NOTIE**		**NIB NQVB**
TTX	**NOCO NOTIE**		**NIB NQVB**
VTX	**NOCO NOTIE**	BTX	**NIB NQVB**
DTE	**TA TNOTCR BASIC**	DEE	**FBKW QVKW GRUBE**

Step 4 Program each phone to forward calls to the pilot number assigned to the voice messaging ports, based on one of the Cisco Unity Connection call transfer types shown in Table 5.

Table 5 Call Transfer Types

Transfer Type	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 user station 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.

Setting Up the PIMG Units

Do the following procedures to set up the PIMG units that are connected to the Siemens Hicom 300 E (European) phone system.

These procedures require that the following tasks have already been completed:

- The phone system is connected to the PIMG units by using analog lines.
- The PIMG units are ready to be connected to the LAN or WAN.
- The PIMG units are connected to a power source.

Fields that are not mentioned in the following procedures must keep their default values. For the default values of all fields, see the documentation for the PIMG units.

To Download the PIMG Firmware Update Files for Analog PIMG Units

Step 1 On a Windows workstation that will have access to the PIMG units, open a web browser and go to the **Cisco Unity PIMG Software Download** page at http://www.cisco.com/cgi-bin/tablebuild.pl/unity-PIMG.

Note To access the software download page, you must be logged on to Cisco.com as a registered user.

Step 2 On the Cisco Unity PIMG Software Download page, click the most recent version of the firmware for analog PIMG units.

Step 3 On the Details page, click **Next**.

Step 4 On the Document page, click **Accept**.

Step 5 In the Enter Network Password dialog box, enter your user name and password, then click **OK**.

Step 6 In the File Download dialog box, click **Save**.

Step 7 In the Save As dialog box, browse to the Windows workstation that will have access the PIMG units, browse to a directory where you want to save the file, and click **Save**.

Step 8 In the Download Complete dialog box, click **Open**. The window for extracting the PIMG firmware update files appears.

Step 9 Click **Extract**.

Step 10 In the Extract dialog box, browse to the directory where you want the extracted files, and click **Extract**.

Step 11 Close the window for the extracting application.

To Set Up the Analog PIMG Units

Step 1 On the Windows workstation, add a temporary route to enable access to the PIMG units.

a. On the Windows Start menu, click **Run**.

b. Enter **cmd**, and press **Enter**. The Command Prompt window appears.

c. At the command prompt, enter **route add 10.12.13.74 <IP Address of Workstation>**, and press **Enter**.

   For example, if the IP address of the workstation is 198.1.3.25, enter “route add 10.12.13.74<space>198.1.3.25” in the Command Prompt window.

d. Close the Command Prompt window.

Step 2 Connect a PIMG unit to the network.

Step 3 In the web browser, go to **http://10.12.13.74**.

Step 4 On the System Login page, enter the following case-sensitive settings.

Table 6 System Login Page Settings

Field	Setting
Username	**admin**
Password	**IpodAdmin**

Step 5 Click **Log On**.

Step 6 On the Configure menu, click **Upgrade**.

Step 7 On the Upgrade page, click **Browse**.

Step 8 In the Choose File dialog box, browse to the directory on the Windows workstation that has the extracted PIMG firmware update files.

Step 9 Click **Ls_<xx>.app** (where <xx> is multiple digits), and click **Open**.

Step 10 On the Upgrade page, click **Install**.

Step 11 After the file is installed, a message prompting you to restart the PIMG unit appears. Click **Cancel**.

> **Caution** Do not restart the PIMG unit until you are instructed to do so later in this procedure, even if the file installation fails. Restarting the PIMG unit at this step may prevent the PIMG unit from functioning correctly.

Step 12 Repeat Step 6 through Step 11 for each of the following files:

- Run_<xx>FskEcho.dsp
- Ls_<xx>.fsh

Step 13 On the Configure menu, click **Upgrade**.

Step 14 On the Upgrade page, under Import, click **Browse**.

Step 15 In the Choose File dialog box, browse to the file Ls_Cfg_Hicom300ECS.ini.

Step 16 Click **Ls_Cfg_Hicom300ECS.ini**, and click **Open**.

Step 17 On the Upgrade page, click **Install**.

Step 18 After the file is installed, a message prompting you to restart the PIMG unit appears. Click **OK**.

Step 19 In the web browser, go to **http://10.12.13.74**.

Step 20 On the System Login page, enter the following case-sensitive settings.

Table 7 System Login Page Settings

Field	Setting
Username	**admin**
Password	**IpodAdmin**

Step 21 Click **Log On**.

Step 22 On the Configure menu, click **Import/Export**.

Step 23 On the Import/Export page, under Export Settings, click **Export Settings**.

Step 24 In the File Download dialog box, click **Save**.

Step 25 In the Save As dialog box, browse to the Windows workstation that will have access to the PIMG units, browse to a directory where you want to save the file, and click **Save**.

Step 26 In the Download Complete dialog box, click **Open**. Notepad opens the file Config.ini that you saved.

Step 27 Locate the line with the following parameter:

```
gwInformUpdatedCallingNumber = no
```

Step 28 Change the value of the parameter to “yes” so that the line reads as follows:

```
gwInformUpdatedCallingNumber = yes
```

Step 29 Save the file, and exit Notepad.

Step 30 On the Configure menu of the PIMG unit, click **Import/Export**.

Step 31 On the Import/Export page, under Import Settings, click **Browse**.

Step 32 In the Choose File dialog box, browse to the file Config.ini that you changed in Step 28.

Step 33 Click **Config.ini**, and click **Open**.

Step 34 On the Import/Export page, click **Import Settings**.

Step 35 When prompted to restart the PIMG unit, click **OK**.

Step 36 On the Configure menu, click **Password**.

Step 37 On the Password page, enter the following settings.

Table 8 Password Page Settings

Field	Setting
Old Password	**IpodAdmin** (This setting is case sensitive.)
New Password	<your new password> (This setting is case sensitive.)
Confirm Password	<your new password> (This setting is case sensitive.)

Step 38 Click **Change**.

Step 39 On the Configure menu, click **System**.

Step 40 On the System page, enter the following settings.

Table 9 System Page Settings

Field	Setting
Operating Mode	**SIP**
Telephony Switch Type	**None**
PCM Coding	**uLaw**

Step 41 Click **Apply Changes**.

Step 42 On the Configure menu, click **Gateway**.

Step 43 On the Gateway page, click the **Gateway Routing** tab.

Step 44 On the Gateway Routing tab, enter the following settings.

Table 10 Gateway Routing Tab Settings

Field	Setting
Fault Tolerance Enabled	**No**
Load Balancing Enabled	**No**

Table 10 Gateway Routing Tab Settings (continued)

Field	Setting
VoIP Endpoint ID: 1	<the IP address of the Cisco Unity Connection server>
VoIP Endpoint ID: 2	<blank>

Step 45 Click **Apply Changes**.

Step 46 Click the **Gateway Advanced** tab.

Step 47 On the Gateway Advanced tab, enter the following settings.

Table 11 Gateway Advanced Tab Settings

Field	Setting
Call Connect Mode	**OnAnswer**
Destination for Unroutable IP Calls	<blank>
Destination for Unroutable PBX Calls	<blank>
Monitor Call Connections	**No**
Maximum Call Party Delay (msecs)	**2000**
Dial Digit on Time (msecs)	**100**
Dial Inter-Digit Time (msecs)	**100**
Dial Pause Time (msecs)	**2000**
Turn MWI On FAC	<the code that turns MWIs on>
Turn MWI Off FAC	<the code that turns MWIs off>
Outbound Call Connect Timeout (msecs)	**10000**
Wait for Ringback/Connect on Blind Transfer	**Yes**
Hunt Group Extension	<the pilot number of Cisco Unity Connection voice messaging ports>
Audio Compression	Click the preferred codec for audio compression: • **G.711 Only** • **G.729 A Preferred**
RTP Digit Relay Mode	**RFC2833**
Signaling Digit Relay Mode	**Off**
Voice Activity Detection	**Off**

Table 11 Gateway Advanced Tab Settings (continued)

Field	Setting
Frame Size	Click the applicable setting: • G.711—**20** • G.729a—**10** **Caution** Failure to use the correct setting will result in recorded messages containing nothing but silence.
Frames Per Packet	Click the applicable setting: • G.711—**1** • G.729a—**2** **Caution** Failure to use the correct setting will result in recorded messages containing nothing but silence.
Call Control QOS Byte	*(PIMG units connect only to a LAN)* **0** *(PIMG units connect to a WAN)* **104** **Note** For details on the setting for a LAN, see the caveat CSCsb96387.
RTP QOS Byte	*(PIMG units connect only to a LAN)* **0** *(PIMG units connect to a WAN)* **184** **Note** For details on the setting for a LAN, see the caveat CSCsb96387.
SNMP Traps Enabled	**No**
E-mail Alarms Enabled	**No**

Step 48 Click **Apply Changes**.

Step 49 Click the **Gateway Capabilities** tab.

Step 50 Depending on how you have planned to use the voice messaging ports, click the applicable setting for each port in the Telephony Port Capability column.

Table 12 Gateway Capabilities Tab Settings

Telephony Port Capability Settings	Voice Messaging Port Usage
Calls-Only	The port will answer incoming calls only and will not dial out (for example, to set MWIs or send message notifications).
MWIs-Only	The port will dial out only (for example, to set MWIs or send message notifications) and will not answer incoming calls.
Both	The port will answer incoming calls and will also dial out (for example, to set MWIs or send message notifications).

Caution In setting up the PIMG unit, do not send calls to 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. Otherwise the integration will not function correctly.

If a port in Cisco Unity Connection is disabled, click **No** in the Telephony Port Enabled column for the corresponding port on this tab. Note that changing a setting in the Telephony Port Enabled column requires restarting the PIMG unit.

Step 51 Click **Apply Changes**.

Step 52 On the Configure menu, click **SIP**.

Step 53 On the SIP page, enter the following settings.

Table 13 SIP Page Settings

Field	Setting
Host and Domain Name	<the domain name of the PIMG unit>
Server Port	**5060**
Transport Type	**UDP**
Call as Domain Name	**No**
Registration Server Address	<blank>
Registration Server Port	**5060**
Registration Expiration (sec)	**3600**
Primary Proxy Server Address	<the IP address of the Cisco Unity Connection server>
Primary Proxy Server Port	**5060** (When you configure more than one PIMG unit, increase this setting by 1 for each successive unit. For example, unit 2 will be 5061, unit 3 will be 5062, and so on. For failover, this setting must match the setting for the Backup Proxy Server Port field.)
Proxy Query Interval	**10**
T1 Time (msecs)	**400**

Table 13 SIP Page Settings (continued)

Field	Setting
T2 Time (msecs)	**3000**
Invite Expiration (sec)	**120**

Step 54 Click **Apply Changes**.

Step 55 On the Configure menu, click **IP**.

Step 56 On the IP page, enter the following settings.

Table 14 IP Page Settings

Field	Setting
Client IP Address	<the new IP address you want to use for the PIMG unit> (This is the IP address that you enter in Cisco Unity Connection Administration when you create the integration.)
Client Subnet Mask	<the new subnet mask, if the subnet mask is different from the default IP address>
Default Network Gateway Address	<the IP address of the default network gateway router that the PIMG units will use>
BOOTP Enabled	**No**

Step 57 Click **Apply Changes**.

Step 58 On the Configure menu, click **Tones**.

Step 59 On the Tones page, click the **Learn** tab.

Caution Destination addresses cannot be duplicated in the same session. Otherwise, the process for learning tones will not succeed. If you do not have enough available phones to learn all the tones at one time, you can run multiple sessions to learn tones individually by checking or unchecking the applicable Acquire Tone check boxes.

Step 60 On the Tones page, for the Dialtone event, confirm that the Acquire Tone check box is checked and leave the Destination Address field blank.

Step 61 On the Tones page, for the Busy Tone event, confirm that the Acquire Tone check box is checked and do the following substeps to verify that the tone is correct.

a. From a available phone, call a second phone.

b. Answer the second phone when it rings, and leave both handsets off so that both phones are busy.

c. From a third phone, dial one of the busy phones.

d. Confirm that you hear a busy tone.

e. Hang up the third phone but leave the handsets for the other two phones off.

Step 62 On the Tones page, in the Destination Address field for Busy Tone, enter the extension that you dialed in Step 61c. from the third phone.

Step 63 On the Tones page, for the Error/Reorder Tone event, confirm that the Acquire Tone check box is checked and do the following substeps to verify that the tone is correct.

a. From an available phone, dial an extension that does not exist.

b. Confirm that you hear the reorder or error tone.

c. Hang up the phone.

Step 64 On the Tones page, in the Destination Address field for Error/Reorder Tone, enter the extension that you dialed in Step 63a.

Step 65 On the Tones page, for the Ringback Tone event, confirm that the Acquire Tone check box is checked and do the following substeps to verify that the tone is correct.

a. From an available phone, dial an extension that does exist

b. Confirm that you hear the ringback tone.

c. Hang up the phone.

Step 66 On the Tones page, in the Destination Address field for Ringback Tone, enter the extension that you dialed in Step 65a.

Step 67 Click **Learn**.

Note When running learn tones, the PIMG unit will restart after learning the first tone. For details, see the caveat CSCsh53791.

Step 68 When the process is complete, check the check box for each newly learned tone and click **Apply**.

Step 69 Hang up the phones that you used in Step 61.

Step 70 On the Configure menu, click **Restart**.

Step 71 On the Restart page, click **Restart Unit Now**.

Step 72 When the PIMG unit has restarted, in the View menu, click **Refresh**.

Step 73 Repeat Step 37 through Step 72 on all remaining PIMG units.

Creating a New Integration with the Siemens Hicom 300 E (European) Phone System

After ensuring that the Siemens Hicom 300 E (European) phone system and Cisco Unity Connection 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 Log on to Cisco Unity Connection Administration.

Step 2 In Cisco Unity Connection Administration, expand **Telephony Integrations**, then click **Phone System**.

Step 3 On the Search Phone Systems page, on the Phone System menu, click **New Phone System**. The Phone System Integration Wizard appears.

Step 4 On the Select Phone System Manufacturer page, in the Manufacturer field, click **Siemens** and click **Next**.

Step 5 On the Select Phone System Model page, in the Model field, click **Siemens Hicom 300 E (European)** and click **Next**.

Step 6 On the Set Up Phone System page, in the Phone System Name field, accept the default name or enter the descriptive name that you want, and click **Next**.

Step 7 On the Select Port Group Template page, in the Port Group Template field, click **Analog PIMG to Siemens Hicom 300 E (European)** and click **Next**.

Step 8 On the Set Up Port Group page, enter the following settings and click **Next**.

Table 15 Settings for the Set Up Port Group Page

Field	Setting
Port Group Name	<a descriptive name for the port group; accept the default name or enter the name that you want>
Number of Ports	**8** (If you want to use fewer than eight voice messaging ports, enter the number of voice messaging ports that you want to use on this PIMG unit.)
IP Address or Host Name	<the IP address of the PIMG unit that you are integrating with Cisco Unity Connection>
Test Address	Click this button to test the IP address that you entered. The results of the test appear in the field to the right of the button.
Port	<the SIP port of the PIMG unit that Cisco Unity Connection will connect to; we recommend that you use the default setting> **Caution** This setting must match the setting for the Server Port field of the PIMG unit. Otherwise the integration will not function correctly.

Step 9 On the Confirm Phone System Settings page, confirm the settings that you have entered and click **Finish**.

Step 10 On the Phone System Creation Summary page, click **Close**.

Step 11 On the Search Phone Systems page, click the display name of the phone system that you created for this phone system integration.

Step 12 On the Phone System Basics page, under Message Waiting Indicator Settings, click **Use Port Memory** and click **Save**.

Step 13 In Cisco Unity Connection Administration, expand **Telephony Integrations**, then click **Port**.

Step 14 On the Search Ports page, click the display name of the first voice messaging port that you created for this phone system integration.

> **Note** By default, the display names for the voice messaging ports are composed of the port group display name followed by incrementing numbers.

Step 15 On the Port Basics page, set the voice messaging port settings as applicable. The fields in the following table are the ones that you can change.

Table 16 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.
Extension	Enter the extension for the port as assigned on the phone system.
Answer Calls	Check this check box to designate the port for answering calls. These calls can be incoming calls from unidentified callers or from users.
Perform Message Notification	Check this check box to designate the port for notifying users of messages. Assign Perform Message Notification to the least busy ports.
Send MWI Requests	Check this check box to designate the port for turning MWIs on and off. Assign Send MWI Requests to the least busy ports.
Allow TRAP Connections	Check this check box so that users can use the port for recording and playback through the phone in Cisco Unity Connection web applications. Assign Allow TRAP Connections to the least busy ports.
Outgoing Hunt Order	Enter the priority order in which Cisco Unity Connection will use the ports when dialing out (for example, if the Perform Message Notification, Send MWI Requests, or Allow TRAP Connections check box is checked). The highest numbers are used first. However, when multiple ports have the same Outgoing Hunt Order number, Cisco Unity Connection will use the port that has been idle the longest.

Step 16 Click **Save**.

Step 17 Click **Next**.

Step 18 Repeat Step 15 through Step 17 for all remaining voice messaging ports for the phone system.

Step 19 If another phone system integration exists, in Cisco Unity Connection Administration, expand **Telephony Integrations**, then click **Trunk**. Otherwise, skip to Step 23.

Step 20 On the Search Phone System Trunks page, on the Phone System Trunk menu, click **New Phone System Trunk**.

Step 21 On the New Phone System Trunk page, enter the following settings for the phone system trunk and click **Save**.

Table 17 Settings for the Phone System Trunk

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

Step 22 Repeat Step 20 and Step 21 for all remaining phone system trunks that you want to create.

Step 23 If prompted to restart Cisco Unity Connection, in the Windows task bar, right-click the **Cisco Unity Connection** icon and click **Restart > Voice Processing Server Role**.

Step 24 When prompted to confirm stopping the Voice Processing server role, click **Yes**.

Step 25 In Cisco Unity Connection Administration, in the Related Links drop-down list, click **Check Telephony Configuration** and click **Go** to confirm the phone system integration settings.

If the test is not successful, the Task Execution Results displays one or more messages with troubleshooting steps. After correcting the problems, test the connection again.

Step 26 In the Task Execution Results window, click **Close**.

Step 27 Log off Cisco Unity Connection Administration.

Testing the Integration

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

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

- The installation guide for the phone system.
- The setup information earlier in this guide.

To Set Up the Test Configuration

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

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

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

Step 3 To create a test user for testing, in Cisco Unity Connection Administration, expand **Users**, then click **Users**.

Step 4 On the Search Users page, on the User menu, click **New User**.

Step 5 On the New User page, enter the following settings.

Table 18 Settings for the New User Page

Field	Setting
User Type	**User with Voice Mailbox**
Based on Template	<the applicable user template>
Alias	**testuser**
First Name	**Test**
Last Name	**User**
Display Name	**Test User**
Extension	<the extension of Phone 1>

Step 6 Click **Save**.

Step 7 On the Edit User Basics page, in the Voice Name field, record a voice name for the test user.

Step 8 In the Phone System field, confirm that the phone system selected is the phone system that Phone 1 is connected to.

Step 9 Uncheck the **Set for Self-enrollment at Next Login** check box.

Step 10 Click **Save**.

Step 11 On the Edit menu, click **Message Waiting Indicators**.

Step 12 On the Message Waiting Indicators page, click the message waiting indicator. If no message waiting indication is in the table, click **Add New**.

Step 13 On the Edit Message Waiting Indicator page, enter the following settings.

Table 19 Settings for the Edit MWI Page

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

Step 14 Click **Save**.

Step 15 On the Edit menu, click **Transfer Options**.

Step 16 On the Transfer Options page, click the active option.

Step 17 On the Edit Transfer Option page, under Transfer Action, click the **Extension** option and enter the extension of Phone 1.

Step 18 In the Transfer Type field, click **Release to Switch**.

Step 19 Click **Save**.

Step 20 Minimize the Cisco Unity Connection Administration window.

Do not close the Cisco Unity Connection Administration window because you will use it again in a later procedure.

Step 21 On the Cisco Unity Connection desktop, double-click the **Tools Depot** icon.

Step 22 In the left pane of the Tools Depot window, expand **Switch Integration Tools**, then double-click **Port Status Monitor**. The Port Status Monitor window appears.

Step 23 On the Ports menu, click **Start All**, and arrange the port monitors so that you can notice which port will handle the calls that you will make.

To Test an External Call with Release Transfer

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

Step 2 In the Port 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 Connection correctly released the call and transferred it to Phone 1.

Step 5 Leaving Phone 1 unanswered, confirm that the state of the port handling the call changes to "Idle." This state means that release transfer is successful.

Step 6 Confirm that, after the number of rings that the phone system is set to wait, the call is forwarded to Cisco Unity Connection and that you hear the greeting for the test user. Hearing the greeting means that the phone system forwarded the unanswered call and the call-forward information to Cisco Unity Connection, which correctly interpreted the information.

Step 7 On the Port Status 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 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 Connection are successfully integrated for turning on MWIs.

To Test Listening to Messages

Step 1 From Phone 1, enter the internal pilot number for Cisco Unity Connection.

Step 2 When asked for your password, enter the password for the test user. Hearing the request for your password means that the phone system sent the necessary call information to Cisco Unity Connection, which correctly interpreted the information.

Step 3 Confirm that you hear the recorded voice name for the test user (if you did not record a voice name for the test user, you will hear the extension number for Phone 1). Hearing the voice name means that Cisco Unity Connection correctly identified the user by the extension.

Step 4 Listen to the message.

Step 5 After listening to the message, delete the message.

Step 6 Confirm that the MWI on Phone 1 is deactivated. The deactivated MWI means that the phone system and Cisco Unity Connection are successfully integrated for turning off MWIs.

Step 7 Hang up Phone 1.

Step 8 On the Port 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 Connection

Step 1 In Cisco Unity Connection Administration, on the Edit Transfer Option page for the test user, in the Transfer Type field, click **Supervise Transfer**.

Step 2 In the Rings to Wait For field, enter **3**.

Step 3 Click **Save**.

Step 4 Minimize the Cisco Unity Connection Administration window.

Do not close the Cisco Unity Connection Administration window because you will use it again in a later procedure.

To Test Supervised Transfer

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

Step 2 On the Port 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).

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

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

Step 7 During the greeting, hang up Phone 2.

Step 8 On the Port 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 9 Exit the Port Status Monitor.

To Delete the Test User

Step 1 In Cisco Unity Connection Administration, expand **Users**, then click **Users**.

Step 2 On the Search Users page, check the check box to the left of the test user.

Step 3 Click **Delete Selected**.

(Multiple Integrations Only) Adding New User Templates

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 will be assigned to this phone system by default.

However, for each additional phone system integration that you create, you must add the applicable new user templates that will assign users to the new phone system. You must add the new templates before you add new users who will be assigned to the new phone system.

For details on adding new user templates, refer to the “Adding, Changing, or Deleting an Account Template” chapter in the *Cisco Unity Connection User Moves, Adds, and Changes Guide* at http://www.cisco.com/en/US/products/ps6509/prod_maintenance_guides_list.html.

For details on selecting a user template when adding a new user, refer to the applicable chapter for adding user accounts in the *Cisco Unity Connection User Moves, Adds, and Changes Guide* at http://www.cisco.com/en/US/products/ps6509/prod_maintenance_guides_list.html.

Appendix: Documentation and Technical Assistance

Conventions

The *Siemens Hicom 300 E (European) PIMG Integration Guide for Cisco Unity Connection 1.2* uses the following conventions.

Table 20 Siemens Hicom 300 E (European) PIMG Integration Guide for Cisco Unity Connection 1.2 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 **Programs > Cisco Unified Serviceability > Real-Time Monitoring Tool**.) • In the navigation bar of Cisco Unity Connection Administration. (Example: In Cisco Unity Connection Administration, expand **System Settings > Advanced**.)
[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 *Siemens Hicom 300 E (European) PIMG Integration Guide for Cisco Unity Connection 1.2* 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 Connection documentation on Cisco.com, see the *Cisco Unity Connection Documentation Guide*. The document is shipped with Cisco Unity Connection and is available at http://www.cisco.com/en/US/products/ps6509/products_documentation_roadmaps_list.html.

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