



Siemens Hicom 300 E CS PIMG Integration Guide for Cisco Unity Connection 1.1

Revised October 12, 2007

This document provides instructions for integrating the Siemens Hicom 300-series E CS 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 phone system, 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 the integration.

Task List to Create the Integration

Use the following task list to set up a new integration with the Siemens Hicom 300 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. If you have a legacy voice messaging system, clear the MWIs for users who are migrating to Cisco Unity Connection. See the [“Clearing MWIs from a Legacy Voice Messaging System” section on page 6](#).
4. Program the Siemens Hicom 300 phone system and extensions. See the [“Programming the Siemens Hicom 300 Phone System” section on page 6](#).
5. Set up the PIMG units. See the [“Setting Up the PIMG Units” section on page 9](#).



6. Create the integration. See the “[Creating a New Integration with the Siemens Hicom 300 Phone System](#)” section on page 17.
7. Test the integration. See the “[Testing the Integration](#)” section on page 19.
8. 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 23.

Requirements

The Siemens Hicom 300 integration supports configurations of the following components:

Phone System

- Siemens Hicom 300-series E CS phone system.
- Software version 9006.4 or later.
- SLMO line cards installed for connecting to the PIMG units.
- 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

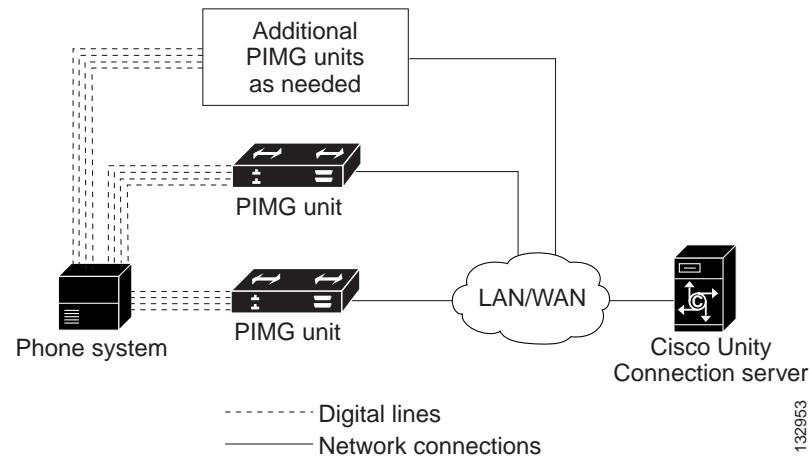
- Cisco Unity Connection installed and ready for the integration, as described in the *Cisco Unity Connection Installation Guide* at http://www.cisco.com/en/US/products/ps6509/prod_installation_guides_list.html.
- A license that enables the applicable number of voice messaging ports.

Integration Description

The Siemens Hicom 300 integration sends call information and voice connections through the digital 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 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.

Clearing MWIs from a Legacy Voice Messaging System

If users do not have existing voice mail on a legacy voice messaging system, skip this section and continue with the next section.

If users have existing voice mail on a legacy voice messaging system, do the following procedure.

To Clear MWIs from a Legacy Voice Messaging System

-
- Step 1** Have all users who are migrating to Cisco Unity Connection delete or listen to their voice messages on the legacy voice messaging system.
- Step 2** Confirm that all MWIs for the users migrating to Cisco Unity Connection are turned off.
-

Programming the Siemens Hicom 300 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 Phone System by Using the Command Line Interface

-
- Step 1** Use the ADD-COS command to add INTDTMF to the class of service that you use. The class of service must have the following parameters while the remaining parameters keep their default values.

Table 2 *ADD-COS Parameter Settings*

Parameter	Required Setting
NEWCOS	<the class of service that you are using>
AVCE	indtmf;

- Step 2** Use the ADD-DPLN command to create an extension for each voice messaging port on the PIMG units. Each extension must have the following parameters set while the remaining parameters keep their default values.

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

Parameter	Required Setting
DGTS	<first extension>&&<last extension>
DPLN	0
CPS	(leave field blank)
DAR	STN
CHECK	N

- Step 3** Use the ADD-SCSU command to create the first voice messaging port on the first PIMG unit. The voice messaging port must have the following parameters while the remaining parameters keep their default values.

Table 4 ADD-SCSU Parameter Settings for Voice Messaging Ports on the PIMG Units

Parameter	Required Setting
STNO	<the extension number>
PEN	<the port equipment number>
DPLN	0
ITR	0
COS1	<the class of service that you are using>
COS2	<the class of service that you are using>
COSX	0
PUBSCR	<the PSTN prefix plus the extension number>
NTYPE	NAT
ACTCDE	000000000000
HTLNIDX	(leave field blank)
DEVFUNC	OPTI
INS	Y
OPTITYPE	OEADPLC
CUI	Y

- Step 4** Repeat [Step 3](#) for each remaining voice messaging port on all PIMG units.
- Step 5** Use the CHA-KEPRO command to program the first voice messaging port on the first PIMG unit. The voice messaging port must have the following parameters while the remaining parameters keep their default values.

Table 5 CHA-KEPRO Parameter Settings for Voice Messaging Ports on the PIMG Units

Parameter	Required Setting
STNO	<the extension number of the voice messaging port>
STD	1
DIGTYPE	OPTIE12

- Step 6** Confirm that the key assignments for the voice messaging port match the key assignments shown in [Table 6](#).

Table 6 Required Key Assignments for Voice Messaging Ports on the PIMG Units

Key	Required Assignment
1	LINE
2	VACANT
3	VACANT
4	VACANT
5	CONF
6	XFER
7	CNCT
8	HOLDM
9	MB
10	SYSSP1
11	SYSSP2
12	VACANT



Caution The key assignments must match the key assignments shown above. Otherwise, the integration will not function correctly.

- Step 7** Repeat [Step 5](#) and [Step 6](#) for each remaining voice messaging port on all PIMG units.
- Step 8** Use the ADD-KCSU command to set the primary line options for the first voice messaging port on the first PIMG unit. The voice messaging port must have the following parameters while the remaining parameters keep their default values.

Table 7 ADD-KCSU Parameter Settings for Voice Messaging Ports on the PIMG Units

Parameter	Required Setting
STNO	<the extension number of the voice messaging port>
TYPE	KEY
PRIMKEY	1
RIOP	Y
ORLNPF	PRIM

Table 7 ADD-KCSU Parameter Settings for Voice Messaging Ports on the PIMG Units

Parameter	Required Setting
TMLNPF	RING
SGLBMOD	Y
ROLRING	STAND
APRIVAT	N
AICS	Y

Step 9 Repeat [Step 8](#) for each remaining voice messaging port on all PIMG units.

Step 10 In the station configuration section, in the CUI (Common User Interface) field, enter **Yes**.

Step 11 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 8](#).

Table 8 Call Transfer Types

Transfer Type	Usage
Release transfer (blind transfer)	Program the phone to forward calls to the pilot number when: <ul style="list-style-type: none"> 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 phone system.

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

- The phone system is connected to the PIMG units by using digital 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 Digital 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 digital (DNI) 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 Digital PIMG Units

- Step 1** On the Windows workstation, add a temporary route to enable access to the PIMG units.
- On the Windows Start menu, click **Run**.
 - Enter **cmd**, and press **Enter**. The Command Prompt window appears.
 - 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.
 - 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 9 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 **Ami<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:
- Ami_<xx>.fsh
 - Run<xx>FskEcho.dsp
 - iNim<xx>.ibt
 - iNim<xx>.ilc
 - iNim<xx>.iap
- Step 13** On the Configure menu, click **Upgrade**.
- Step 14** On the Upgrade page, click **Browse**.
- Step 15** In the Choose File dialog box, browse to the file DNI_Cfg_Generic.ini .
- Step 16** Click **DNI_Cfg_Generic.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 10 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 11 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 12 System Page Settings

Field	Setting
Operating Mode	SIP
Telephony Switch Type	Optiset_300ECS
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 13 Gateway Routing Tab Settings

Field	Setting
Fault Tolerance Enabled	No
Load Balancing Enabled	No
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 14 Gateway Advanced Tab Settings





Field	Setting
Call Connect Mode	OnAnswer
Destination for Unroutable PBX Calls	<the extension of an attendant who will receive calls to Cisco Unity Connection that are unanswered>
Turn MWI On FAC	##1
Turn MWI Off FAC	##8
Wait for Ringback/Connect on Blind Transfer	Yes
Hunt Group Extension	<the pilot number for the Cisco Unity Connection voice messaging ports>
Audio Compression	Click the preferred codec for audio compression: <ul style="list-style-type: none"> • G.711 Only • G.729 A Preferred
Signaling Digit Relay Mode	Off
Voice Activity Detection	Off
Frame Size	Click the applicable setting: <ul style="list-style-type: none"> • G.711—20 • G.729a—10 <div style="text-align: center;"></div> <p>Caution Failure to use the correct setting will result in recorded messages containing nothing but silence.</p>

Table 14 Gateway Advanced Tab Settings (continued)

Field	Setting
Frames Per Packet	Click the applicable setting: <ul style="list-style-type: none"> • 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 .


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 15 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 16 *SIP Page Settings*

Field	Setting
Host and Domain Name	<the domain name of the PIMG unit>
Server Port	5060
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.)
Proxy Query Interval	10
T1 Time	400
T2 Time	3000

- Step 54** Click **Apply Changes**.
- Step 55** On the Configure menu, click **IP**.
- Step 56** On the IP page, enter the following settings.


Table 17 *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 will 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.
- From a available phone, call a second phone.
 - Answer the second phone when it rings, and leave both handsets off so that both phones are busy.
 - From a third phone, dial one of the busy phones.
 - Confirm that you hear a busy tone.
 - 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.
- From an available phone, dial an extension that does not exist.
 - Confirm that you hear the reorder or error tone.
 - 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.
- From an available phone, dial an extension that does exist
 - Confirm that you hear the ringback tone.
 - 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 2](#) through [Step 72](#) on all remaining PIMG units.
-


Creating a New Integration with the Siemens Hicom 300 Phone System

After ensuring that the Siemens Hicom 300 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** 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 **Siemens Hicom 300 ECS via PIMG** and click **Next**.
- Step 8** On the Set Up Port Group page, enter the following settings and click **Next**.

Table 18 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 19 *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 20 **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 21 *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 22 *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 CS PIMG Integration Guide for Cisco Unity Connection 1.1* uses the following conventions.

Table 23 *Siemens Hicom 300 E CS PIMG Integration Guide for Cisco Unity Connection 1.1 Conventions*

Convention	Description
boldfaced text	Boldfaced text is used for: <ul style="list-style-type: none"> 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: <ul style="list-style-type: none"> 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.)

Table 23 *Siemens Hicom 300 E CS PIMG Integration Guide for Cisco Unity Connection 1.1 Conventions (continued)*

Convention	Description
[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 CS PIMG Integration Guide for Cisco Unity Connection 1.1* 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>

CCVP, the Cisco logo, and Welcome to the Human Network are trademarks of Cisco Systems, Inc.; Changing the Way We Work, Live, Play, and Learn is a service mark of Cisco Systems, Inc.; and Access Registrar, Aironet, Catalyst, CCDA, CCDP, CCIE, CCIP, CCNA, CCNP, CCSP, Cisco, the Cisco Certified Internetwork Expert logo, Cisco IOS, Cisco Press, Cisco Systems, Cisco Systems Capital, the Cisco Systems logo, Cisco Unity, Enterprise/Solver, EtherChannel, EtherFast, EtherSwitch, Fast Step, Follow Me Browsing, FormShare, GigaDrive, HomeLink, Internet Quotient, IOS, iPhone, IP/TV, iQ Expertise, the iQ logo, iQ Net Readiness Scorecard, iQuick Study, LightStream, Linksys, MeetingPlace, MGX, Networkers, Networking Academy, Network Registrar, PIX, ProConnect, ScriptShare, SMARTnet, StackWise, The Fastest Way to Increase Your Internet Quotient, and TransPath are registered trademarks of Cisco Systems, Inc. and/or its affiliates in the United States and certain other countries.

All other trademarks mentioned in this document or Website are the property of their respective owners. The use of the word partner does not imply a partnership relationship between Cisco and any other company. (0711R)

Any Internet Protocol (IP) addresses used in this document are not intended to be actual addresses. Any examples, command display output, and figures included in the document are shown for illustrative purposes only. Any use of actual IP addresses in illustrative content is unintentional and coincidental.

© 2007 Cisco Systems, Inc. All rights reserved.

