



## CHAPTER 13

# Setting Up a Siemens Hicom 300 E (North American) Digital PIMG Integration with Cisco Unity Connection

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For detailed instructions for setting up a Siemens Hicom 300-series E (North American) digital PIMG integration with Cisco Unity Connection, see the following sections in this chapter:

- [Task List to Create the Integration with the Siemens Hicom 300 PIMG Phone System, page 13-1](#)
- [Requirements, page 13-2](#)
- [Clearing MWIs from a Legacy Voice Messaging System, page 13-3](#)
- [Programming the Siemens Hicom 300 E \(North American\) Phone System for Integrating with Cisco Unity Connection, page 13-3](#)
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## Task List to Create the Integration with the Siemens Hicom 300 PIMG Phone System

Before doing the following tasks to integrate Cisco Unity Connection with the Siemens Hicom 300-series E (North American) phone system by using PIMG units (media gateways), confirm that the Cisco Unity Connection server is ready for the integration by completing the applicable tasks in the *Cisco Unity Connection Installation Guide*.

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 13-2](#).
2. Plan how the voice messaging ports will be used by Cisco Unity Connection. See [Chapter 2, “Planning How the Voice Messaging Ports Will Be Used by Cisco Unity Connection.”](#)
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 13-3](#).
4. Program the Siemens Hicom 300 phone system and extensions. See the [“Programming the Siemens Hicom 300 E \(North American\) Phone System for Integrating with Cisco Unity Connection” section on page 13-3](#).
5. Set up the PIMG units. See the [“Setting Up the Digital PIMG Units” section on page 13-6](#).

6. Create the integration. See the “[Creating a New Integration with the Siemens Hicom 300 Phone System](#)” section on page 13-21.
7. Test the integration. See [Chapter 14, “Testing the Integration.”](#)
8. If this integration is a second or subsequent integration, add the applicable new user templates for the new phone system. See [Chapter 15, “Adding New User Templates for Multiple Integrations.”](#)

## Requirements

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

### Phone System

- Siemens Hicom 300-series E CS (North American) 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, see [Chapter 1, “Introduction.”](#)
- The voice messaging ports in the phone system connected by digital lines to the ports on the PIMG units.

To simplify troubleshooting, we recommend that you connect the voice messaging ports on the phone system to the ports on the PIMG units in a planned manner. For example, connect the first phone system voice messaging port to the first port on the first PIMG unit, connect the second phone system voice messaging port to the second port on the first PIMG unit, and so on. Alternatively, if you have multiple PIMG units, you can reduce answer times in the event of a PIMG unit failure by connecting the phone system ports to the PIMG units in a round-robin fashion. For example, connect the first phone system voice messaging port to the first port on the first PIMG unit, connect the second phone system voice messaging port to the first port on the second 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 one-way 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](http://www.cisco.com/en/US/products/ps6509/prod_installation_guides_list.html).
- A license that enables the applicable number of voice messaging ports.

**Centralized Voice Messaging**

Cisco Unity Connection supports centralized voice messaging through the phone system, which supports various inter-phone system networking protocols including proprietary protocols such as Avaya DCS, Nortel MCDN, or Siemens CorNet, and standards-based protocols such as QSIG or DPNSS. Note that centralized voice messaging is a function of the phone system and its inter-phone system networking, not voicemail. Connection will support centralized voice messaging as long as the phone system and its inter-phone system networking are properly configured. For details, see the “Centralized Voice Messaging” section in the “Integrating Cisco Unity Connection with the Phone System” chapter of the *Cisco Unity Design Guide Release 8.x* at [http://www.cisco.com/en/US/docs/voice\\_ip\\_comm/connection/8x/design/guide/8xcucdtx.html](http://www.cisco.com/en/US/docs/voice_ip_comm/connection/8x/design/guide/8xcucdtx.html).

## Clearing MWIs from a Legacy Voice Messaging System

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

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

**To Clear MWIs from a Legacy Voice Messaging System**

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- 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 subscribers migrating to Cisco Unity Connection are turned off.
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## Programming the Siemens Hicom 300 E (North American) Phone System for Integrating with Cisco Unity Connection

The following programming instructions are provided as an example only. The specific programming for your phone system may vary depending on its configuration.

**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.

**To Program the Siemens Hicom 300 E (North American) 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 13-1** ADD-COS Parameter Settings

Parameter	Setting
NEWCOS	Enter the class of service that you are using.
AVCE	Enter <b>indtmf;</b> .

- 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 13-2** ADD-DPLN Parameter Settings for Voice Messaging Ports on the PIMG Units

Parameter	Setting
DGTS	Enter <first extension>&&<last extension>.
DPLN	Enter 0.
CPS	Leave this field blank.
DAR	Enter STN.
CHECK	Enter 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 13-3** ADD-SCSU Parameter Settings for Voice Messaging Ports on the PIMG Units

Parameter	Setting
STNO	Enter the extension number.
PEN	Enter the port equipment number.
DPLN	Enter 0.
ITR	Enter 0.
COS1	Enter the class of service that you are using.
COS2	Enter the class of service that you are using.
COSX	Enter 0.
PUBSCR	Enter the PSTN prefix plus the extension number.
NTYPE	Enter NAT.
ACTCDE	Enter 000000000000.
HTLNIDX	Leave this field blank.
DEVFUNC	Enter OPTI.
INS	Enter Y.
OPTITYPE	Enter OEADPLC.
CUI	Enter 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 13-4** CHA-KEPRO Parameter Settings for Voice Messaging Ports on the PIMG Units

Parameter	Setting
STNO	Enter the extension number of the voice messaging port.
STD	Enter <b>1</b> .
DIGTYPE	Enter <b>OPTIE12</b> .

- Step 6** Confirm that the key assignments for the voice messaging port match the key assignments shown in [Table 13-5](#).

**Table 13-5** Required Key Assignments for Voice Messaging Ports on the PIMG Units

Key	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 13-6** ADD-KCSU Parameter Settings for Voice Messaging Ports on the PIMG Units

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

**Table 13-6** ADD-KCSU Parameter Settings for Voice Messaging Ports on the PIMG Units

Parameter	Setting
TMLNPF	Enter <b>RING</b> .
SGLBMOD	Enter <b>Y</b> .
ROLRING	Enter <b>STAND</b> .
APRIVAT	Enter <b>N</b> .
AICS	Enter <b>Y</b> .

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

**Step 10** In the station configuration section, in the 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 13-7](#).

**Table 13-7** 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"> <li>The extension is busy.</li> <li>The call is not answered.</li> </ul>
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 Digital PIMG Units

Do the following procedures to set up the digital 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 manufacturer documentation for the PIMG units.

### To Download the PIMG Firmware Update Files for Digital PIMG Units

**Step 1** On a Windows workstation with a high-speed Internet connection that will have access to the PIMG units, go to the Voice and Unified Communications Downloads page at <http://tools.cisco.com/support/downloads/pub/Redirect.x?mdfid=278875240>.



**Note** To access the software download page, you must be signed in to Cisco.com as a registered user.

This procedure describes the steps when using Internet Explorer as your web browser. If you are using a different web browser, the steps may differ.

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- Step 2** In the tree control on the Downloads page, expand **Unified Communications Applications > Voice Mail and Unified Messaging > Cisco Unity**, and select **Cisco Unity Telephony Integration**.
- Step 3** On the Log In page, enter your username and password, then select **Log In**.
- Step 4** On the Select a Release page, under Latest Releases, select the most recent release.
- Step 5** In the right column, select the version of the firmware for digital PIMG units.
- Step 6** On the Download Image page, select **Download**.
- Step 7** On the Supporting Document(s) page, select **Agree**.
- Step 8** In the File Download dialog box, select **Save**.
- Step 9** 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 select **Save**.
- Step 10** In the Download Complete dialog box, select **Open**. The window for extracting the PIMG firmware update files appears.
- Step 11** Select **Extract**.
- Step 12** In the Extract dialog box, browse to the directory where you want the extracted files, and select **Extract**.
- Step 13** Close the window for the extracting application.
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#### **To Set Up the Digital PIMG Units (Firmware Version 6.x)**

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- Step 1** On the Windows workstation, add a temporary route to enable access to the PIMG units.
- On the Windows Start menu, select **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** To sign in, enter the following case-sensitive settings.

**Table 13-8 Sign-in Settings**

Field	Setting
Username	Enter <b>admin</b> .
Password	Enter <b>IpodAdmin</b> .

- Step 5** Select **OK**.
- Step 6** On the System menu, select **Upgrade**.
- Step 7** On the Upgrade page, select **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** Select **Ami<xx>.app** (where <xx> is multiple digits), and select **Open**.
- Step 10** On the Upgrade page, select **Install File**.
- Step 11** After the file is installed, a message prompting you to restart the PIMG unit appears. Select **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 Configuration menu, select **Import/Export**.
- Step 14** On the Import/Export page, select **Browse**.
- Step 15** In the Choose File dialog box, browse to the file DNI\_Cfg\_Generic.ini.
- Step 16** Select **DNI\_Cfg\_Generic.ini**, and select **Open**.
- Step 17** On the Import/Export page, select **Import File**.
- Step 18** After the file is imported, a message prompting you to restart the PIMG unit appears. Select **OK**.
- Step 19** In the web browser, go to **http://10.12.13.74**.
- Step 20** To sign in, enter the following case-sensitive settings.

**Table 13-9 Sign-in Settings**

Field	Setting
Username	Enter <b>admin</b> .
Password	Enter <b>IpodAdmin</b> .

- Step 21** Select **OK**.



**Step 22** Do the following substeps to configure an RTP port range of 16384 to 32767.



**Caution**

You must set the RTP port range for the PIMG units if your system uses an RTP port range of 16384 to 32767. Otherwise, Cisco Unity Connection will not be able to answer calls, and callers will hear ringing or silence.



**Note**

The default RTP port range for PIMG units is 49000 to 50000. Some Cisco Unity Connection configurations require a different RTP port range.

- a. On the Configuration menu, select **Import/Export**.
- b. On the Import/Export page, under Export Settings, select **Export All Settings**.
- c. In the File Download dialog box, select **Save**.
- d. In the Save As dialog box, browse to the Windows workstation that has access to the PIMG units, browse to a directory where you want to save the file, and select **Save**.
- e. In the Download Complete dialog box, select **Open**. Notepad opens the file Config.ini that you saved.
- f. Locate the line with the following parameter:  

```
gwRTPStartPort
```
- g. Change the value of the parameter to **16384** so that the line reads as follows:  

```
gwRTPStartPort = 16384
```
- h. Locate the line with the following parameter:  

```
gwRTPEndPort
```
- i. Change the value of the parameter to **32767** so that the line reads as follows:  

```
gwRTPEndPort = 32767
```
- j. Save the file, and exit Notepad.
- k. On the Configuration menu of the PIMG unit, select **Import/Export**.
- l. On the Import/Export page, under Browse for Import File, select **Browse**.
- m. In the Choose File dialog box, browse to the file Config.ini that you saved.
- n. Select **Config.ini**, and select **Open**.
- o. On the Import/Export page, select **Import File**.
- p. When prompted to restart the PIMG unit, select **OK**.
- q. When the PIMG unit has restarted, in the web browser, go to **http://10.12.13.74**.
- r. To sign in, enter the following case-sensitive settings.

**Table 13-10 Sign-in Settings**

Field	Setting
Username	Enter <b>admin</b> .
Password	Enter <b>IpodAdmin</b> .

s. Select **OK**.

**Step 23** On the System menu, select **Password**.

**Step 24** On the Change Password page, enter the following settings.

**Table 13-11 Change Password Page Settings**

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

**Step 25** Select **Change**.

**Step 26** On the Configuration menu, select **Routing Table**.

**Step 27** On the Routing Table page, under Router Configuration, select **VoIP Host Groups**.

**Step 28** Under VoIP Host Groups, enter the following settings for the first VoIP Host Group.

**Table 13-12 First VoIP Host Group Settings**

Field	Settings
Name	Accept the default or enter another descriptive name of the VoIP host group.
Load-Balanced	(Cisco Unity Connection without a cluster) Select <b>False</b> . (Cisco Unity Connection with a cluster configured) Select <b>False</b> .
Fault-Tolerant	(Cisco Unity Connection without a cluster) Select <b>False</b> . (Cisco Unity Connection with a cluster configured) Select <b>True</b> .

**Step 29** For Cisco Unity Connection without a cluster, under Host List, enter the host name or IP address of the Cisco Unity Connection server and the server port in the format <host name or IP address>:5060.

For Cisco Unity Connection with a cluster configured, under Host List, enter the host name or IP address of the subscriber Cisco Unity Connection server (the second Cisco Unity Connection server that you installed) and the server port in the format <host name or IP address>:5060.

**Step 30** For Cisco Unity Connection without a cluster, continue to [Step 32](#). For Cisco Unity Connection with a cluster configured, select **Add Host**.

**Step 31** In the second field, enter the host name or IP address of the publisher Cisco Unity Connection server (the first Cisco Unity Connection server that you installed) and the server port in the format <host name or IP address>:5060.



**Caution** Do not add a third host under Host List or a second host group under VoIP Host Groups. Otherwise, the Cisco Unity Connection cluster may not function correctly.

- Step 32** Select **Submit**.
- Step 33** Under Router Configuration, select **TDM Trunk Groups**.
- Step 34** Under TDM Trunk Groups, select **Add Trunk Group**.
- Step 35** Under TDM Trunk Groups, enter the following settings for the first TDM trunk group.

**Table 13-13 First TDM Trunk Group Settings (Inbound Calls)**

Field	Settings
Name	Enter <b>Inbound_PBX_Calls</b> or another unique name. This TDM trunk group will handle all incoming calls from the phone system.
Selection Direction	Select <b>Ascending</b> .
Selection Mode	Select <b>Linear</b> .
Port/Channel Content	Enter the numbers of the PIMG ports that will handle inbound calls. For example, enter "*" for all PIMG ports, or enter "1-6" for the first six PIMG ports.

- Step 36** Under TDM Trunk Groups, select **Add Trunk Group**.
- Step 37** Enter the following settings for the second TDM trunk group.

**Table 13-14 Second TDM Trunk Group Settings (MWIs)**

Field	Settings
Name	Enter <b>MWIs</b> or another unique name. This TDM trunk group will handle outbound MWI calls (where applicable).
Selection Direction	Select <b>Ascending</b> .
Selection Mode	Select <b>Cyclic</b> .
Port/Channel Content	Enter the numbers of the PIMG ports that will MWIs. For example, enter "*" for all PIMG ports, or enter "7,8" for the last two PIMG ports.

- Step 38** Under TDM Trunk Groups, select **Add Trunk Group**.
- Step 39** Enter the following settings for the third TDM trunk group.

**Table 13-15 Third TDM Trunk Group Settings (Outbound Calls)**

Field	Settings
Name	Enter <b>Outbound_PBX_Calls</b> or another unique name. This TDM trunk group will handle all outbound calls from Cisco Unity Connection.
Selection Direction	Select <b>Descending</b> .

**Table 13-15** Third TDM Trunk Group Settings (Outbound Calls) (continued)

Field	Settings
Selection Mode	Select <b>Linear</b> .
Port/Channel Content	Enter * for all channels in all ports. Enter the numbers of the PIMG ports that will handle outbound (dialout) calls. For example, enter “*” for all PIMG ports, or enter “7,8” for the last two PIMG ports.

**Step 40** Select **Submit**.

**Step 41** Under Router Configuration, select **Inbound VoIP Rules**.

**Step 42** Under Inbound VoIP Rules, uncheck the **Enabled** check box for the default rule.

**Step 43** Select **Add Rule**.

**Step 44** Under Inbound VoIP Rules, enter the following settings for the first new inbound VoIP rule.

**Table 13-16** First New Inbound VoIP Rule Settings (MWIs)

Field	Settings
Enable	Check this check box.
Rule Label	Enter <b>MWI_Calls</b> or another name. This inbound VoIP rule will handle all MWI calls from Cisco Unity Connection.
Request Type	Select <b>Message</b> .
Originating VoIP Host Address	Enter *.

**Step 45** Under Inbound VoIP Request Matching, enter the following settings.



**Caution** The rule that you created in [Step 44](#) must be selected. Otherwise, any changes you make will apply to another inbound VoIP rule.

**Table 13-17** Inbound VoIP Request Matching Settings

Field	Settings
Calling Number	Enter *.
Calling Name	Enter *.
Called Number	Enter *.
Called Name	Enter *.
Redirect Number	Enter *.
Redirect Name	Enter *.

**Step 46** Under Outbound Routes, enter the following settings.



**Caution** The rule that you created in [Step 44](#) must be selected. Otherwise, any changes you make will apply to another rule.

**Table 13-18 Outbound Routes Settings**

Field	Settings
<b>Device Selection</b>	
Outbound Destination	Select <b>TDM</b> .
Trunk Group	Select the name of the TDM trunk group that you created for MWIs in <a href="#">Step 37</a> . For example, select “MWIs.”
<b>CPID Manipulation</b>	
Calling Number	Enter <b>S</b> .
Calling Name	Enter <b>S</b> .
Called Number	Enter <b>D</b> .
Called Name	Enter <b>D</b> .
Redirect Number	Enter <b>R</b> .
Redirect Name	Enter <b>R</b> .
<b>Select Primary/Alternate Route</b>	
Primary	Select <b>Primary</b> .

**Step 47** Under Inbound VoIP Rules, select **Add Rule**.

**Step 48** Under Inbound VoIP Rules, enter the following settings for the second new inbound VoIP rule.

**Table 13-19 Second New Inbound VoIP Rule Settings (Outbound Calls)**

Field	Settings
Enable	Check this check box.
Rule Label	Enter <b>Outbound_UC_Calls</b> or another name. This inbound VoIP rule will handle all outbound calls from Cisco Unity Connection.
Request Type	Select <b>Call</b> .
Originating VoIP Host Address	Enter *.

**Step 49** Under Inbound VoIP Request Matching, enter the following settings.



**Caution** The rule that you created in [Step 48](#) must be selected. Otherwise, any changes you make will apply to another rule.

**Table 13-20 Inbound VoIP Request Matching Settings**

Field	Settings
Calling Number	Enter *.
Calling Name	Enter *.
Called Number	Enter *.
Called Name	Enter *.
Redirect Number	Enter *.
Redirect Name	Enter *.

**Step 50** Under Outbound Routes, enter the following settings.



**Caution** The rule that you created in [Step 48](#) must be selected. Otherwise, any changes you make will apply to another rule.

**Table 13-21 Outbound Routes Settings**

Field	Settings
<b>Device Selection</b>	
Outbound Destination	Select <b>TDM</b> .
Trunk Group	Select the name of the TDM trunk group that you created for outbound calls in <a href="#">Step 39</a> . For example, select “Outbound_PBX_Calls.”
<b>CPID Manipulation</b>	
Calling Number	Enter <b>S</b> .
Calling Name	Enter <b>S</b> .
Called Number	Enter <b>D</b> .
Called Name	Enter <b>D</b> .
Redirect Number	Enter <b>R</b> .
Redirect Name	Enter <b>R</b> .
<b>Select Primary/Alternate Route</b>	
Primary	Select <b>Primary</b> .

**Step 51** Select **Submit**.

**Step 52** Under Router Configuration, select **Inbound TDM Rules**.

**Step 53** Under Inbound TDM Rules, enter the following settings for the first inbound TDM rule.

**Table 13-22** First Inbound TDM Rule Settings

Field	Settings
Enable	Check this check box.
Rule Label	Enter <b>Inbound_Rule_1</b> or another name. This inbound TDM rule will handle all incoming calls from the phone system.
Request Type	Select <b>Call</b> .
Trunk Group	Select the name of the TDM trunk group that you created for incoming calls from the phone system in <a href="#">Step 35</a> . For example, select “Inbound_PBX_Calls.”

**Step 54** Under Inbound TDM Request Matching, enter the following settings.



**Caution** The rule that you created in [Step 53](#) must be selected. Otherwise, any changes you make will apply to another rule.

**Table 13-23** Inbound TDM Request Matching Settings

Field	Settings
Calling Number	Enter the applicable matching rule that will be used. For example, enter “*” for all.
Calling Name	Enter the applicable matching rule that will be used. For example, enter “*” for all.
Called Number	Enter the applicable matching rule that will be used. For example, enter “*” for all.
Called Name	Enter the applicable matching rule that will be used. For example, enter “*” for all.
Redirect Number	Enter the applicable matching rule that will be used. For example, enter “*” for all.
Redirect Name	Enter the applicable matching rule that will be used. For example, enter “*” for all.

**Step 55** Under Outbound Routes, enter the following settings.



**Caution** The rule that you created in [Step 53](#) must be selected. Otherwise, any changes you make will apply to another rule.

**Table 13-24** Outbound Routes Settings

Field	Settings
<b>Device Selection</b>	
Outbound Destination	Select <b>VoIP</b> .

**Table 13-24 Outbound Routes Settings (continued)**

Field	Settings
Host Group	Select the name of the VoIP host group that you created for Cisco Unity Connection in <a href="#">Step 28</a> .
<b>CPID Manipulation</b>	
Calling Number	Enter <b>S</b> .
Calling Name	Enter <b>S</b> .
Called Number	Enter <b>D</b> .
Called Name	Enter <b>D</b> .
Redirect Number	Enter <b>R</b> .
Redirect Name	Enter <b>R</b> .
<b>Select Primary/Alternate Route</b>	
Primary	Select <b>Primary</b> .

- Step 56** If you want to create more Inbound TDM rules, under Inbound TDM Rules, select **Add Rule**. Otherwise, continue to [Step 58](#).
- Step 57** Repeat [Step 53](#) through [Step 56](#) for all remaining inbound TDM rules that you want to create.
- Step 58** Select **Submit**.
- Step 59** On the Configuration menu, select **TDM > Digital**.
- Step 60** On the Digital Telephony page, in the Telephony Switch Type field, select **Optiset\_300ECS**.
- Step 61** Select **Submit**.
- Step 62** On the Configuration menu, select **TDM > General**.
- Step 63** On the TDM General Settings page, enter the following settings.

**Table 13-25 TDM General Settings Page Settings**

Field	Settings
PCM Coding	Select <b>uLaw</b> .
Minimum Call Party Delay (ms)	Enter <b>500</b> .
Maximum Call Party Delay (ms)	Enter <b>2000</b> .
Dial Digit on Time (ms)	Enter <b>100</b> .
Dial Inter-Digit Time (ms)	Enter <b>100</b> .
Dial Pause Time (ms)	Enter <b>2000</b> .
Turn MWI On FAC	Leave this field blank.
Turn MWI Off FAC	Leave this field blank.
Outbound Call Connect Timeout (ms)	Enter <b>10000</b> .



**Table 13-25 TDM General Settings Page Settings (continued)**

Field	Settings
Wait for Ringback/Connect on Blind Transfer	Select <b>Yes</b> .
Hunt Group Extension	Enter the pilot number of the Cisco Unity Connection voice messaging ports.

**Step 64** Select **Submit**.

**Step 65** On the Configuration menu, select **TDM > Port Enable**.

**Step 66** On the TDM Port Enabling page, select **No** for the ports that you want to disable on the PIMG unit.

**Step 67** Confirm that **Yes** is selected for all other ports on the PIMG unit.

**Step 68** Select **Submit**.

**Step 69** On the Configuration menu, select **VoIP > General**.

**Step 70** On the VoIP General Settings page, enter the following settings.

**Table 13-26 VoIP General Settings Page Settings**

Field	Setting
<b>User-Agent</b>	
Host and Domain Name	Enter the host and domain name of the PIMG unit.
Transport Type	Select <b>UDP</b> .
Call as Domain Name	Select <b>No</b> .
Invite Expiration (sec)	Enter <b>120</b> .
<b>Server</b>	
DNS Server Address	Enter the IP Address of the Domain Name Server that the PIMG unit will use.
Registration Server Address	Leave this field blank.
Registration Server Port	Enter <b>5060</b> .
Registration Expiration (sec)	Enter <b>3600</b> .
<b>TCP/UDP</b>	
UDP/TCP Transports Enabled	Select <b>Yes</b> .
TCP/UDP Server Port	Enter <b>5060</b> .
<b>Proxy</b>	
Primary Proxy Server Address	Leave this field blank.

**Table 13-26 VoIP General Settings Page Settings (continued)**

Field	Setting
Primary Proxy Server Port	Not applicable. Leave the default setting.
Backup Proxy Server Address	Not applicable. Leave the default setting.
Backup Proxy Server Port	Not applicable. Leave the default setting.
Proxy Query Interval	Enter <b>10</b> .
<b>Timing</b>	
T1 Time (ms)	Enter <b>400</b> .
T2 Time (ms)	Enter <b>3000</b> .
<b>Monitoring</b>	
Monitor Call Connections	Select <b>No</b> .

**Step 71** Select **Submit**.



**Step 72** On the Configuration menu, select **VoIP > Media**.

**Step 73** On the VoIP Media Settings page, enter the following settings.

**Table 13-27 VoIP Media Settings Page Settings**

Field	Settings
<b>Audio</b>	
Audio Compression	Select the preferred codec for audio compression.
RTP Digit Relay Mode	Select <b>RFC2833</b> .
Signaling Digit Relay Mode	Select <b>Off</b> .
Voice Activity Detection	Select <b>Off</b> .

**Table 13-27** VoIP Media Settings Page Settings (continued)

Field	Settings
Frame Size	Select the applicable setting: <ul style="list-style-type: none"> <li>• G.711—<b>20</b></li> <li>• G.729AB—<b>10</b></li> </ul>  <b>Caution</b> Failure to use the correct setting will result in recorded messages containing nothing but silence.
Frames Per Packet	Select the applicable setting: <ul style="list-style-type: none"> <li>• G.711—<b>1</b></li> <li>• G.729AB—<b>2</b></li> </ul>  <b>Caution</b> Failure to use the correct setting will result in recorded messages containing nothing but silence.

**Step 74** Select **Submit**.

**Step 75** On the Configuration menu, select **VoIP > QOS**.

**Step 76** On the VoIP QOS Configuration page, enter the following settings.

**Table 13-28** VoIP QOS Configurative Page Settings

Field	Settings
Call Control QOS Byte	Enter <b>104</b> .
RTP QOS Byte	Enter <b>184</b> .

**Step 77** Select **Submit**.

**Step 78** On the Configuration menu, select **IP**.

**Step 79** On the IP Settings page, enter the following settings.

**Table 13-29** IP Settings Page Settings

Field	Settings
Client IP Address	Enter 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	Enter the new subnet mask, if the subnet mask is different from the default IP address.
Default Network Gateway Address	Enter the IP address of the default network gateway router that the PIMG units will use.
BOOTP Enabled	Select <b>No</b> .

- Step 80** Select **Submit**.
- Step 81** On the Configuration menu, select **Tone Detection**.
- Step 82** On the Tone Detection page, under Call Progress Tone - Learn, in the Learn Tone Event field, select **Busy** and do the following substeps to verify that the tone is correct.
- From an 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 83** Under Call Progress Tone - Learn, in the Dial String field, enter the extension that you dialed in [Step 82c](#) from the third phone.
- Step 84** Select **Learn**.
- Step 85** On the Tone Detection page, under Call Progress Tone - Learn, in the Learn field, select **Error** 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 86** Under Call Progress Tone - Learn, in the Dial String field, enter the extension that you dialed in [Step 85a](#).
- Step 87** Select **Learn**.
- Step 88** On the Tone Detection page, under Call Progress Tone - Learn, in the Learn field, select **Ringback** 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 89** Under Call Progress Tone - Learn, in the Dial String field, enter the extension that you dialed in [Step 88a](#).
- Step 90** Select **Learn**.
- Step 91** Select **Submit**.
- Step 92** Hang up the phones that you used in [Step 82](#).
- Step 93** On the System menu, select **Restart**.
- Step 94** On the Restart page, select **Restart Unit Now**.
- Step 95** Repeat [Step 2](#) through [Step 94](#) 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, the PIMG units, and Cisco Unity Connection are ready for the integration, do the following procedure to set up the integration and to enter the port settings.

### To Create an Integration

- Step 1** Sign in to Cisco Unity Connection Administration.
- Step 2** In Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Phone System**.
- Step 3** On the Search Phone Systems page, under Display Name, select the name of the default phone system.
- Step 4** On the Phone System Basics page, in the Phone System Name field, enter the descriptive name that you want for the phone system.
- Step 5** If you want to use this phone system as the default for TRaP connections so that administrators and users without voicemail boxes can record and playback through the phone in Cisco Unity Connection web applications, check the **Default TRAP Switch** check box. If you want to use another phone system as the default for TRaP connections, uncheck this check box.
- Step 6** Select **Save**.
- Step 7** On the Phone System Basics page, in the Related Links drop-down box, select **Add Port Group** and select **Go**.
- Step 8** On the New Port Group page, enter the applicable settings and select **Save**.

**Table 13-30 Settings for the New Port Group Page**

Field	Setting
Phone System	Select the name of the phone system that you entered in <a href="#">Step 4</a> .
Create From	Select <b>Port Group Template</b> and select <b>SIP to DMG/PIMG/TIMG</b> in the drop-down box.
Display Name	Enter a descriptive name for the port group. You can accept the default name or enter the name that you want.
SIP Security Profile	Select <b>5060</b> .
SIP Transport Protocol	Select the SIP transport protocol that Cisco Unity Connection will use.
IPv4 Address or Host Name ( <i>Cisco Unity Connection 8.5 and later</i> )	Enter the IP address of the PIMG unit that you are integrating with Cisco Unity Connection.
IPv6 Address or Host Name ( <i>Cisco Unity Connection 8.5 and later</i> )	Do not enter a value in this field. IPv6 is not supported for PIMG integrations.
IP Address or Host Name ( <i>Cisco Unity Connection 8.0</i> )	Enter the IP address of the PIMG unit that you are integrating with Cisco Unity Connection.
Port	Enter the SIP port of the PIMG unit that Cisco Unity Connection will connect to. We recommend that you use the default setting.
	 <p><b>Caution</b> This name must match the setting in the TCP/UDP Server Port field on the Configuration &gt; VoIP &gt; General page of the PIMG unit. Otherwise, the integration will not function correctly.</p>

- Step 9** In the Related Links drop-down box, select **Add Ports** and select **Go**.
- Step 10** On the New Port page, enter the following settings and select **Save**.

**Table 13-31 Settings for the New Port Page**

Field	Considerations
Enabled	Check this check box.
Number of Ports	Enter <b>8</b> .  (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.)  <b>Note</b> For a Cisco Unity Connection cluster, the Cisco Unity Connection server must have the number of voice messaging ports that are set up on the phone system for the PIMG integration so that this Cisco Unity Connection server can handle all voice messaging traffic for the Cisco Unity Connection cluster if one of the servers stops functioning. For example, if the phone system is set up with 16 voice messaging ports, this Cisco Unity Connection server must have 16 voice messaging ports.
Phone System	Select the name of the phone system that you entered in <a href="#">Step 4</a> .
Port Group	Select the name of the port group that you added in <a href="#">Step 8</a> .

- Step 11** On the Search Ports page, select the display name of the first voice messaging port that you created for this phone system integration.



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

- Step 12** 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 13-32 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 13-32** 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.

**Step 13** Select **Save**.

**Step 14** Select **Next**.

**Step 15** Repeat [Step 12](#) through [Step 14](#) for all remaining voice messaging ports for the phone system.

**Step 16** In Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Phone System**.

**Step 17** On the Search Phone Systems page, under Display Name, select the name of the phone system that you entered in [Step 4](#).

**Step 18** Repeat [Step 7](#) through [Step 17](#) for each remaining PIMG unit that will be integrated with Cisco Unity Connection.



**Note** Each PIMG unit is connected to one port group with the applicable voice messaging ports. For example, a system that uses five PIMG units requires five port groups, one port group for each PIMG unit.

**Step 19** If another phone system integration exists, in Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Trunk**. Otherwise, skip to [Step 23](#).

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

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

**Table 13-33** Settings for the Phone System Trunk

Field	Setting
From Phone System	Enter the display name of the phone system that you are creating a trunk for.
To Phone System	Enter the display name of the previously existing phone system that the trunk will connect to.
Trunk Access Code	Enter 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** In the Related Links drop-down list, select **Check Telephony Configuration** and select **Go** to confirm the phone system integration settings.

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

**Step 24** In the Task Execution Results window, select **Close**.

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