



Avaya Definity G3 PIMG Integration Guide for Cisco Unity Connection 1.1

Revised October 12, 2007

This document provides instructions for integrating the Avaya Definity G3-series 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 Avaya Definity G3 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 an integration.

Task List to Create the Integration

Use the following task list to set up a new integration with the Avaya Definity G3 phone system. If you are installing a new Cisco Unity Connection server by using the *Cisco Unity Connection Installation Guide*, you may have already completed some of the following tasks.

1. Review the system and equipment requirements to confirm that all phone system and Cisco Unity Connection server requirements have been met. See the [“Requirements” section on page 2](#).
2. Plan how the voice messaging ports will be used by Cisco Unity Connection. See the [“Planning How the Voice Messaging Ports Will Be Used by Cisco Unity Connection” section on page 4](#).
3. Program the Avaya Definity G3 phone system and extensions. See the [“Programming the Avaya Definity G3 Phone System” section on page 6](#).
4. Set up the PIMG units. See the [“Setting Up the PIMG Units” section on page 8](#).
5. Create the integration. See the [“Creating a New Integration with the Avaya Definity G3 Phone System” section on page 15](#).
6. Test the integration. See the [“Testing the Integration” section on page 17](#).



- 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 21](#).

Requirements

The Avaya Definity G3 integration supports configurations of the following components:

Phone System

- Avaya Definity G3-series phone system. The voice messaging ports that connect to the PIMG units must be compatible with the 7434ND, 8434, 8434D, or 8434DX digital phones.



Caution You must use 2-wire line cards for these voice messaging ports. Otherwise, the voice messaging ports will not be compatible with the 7434ND, 8434, 8434D, or 8434DX digital phones.

- Software version 4 or later.
- One or more of the applicable PIMG units. For details, refer to the “Supported Circuit-Switched Phone System Integrations” section of *Cisco Unity Connection System Requirements, and Supported Hardware and Software* at http://www.cisco.com/en/US/products/ps6509/prod_installation_guides_list.html.
- The voice messaging ports in the phone system connected by digital lines to the ports on the PIMG units.

We recommend that you connect the voice messaging ports on the phone system to the ports on the PIMG units in a planned manner to simplify troubleshooting. For example, the first phone system voice messaging port connects to the first port on the first PIMG unit, the second phone system voice messaging port connects to the second port on the first PIMG unit, and so on.

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

Cisco Unity Connection Server

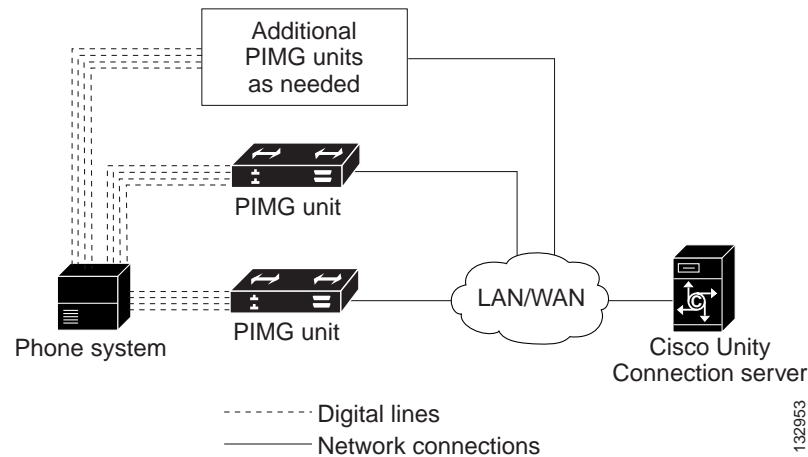
- Cisco Unity Connection installed and ready for the integration, as described in the *Cisco Unity Connection Installation Guide* at http://www.cisco.com/en/US/products/ps6509/prod_installation_guides_list.html.
- A license that enables the applicable number of voice messaging ports.

Integration Description

The Avaya Definity G3 PIMG 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 Avaya Definity G3 integration with Cisco Unity Connection provides the following features:

- Call forward to personal greeting
- Call forward to busy greeting
- Caller ID
- Easy message access (a subscriber can retrieve messages without entering an ID because Cisco Unity Connection identifies the subscriber based on the extension from which the call originated; a password may be required)

- Identified subscriber messaging (Cisco Unity Connection identifies the subscriber who leaves a message during a forwarded internal call, based on the extension from which the call originated)
- Message waiting indication (MWI)

Integrations with Multiple Phone Systems

Cisco Unity Connection can be integrated with multiple phone systems at one time. For information on and instructions for integrating Cisco Unity Connection with multiple phone systems, refer to the *Multiple Phone System Integration Guide* at http://www.cisco.com/en/US/products/ps6509/products_installation_and_configuration_guides_list.html.

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

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

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

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

Table 1 *Settings for the Voice Messaging Ports*

Field	Considerations
Enabled	Check this check box to enable the port. The port is enabled during normal operation. Uncheck this check box to disable the port. When the port is disabled, calls to the port get a ringing tone but are not answered. Typically, the port is disabled only by the installer during testing.
Extension	Enter the extension for the port as assigned on the phone system.
Answer Calls	Check this check box to designate the port for answering calls. These calls can be incoming calls from unidentified callers or from users.
Perform Message Notification	Check this check box to designate the port for notifying users of messages. Assign Perform Message Notification to the least busy ports.
Send MWI Requests	Check this check box to designate the port for turning MWIs on and off. Assign Send MWI Requests to the least busy ports.

Table 1 **Settings for the Voice Messaging Ports (continued)**

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

The Number of Voice Messaging Ports to Install

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

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

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

The Number of Voice Messaging Ports That Will Answer Calls

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

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

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

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

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

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



Caution

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

Preparing for Programming the Phone System

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

Programming the Avaya Definity G3 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 Avaya Definity G3 Phone System

- Step 1** Create a coverage path which contains the PIMG unit hunt group number as the coverage point.
- Step 2** Assign the coverage path that you created in [Step 1](#) to the user stations that must forward to the voice messaging ports on the PIMG units when calls are not answered or when the user station is busy, based on one of the Cisco Unity Connection call transfer types shown in [Table 2](#).

Table 2 Call Transfer Types

Transfer Type	Usage
Release transfer (blind transfer)	Program the user station 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.

- Step 3** Use the Add Station <extension number> command (for example, Add Station 2999) to assign an extension number for each voice messaging port, which is a digital line that connects to the PIMG unit. Set the voice messaging port options ([Table 3](#)) and button assignments ([Table 4](#)), and press **Enter**.



Note We recommend that you distribute the voice messaging ports among multiple phone system line cards so that call processing can continue even if a line card becomes inactive.

Table 3 *Voice Messaging Port Options for All Lines*


Option	Setting
Extension	<the extension number of the digital line>
Type	<p>7434ND (recommended)</p> <p>You can also use the following settings. However, you must include the extension with each subscriber name when programming the phone system (for example, “3001 Jane Doe”). Otherwise, the integration will not function correctly.</p> <ul style="list-style-type: none"> • 8434 • 8434D • 8434DX <p> Note If the setting that you want is not available, contact the phone system manufacturer.</p>
LWC Reception	msa
LWC Activation?	y
Restrict Last Appearance?	y
Display Module	y

Table 4 *Button Assignments for All Lines*

Button Assignment	Setting
1	call-appr
2	call-appr
9	lwc-store
10	lwc-cancel

Step 4 Use the Add Hunt <hunt group number> command (for example, Add Hunt 1) to assign a the voice messaging ports to a hunt group. Set the following options.

Table 5 *Hunt Group Options*

Option	Setting
Group Number	<the hunt group number>
Group Extension	<the pilot number for the hunt group>
Group Type	ucd

Table 5 Hunt Group Options (continued)

Option	Setting
Group Name	<the display name for the hunt group>
Queue?	N

Step 5 Enter the group member assignments for the voice messaging ports that will answer calls and press **Enter**.

If you plan to set the voice messaging ports to either answer calls or to dial out (for example, to set MWIs), make sure that you include in the hunt group only the voice messaging ports that will be set to answer calls.

For smaller systems, include in the hunt group all voice messaging ports when the ports will be set to both answer calls and dial out (for example, to set MWIs).

Setting Up the PIMG Units

Do the following procedures to set up the PIMG units that are connected to the Avaya Definity G3 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 unit.

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 6 System Login Page Settings

Field	Setting
Username	admin
Password	IpodAdmin

- Step 5** Click **Log On**.
- Step 6** On the Configure menu, click **Upgrade**.
- Step 7** On the Upgrade page, click **Browse**.
- Step 8** In the Choose File dialog box, browse to the directory on the Windows workstation that has the extracted PIMG firmware update files.
- Step 9** Click **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_Lucent.ini.
- Step 16** Click **DNI_Cfg_Lucent.ini**, and click **Open**.
- Step 17** On the Upgrade page, click **Install**.
- Step 18** After the file is installed, a message prompting you to restart the PIMG unit appears. Click **OK**.
- Step 19** In the web browser, go to **http://10.12.13.74**.
- Step 20** On the System Login page, enter the following case-sensitive settings.

Table 7 System Login Page Settings

Field	Setting
Username	admin
Password	IpodAdmin

- Step 21** Click **Log On**.
- Step 22** On the Configure menu, click **Password**.
- Step 23** On the Password page, enter the following settings.

Table 8 Password Page Settings

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

- Step 24** Click **Change**.
- Step 25** On the Configure menu, click **System**.
- Step 26** On the System page, enter the following settings.

Table 9 System Page Settings

Field	Setting
Operating Mode	SIP

Table 9 System Page Settings (continued)

Field	Setting
Telephony Switch Type	Lucent
PCM Coding	uLaw

- Step 27** Click **Apply Changes**.
- Step 28** On the Configure menu, click **Gateway**.
- Step 29** On the Gateway page, click the **Gateway Routing** tab.
- Step 30** On the Gateway Routing tab, enter the following settings.

Table 10 Gateway Routing Tab Settings





Field	Setting
Fault Tolerance Enabled	No
Load Balancing Enabled	No
VoIP Endpoint ID: 1	<the IP address of the Cisco Unity Connection server>
VoIP Endpoint ID: 2	<blank>

- Step 31** Click **Apply Changes**.
- Step 32** Click the **Gateway Advanced** tab.
- Step 33** On the Gateway Advanced tab, enter the following settings.

Table 11 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	<blank>
Turn MWI Off FAC	<blank>
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

Table 11 Gateway Advanced Tab Settings (continued)

Field	Setting
Voice Activity Detection	Off
Frame Size	<p>Click the applicable setting:</p> <ul style="list-style-type: none"> • G.711—20 • G.729a—10 <p> Caution Failure to use the correct setting will result in recorded messages containing nothing but silence.</p>
Frames Per Packet	<p>Click the applicable setting:</p> <ul style="list-style-type: none"> • G.711—1 • G.729a—2 <p> Caution Failure to use the correct setting will result in recorded messages containing nothing but silence.</p>
Call Control QOS Byte	<p>(PIMG units connect only to a LAN) 0</p> <p>(PIMG units connect to a WAN) 104</p> <p> Note For details on the setting for a LAN, see the caveat CSCsb96387.</p>
RTP QOS Byte	<p>(PIMG units connect only to a LAN) 0</p> <p>(PIMG units connect to a WAN) 184</p> <p> Note For details on the setting for a LAN, see the caveat CSCsb96387.</p>

Step 34 Click **Apply Changes**.

Step 35 Click the **Gateway Capabilities** tab.

Step 36 Depending on how you have planned to use the voice messaging ports, click the applicable setting for each port in the Telephony Port Capability column.

Table 12 Gateway Capabilities Tab Settings

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



Caution In setting up the PIMG unit, do not send calls to ports in Cisco Unity that cannot answer calls (voice messaging ports that are not set to Answer Calls). For example, if a voice messaging port is set only to Dialout MWI, do not send calls to it. Otherwise the integration will not function correctly.

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

- Step 37** Click **Apply Changes**.
- Step 38** On the Configure menu, click **SIP**.
- Step 39** On the SIP page, enter the following settings.

Table 13 *SIP Page Settings*

Field	Setting
Host and Domain Name	<the domain name of the PIMG unit>
Server Port	5060
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 40** Click **Apply Changes**.
- Step 41** On the Configure menu, click **IP**.
- Step 42** On the IP page, enter the following settings.

Table 14 *IP Page Settings*

Field	Setting
Client IP Address	<the new IP address you want to use for the PIMG unit> (This is the IP address that you 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 43** Click **Apply Changes**.

Step 44 On the Configure menu, click **Tones**.

Step 45 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 46 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 47 On the Tones page, for the Busy Tone event, confirm that the Acquire Tone check box is checked and do the following substeps to verify that the tone is correct.

- a. From a available phone, call a second phone.
- b. Answer the second phone when it rings, and leave both handsets off so that both phones are busy.
- c. From a third phone, dial one of the busy phones.
- d. Confirm that you hear a busy tone.
- e. Hang up the third phone but leave the handsets for the other two phones off.

Step 48 On the Tones page, in the Destination Address field for Busy Tone, enter the extension that you dialed in [Step 47c](#). from the third phone.

Step 49 On the Tones page, for the Error/Reorder Tone event, confirm that the Acquire Tone check box is checked and do the following substeps to verify that the tone is correct.

- a. From an available phone, dial an extension that does not exist.
- b. Confirm that you hear the reorder or error tone.
- c. Hang up the phone.

Step 50 On the Tones page, in the Destination Address field for Error/Reorder Tone, enter the extension that you dialed in [Step 49a](#).

Step 51 On the Tones page, for the Ringback Tone event, confirm that the Acquire Tone check box is checked and do the following substeps to verify that the tone is correct.

- a. From an available phone, dial an extension that does exist
- b. Confirm that you hear the ringback tone.
- c. Hang up the phone.

Step 52 On the Tones page, in the Destination Address field for Ringback Tone, enter the extension that you dialed in [Step 51a](#).

Step 53 Click **Learn**.



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

Step 54 When the process is complete, check the check box for each newly learned tone and click **Apply**.

Step 55 Hang up the phones that you used in [Step 47](#).

Step 56 On the Configure menu, click **Restart**.

Step 57 On the Restart page, click **Restart Unit Now**.

- Step 58** When the PIMG unit has restarted, in the View menu, click **Refresh**.
- Step 59** Repeat [Step 2](#) through [Step 58](#) on all remaining PIMG units.


Creating a New Integration with the Avaya Definity G3 Phone System

After ensuring that the Avaya Definity G3 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 **Avaya** and click **Next**.
- Step 5** On the Select Phone System Model page, in the Model field, click **Avaya Definity G3** 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 **Avaya Definity G3 via PIMG** and click **Next**.
- Step 8** On the Set Up Port Group page, enter the following settings and click **Next**.

Table 15 Settings for the Set Up Port Group Page

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

- Step 9** On the Confirm Phone System Settings page, confirm the settings that you have entered and click **Finish**.
- Step 10** On the Phone System Creation Summary page, click **Close**.
- Step 11** On the Search Phone Systems page, click the display name of the phone system that you created for this phone system integration.
- Step 12** On the Phone System Basics page, under Message Waiting Indicator Settings, click **Use Port Memory** and click **Save**.
- Step 13** In Cisco Unity Connection Administration, expand **Telephony Integrations**, then click **Port**.
- Step 14** On the Search Ports page, click the display name of the first voice messaging port that you created for this phone system integration.



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

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

Table 16 Settings for the Voice Messaging Ports

Field	Considerations
Enabled	Check this check box to enable the port. The port is enabled during normal operation. Uncheck this check box to disable the port. When the port is disabled, calls to the port get a ringing tone but are not answered. Typically, the port is disabled only by the installer during testing.
Extension	Enter the extension for the port as assigned on the phone system.
Answer Calls	Check this check box to designate the port for answering calls. These calls can be incoming calls from unidentified callers or from users.
Perform Message Notification	Check this check box to designate the port for notifying users of messages. Assign Perform Message Notification to the least busy ports.
Send MWI Requests	Check this check box to designate the port for turning MWIs on and off. Assign Send MWI Requests to the least busy ports.
Allow TRAP Connections	Check this check box so that users can use the port for recording and playback through the phone in Cisco Unity Connection web applications. Assign Allow TRAP Connections to the least busy ports.
Outgoing Hunt Order	Enter the priority order in which Cisco Unity Connection will use the ports when dialing out (for example, if the Perform Message Notification, Send MWI Requests, or Allow TRAP Connections check box is checked). The highest numbers are used first. However, when multiple ports have the same Outgoing Hunt Order number, Cisco Unity Connection will use the port that has been idle the longest.

- Step 16** Click **Save**.
- Step 17** Click **Next**.
- Step 18** Repeat [Step 15](#) through [Step 17](#) for all remaining voice messaging ports for the phone system.
- Step 19** If another phone system integration exists, in Cisco Unity Connection Administration, expand **Telephony Integrations**, then click **Trunk**. Otherwise, skip to [Step 23](#).

- Step 20** On the Search Phone System Trunks page, on the Phone System Trunk menu, click **New Phone System Trunk**.
- Step 21** On the New Phone System Trunk page, enter the following settings for the phone system trunk and click **Save**.

Table 17 Settings for the Phone System Trunk

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

- Step 22** Repeat [Step 20](#) and [Step 21](#) for all remaining phone system trunks that you want to create.
- Step 23** If prompted to restart Cisco Unity Connection, in the Windows task bar, right-click the **Cisco Unity Connection** icon and click **Restart > Voice Processing Server Role**.
- Step 24** When prompted to confirm stopping the Voice Processing server role, click **Yes**.
- Step 25** In Cisco Unity Connection Administration, in the Related Links drop-down list, click **Check Telephony Configuration** and click **Go** to confirm the phone system integration settings.
- If the test is not successful, the Task Execution Results displays one or more messages with troubleshooting steps. After correcting the problems, test the connection again.
- Step 26** In the Task Execution Results window, click **Close**.
- Step 27** Log off Cisco Unity Connection Administration.



Note If your recording volume needs adjusting, you can change the recording gain for the PIMG units. For instructions, see the [“Appendix: Adjusting the Recording Gain for PIMG Units”](#) section on page 21.

Testing the Integration

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

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

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

To Set Up the Test Configuration

- Step 1** Set up two test extensions (Phone 1 and Phone 2) on the same phone system that Cisco Unity Connection is connected to.
- Step 2** Set Phone 1 to forward calls to the Cisco Unity Connection pilot number when calls are not answered.



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

- Step 3** To create a test user for testing, in Cisco Unity Connection Administration, expand **Users**, then click **Users**.
- Step 4** On the Search Users page, on the User menu, click **New User**.
- Step 5** On the New User page, enter the following settings.

Table 18 *Settings for the New User Page*

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

- Step 6** Click **Save**.
- Step 7** On the Edit User Basics page, in the Voice Name field, record a voice name for the test user.
- Step 8** In the Phone System field, confirm that the phone system selected is the phone system that Phone 1 is connected to.
- Step 9** Uncheck the **Set for Self-enrollment at Next Login** check box.
- Step 10** Click **Save**.
- Step 11** On the Edit menu, click **Message Waiting Indicators**.
- Step 12** On the Message Waiting Indicators page, click the message waiting indicator. If no message waiting indication is in the table, click **Add New**.
- Step 13** On the Edit Message Waiting Indicator page, enter the following settings.

Table 19 *Settings for the Edit MWI Page*

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

- Step 14** Click **Save**.
- Step 15** On the Edit menu, click **Transfer Options**.
- Step 16** On the Transfer Options page, click the active option.

- Step 17** On the Edit Transfer Option page, under Transfer Action, click the **Extension** option and enter the extension of Phone 1.
- Step 18** In the Transfer Type field, click **Release to Switch**.
- Step 19** Click **Save**.
- Step 20** Minimize the Cisco Unity Connection Administration window.
Do not close the Cisco Unity Connection Administration window because you will use it again in a later procedure.
- Step 21** On the Cisco Unity Connection desktop, double-click the **Tools Depot** icon.
- Step 22** In the left pane of the Tools Depot window, expand **Switch Integration Tools**, then double-click **Port Status Monitor**. The Port Status Monitor window appears.
- Step 23** On the Ports menu, click **Start All**, and arrange the port monitors so that you can notice which port will handle the calls that you will make.
-

To Test an External Call with Release Transfer

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

To Test Listening to Messages

- Step 1** From Phone 1, enter the internal pilot number for Cisco Unity Connection.
- Step 2** When asked for your password, enter the password for the test user. Hearing the request for your password means that the phone system sent the necessary call information to Cisco Unity Connection, which correctly interpreted the information.

- Step 3** Confirm that you hear the recorded voice name for the test user (if you did not record a voice name for the test user, you will hear the extension number for Phone 1). Hearing the voice name means that Cisco Unity Connection correctly identified the user by the extension.
 - Step 4** Listen to the message.
 - Step 5** After listening to the message, delete the message.
 - Step 6** Confirm that the MWI on Phone 1 is deactivated. The deactivated MWI means that the phone system and Cisco Unity Connection are successfully integrated for turning off MWIs.
 - Step 7** Hang up Phone 1.
 - Step 8** On the Port Status Monitor, confirm that the state of the port handling the call changes to “Idle.” This state means that the port was successfully released when the call ended.
-

To Set Up Supervised Transfer on Cisco Unity Connection

- Step 1** In Cisco Unity Connection Administration, on the Edit Transfer Option page for the test user, in the Transfer Type field, click **Supervise Transfer**.
 - Step 2** In the Rings to Wait For field, enter **3**.
 - Step 3** Click **Save**.
 - Step 4** Minimize the Cisco Unity Connection Administration window.
Do not close the Cisco Unity Connection Administration window because you will use it again in a later procedure.
-

To Test Supervised Transfer

- Step 1** From Phone 2, enter the access code necessary to get an outside line, then enter the number outside callers use to dial directly to Cisco Unity Connection.
 - Step 2** On the Port Status Monitor, note which port handles this call.
 - Step 3** When you hear the opening greeting, enter the extension for Phone 1. Hearing the opening greeting means that the port is configured correctly.
 - Step 4** Confirm that Phone 1 rings and that you do not hear a ringback tone on Phone 2. Instead, you should hear the indication your phone system uses to mean that the call is on hold (for example, music).
 - Step 5** Leaving Phone 1 unanswered, confirm that the state of the port handling the call remains “Busy.” This state and hearing an indication that you are on hold mean that Cisco Unity Connection is supervising the transfer.
 - Step 6** Confirm that, after three rings, you hear the greeting for the test user. Hearing the greeting means that Cisco Unity Connection successfully recalled the supervised-transfer call.
 - Step 7** During the greeting, hang up Phone 2.
 - Step 8** On the Port Status Monitor, confirm that the state of the port handling the call changes to “Idle.” This state means that the port was successfully released when the call ended.
 - Step 9** Exit the Port Status Monitor.
-

To Delete the Test User

- Step 1 In Cisco Unity Connection Administration, expand **Users**, then click **Users**.
 - Step 2 On the Search Users page, check the check box to the left of the test user.
 - Step 3 Click **Delete Selected**.
-

(Multiple Integrations Only) Adding New User Templates

When you create the first phone system integration, this phone system is automatically selected in the default user template. The users that you add after creating this phone system integration will be assigned to this phone system by default.

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

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

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

Appendix: Adjusting the Recording Gain for PIMG Units

If recordings are too quiet or too loud, do the following procedure to adjust the recording gain for the PIMG units.

To Adjust the Recording Gain for PIMG Units

- Step 1 In a web browser, go to **http://<IP address of PIMG unit>**.
- Step 2 On the System Login page, enter the user name and password.
- Step 3 Click **Log On**.
- Step 4 Go to **http://<IP address of PIMG unit>/dsp.htm**.
- Step 5 In the TDM to IP Gain field, increase the value to make recordings louder, or lower the value to make recordings quieter.



Caution Increasing the recording gain too much can cause the PIMG unit to fail to recognize DTMF tones, which will result in integration problems. If DTMF recognition problems occur after increasing the recording gain, decrease the value in the TDM to IP Gain field.

Step 6 Click **Apply Changes**.

Step 7 Repeat [Step 1](#) through [Step 6](#) for all remaining PIMG units.

Appendix: Documentation and Technical Assistance

Conventions

The *Avaya Definity G3 PIMG Integration Guide for Cisco Unity Connection 1.1* uses the following conventions.

Table 20 *Avaya Definity G3 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.)
[x] (square brackets)	Square brackets enclose an optional element (keyword or argument). (Example: [reg-e164])

Table 20 *Avaya Definity G3 PIMG Integration Guide for Cisco Unity Connection 1.1 Conventions (continued)*

Convention	Description
[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 *Avaya Definity G3 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>

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