



# Ericsson MD-110 Serial PIMG Integration Guide for Cisco Unity 4.0

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This document provides instructions for integrating the Ericsson MD-110 phone system with Cisco Unity by using the Intel NetStructure PBX-IP Media Gateway (PIMG) and an RS-232 serial cable.

## Integration Tasks

Before doing the following tasks to integrate Cisco Unity with the Ericsson MD-110 phone system by using the Intel NetStructure PBX-IP Media Gateway (PIMG), confirm that the Cisco Unity server is ready for the integration by completing the applicable tasks in the applicable Cisco Unity installation guide.

The following task lists describe 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 Ericsson MD-110 phone system. If you are installing a new Cisco Unity server by using the applicable Cisco Unity installation guide, you may have already completed some of the following tasks.

1. Review the system and equipment requirements to confirm that all phone system and Cisco Unity server requirements have been met. See the [“Requirements” section on page 2](#).
2. Plan how the voice messaging ports will be used by Cisco Unity. See the [“Planning How the Voice Messaging Ports Will Be Used by Cisco Unity” section on page 5](#).
3. Program the Ericsson MD-110 phone system and extensions. See the [“Programming the Ericsson MD-110 Phone System” section on page 7](#).
4. Set up the PIMG units. See the [“Setting Up the PIMG Units” section on page 9](#).
5. Create the integration. See the [“Creating a New Integration with the Ericsson MD-110 Phone System” section on page 17](#).



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**Caution**

Do not edit the phone configuration file (also known as the switch ini file) to customize this integration. If you change the settings in this file, the integration may not function correctly.

6. Test the integration. See the “Testing the Integration” section on page 21.
7. (*Cisco Unity 4.1 and later*) If you have a secondary server for Cisco Unity failover, integrate the secondary server. See the “Integrating a Secondary Server for Cisco Unity Failover” section on page 25.

## Requirements

The Ericsson MD-110 integration supports configurations of the following components:

### Phone System

- Ericsson MD-110.
- Software level BC6 or later.
- ICU card installed to provide the serial data port.
- One or more of the applicable PIMG units. For details, refer to the “Supported Circuit-Switched Phone System Integrations” section in the applicable *Supported Hardware and Software, and Support Policies* at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod\\_installation\\_guides\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod_installation_guides_list.html).
- The serial data port on the phone system connected to the serial port on the master PIMG unit with an RS-232 serial cable.

We recommend that the serial cable have the following construction:

- A maximum of 50 feet (15.24 m) in length
  - 24 AWG stranded conductors
  - Low capacitance—for example, no more than 12 pF/ft (39.4 pF/m) between conductors
  - At least 65 percent braided shield over aluminized polymer sleeve around conductors
  - UL-recognized overall cable jacket insulation with low dielectric constant
  - Braided shield fully terminated to and enclosed by a metal connector backshell
  - Gold-plated connector contacts
- The voice messaging ports in the phone system connected by analog 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 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.

**Caution**

If you use G.729a codec formatting over a WAN and your PIMG units use firmware release 4 SU6 or earlier, you must disable comfort noise. Otherwise, callers will hear loud comfort noise at certain points. For details, see the “[Appendix: PIMG Integrations Over a WAN That Use the G.729a Codec May Need to Disable Comfort Noise](#)” section on page 28.

- 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.

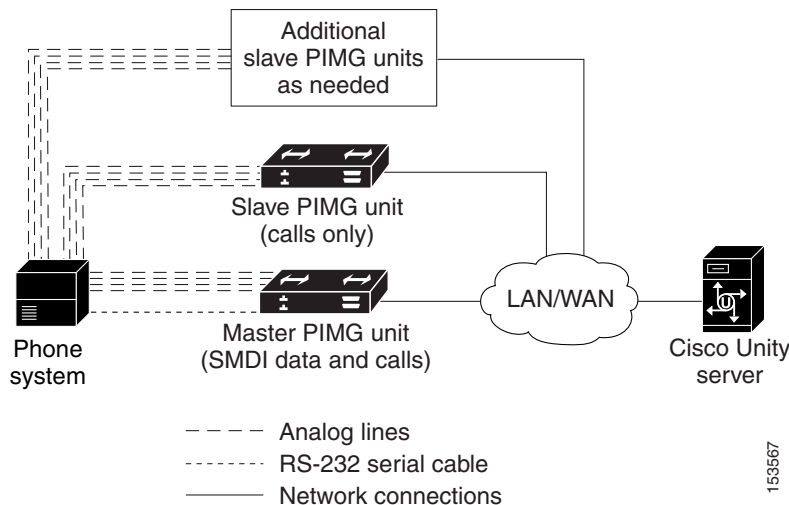
**Cisco Unity Server**

- Cisco Unity installed and ready for the integration, as described in the applicable Cisco Unity installation guide at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod\\_installation\\_guides\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod_installation_guides_list.html).
- A license that enables the applicable number of voice messaging ports.

## Integration Description

The Ericsson MD-110 PIMG integration sends call information and MWI requests through the data link, which is an RS-232 serial cable that connects the phone system and the master PIMG unit. Voice connections are sent through the analog lines between the phone system and the PIMG units. The PIMG units communicate with the Cisco Unity 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



## Call Information

The phone system sends the following information through the data link:

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

**Note**

The Ericsson MD-110 phone system also sends requests to turn on and turn off MWIs through the data link.

## Integration Functionality

The Ericsson MD-110 integration with Cisco Unity provides the following integration features:

- Call forward to personal greeting
- Call forward to busy greeting
- Caller ID
- Easy message access (a subscriber can retrieve messages without entering an ID because Cisco Unity identifies the subscriber based on the extension from which the call originated; a password may be required)
- Identified subscriber messaging (Cisco Unity identifies the subscriber who leaves a message during a forwarded internal call, based on the extension from which the call originated)
- Message waiting indication (MWI)

## Integrations with Multiple Phone Systems

Depending on the version, Cisco Unity can be integrated with two or more phone systems:

- Cisco Unity 4.0 and 4.1 can be integrated with a maximum of two phone systems at one time. For information on and instructions for integrating Cisco Unity with two phone systems, refer to the *Dual Phone System Integration Guide* at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod\\_configuration\\_guide09186a0080211b2e.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod_configuration_guide09186a0080211b2e.html).
- Cisco Unity 4.2 and later can be integrated with two or more phone systems at one time. For information on the maximum supported combinations and instructions for integrating Cisco Unity with multiple phone systems, refer to the *Multiple Phone System Integration Guide* at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod\\_configuration\\_guide09186a00806192a3.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod_configuration_guide09186a00806192a3.html).

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

Before programming the phone system, you need to plan how the voice messaging ports will be used by Cisco Unity. 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), to make AMIS deliveries, and to make telephone record and playback (TRAP) connections.

The following table describes the voice messaging port settings in Cisco Unity that can be set in UTIM, and that are displayed as read-only text on the System > Ports page of the Cisco Unity Administrator.

**Table 1**      **Settings for the Voice Messaging Ports**

Field	Considerations
Extension	Enter the extension for the port as assigned on the phone system.
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.
Answer Calls	Check this check box to designate the port for answering calls. These calls can be incoming calls from unidentified callers or from subscribers.
Message Notification	Check this check box to designate the port for notifying subscribers of messages. Assign Message Notification to the least busy ports.
Dialout MWI <i>(not used by serial or SMDI integrations)</i>	Check this check box to designate the port for turning MWIs on and off. Assign Dialout MWI to the least busy ports.
AMIS Delivery <i>(available with the AMIS licensed feature only)</i>	Check this check box to designate the port for making outbound AMIS calls to deliver voice messages from Cisco Unity subscribers to users on another voice messaging system. Cisco Unity supports the Audio Messaging Interchange Specification (AMIS) protocol, which provides an analog mechanism for transferring voice messages between different voice messaging systems.  This setting affects outbound AMIS calls only. All ports are used for incoming AMIS calls. Because the transmission of outgoing AMIS messages may tie up voice ports for long periods of time, you may want to adjust the schedule on the Network > AMIS > Schedule page so that outgoing AMIS calls are placed during closed hours or at times when Cisco Unity is not processing many calls.
TRAP Connection	Check this check box so that subscribers can use the phone as a recording and playback device in Cisco Unity web applications and e-mail clients. Assign TRAP Connection to the least busy ports.

## The Number of Voice Messaging Ports to Install

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

- The number of calls Cisco Unity will answer when call traffic is at its peak.
- The expected length of each message that callers will record and that subscribers will listen to.
- The number of subscribers.
- The number of 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 AMIS delivery calls.
- The number of TRAP connections needed when call traffic is at its peak. (TRAP connections are used by Cisco Unity web applications to play back and record over the phone.)
- The number of calls that will use the automated attendant and call handlers when call traffic is at its peak.

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

#### **The Number of Voice Messaging Ports That Will Answer Calls**

The calls that the voice messaging ports answer can be incoming calls from unidentified callers or from subscribers. 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 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 subscribers by phone, pager, or e-mail of messages that have arrived.
- Turn MWIs on and off for subscriber extensions.
- Make outbound AMIS calls to deliver voice messages from Cisco Unity subscribers to users on another voice messaging system. (This action is available only with the AMIS licensed feature.)
- Make a TRAP connection so that subscribers can use the phone as a recording and playback device in Cisco Unity web applications.

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 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.

#### **Preparing for Programming the Phone System**

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

# Programming the Ericsson MD-110 Phone System

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

Do the following procedure.

## To Program the Ericsson MD-110 Phone System

- Step 1** Check the software version on the phone system and the country variant that is configured. This information may be useful for troubleshooting problems with the integration. For example:
- ```
<cadap;                               ==> Check the software version and Country variant
CALENDAR DATA

IDENTITY=ACM1
VERSION=CXP1010101/4/TSWSP02/R3A ==> 01 == Standard Application
```
- Step 2** Configure the analog extensions for the voice messaging ports similarly to the following example:
- ```
<excap:dir=1063&1064;                 ==> print analog extension
EXTENSION CATEGORY FIELDS

DIR      TRAF      SERV      CDIV      ROC      TRM      ADC      BSEC
1063     00151515  0201120600 000151000 0000001  0        010001701 0
1064     00151515  0201120600 000151000 0000001  0        010001701 0
```
- Step 3** Configure the hunt group number for the voice messaging ports on the PIMG units similarly to the following example:
- ```
<vmfup:grp=all;                       ==> print VM HuntGroup number
VOICE MAIL GROUP DATA
GRP      IFCIND
4500     1
```
- Step 4** Associate the voice messaging port extensions with the hunt group number similarly to the following example:
- ```
<vmfup:dir=all;                       ==> print VM port DN associated with the HuntGroup number
VOICE MAIL PORT DATA
DIR      PORT      IFCIND
1063     1063      1
1064     1064      1
```
- Step 5** Configure the voice mail function information similarly to the following:
- ```
<vmfup:ifcind=1;                      ==> print VM function information
VOICE MAIL FUNCTION DATA
IFCIND   VMF      POFMT
1        EXTN3   4
```
- Step 6** Define the I/O device interface for voice mail similarly to the following:
- ```
<ioddp;                               ==> print I/O Device interface
I/O DEVICE DATA
NODE      IODEV/SUBFS      BPOS/EQU      I/O-BUS TYPE/USAGE      STATUS      AUTH
SYN      V-MAIL      001-0-60-3   -      OUT      IN SERVICE
```
- Step 7** Configure the I/O device function for voice mail similarly to the following example:
- ```
<iofd;                                 ==> print I/O Device Function
I/O FUNCTIONS DEFINITION
IODEV      TRD      NDC      CALLS WAIT DELAY SUPER
```

```
V-MAIL          BOTH          2      30      5      YES
```

**Step 8** Configure the serial connection parameters for voice mail similarly to the following example:

```
<ioifp;                               ==> print I/O Device serial interface parameters VM
I/O INTERFACE CHARACTERISTICS
IODEV          IFACE  BAUDR   WORDL   PARITY  STPBIT  PROC
SYSTEMINAL    V24    ALL    8       NONE   1       ECHO
V-MAIL        V24    9600  8       NONE   1       ECHO
```

**Step 9** Configure MWIs to be sent over the serial cable similarly to the following example:

```
<icmwp:sid=1;                          ==> print MWI ext using on serial interface
INFORMATION COMPUTER MESSAGE WAITING DATA
SID  DTXT          KFCN  DIG
1    4500          MWC   4500
```

**Step 10** Configure the filler information similarly to the following example:

```
<ICFUP:ifcind=1;                       ==> print the filler information
INFORMATION COMPUTER COMMON FUNCTIONS DATA

MESSAGE WAITING FUNCTIONALITY IS ALL

INFORMATION COMPUTER EQUIPMENT DATA

IFCIND  IODEV          EQU          RATE  DFMT  UPDFCN  PARITY  CCHECK
1       V-MAIL          FILLER=48  ICEXG=NONE  USER=NONE
```

**Step 11** Configure the hookflash timer similarly to the following example:

```
<aspap:parnum=253;                      ==> print Hookflash timer
APPLICATION SYSTEM PARAMETERS
PARNUM  PARVAL
253    110
```

**Step 12** Configure the on-hook timer similarly to the following example:

```
<aspap:parnum=252;                      ==> print On-hook timer
APPLICATION SYSTEM PARAMETERS
PARNUM  PARVAL
252    225
```

**Step 13** Program each phone to forward calls to the hunt group number (defined in [Step 3](#)), based on one of the Cisco Unity call transfer types shown in [Table 2](#).

**Table 2** Call Transfer Types

| Transfer Type                        | Usage                                                                                                                                                                  |
|--------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Release transfer<br>(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 phone to forward calls to the pilot number only when the call is not answered. 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 Ericsson MD-110 phone system.

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

- The phone system is connected to the PIMG units by using analog lines and the applicable RS-232 serial cable.
- 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 Analog PIMG Units

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- 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.

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- Step 2** On the Cisco Unity PIMG Software Download page, click the most recent version of the firmware for analog PIMG units.
- Step 3** On the Details page, click **Next**.
- Step 4** On the Document page, click **Accept**.
- Step 5** In the Enter Network Password dialog box, enter your user name and password, then click **OK**.
- Step 6** In the File Download dialog box, click **Save**.
- Step 7** In the Save As dialog box, browse to the Windows workstation that will have access the PIMG units, browse to a directory where you want to save the file, and click **Save**.
- Step 8** In the Download Complete dialog box, click **Open**. The window for extracting the PIMG firmware update files appears.
- Step 9** Click **Extract**.
- Step 10** In the Extract dialog box, browse to the directory where you want the extracted files, and click **Extract**.
- Step 11** Close the window for the extracting application.
- 

## To Set Up the Analog PIMG Units

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- Step 1** On the Windows workstation, add a temporary route to enable access to the PIMG units.
- a. On the Windows Start menu, click **Run**.
  - b. Enter **cmd**, and press **Enter**. The Command Prompt window appears.
  - c. At the command prompt, enter **route add 10.12.13.74 <IP Address of Workstation>**, and press **Enter**.

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

d. Close the Command Prompt window.

**Step 2** Connect a PIMG unit to the network.

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

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

**Table 3** System Login Page Settings

| Field    | Setting   |
|----------|-----------|
| Username | admin     |
| Password | IpodAdmin |

**Step 5** Click **Log On**.

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

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

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

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

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

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



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

**Step 12** Repeat [Step 6](#) through [Step 11](#) for the file Run\_<xx>.dsp.

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

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

**Step 15** In the Choose File dialog box, browse to the file Ls\_<xx>.fsh.

**Step 16** Click **Ls\_<xx>.fsh**, 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 4** 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 5 Password Page Settings**

| Field            | Setting                                                  |
|------------------|----------------------------------------------------------|
| Old Password     | <b>IpodAdmin</b><br>(This setting is case sensitive.)    |
| New Password     | <your new password><br>(This setting is case sensitive.) |
| Confirm Password | <your new password><br>(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 6 System Page Settings**

| Field                 | Setting     |
|-----------------------|-------------|
| Operating Mode        | <b>SIP</b>  |
| Telephony Switch Type | <b>None</b> |
| PCM Coding            | <b>uLaw</b> |

- Step 27** Click **Apply Changes**.
- Step 28** On the Configure menu, click **Serial Protocol**.
- Step 29** On the Serial Protocol page, enter the following settings.

**Table 7 Serial Protocol Page Settings**

| Field                     | Setting                                                                                                                                                                                                                                                                                                                                                                                                                                   |
|---------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Serial Mode               | Click the applicable setting: <ul style="list-style-type: none"> <li>• <b>Master</b>—This PIMG unit is connected to the data link serial cable from the phone system. There can be only one master PIMG unit in a phone system integration.</li> <li>• <b>Slave</b>—This PIMG unit is not connected to the data link serial cable from the phone system. There can be multiple slave PIMG units in a phone system integration.</li> </ul> |
| Serial Interface Protocol | <b>MD110</b>                                                                                                                                                                                                                                                                                                                                                                                                                              |
| Cpid Len                  | Click the applicable setting. Typically, the settings are 7 or 10.                                                                                                                                                                                                                                                                                                                                                                        |
| Cpid Padding String       | Leave this field blank or enter the applicable string.                                                                                                                                                                                                                                                                                                                                                                                    |

**Table 7** Serial Protocol Page Settings (continued)

| Field                       | Setting                                                                                                                                                                                                                     |
|-----------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Voice Mail Port Len         | <b>2</b>                                                                                                                                                                                                                    |
| System Number               | <b>1</b>                                                                                                                                                                                                                    |
| MWI Response Timeout        | <b>2000</b>                                                                                                                                                                                                                 |
| IP Address of Serial Server | If the PIMG unit is the master, leave this field blank.<br>If the PIMG unit is a slave, enter the IP address of the master PIMG unit (the PIMG unit that is connected to the data link serial cable from the phone system). |
| Serial Cpid Expiration      | <b>2000</b>                                                                                                                                                                                                                 |
| Logical Extension Number    | Enter the extension number for each port on the PIMG unit.                                                                                                                                                                  |

**Step 30** Click **Apply Changes**.

**Step 31** On the Configure menu, click **Gateway**.

**Step 32** On the Gateway page, click the **Gateway Routing** tab.

**Step 33** On the Gateway Routing tab, enter the following settings.

**Table 8** Gateway Routing Tab Settings

| Field                   | Setting                                                                                                                                                                                                                            |
|-------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Fault Tolerance Enabled | (Cisco Unity without failover) <b>No</b><br>(Cisco Unity with failover configured) <b>Yes</b>                                                                                                                                      |
| Load Balancing Enabled  | <b>No</b>                                                                                                                                                                                                                          |
| VoIP Endpoint ID:<br>1  | (Cisco Unity without failover) <the IP address of the Cisco Unity server><br>(Cisco Unity with failover configured) <the name the primary Cisco Unity server; this setting must match the Contact Line Name field setting in UTIM> |
| VoIP Endpoint ID:<br>2  | (Cisco Unity without failover) <blank><br>(Cisco Unity with failover configured) <the name the secondary Cisco Unity server; this setting must match the Contact Line Name field setting in UTIM>                                  |

**Step 34** Click **Apply Changes**.


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

**Step 36** On the Gateway Advanced tab, enter the following settings.




**Table 9** Gateway Advanced Tab Settings

| Field                               | Setting         |
|-------------------------------------|-----------------|
| Call Connect Mode                   | <b>OnAnswer</b> |
| Destination for Unroutable IP Calls | <blank>         |

**Table 9** Gateway Advanced Tab Settings (continued)

| Field                                       | Setting                                                                                                                                                                                                                                                                                                                                                                            |
|---------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Destination for Unroutable PBX Calls        | <blank>                                                                                                                                                                                                                                                                                                                                                                            |
| Monitor Call Connections                    | <b>No</b>                                                                                                                                                                                                                                                                                                                                                                          |
| Maximum Call Party Delay (msecs)            | <b>2000</b>                                                                                                                                                                                                                                                                                                                                                                        |
| Dial Digit on Time (msecs)                  | <b>100</b>                                                                                                                                                                                                                                                                                                                                                                         |
| Dial Inter-Digit Time (msecs)               | <b>100</b>                                                                                                                                                                                                                                                                                                                                                                         |
| Dial Pause Time (msecs)                     | <b>2000</b>                                                                                                                                                                                                                                                                                                                                                                        |
| Turn MWI On FAC                             | <blank>                                                                                                                                                                                                                                                                                                                                                                            |
| Turn MWI Off FAC                            | <blank>                                                                                                                                                                                                                                                                                                                                                                            |
| Outbound Call Connect Timeout (msecs)       | <b>10000</b>                                                                                                                                                                                                                                                                                                                                                                       |
| Wait for Ringback/Connect on Blind Transfer | <b>Yes</b>                                                                                                                                                                                                                                                                                                                                                                         |
| Hunt Group Extension                        | <the pilot number for the Cisco Unity voice messaging ports>                                                                                                                                                                                                                                                                                                                       |
| Audio Compression                           | Click the preferred codec for audio compression: <ul style="list-style-type: none"> <li>• <b>G.711 Only</b></li> <li>• <b>G.729 A Preferred</b></li> </ul>                                                                                                                                                                                                                         |
| RTP Digit Relay Mode                        | <b>RFC2833</b>                                                                                                                                                                                                                                                                                                                                                                     |
| Signaling Digit Relay Mode                  | <b>Off</b>                                                                                                                                                                                                                                                                                                                                                                         |
| Voice Activity Detection                    | <b>Off</b>                                                                                                                                                                                                                                                                                                                                                                         |
| Frame Size                                  | Click the applicable setting: <ul style="list-style-type: none"> <li>• G.711—<b>20</b></li> <li>• G.729a—<b>10</b></li> </ul> <div style="text-align: center;"></div> <p><b>Caution</b> Failure to use the correct setting will result in recorded messages containing nothing but silence.</p> |

**Table 9** Gateway Advanced Tab Settings (continued)

| Field                 | Setting                                                                                                                                                                                                                                                                                                                      |
|-----------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Frames Per Packet     | <p>Click the applicable setting:</p> <ul style="list-style-type: none"> <li>G.711—1</li> <li>G.729a—2</li> </ul> <p> <b>Caution</b> 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) <b>0</b></p> <p>(PIMG units connect to a WAN) <b>104</b></p> <p> <b>Note</b> For details on the setting for a LAN, see the caveat <a href="#">CSCsb96387</a>.</p>                                     |
| RTP QOS Byte          | <p>(PIMG units connect only to a LAN) <b>0</b></p> <p>(PIMG units connect to a WAN) <b>184</b></p> <p> <b>Note</b> For details on the setting for a LAN, see the caveat <a href="#">CSCsb96387</a>.</p>                                     |
| SNMP Traps Enabled    | <b>No</b>                                                                                                                                                                                                                                                                                                                    |
| E-mail Alarms Enabled | <b>No</b>                                                                                                                                                                                                                                                                                                                    |

**Step 37** Click **Apply Changes**.

**Step 38** Click the **Gateway Capabilities** tab.

**Step 39** 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 10** 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 send message notifications). |
| MWIs-Only                          | The port will dial out only (for example, to send message notifications) and will not answer incoming calls. |
| Both                               | The port will answer incoming calls and will also dial out (for example, to send message notifications).     |



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 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 40** Click **Apply Changes**.
- Step 41** On the Configure menu, click **SIP**.
- Step 42** On the SIP page, enter the following settings.

**Table 11**      **SIP Page Settings**

| <b>Field</b>                 | <b>Setting</b>                                                                                                                                                                                                                                                                                                                                                                                                       |
|------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Host and Domain Name         | <the domain name of the PIMG unit>                                                                                                                                                                                                                                                                                                                                                                                   |
| Server Port                  | <b>5060</b>                                                                                                                                                                                                                                                                                                                                                                                                          |
| Primary Proxy Server Address | <i>(Cisco Unity without failover)</i> <the IP address of the Cisco Unity server><br><i>(Cisco Unity with failover configured)</i> <the IP address of the primary Cisco Unity server>                                                                                                                                                                                                                                 |
| Primary Proxy Server Port    | <b>5060</b><br><br>(When you configure more than one PIMG unit, increase this setting by 1 for each successive unit. For example, unit 2 will be 5061, unit 3 will be 5062, and so on. For failover, this setting must match the setting for the Backup Proxy Server Port field.)                                                                                                                                    |
| Backup Proxy Server Address  | <i>(Cisco Unity without failover)</i> Not applicable; leave the default setting.<br><i>(Cisco Unity with failover configured)</i> <the IP address of the secondary Cisco Unity server>                                                                                                                                                                                                                               |
| Backup Proxy Server Port     | <i>(Cisco Unity without failover)</i> Not applicable; leave the default setting.<br><i>(Cisco Unity with failover configured)</i> <b>5060</b><br><br>(When you configure more than one PIMG unit, increase this setting by 1 for each successive unit. For example, unit 2 will be 5061, unit 3 will be 5062, and so on. For failover, this setting must match the setting for the Primary Proxy Server Port field.) |
| Proxy Query Interval         | <b>10</b>                                                                                                                                                                                                                                                                                                                                                                                                            |
| T1 Time                      | <b>400</b>                                                                                                                                                                                                                                                                                                                                                                                                           |
| T2 Time                      | <b>3000</b>                                                                                                                                                                                                                                                                                                                                                                                                          |

- Step 43** Click **Apply Changes**.
- Step 44** On the Configure menu, click **IP**.
- Step 45** On the IP page, enter the following settings.

**Table 12** IP Page Settings

| Field                           | Setting                                                                                                                                         |
|---------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------|
| Client IP Address               | <the new IP address you want to use for the PIMG unit><br>(This is the IP address that you will enter in UTIM 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                   | <b>No</b>                                                                                                                                       |

- Step 46** Click **Apply Changes**.
- Step 47** On the Configure menu, click **Tones**.
- Step 48** 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 49** 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 50** 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 51** On the Tones page, in the Destination Address field for Busy Tone, enter the extension that you dialed in [Step 50c](#). from the third phone.
- Step 52** 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 53** On the Tones page, in the Destination Address field for Error/Reorder Tone, enter the extension that you dialed in [Step 52a](#).
- Step 54** 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

- b. Confirm that you hear the ringback tone.
- c. Hang up the phone.

**Step 55** On the Tones page, in the Destination Address field for Ringback Tone, enter the extension that you dialed in [Step 54a](#).

**Step 56** Click **Learn**.



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

**Step 57** When the process is complete, check the check box for each newly learned tone and click **Apply**.

**Step 58** Hang up the phones that you used in [Step 50](#).

**Step 59** On the Configure menu, click **Restart**.

**Step 60** On the Restart page, click **Restart Unit Now**.

**Step 61** When the PIMG unit has restarted, in the View menu, click **Refresh**.

**Step 62** Repeat [Step 2](#) through [Step 61](#) on all remaining PIMG units.

## Creating a New Integration with the Ericsson MD-110 Phone System

After ensuring that the Ericsson MD-110 phone system and the Cisco Unity server are ready for the integration, do the following procedures to set up the integration and to enter the port settings.

### To Create an Integration

- Step 1** If UTIM is not already open, on the Windows Start menu of the Cisco Unity server, click **Programs > Cisco Unity > Manage Integrations**. UTIM appears.
- Step 2** In the left pane of the UTIM window, click **Cisco Unity Server**.
- Step 3** On the Integration menu of the UTIM window, click **New**. The Telephony Integration Setup Wizard appears.
- Step 4** On the Welcome page, click **Circuit-switched via Intel PIMG** and click **Next**.
- Step 5** On the Name the Phone System Integration page, accept the default name or enter the phone system name to identify this integration, then click **Next**.
- Step 6** On the Enter PIMG Settings page, click **Add**.
- Step 7** In the Add PIMG dialog box, enter the following settings, then click **OK**.

**Table 13** Settings for the Add PIMG Dialog Box

| Field        | Setting                                                                    |
|--------------|----------------------------------------------------------------------------|
| Display Name | <accept the default name or enter another name to identify this PIMG unit> |
| IP Address   | <the IP address of this PIMG unit>                                         |

**Table 13** Settings for the Add PIMG Dialog Box (continued)

| Field                         | Setting                                                                                                                                               |
|-------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------|
| SIP Port                      | 5060                                                                                                                                                  |
| Phone Lines (Ports) Connected | 8<br><if you want to use fewer than eight voice messaging ports, enter the number of ports (or phone lines) that you want to use with this PIMG unit> |

**Step 8** Repeat [Step 6](#) and [Step 7](#) for each remaining PIMG unit that you are connecting to the Cisco Unity server.

You can press the following buttons to modify, delete, or verify the PIMG units that you are connecting to the Cisco Unity server.

**Table 14** Buttons on the Enter PIMG Settings Page

| Button       | Action                                                                                                |
|--------------|-------------------------------------------------------------------------------------------------------|
| Add          | Displays the Add PIMG dialog box to add another PIMG unit to the integration.                         |
| Modify       | Displays the Modify PIMG dialog box so that you can modify the settings of the selected PIMG unit.    |
| Delete       | Deletes the selected PIMG unit from the integration.                                                  |
| Ping Servers | Confirms that the IP address is correct for all PIMG units.                                           |
| Licensing    | Displays a list of the licensed, used, and available voice messaging ports on the Cisco Unity server. |

**Step 9** On the Enter PIMG Settings page, click **Next**.

**Step 10** On the PIMG Integration with the PBX page, click **Yes**.

**Step 11** In the This PIMG Is the Serial Master field, click the name of the PIMG unit that is connected to the serial cable from the phone system, then click **Next**.

**Step 12** On the Configure Cisco Unity SIP Settings page, enter the following settings, then click **Next**.

**Table 15** Settings for the Configure Cisco Unity SIP Settings Page

| Field                | Setting                                                                                                                                                                                                                                                                                                                                                                                                                                                             |
|----------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Contact Line Name    | <i>(Cisco Unity without failover)</i> <the voice messaging line name that subscribers use to contact Cisco Unity and that Cisco Unity will use to register with the PIMG units><br><br><i>(Cisco Unity with failover configured)</i> <the name the primary Cisco Unity server; this setting must match the Port X Endpoint parameter settings in the PIMG administration; this setting must be the same for both the primary and the secondary Cisco Unity servers> |
| Cisco Unity SIP Port | <the IP port on Cisco Unity that callers and the SIP server use to connect to voice mail; we recommend using the default setting>                                                                                                                                                                                                                                                                                                                                   |
| Preferred Codec      | <the codec Cisco Unity will first attempt to use on outgoing calls>                                                                                                                                                                                                                                                                                                                                                                                                 |

**Step 13** If other integrations already exist, the Enter Trunk Access Code page appears. Enter the extra digits that Cisco Unity must use to transfer calls through the gateway to extensions on the other phone systems with which it is integrated. Then click **Next**.

**Step 14** (*Cisco Unity 4.2 and later only*) On the Reassign Subscribers page, any subscribers whose phone system integration has been deleted and who are not currently assigned to a phone system integration will appear in the list.

If no subscribers appear in the list, click **Next** and continue to [Step 15](#).

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

**Table 16** *Selection Controls for the Reassign Subscribers Page*

| Selection Control | Effect                                                                                                                                                                                                                                                                                              |
|-------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Check All         | Checks the check boxes for all subscribers in the list.                                                                                                                                                                                                                                             |
| Uncheck All       | Unchecks the check boxes for all subscribers in the list.                                                                                                                                                                                                                                           |
| Toggle Selected   | For the subscribers highlighted in the list, toggles between checking and unchecking the check boxes.<br><br>If some highlighted subscriber check boxes are checked and others are unchecked, clicking this button will check all the check boxes. Clicking again will uncheck all the check boxes. |

**Step 15** (*Cisco Unity 4.2 and later only*) On the Reassign Call Handlers page, any call handlers whose phone system integration has been deleted and that are not currently assigned to a phone system integration will appear in the list.

If no call handlers appear in the list, click **Next** and continue to [Step 16](#).

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

**Table 17** *Selection Controls for the Reassign Call Handlers Page*

| Selection Control | Effect                                                                                                                                                                                                                                                                                                  |
|-------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Check All         | Checks the check boxes for all call handlers in the list.                                                                                                                                                                                                                                               |
| Uncheck All       | Unchecks the check boxes for all call handlers in the list.                                                                                                                                                                                                                                             |
| Toggle Selected   | For the call handlers highlighted in the list, toggles between checking and unchecking the check boxes.<br><br>If some highlighted call handler check boxes are checked and others are unchecked, clicking this button will check all the check boxes. Clicking again will uncheck all the check boxes. |

**Step 16** On the Completing page, verify the settings you entered, then click **Finish**.

**Step 17** At the prompt to restart the Cisco Unity services, click **Yes**. The Cisco Unity services restart.

Alternatively, you can restart the Cisco Unity services in UTIM on the Tools menu by clicking **Restart Cisco Unity**.

### To Enter the Voice Messaging Port Settings for the Integration

- Step 1** After the Cisco Unity services restart, on the View menu, click **Refresh**.
- Step 2** In the left pane of the UTIM window, expand the phone system integration that you are creating.
- Step 3** In the left pane, click the name of the first PIMG unit.
- Step 4** In the right pane, click the **Ports** tab.
- Step 5** Enter the settings shown in [Table 18](#) for the voice messaging ports.

For best performance, use the first voice messaging ports for incoming calls and the last ports to dial out. This helps minimize the possibility of a collision, in which an incoming call arrives on a port at the same time that Cisco Unity takes the port off-hook to dial out.



#### Caution

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

**Table 18** Settings for the Voice Messaging Ports

| Field                                                                   | Considerations                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                       |
|-------------------------------------------------------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Extension                                                               | Enter the extension for the port as assigned on the phone system.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                    |
| Enabled                                                                 | Check this check box to enable the port. The port is enabled during normal operation.<br>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.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                     |
| Answer Calls                                                            | Check this check box to designate the port for answering calls. These calls can be incoming calls from unidentified callers or from subscribers.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                     |
| Message Notification                                                    | Check this check box to designate the port for notifying subscribers of messages. Assign Message Notification to the least busy ports.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                               |
| Dialout MWI<br><i>(not used by serial or SMDI integrations)</i>         | Check this check box to designate the port for turning MWIs on and off. Assign Dialout MWI to the least busy ports.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                  |
| AMIS Delivery<br><i>(available with the AMIS licensed feature only)</i> | Check this check box to designate the port for making outbound AMIS calls to deliver voice messages from Cisco Unity subscribers to users on another voice messaging system. Cisco Unity supports the Audio Messaging Interchange Specification (AMIS) protocol, which provides an analog mechanism for transferring voice messages between different voice messaging systems.<br><br>This setting affects outbound AMIS calls only. All ports are used for incoming AMIS calls. Because the transmission of outgoing AMIS messages may tie up voice ports for long periods of time, you may want to adjust the schedule on the Network > AMIS > Schedule page so that outgoing AMIS calls are placed during closed hours or at times when Cisco Unity is not processing many calls. |
| TRAP Connection                                                         | Check this check box so that subscribers can use the phone as a recording and playback device in Cisco Unity web applications and e-mail clients. Assign TRAP Connection to the least busy ports.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                    |

- Step 6** Click **Save**.
- Step 7** Click the **SIP Info** tab.
- Step 8** (*Cisco Unity 4.2 and later*) Uncheck the **Register with SIP Server** check box and click **Save**.  
(*Cisco Unity 4.0 and 4.1*) Uncheck the **Register with Proxy Server** check box and click **Save**.
- Step 9** At the prompt to restart the Cisco Unity services, click **No**.
- Step 10** Repeat [Step 3](#) through [Step 9](#) for all remaining PIMG units.
- Step 11** In the left pane, click **Properties** for the phone system.
- Step 12** In the right pane, click the **PIMG** tab.
- Step 13** Under Set Messaging Waiting Indicators (MWI) Using This Method, confirm that the Out-of-Band - SIP NOTIFY option is selected.
- Step 14** Click **Save**.
- Step 15** At the prompt to restart the Cisco Unity services, click **Yes**.
- Step 16** After the Cisco Unity services restart, exit UTIM.

**Caution**

Do not edit the phone configuration file (also known as the switch ini file) to customize this integration. If you change the settings in this file, the integration may not function correctly.

---

## Testing the Integration

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

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

- The installation guide for the phone system.
- *Cisco Unity Troubleshooting Guide*, available at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod\\_troubleshooting\\_guides\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod_troubleshooting_guides_list.html).
- The setup information earlier in this guide.

### To Set Up the Test Configuration

---

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

**Caution**

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

---

- Step 3** In the Cisco Unity Administrator, create a test subscriber to use for testing by doing the applicable substeps below.

If your message store is Microsoft Exchange, do the following:

- a. In the Cisco Unity Administrator, go to the **Subscribers > Subscribers > Profile** page.
- b. Click the **Add** icon.
- c. Select **New Exchange Subscriber**.
- d. On the Add Subscriber page, enter the applicable information.
- e. Click **Add**.

If your message store is IBM Lotus Domino, do the following:

- a. In the Cisco Unity Administrator, go to the **Subscribers > Subscribers > Profile** page.
- b. Click the **Add** icon.
- c. Click **Notes**.
- d. In the Address Book list, confirm that the address book listed is the one that contains the user data that you want to import.  
If the address book that you want to use is not listed, go to the **System > Configuration > Subscriber Address Books** page and add a different address book.
- e. In the Find Domino Person By list, indicate whether to search by short name, first name, or last name.
- f. Enter the applicable short name or name. You also can enter \* to display a list of all users, or enter one or more characters followed by \* to narrow your search.
- g. Click **Find**.
- h. On the list of matches, click the name of the user to import.
- i. On the Add Subscriber page, enter the applicable information.
- j. Click **Add**.

**Step 4** In the Extension field, enter the extension of Phone 1.

**Step 5** In the Active Schedule field, click **All Hours - All Days**.

**Step 6** Click the **Save** icon.

**Step 7** In the navigation bar, click **Call Transfer** to go to the Subscribers > Subscribers > Call Transfer page for the test subscriber.

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

**Step 8** Under Transfer Incoming Calls, click **Yes, Ring Subscriber’s Extension**, and confirm that the extension number is for Phone 1.

**Step 9** Under Transfer Type, click **Release to Switch**.

**Step 10** Click the **Save** icon.

**Step 11** In the navigation bar, click **Messages** to go to the Subscribers > Subscribers > Messages page for the test subscriber.

**Step 12** Under Message Waiting Indicators (MWIs), check **Use MWI for Message Notification**.

**Step 13** In the Extension field, enter **x**.

**Step 14** Click the **Save** icon.

**Step 15** Open the Status Monitor by doing one of the following:

- In Internet Explorer, go to **http://<Cisco Unity server name>/web/sm**.

- Double-click the desktop shortcut to the Status Monitor.
  - In the status bar next to the clock, right-click the Cisco Unity tray icon and click **Status Monitor**.
- 

### To Test an External Call with Release Transfer

---

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

### To Test Listening to Messages

---

- Step 1** From Phone 1, enter the internal pilot number for Cisco Unity.
- Step 2** When asked for your password, enter the default password. Hearing the request for your password means that the phone system sent the necessary call information to Cisco Unity, which correctly interpreted the information.
- Step 3** Confirm that you hear the recorded voice name for the test subscriber (if you did not record a voice name for the test subscriber, you will hear the extension number for Phone 1). Hearing the voice name means that Cisco Unity correctly identified the subscriber by the extension.
- Step 4** When asked whether you want to listen to your message, press **1**.
- Step 5** After listening to the message, press **3** to delete the message.
- Step 6** Confirm that the MWI on Phone 1 is deactivated. The deactivated MWI means that the phone system and Cisco Unity are successfully integrated for turning off MWIs.
- Step 7** Hang up Phone 1.

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

#### To Set Up Supervised Transfer on Cisco Unity

---

- Step 1** In the Cisco Unity Administrator, go to the **Subscribers > Subscribers > Call Transfer** page.
- If the name of the test subscriber is not displayed, click the **Find** icon (the magnifying glass) in the title bar, then click **Find**, and select the name of the test subscriber in the list that appears.
- For more information on transfer settings, refer to the “Subscriber Template Call Transfer Settings” section in the Cisco Unity Administrator Help.
- Step 2** Under Transfer Type, click **Supervise Transfer**.
- Step 3** Set the Rings to Wait For field to **3**.
- Step 4** Click the **Save** icon.
- 

#### To Test Supervised Transfer

---

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

#### To Delete the Test Subscriber

---

- Step 1** In the Cisco Unity Administrator, go to the **Subscribers > Subscribers > Profile** page.
- If the name of the test subscriber is not displayed, click the **Find** icon (the magnifying glass) in the title bar, then click **Find**, and select the name of the test subscriber in the list that appears.
- Step 2** In the title bar, click the **Delete Subscriber** icon (the X).
- Step 3** Click **Delete**.
-

# Integrating a Secondary Server for Cisco Unity Failover

For Cisco Unity 4.1 and later, the Cisco Unity failover feature enables a secondary server to provide voice messaging services when the primary server becomes inactive. For information on installing a secondary server for failover, refer to the applicable Cisco Unity installation guide, available at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod\\_installation\\_guides\\_list.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod_installation_guides_list.html).

The Cisco Unity failover feature is not available when this phone system is integrated with Cisco Unity 4.0 through PIMG units.

For information on failover, refer to the *Cisco Unity Failover Configuration and Administration Guide*. The Domino version of the guide is available at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products\\_feature\\_guide\\_book09186a00803f70f3.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products_feature_guide_book09186a00803f70f3.html). The Exchange version of the guide is available at [http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products\\_feature\\_guide\\_book09186a00801b9241.html](http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products_feature_guide_book09186a00801b9241.html).

## Requirements

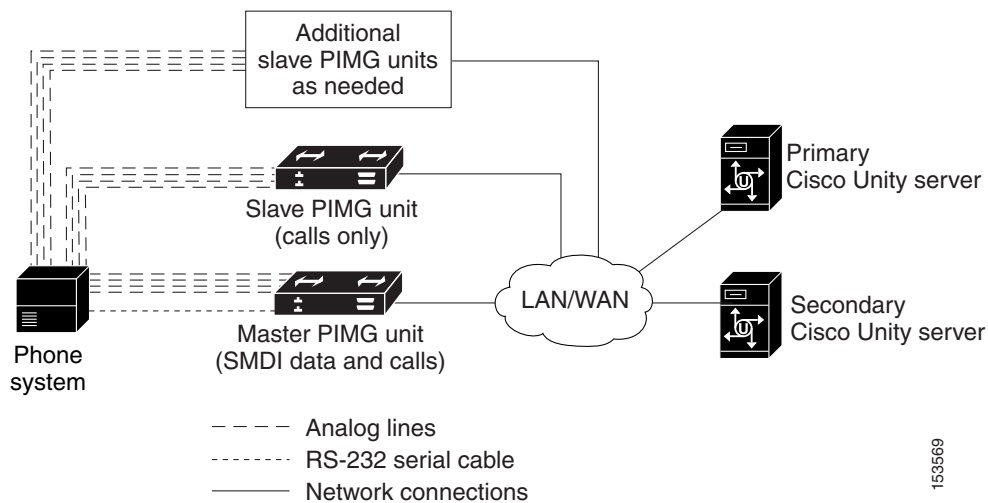
The following components are required to integrate a secondary server:

- One secondary server for each primary server installed and ready for the integration, as described in the applicable Cisco Unity installation guide and earlier in this integration guide.
- A license that enables failover.

## Integration Description

The Ericsson MD-110 phone system sends call information and MWI requests through the data link, which consists of an RS-232 serial cable between the phone system and the master PIMG unit. Voice connections are sent through the analog lines between the phone system and the PIMG units. The PIMG units communicate with the primary and secondary servers through the LAN or WAN by using Session Initiation Protocol (SIP). [Figure 2](#) shows the required connections.

**Figure 2** Connections Between the Phone System and the Cisco Unity Servers



The primary and secondary servers act in the following manner:

- When the primary server is operating normally, the secondary server is inactive.
- When the primary server becomes inactive, the secondary server becomes active.
- When the primary server becomes active again, the secondary server becomes inactive.

## Setting Up the Secondary Server for Failover

Do the following procedure to integrate the secondary server.

### To Set Up the Secondary Server for Failover

- Step 1** Install a secondary server with the same configuration as the primary server. For installation instructions, refer to the applicable Cisco Unity installation guide.
- Step 2** On the Windows Start menu of the secondary server, click **Programs > Cisco Unity > Manage Integrations**. The UTIM window appears.
- Step 3** On the Integration menu of the UTIM window, click **New**. The Telephony Integration Setup Wizard appears.
- Step 4** Enter the settings to match the integration settings on the primary server.




**Note** We recommend not reassigning any unassigned subscribers and call handlers to the new integration, if you are asked by the wizard. Failover replication will automatically assign the correct integration.

- Step 5** At the prompt to restart the Cisco Unity services, click **Yes**.



**Note** When restarting the Cisco Unity services, use the UTIM prompt instead of the Cisco Unity icon in the Windows taskbar. The taskbar icon does not restart all of the Cisco Unity services.

- Step 6** After Cisco Unity restarts, on the Windows Start menu of the Cisco Unity server, click **Programs > Cisco Unity > Manage Integrations**. UTIM appears.
- Step 7** In the left pane of the UTIM window, click the phone system integration that you created in [Step 3](#).
- Step 8** For Cisco Unity 4.0 and 4.1, continue to [Step 9](#).  
For Cisco Unity 4.2 and later, do the following substeps.
- In the right pane, click **Properties**.
  - On the Integration tab, compare the setting of the Integration ID field for the secondary server to the setting of the Integration ID field for the primary server.
  - If the integration IDs of the phone system on the primary and secondary server are the same, continue to [Step 9](#).  
If the integration IDs of the phone system on the primary and secondary servers are different, on the secondary server, click **Modify Integration ID**.
  - When cautioned that subscribers associated with the current Integration ID setting will not be automatically associated with the new Integration ID setting, click **OK**.
  - In the Modify Integration ID dialog box, in the Enter New Integration ID field, enter the Integration ID setting for the phone system on the primary server and click **OK**.
  - Click **Save**.
  - At the prompt to restart the Cisco Unity services, click **Yes**.
  - In the left pane, click the phone system integration that you created in [Step 3](#).
- Step 9** In the right pane, click the **Ports** tab.
- Step 10** Enter the port settings to match the port settings on the primary server.
-  **Caution** In programming the phone system, do not send calls to voice messaging ports in Cisco Unity that cannot answer calls (voice messaging ports that are not set to Answer Calls). For example, if a voice messaging port is set only to Message Notification, do not send calls to it.
- Step 11** Click **Save**.
- Step 12** Repeat [Step 9](#) through [Step 11](#) for each remaining PIMG unit in the phone system integration.
- Step 13** Exit UTIM.
-

# Appendix: PIMG Integrations Over a WAN That Use the G.729a Codec May Need to Disable Comfort Noise



## Note

This appendix applies only to systems that use PIMG units with PIMG firmware release 4, SU6 or earlier.

When the PIMG units connect to a WAN, use the G.729a codec, and have firmware release 4 SU6 or earlier, you must disable comfort noise. Otherwise, callers will hear loud comfort noise after pressing a DTMF key or between prompts when recording a message.

## To Disable Comfort Noise

**Step 1** On the Cisco Unity server, on the Start menu, click **Run**.

**Step 2** In the Run dialog box, enter **regedit** and click **OK**.



## Caution

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

**Step 3** If you do not have a current backup of the registry, save the registry settings to a file by doing the following depending on the Windows version:

|                     |                                                   |
|---------------------|---------------------------------------------------|
| <b>Windows 2003</b> | Click <b>File &gt; Export</b> .                   |
| <b>Windows 2000</b> | Click <b>Registry &gt; Export Registry File</b> . |

**Step 4** Expand the registry key HKEY\_LOCAL\_MACHINE\System\CurrentControlSet\Services\Avaudio\Parameters and double-click the **ComfortNoise** value in the right pane.

**Step 5** In the Edit DWORD Value dialog box, under Base, click **Decimal**.

**Step 6** In the Value Data field, enter **128**.

**Step 7** Click **OK**.

**Step 8** Exit the Registry Editor.

**Step 9** Restart the Cisco Unity server.

- Step 10** If you are using failover, repeat this procedure to apply the registry setting to the secondary Cisco Unity server.

## Appendix: Documentation and Technical Assistance

### Conventions

The *Ericsson MD-110 Serial PIMG Integration Guide for Cisco Unity 4.0* uses the following conventions.

**Table 19** *Ericsson MD-110 Serial PIMG Integration Guide for Cisco Unity 4.0 Conventions*

| Convention                 | Description                                                                                                                                                                                                                                                                                                                                                                             |
|----------------------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| boldfaced text             | Boldfaced text is used for: <ul style="list-style-type: none"> <li>Key and button names. (Example: Click <b>OK</b>.)</li> <li>Information that you enter. (Example: Enter <b>Administrator</b> in the User Name box.)</li> </ul>                                                                                                                                                        |
| < ><br>(angle brackets)    | Angle brackets are used around parameters for which you supply a value. (Example: In the Command Prompt window, enter <b>ping &lt;IP address&gt;</b> .)                                                                                                                                                                                                                                 |
| -<br>(hyphen)              | Hyphens separate keys that must be pressed simultaneously. (Example: Press <b>Ctrl-Alt-Delete</b> .)                                                                                                                                                                                                                                                                                    |
| ><br>(right angle bracket) | A right angle bracket is used to separate selections that you make: <ul style="list-style-type: none"> <li>On menus. (Example: On the Windows Start menu, click <b>Settings &gt; Control Panel &gt; Phone and Modem Options</b>.)</li> <li>In the navigation bar of the Cisco Unity Administrator. (Example: Go to the <b>System &gt; Configuration &gt; Settings</b> page.)</li> </ul> |
| [x]<br>(square brackets)   | Square brackets enclose an optional element (keyword or argument). (Example: [reg-e164])                                                                                                                                                                                                                                                                                                |
| [x   y]<br>(vertical line) | Square brackets enclosing keywords or arguments separated by a vertical line indicate an optional choice. (Example: [transport tcp   transport udp])                                                                                                                                                                                                                                    |
| {x   y}<br>(braces)        | Braces enclosing keywords or arguments separated by a vertical line indicate a required choice. (Example: {tcp   udp})                                                                                                                                                                                                                                                                  |

The *Ericsson MD-110 Serial PIMG Integration Guide for Cisco Unity 4.0* also uses the following conventions:

**Note**


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Means *reader take note*. Notes contain helpful suggestions or references to material not covered in the document.

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**Caution**


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Means *reader be careful*. In this situation, you might do something that could result in equipment damage or loss of data.

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For descriptions and URLs of Cisco Unity documentation on Cisco.com, see the *About Cisco Unity Documentation*. The document is shipped with Cisco Unity and is available at [http://www.cisco.com/univercd/cc/td/doc/product/voice/c\\_unity/about/aboutdoc.htm](http://www.cisco.com/univercd/cc/td/doc/product/voice/c_unity/about/aboutdoc.htm).

## 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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