



# Intecom Pointspan 6880 Serial SMDI TIMG Integration Guide for Cisco Unity 4.2

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This document provides instructions for integrating the Intecom Pointspan 6880 phone system with Cisco Unity 4.2 by using TIMG units (T1 media gateways), T1 digital lines (DS1 or “dry T1” digital lines only), and an RS-232 serial cable.

## Integration Tasks

Before doing the following tasks to integrate Cisco Unity with the Intecom Pointspan 6880 phone system by using the TIMG units (T1 media gateways), 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 list describes the process for creating the integration.

## Task List to Create the Integration

Use the following task list to set up a new integration with the Intecom Pointspan 6880 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 4](#).
3. Program the Intecom Pointspan 6880 phone system and extensions. See the [“Programming the Intecom Pointspan 6880 Phone System” section on page 6](#).
4. Set up the TIMG units. See the [Setting Up the TIMG Units, page 8](#).
5. *(Cisco Unity 4.2(1) only)* Download and install the required Cisco Unity ES file. See the [Downloading and Installing the Required Cisco Unity ES File \(Cisco Unity 4.2\(1\) Only\), page 19](#).
6. Create the integration. See the [Creating an Integration with the Phone System, page 20](#).



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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.

7. Test the integration. See the [Testing the Integration, page 24](#).
8. 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 27](#).

## Requirements

The Intecom Pointspan 6880 integration supports configurations of the following components:

### Phone System

- The Intecom Pointspan 6880 phone system.
- Software version 3.4K or later.
- PDI model 520-1000-004.
- T1 digital trunk interface card model 300-0289-001.
- The firmware must be configured to support T1 line-side signaling.
- T1 CAS connections to the TIMG units by using the FXS/FSO or E&M protocol.
- One or more TIMG units (media gateways). For details, refer to the “Supported Phone System Integrations Through PIMG/TIMG Units” section in the applicable *Supported Hardware and Software, and Support Policies for Cisco Unity* 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 TIMG unit with an RS-232 serial cable (which is available from Cisco).

Specifications for the serial cable are in *Connecting PBX-IP Media Gateway (PIMG) to the Serial Port of a PBX* at <http://www.dialogic.com/support/helpweb/mg/tn117.htm>.

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 T1 digital lines (DS1 or “dry T1” digital lines only) to the ports on the TIMG units.

**Caution**

T1 (or “wet T1”) connections to the PSTN must be through an MTU, CSU, or other device that provides line isolation. Otherwise, the TIMG units may be damaged.

- The TIMG units connected to the same LAN or WAN that Cisco Unity is connected to.

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

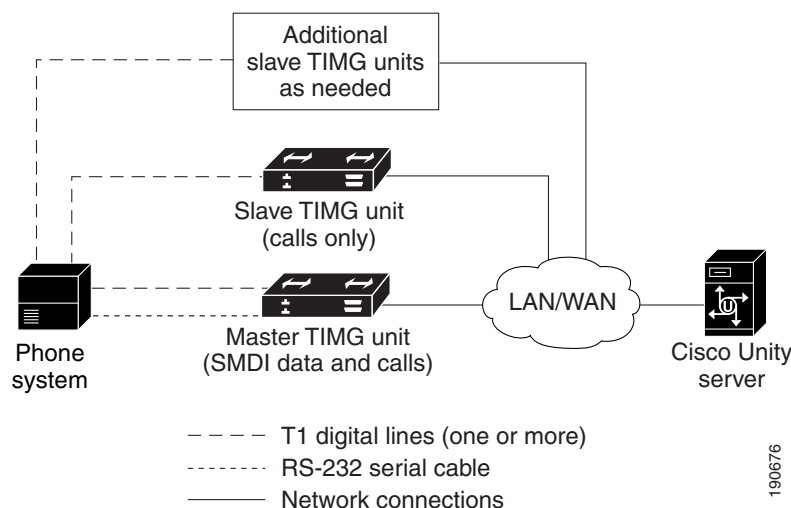
### Cisco Unity Server

- Cisco Unity 4.2 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).
- For Cisco Unity 4.2(1) only, the latest ES installed.
- A license that enables the applicable number of voice messaging ports.

## Integration Description

The Intecom Pointspan 6880 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 TIMG unit. Voice connections are sent through the T1 digital lines between the phone system and the TIMG units. The TIMG 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 with forwarded calls:

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

Cisco Unity 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.

## Integration Functionality

The Intecom Pointspan 6880 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)

### Supervised Transfer for Calls

When supervised transfer is enabled and a caller hangs up while being transferred to another phone, the called phone will ring briefly even though there is no call. If the ringing phone is answered, there will be no call, only dead air will be heard.

## Integrations with Multiple Phone Systems

Cisco Unity 4.2 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 for Cisco Unity 4.2* at [http://cisco.com/en/US/products/sw/voicesw/ps2237/products\\_installation\\_and\\_configuration\\_guides\\_list.html](http://cisco.com/en/US/products/sw/voicesw/ps2237/products_installation_and_configuration_guides_list.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 integrations)</i>	Do not check this check box. Otherwise, the integration may not function correctly.
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 Intecom Pointspan 6880 Phone System

Instruct the phone system technician to set up the phone system in the manner directed in the following procedure.

#### To Program the Intecom Pointspan 6880 Phone System for a Serial TIMG Integration with Cisco Unity

- Step 1** Program the phone lines connecting to the voice messaging ports on the TIMG units as a multiline hunt group.

Make sure that the phone system sends calls only to Cisco Unity voice messaging ports that are set to Answer Calls. Calls sent to a voice port not set to Answer Calls cannot be answered by Cisco Unity and may cause other problems.

**Step 2** Enable hookflash transfer capability on each phone line that connects to the voice messaging ports on the TIMG units.

**Step 3** Enable caller ID (through the serial data link to the master TIMG unit) on each subscriber extension.



**Caution** The phone system just provide caller ID through the serial data link to the master TIMG unit. Otherwise, some integration features will not function correctly.

**Step 4** For each subscriber extension, set the call forwarding options to the following:

- Unrestricted source
- Forward when the extension is not answered
- Forward when the extension is busy

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

#### Example of Programming for the Intecom Pointspan 6880 Phone System

```

** TRUNK GROUP DEFINITION
***...TRUNK GROUP.....17
***...TRUNK GROUP TYPE.....UNIVERSAL
TCI...TRUNK CLASS IDENTIFICATION.....LOCAL CO
UGP...USER GROUP NUMBER.....11
VTT...VOICE TRUNK TRANSFER ENABLED.....YES
CDT...TRANSFER COLLECT DIGIT TABLE #....NONE
DCS...DEFAULT CLASS OF SERVICE.....0
CNC...NATIONAL CALLING PARTY # CONTENTS..USER GROUP
CNI...USER GROUP CALLING PARTY #.....4 = 206-555-1212
DCP...DISPLAY CALLING PARTY NUMBER.....YES
PND...PRIVATE NETWORK.....NO
ITY...INTEMAIL TYPE.....INTEMAIL TYPE II
IST...DOES InteMail SUPERVISE TRANSFERS?.NO
IML...InteMail USER ID LENGTH.....5
IIN...INTEMAIL INTERFACE NUMBER.....2
VNP...InteMail NUMBER FORMAT.....DIRN
OAM...OAI ASSOCIATED MEMBER.....NO
BTG...BROADCAST TRUNK GROUP.....NO
TCM...TRAVELING CLASSMARK.....NO
FTH...FAILURE THRESHOLD.....3
RDT...RESEIZE DELAY TIME.....MSEC:200
CHT...TRUNK MONITOR MINIMUM HOLD TIME....0
DET...DISTANT END RELEASE TIME.....SEC:55
DCT...DATA CALLS ALLOWED.....NO
SWM...SEIZE WHEN MOS.....NO
TCH...TRUNK CALL HANDLING.....INTERNAL
NDS...DISCONNECT SUPERVISION.....YES
IGG...IGNORE GLARE.....NO
GDT...GLARE DETECT TIME.....MSEC:100
XFT...DISTANT IBX ALLOWS FEATURE TRANSP..NO
DPT...DTMF PASSTHROUGH TIMING INDEX.....NONE
.....TRUNK DIRECTION.....BOTH WAYS
*** INCOMING PARAMETERS
STY...INCOMING CALL ORIGINATION TYPE....T1 OFF PREM....OPX (OFF PREMISE)

```

```

TYP...INCOMING TRUNK TYPE.....DIALED
ICM...INCOMING CALL MESSAGE #.....17
IDS...INCOMING DIGIT SEQUENCE.....DESTINATION NUMBER ONLY
IRD...RESPONSE TO DESTINATION NUMBER.....NONE
IRC...RESPONSE TO CALLING PARTY NUMBER...NONE
IIT...INCOMING INFO DIGIT TYPE/LENGTH...NONE
WPR...WHISPER MESSAGE SOURCE GROUP.....NONE
APA...TRUNK GROUP AUTHORIZATION TYPE....NONE
PVA...PRE-VALIDATE AUTHORIZATION CODE...NO
RSC...RESET COUNT.....1
LVL...PREDEFINED LEVEL CODE.....NONE
TNE...TONE TABLE ENTRY NUMBER.....NONE
MOD...INCOMING DIAL MODE.....DTMF
RGF...DTMF RECEIVER GROUP.....52
TOO...TIMEOUT TO ATTENDANT.....NO
MCL...MULTIPLE CALLING ALLOWED.....NO
RAC...REUSE AUTH FOR MULT. CALLS.....NO
GAC...GROUP AUTH REQUIRED FOR TRUNKS....NO
SAC...SYSTEM ACCESS CODE.....NONE
CWR...CALLWAIT RINGBACK.....NO
UCT...TRUNK UPDATE CDR ON TRANSFER.....ALL
CPT...CALL PROGRESSING TONES:.....IBX PROVIDED
RIO...RESPONSE TO INCOMING ORIGINATION..NONE
IUG...InteMail LAMP MESSAGE USER GROUPS..ALL
NUG...INTER-USER GROUP NNP USER GROUPS..ALL
TCT...STATION CALL RESTRICTION ENABLED..NO
8NC...800 TO 4D SPEED NUMBER CONVERSION..NO
NWT...CALL PARTY NAME WAIT TIME.....NONE
*** OUTGOING PARAMETERS
MSG...MODEM SIGNALLING.....YES
TXA...DIRECT TGRP SELECT ALLOWED.....YES
ATG...ANNOUNCEMENT TRUNK GROUP.....NO
SLC...TRUNK SELECTION.....TOP DOWN
ICA...INTER-LATA CARRIER.....10XXX
OPS...OUTGOING OUTPUTSING SEQUENCE.....DESTINATION NUMBER ONLY
DIAGNOSTIC PARAMETERS: Y or N.....N

```

## Setting Up the TIMG Units

Do the following procedures to set up the analog TIMG units (media gateways) that are connected to the phone system.

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

- The phone system is connected to the TIMG units by using T1 digital lines and the applicable RS-232 serial cable.
- The TIMG units are ready to be connected to the LAN or WAN.
- The TIMG units are connected to a power source.

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

### To Download the TIMG Firmware Update Files for TIMG Units

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



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

- Step 2** On the Cisco Unity PIMG Software Download page, click the most recent version of the firmware for your TIMG 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 TIMG 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 TIMG 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 TIMG Units

- Step 1** On the Windows workstation, add a temporary route to enable access to the TIMG units.
- On the Windows Start menu, click **Run**.
  - Enter **cmd**, and press **Enter**. The Command Prompt window appears.
  - At the command prompt, enter **route add 10.12.13.74 <IP Address of Workstation>**, and press **Enter**.  
For example, if the IP address of the workstation is 198.1.3.25, enter “route add 10.12.13.74<space>198.1.3.25” in the Command Prompt window.
  - Close the Command Prompt window.
- Step 2** Connect a TIMG 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 2** System Login Page Settings

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

- 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 TIMG 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 TIMG unit appears. Click **Cancel**.



**Caution** Do not restart the TIMG unit until you are instructed to do so later in this procedure, even if the file installation fails. Restarting the TIMG unit at this step may prevent the TIMG 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 TIMG 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 3** System Login Page Settings

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

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

**Table 4** Password Page Settings

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

- Step 24** Click **Change**.
- Step 25** On the Configure menu, click **IP**.

**Step 26** On the IP page, enter the following settings for LAN1.

**Table 5** *IP Page Settings for LAN1*

Field	Setting
Client IP Address	Enter the new IP address that you want to use for the TIMG unit. (This is the IP address that you will enter in UTIM when you create the integration.)
Client Subnet Mask	Enter the new subnet mask, if the subnet mask is different from the default IP address.
Default Network Gateway Address	Enter the IP address of the default network gateway router that the TIMG units will use.
BOOTP Enabled	If you are using DHCP, click <b>Yes</b> . If you are not using DHCP, click <b>No</b> .

**Step 27** Click **Apply Changes**.

**Step 28** On the Configure menu, click **System**.

**Step 29** On the System page, enter the following settings.

**Table 6** *System Page Settings for the System and Telephony Groups*

Field	Setting
Operating Mode	Click <b>SIP</b> .
PCM Coding	Click <b>uLaw</b> .

**Step 30** Determine which serial port on the TIMG unit you will use to connect the data link serial cable from the phone system, then enter the following settings in the applicable group.

**Table 7** *System Page Settings for Serial Port Groups*

Field	Setting
Serial Port Baud Rate	Click the setting that is configured on the phone system. The default setting is 9600.
Serial Port Parity	Click the setting that is configured on the phone system. The default setting is None.
Serial Port Data Bits	Click the setting that is configured on the phone system. The default setting is 8.
Serial Port Stop Bits	Click the setting that is configured on the phone system. The default setting is 1.

**Step 31** Click **Apply Changes**.

**Step 32** On the Configure menu, click **Gateway**.

**Step 33** On the Gateway page, click the **Gateway Routing** tab.

**Step 34** On the Gateway Routing tab, enter the following settings.

**Table 8 Gateway Routing Tab Settings**

Field	Setting
Fault Tolerance Enabled	Click <b>Yes</b> .
Load Balancing Enabled	Enter <b>No</b> .
VoIP Endpoint ID: 1	<i>(Cisco Unity without failover)</i> Enter the IP address for the Cisco Unity server. <i>(Cisco Unity with failover configured)</i> Enter the IP address for the primary Cisco Unity server.
VoIP Endpoint ID: 2	<i>(Cisco Unity without failover)</i> Leave this field blank. <i>(Cisco Unity with failover configured)</i> Enter the IP address for the secondary Cisco Unity server.

**Step 35** Click **Apply Changes**.



**Step 36** On the Gateway page, click the **Gateway Advanced** tab.

**Step 37** On the Gateway Advanced tab, enter the following settings.

**Table 9 Gateway Advanced Tab Settings**

Field	Settings
<b>Advanced Call Routing</b>	
Call Connect Mode	Click <b>OnAnswer</b> .
Send DNIS to VoIP Endpoint	Click <b>No</b> .
Destination for Unroutable IP Calls	Leave this field blank.
Destination for Unroutable PBX Calls	Leave this field blank.
Monitor Call Connections	Click <b>No</b> .
<b>Telephony</b>	
Minimum Call Party Delay	Enter <b>0</b> .
Maximum Call Party Delay	Enter <b>2000</b> .
Dial Digit on Time	Enter <b>100</b> .
Dial Inter-Digit Time	Enter <b>100</b> .
Dial Pause Time	Enter <b>2000</b> .
Turn MWI On FAC	Leave this field blank.
Turn MWI Off FAC	Leave this field blank.
Dial Send Key	Click <b>None</b> .

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

Field	Settings
Outbound Call Connect Timeout	Enter <b>10000</b> .
Wait for Ringback/Connect on Blind Transfer	Click <b>Yes</b> .
Hunt Group Extension	Enter the pilot number for the Cisco Unity voice messaging ports.
<b>Audio</b>	
Audio Compression	Click the preferred codec for audio compression: <ul style="list-style-type: none"> <li>• <b>G.711</b>—The TIMG unit will use only the G.711 codec.</li> <li>• <b>G.729</b>—The TIMG unit will prefer the G.729a codec but can also use the G.711 codec.</li> </ul>
RTP Digit Relay Mode	Click <b>RFC2833</b> .
Signaling Digit Relay Mode	Click <b>Off</b> .
Voice Activity Detection	Click <b>On</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="border: 1px solid black; padding: 2px; margin-top: 5px;">  <b>Caution</b> Failure to use the correct setting will result in recorded messages containing nothing but silence. </div>
Frames Per Packet	Click the applicable setting: <ul style="list-style-type: none"> <li>• G.711—<b>1</b></li> <li>• G.729a—<b>2</b></li> </ul> <div style="border: 1px solid black; padding: 2px; margin-top: 5px;">  <b>Caution</b> Failure to use the correct setting will result in recorded messages containing nothing but silence. </div>
<b>Quality of Service</b>	
Call Control QOS Byte	<i>(TIMG units connect only to a LAN)</i> Enter <b>0</b> (equivalent to DSCP AF31). <i>(TIMG units connect to a WAN)</i> Enter <b>104</b> (equivalent to DSCP AF31).
RTP QOS Byte	<i>(TIMG units connect only to a LAN)</i> Enter <b>184</b> (equivalent to DSCP EF). <i>(TIMG units connect to a WAN)</i> Enter <b>184</b> (equivalent to DSCP EF).
<b>Traps and Alarms</b>	
SNMP Traps Enabled	Click <b>No</b> .
E-mail Alarms Enabled	Click <b>No</b> .

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

Field	Settings
HTTP Server Enabled	Click <b>Yes</b> .
HTTPs Server Enabled	Click <b>No</b> .

**Step 38** Click **Apply Changes**.

**Step 39** On the Gateway page, click the **Gateway Capabilities** tab.

**Step 40** Enter the following settings for all ports that are used by voice messaging ports on Cisco Unity.

**Table 10 Gateway Capabilities Tab Settings**

Field	Setting
Telephony Port Capability	Click <b>Both</b> .
Telephony Port Enabled	For ports that are used by voice messaging ports on Cisco Unity, click <b>Yes</b> . For ports that are not used, click <b>No</b> .

**Step 41** Click **Apply Changes**.

**Step 42** On the Configure menu, click **T1E1**.

**Step 43** On the T1E1 page, click the **T1/E1 Mode** tab.

**Step 44** On the T1/E1 Mode tab, enter the following settings.

**Table 11 T1/E1 Mode Tab Settings**

Field	Setting
Line Mode	Click <b>T1</b> .
Signaling Mode	Click <b>CAS</b> .
Interface Mode	Click <b>Terminal</b> .

**Step 45** Click **Apply Changes**.

**Step 46** Click the **T1-CAS Protocol** tab.

**Step 47** On the T1-CAS Protocol tab, enter the following settings.

**Table 12 T1-CAS Protocol Tab Settings**

Field	Setting
T1 CAS Protocol	Enter the setting that matches the phone system programming.
Line Encoding	Enter the setting that matches the phone system programming.
Framing	Enter the setting that matches the phone system programming.
Selects Transmit Pulse Waveform	Enter the setting that matches the phone system programming.
Flash Hook	Enter the setting that matches the phone system programming.

**Table 12** *T1-CAS Protocol Tab Settings (continued)*

Field	Setting
Consult Call Dialtone Drop Code	Enter the setting that matches the phone system programming.
Consult Call Proceeding Drop Code	Enter the setting that matches the phone system programming.
Consult Call Busy Drop Code	Enter the setting that matches the phone system programming.
Consult Call Error Drop Code	Enter the setting that matches the phone system programming.
Consult Call Connected Drop Code	Enter the setting that matches the phone system programming.
Consult Call Disconnected Drop Code	Enter the setting that matches the phone system programming.
MWI confirmation Tone	Click <b>No</b> .
CPID Type	Click <b>TypeII_CPID</b> .
Initial Wait for Inband CPID	Enter <b>100</b> .
Inband CPID Complete Timeout	Enter <b>300</b> .

**Step 48** Click **Apply Changes**.

**Step 49** On the Configure menu, click **Serial Protocol**.

**Step 50** On the Serial Protocol page, enter the following settings.

**Table 13** *Serial Protocol Page Settings*

Field	Setting
Serial Mode (Master/Slave)	Click the applicable setting: <ul style="list-style-type: none"> <li>• <b>Master</b>—Click this setting when this TIMG unit is connected to the data link serial cable from the phone system. There can be only one master TIMG unit in a phone system integration.</li> <li>• <b>Slave</b>—Click this setting when this TIMG unit is not connected to the data link serial cable from the phone system. There can be multiple slave TIMG units in a phone system integration.</li> </ul>
Serial Interface Protocol	Click <b>SMDI</b> .
MWI Response Timeout	Enter <b>2000</b> .
IP Address of Serial Server	If the TIMG unit is the master, leave this field blank. If the TIMG unit is a slave, enter the IP address of the master TIMG unit (the TIMG unit that is connected to the data link serial cable from the phone system).

**Table 13** Serial Protocol Page Settings (continued)

Field	Setting
Serial Cpid Expiration	Enter <b>5000</b> .
Logical Extension Number	Enter the extension number for each port on the TIMG unit.

**Step 51** Click **Apply Changes**.

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

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

**Table 14** SIP Page Settings

Field	Setting
Host and Domain Name	Enter the domain name of the TIMG unit.
Transport Type	Click <b>UDP</b> .
Call as Domain Name	Click <b>No</b> .
SIPS URI Scheme Enabled	Click <b>No</b> .
Invite Expiration	Enter <b>120</b> .
DNS Server Address	Enter the IP address of the DNS server.
Registration Server Address	<i>(Cisco Unity without failover)</i> Enter the IP address of the Cisco Unity server.  <i>(Cisco Unity with failover configured)</i> Enter the IP address of the primary Cisco Unity server.
Registration Server Port	For the first TIMG unit, enter <b>5060</b> .  When you configure more than one TIMG 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.
Registration Expiration	Enter <b>3600</b> .
UDP/TCP Transports Enabled	Click <b>Yes</b> .
TCP/UDP Server Port	Enter <b>5060</b> .
Primary Proxy Server Address	<i>(Cisco Unity without failover)</i> Enter the IP address of the Cisco Unity server.  <i>(Cisco Unity with failover configured)</i> Enter the IP address of the primary Cisco Unity server.
Primary Proxy Server Port	For the first TIMG unit, enter <b>5060</b> .  When you configure more than one TIMG 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.

**Table 14** SIP Page Settings (continued)

Field	Setting
Backup Proxy Server Address	(Cisco Unity without failover) Not applicable. Leave the default setting. (Cisco Unity with failover configured) Enter the IP address of the secondary Cisco Unity server.
Backup Proxy Server Port	(Cisco Unity without failover) Not applicable. Leave the default setting. (Cisco Unity with failover configured) For the first TIMG unit, enter <b>5060</b> . When you configure more than one TIMG 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. This setting must match the setting for the Primary Proxy Server Port field.
Proxy Query Interval	Enter <b>10</b> .
T1 Time	Enter <b>500</b> .
T2 Time	Enter <b>4000</b> .
T4 Time	Enter <b>5000</b> .

- Step 54** Click **Apply Changes**.
- Step 55** On the Configure menu, click **Tones**.
- Step 56** 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 57** 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 58** 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 59** On the Tones page, in the Destination Address field for Busy Tone, enter the extension that you dialed in [Step 58c](#). from the third phone.
- Step 60** 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 61** On the Tones page, in the Destination Address field for Error/Reorder Tone, enter the extension that you dialed in [Step 60a](#).
- Step 62** On the Tones page, for the Ringback Tone event, confirm that the Acquire Tone check box is checked and do the following substeps to verify that the tone is correct.
- From an available phone, dial an extension that does exist.
  - Confirm that you hear the ringback tone.
  - Hang up the phone.
- Step 63** On the Tones page, in the Destination Address field for Ringback Tone, enter the extension that you dialed in [Step 62a](#).
- Step 64** Click **Learn**.
- Step 65** When the process is complete, check the check box for each newly learned tone and click **Apply**.
- Step 66** Hang up the phones that you used in [Step 58](#).
- Step 67** On the Configure menu, click **Restart**.
- Step 68** On the Restart page, click **Restart Unit Now**.
- Step 69** When the TIMG unit has restarted, in the View menu, click **Refresh**.
- Step 70** Repeat [Step 2](#) through [Step 69](#) on all remaining TIMG units.

---

#### To Configure the RS-232 Serial Cable

- Step 1** Use the following pinout to configure the RS-232 serial cable by using an RJ-45-to-DB-9 terminal adaptor connector.

**Table 1-15** Pinout for the RS-232 Serial Cable

DB-9 Pin	Serial Port Pin Definition from the Phone System
1	DCD (data carrier detect)
2	RX (transmit)
3	TX (receive)
4	DTR (data terminal ready)
5	GND (signal ground)
6	DSR (data set ready)
7	RTS (request to send)
8	CTS (clear to send)
9	(no connection)

---

# Downloading and Installing the Required Cisco Unity ES File (Cisco Unity 4.2(1) Only)

For Cisco Unity 4.2(1) only, you must download and install the latest ES. Do the following procedure.

## To Download and Install the Required Cisco Unity ES File (Cisco Unity 4.2(1) Only)

- Step 1** On the Cisco Unity server, open a web browser and go to the Cisco Unity 4.2 Software Download page at <http://www.cisco.com/cgi-bin/tablebuild.pl/unity-42>.



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

- Step 2** On the Cisco Unity 4.2 Software Download page, click the link for the latest ES.
- 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 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 ES 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.
- Step 12** Browse to the directory where you saved the extracted ES file.
- Step 13** Double-click the ES file and follow the on-screen instructions to install the ES.
- Step 14** On Windows Start menu, click **Run**.
- Step 15** 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.) If you have any questions about changing registry key settings, contact Cisco TAC.

- Step 16** In the Registry Editor window, expand the key  
HKEY\_LOCAL\_MACHINE\Software\Active Voice\AvSkinny.
- Step 17** Double-click **Total Devices**.
- Step 18** In the Edit DWORD Value dialog box, in the Value Data field, enter **120**.
- Step 19** Under Base, click **Decimal** and click **OK**.
- Step 20** Close the Registry Editor.
- Step 21** Restart the Cisco Unity server.


## Creating an Integration with the Phone System

After ensuring that the phone system, the TIMG units, 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 16** Settings for the Add PIMG Dialog Box

Field	Setting
Display Name	Accept the default name or enter another name to identify this TIMG unit.
PIMG Type	Click the applicable setting: <ul style="list-style-type: none"> <li>• <b>TIMG (Single Span)</b></li> <li>• <b>TIMG (Dual Span)</b></li> <li>• <b>TIMG (Quad Span)</b></li> </ul>
IP Address	Enter the IP address of this TIMG unit.
SIP Port	Enter <b>5060</b> .   <b>Caution</b> You must enter the documented setting. Otherwise, MWIs will not function correctly.
Phone Lines (Ports) Connected	Enter the applicable setting: <ul style="list-style-type: none"> <li>• <b>24</b> (for single-span TIMG units)</li> <li>• <b>48</b> (for dual-span TIMG units)</li> <li>• <b>96</b> (for quad-span TIMG units)</li> </ul> <p>If you want to use fewer than the default number of voice messaging ports, enter the number of ports (or phone lines) that you want to use with this TIMG unit.</p>

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


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

**Table 17** Buttons on the Enter PIMG Settings Page

Field	Setting
Add	Displays the Add PIMG dialog box to add another TIMG unit to the integration.
Modify	Displays the Modify PIMG dialog box so that you can modify the settings of the selected TIMG unit.
Delete	Deletes the selected TIMG unit from the integration.
Ping Servers	Confirms that the IP address is correct for all TIMG units that are used by this integration.
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 TIMG 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 18** Settings for the Configure Cisco Unity SIP Settings Page

Field	Setting
Contact Line Name	<p>(Cisco Unity without failover) Enter the voice messaging line name that subscribers use to contact Cisco Unity and that Cisco Unity will use to register with the TIMG units.</p> <p>(Cisco Unity with failover configured) Enter the name the primary Cisco Unity server. This setting must match the Port X Endpoint parameter settings in the TIMG administration. This setting must be the same for both the primary and the secondary Cisco Unity servers.</p>
Cisco Unity SIP Port	<p>Enter the IP port on Cisco Unity that callers and the SIP server use to connect to voice mail. We recommend using the default setting.</p> <p> <b>Caution</b> The setting must match the setting on the TIMG units. If there are multiple phone system integrations with Cisco Unity, this setting must be unique for all integrations. Otherwise, the integration will not function correctly.</p>
Preferred Codec	Click the codec that 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** 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 19**      **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.  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** 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 20**      **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.  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

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**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 TIMG unit.

**Step 4** In the right pane, click the **Ports** tab.

**Step 5** Enter the settings shown in [Table 21](#) 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 21** 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 integrations)</i>	Do not check this check box. Otherwise, the integration may not function correctly.
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.

**Step 6** Click **Save**.

**Step 7** Click the **SIP Info** tab.

**Step 8** Uncheck the **Register with SIP 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 TIMG 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

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- 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

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).

For information on failover, refer to the *Cisco Unity Failover Configuration and Administration Guide* at [http://cisco.com/en/US/products/sw/voicesw/ps2237/products\\_feature\\_guides\\_list.html](http://cisco.com/en/US/products/sw/voicesw/ps2237/products_feature_guides_list.html).

## Requirements

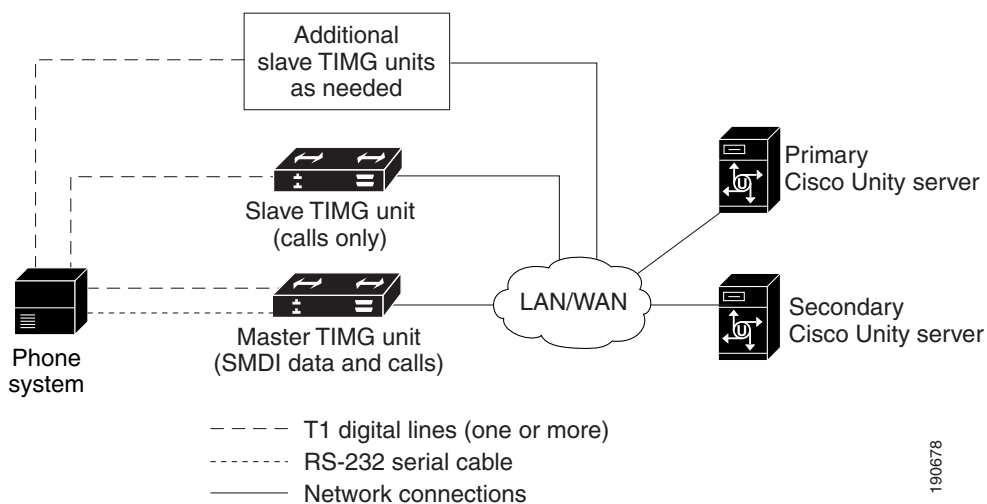
The following components are required to integrate a secondary server:

- 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 Intecom Pointspan 6880 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 TIMG unit. Voice connections are sent through the T1 digital lines (DS1 or “dry T1” digital lines only) between the phone system and the TIMG units. The TIMG 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 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** In the right pane, click **Properties**.
- Step 9** 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.
- Step 10** If the integration IDs of the phone system on the primary and secondary server are the same, continue to [Step 16](#).
- If the integration IDs of the phone system on the primary and secondary servers are different, on the secondary server, click **Modify Integration ID**.
- Step 11** When cautioned that subscribers associated with the current Integration ID setting will not be automatically associated with the new Integration ID setting, click **OK**.
- Step 12** 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**.
- Step 13** Click **Save**.
- Step 14** At the prompt to restart the Cisco Unity services, click **Yes**.
- Step 15** In the left pane, click the phone system integration that you created in [Step 3](#).
- Step 16** In the right pane, click the **Ports** tab.
- Step 17** 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 Dialout MWI, do not send calls to it.
- Step 18** Click **Save**.
- Step 19** Repeat [Step 16](#) through [Step 18](#) for each remaining TIMG unit in the phone system integration.

Step 20 Exit UTIM.

## Appendix: Documentation and Technical Assistance

### Conventions

The *Intecom Pointspan 6880 Serial SMDI TIMG Integration Guide for Cisco Unity 4.2* uses the following conventions.

**Table 22** *Intecom Pointspan 6880 Serial SMDI TIMG Integration Guide for Cisco Unity 4.2 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>
< > (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> .)
- (hyphen)	Hyphens separate keys that must be pressed simultaneously. (Example: Press <b>Ctrl-Alt-Delete</b> .)
> (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] (square brackets)	Square brackets enclose an optional element (keyword or argument). (Example: [reg-e164])
[x   y] (vertical line)	Square brackets enclosing keywords or arguments separated by a vertical line indicate an optional choice. (Example: [transport tcp   transport udp])
{x   y} (braces)	Braces enclosing keywords or arguments separated by a vertical line indicate a required choice. (Example: {tcp   udp})

The *Intecom Pointspan 6880 Serial SMDI TIMG Integration Guide for Cisco Unity 4.2* also uses the following conventions:

**Note**

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

**Caution**

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

For descriptions and URLs of Cisco Unity documentation on Cisco.com, see the *Documentation Guide for Cisco Unity*. The document is shipped with Cisco Unity and is available at

[http://cisco.com/en/US/products/sw/voicesw/ps2237/products\\_documentation\\_roadmap09186a00801179df.html](http://cisco.com/en/US/products/sw/voicesw/ps2237/products_documentation_roadmap09186a00801179df.html).

## Obtaining Documentation, Obtaining Support, and Security Guidelines

For information on obtaining documentation, obtaining support, providing documentation feedback, security guidelines, and also recommended aliases and general Cisco documents, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

<http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html>

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