



Nortel SL-100 Serial SMDI TIMG Integration Guide for Cisco Unity 4.2

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This document provides instructions for integrating the Nortel SL-100 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 Nortel SL-100 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 Nortel SL-100 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 Nortel SL-100 phone system and extensions. See the [“Programming the Nortel SL-100 Phone System” section on page 6](#).
4. Set up the TIMG units. See the [Setting Up the TIMG Units, page 14](#).
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 25](#).
6. Create the integration. See the [Creating an Integration with the Phone System, page 26](#).



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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 31](#).
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 34](#).

Requirements

The Nortel SL-100 integration supports configurations of the following components:

Phone System

- The Nortel SL-100 phone system.
- Software version SE-06 or later.
- T1 digital trunk interface card (NT5D11).
- 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.
- 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.

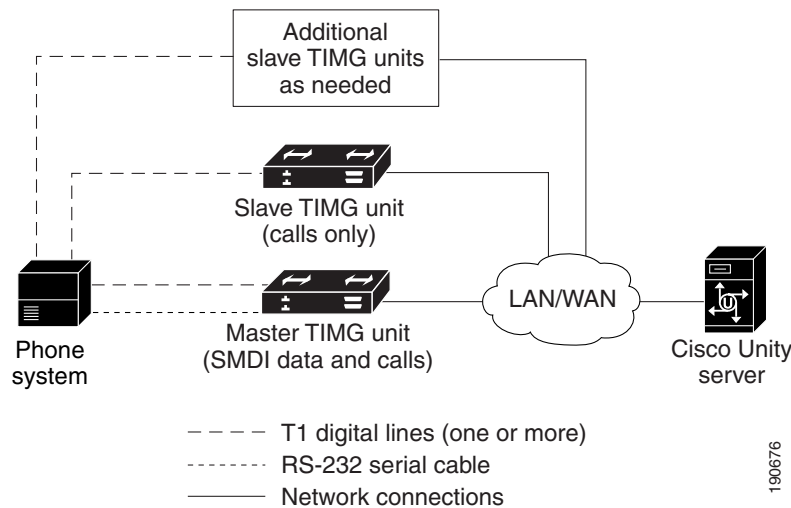
Cisco Unity Server

- 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.
- 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 Nortel SL-100 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 Nortel SL-100 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.

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 Nortel SL-100 Phone System

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

To Program the Nortel SL-100 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.



Note The following programming was accomplished through the maintenance administration position (MAP) terminal.

Example of Programming for the Nortel SL-100 Phone System

1. On the MAP terminal, enter **table mpc** and press **Enter**.
2. Enter **add** and press **Enter**.
3. On Table MPC, enter the following settings.

Table 2 Settings for Table MPC

Field	Setting
MPCNO	Enter the MPC number used for SMDI.
MPCIOC	Enter the number associated with the MPC (SMDI) card.
IOCCCT	Enter the slot position on the IOC shelf multiplied by 4, from 0 to 32.
EQ	Enter the NT product engineering code for the MPC card in the format 1X89zz or FX30zz, where zz are the two letters at the end of the product code.
DLDFILE	In the format MPCAxxyy, enter the name of the 8-character download file for SMDI and MPCA.

4. Enter **table mpclink** and press **Enter**.
5. Enter **add** and press **Enter**.
6. On Table MPCLINK, enter the following settings.

Table 3 Settings for Table MPCLINK

Field	Subfield	Setting
LINKKEY	MPCNO	Enter the MPC number used for SMDI (the same number entered in Table MPC).
	LINKNO	Enter the MPC link number for SMDI application with ASYNC protocol.
LINKALM		Enter Y to activate the MPCLINK alarm for system busy (SYSB) MPC links. Enter N if you do not want to activate the MPCLINK alarm for system busy (SYSB) MPC links. Note If you enter N , the system does not generate MPC908 (MPC link state transition) logs.
PROTOCOL		Enter ASYNC .
LINKNABL		Enter 765 .
PARAM		Enter APLDEFN .
ADEFN		Enter SMDI .
PARAM		Enter BAUDRATE .
RATE		Enter B9600 .
PARAM		Enter PARITY .
PRTY		Enter EVEN .
PARAM		The following are among the optional parameters: L1IDLY, L2IDLY, LNKDOWN. If you enter a parameter, you are then prompted to enter a value for it. L1IDLY and L2IDLY timers can be used in offices with heavy SMDI/VMS traffic to shorten the amount of time the MPC can delay sending an MWI to the switch. (The default is 3 seconds.) The LNKDOWN timer adjusts the length of time the switch takes to recognize LINK failure and sets the LINK to SYSB. (The default is 2 seconds.)
CHARBITS		Enter BIT7 .

7. Enter **table sllnkdev** and press **Enter**.
8. Enter **add** and press **Enter**.
9. On Table SLLNKDEV, enter the following settings.

Table 4 Settings for Table SLLNKDEV

Field	Setting
DEVNAME	Enter a unique device name.
DEVICE	Enter 1X89 .
MPCNO	Enter the value MPC number that was specified in Table MPC.
LINKNO	Enter the value for the MPC link number that was specified in Table MPCLINK.
XLATION	Enter NONE .

Table 4 **Settings for Table SLLNKDEV (continued)**

Field	Setting
PROTOCOL	Enter NONE .
DIRECTION	Enter INOUTLK .
XFER	Enter SMDIDATA .
OPTION	Enter NUMOFDIGS .
NUMDIGS	The number of digits sent by the phone system to the voice messaging system through the SMDI link. Note The entry value chosen should match the dialing plan configured on the phone system.
OPTION	Enter CGNADDRDN .
OPTION	Enter \$.

10. Enter **table ofrt** and press **Enter**.

You use Table OFRT to set up a treatment for unanswered calls. The following example shows settings for routing unanswered calls back to the voice messaging system.

11. Enter **add** and press **Enter**.**12.** On Table OFRT, enter the following settings.**Table 5** **Settings for Table OFRT**

Field	Setting
RTE	If the record is the first in the route list, enter the route reference number assigned to the route list. Otherwise, leave this field blank.
RTSEL	Enter the route selector.
SNPA	Enter the serving NPA (area code) of the DN.
TYPCALL	Enter the type of call: <ul style="list-style-type: none"> • DD • NP • OA
ORIGSCRE	Enter LCL (for local) or NLCL (for non-local).
REPLDIGS	Enter up to 11 replace digits.
CANCNORC	Enter Y or N to indicate whether to cancel normal change.
BILLCODE	Enter the billing code. If there is no billing number, enter N .

13. Enter **table digcol** and press **Enter**.

You use Table DIGCOL to set up the action that the line module must take with the first digit that is dialed.

14. Enter **add** and press **Enter**.**15.** On Table DIGCOL, enter the following settings.

Table 6 Settings for Table DIGCOL

Field	Subfield	Setting
DGKEY	DATNAME	Enter the character assigned to the block of data in Table DIGCOL.
	DIGIT	Enter a numeric value from 0–9, STAR (*), or OCT (#) to specify the digit that is applicable to the record.
DGDATA	DGCOLSEL	Enter COL for the collection of more digits.
COLDATA	TMODE	Enter S for short timing mode.
	NUMDIGS	Number of digits. If the TMODE value is S, specify the number of digits for which short timing is required after the receipt of each digit. The number of digits specified, which does not include the initial digit, must be no greater than three for short timing.

16. Enter **table ucdgroup** and press **Enter**.

You use Table UCDGRP to set up the UCD group.



Note The UCD group must have a unique primary DN.

17. Enter **add** and press **Enter**.
18. On Table UCDGRP, enter the following settings.

Table 7 Settings for Table UCDGRP

Field	Setting
UCDNAME	This is the name of the UCD group. It can be up to 16 characters in length. The first eight characters must be unique.
ACD	Enter N .
CUSTGRP	Name of the customer group to which the UCD group belongs.
UCDRNGTH	Ringling threshold, in one-second intervals, after which an unanswered call to a UCD agent is forwarded to the route specified in the THROUT field. The range is 0–63.
TABNAME	Enter OFRT .
INDEX	Enter the number assigned to the route list in Table OFRT (1–1023).
TABNAME	Enter OFRT for the table to which translations are routed.
INDEX	Enter the number assigned to the route list in Table OFRT (1–1023).
PRIOPRO	Enter Maximum time, in seconds, a call can wait in a UCD group (0–255).
MAXPOS	Enter the maximum number of UCD agent positions that can be active at one time. This number should be the number of ports on the all TIMG units that are connected to the phone system.
DBG	Delayed billing. Enter Y if billing starts when the call is answered by a UCD agent. Enter N if billing starts when the caller receives a recorded announcement.
DEFPRIO	Enter 0 .
RLSCNT	Enter 0 .

Table 7 Settings for Table UCDGRP (continued)

Field	Setting
MAXWAIT	Enter the maximum time, in seconds, that a call waits in the incoming call queue before being answered (0–1800).
MAXCQSIZ	Enter the maximum number of calls that can be in the incoming queue waiting for an idle channel (0–511).
OPTION	Enter UCD_SMDI .
SMDI_LINK	Enter the terminal designation defined in Table SLLNKDEV.
SMDI_DESK_NO	Enter the message desk number (1–63). If you have more than one UCD group, one of them must be set to 63. We recommend that the first UCD group on a data link be set to 63. The second is set to 62, and so on. Note If CRR (Call Request Retrieval) is used, all requests are made to the UCD group with SMDI_DSK_NO = 63.

19. Enter **table dnroute** and press **Enter**.

You use Table DNROUTE to set up the UCD group.



Note The UCD group must have a unique primary DN.

20. Enter **add** and press **Enter**.

21. On Table DNROUTE, enter the following settings.

Table 8 Settings for Table DNROUTE

Field	Subfield	Setting
DNNM	AREACODE	Enter the DN for the UCD group specified as the UCDGRP.
	OFCCODE	Note The UCD DN must be a dialable number from an agent on the phone system so that dialing plans and translation tables do not conflict.
	STNCODE	
DN_SEL		Enter FEAT .
FEATURE		Enter UCD .
UCDGRP		Enter the value for the UCDNAME field that is defined in Table UCDGROUP.
DNTYPE		Enter PRIM .
TOLLPRIO		Enter 0 .

22. Enter **table lninv** and press **Enter**.

You use Table LNINV to assign card slots on the line or remote line module.

23. Enter **add** and press **Enter**.

24. On Table LNINV, enter the following settings.

Table 9 Settings for Table LNINV

Field	Setting
LEN	Enter the line equipment number of the card slot.
CARDCODE	Enter 5d11ae .
PADGRP	Enter the name of the appropriate pad group that appears in the PADDATA table.
STATUS	Enter WORKING .
GND	Enter Y .
BNV	Enter NL .
MNO	Enter Y .
CARDTYPE	Enter NIL .

25. Enter **servord** and press **Enter**.

You will add agents to the UCD group by entering the following inputs at the prompts.

Table 10 Inputs for Adding Agents to the UCD Group

Prompt	Input	Description
SO:	NEW	
SONUMBER:	Press Enter .	When to invoke this service. Pressing Enter starts the service at the current date and time.
DN:		The Directory Number of the line. Use ten-digit DNs.
LCC_ACC:	IBN	The line class code of service.
GROUP:		The name of the IBN customer group to which the line belongs. For example, covm.
SUBGRP:		The subgroup number. For example, 0.
NCOS:		The network class of service. For example, 1.
SNPA:		The serving NPA (area code) of the DN.
LEN_OR_LT D:		The line equipment number of the line. For example, 4 0 1 0 (separated by spaces).
OPTION:	COD	The cut-off on disconnect.
OPTION:	UCD	Uniform call distribution.
OPTION:	DGT	Digitone.
OPTION:	3WC	Three-way calling.
OPTION:	CXR	Call Transfer.
CXFERTYP:	CTALL	Call Transfer Type. CTALL = transfer all calls.
CXRRCL:	N	Call Transfer Recall.
METHOD:	STD	Method of Call Transfer: Std = Std Call Transfer method.
OPTION:	SMDI	Simplified message desk interface.

Table 10 *Inputs for Adding Agents to the UCD Group (continued)*

Prompt	Input	Description
LINENO:		The UCD terminal number. This is the line number associated with the SMDI channel. This parameter must be unique for each agent in the associated UCDGRP.
UCDGRP:		The UCDNAME from the UCDGRP table. This is the UCD group to which you are adding the agent.
AUTO_LOG:	Y	Autologon capability required.
OPTION:	\$	The data you have entered appears.
	Y	Enter Y .

26. Enter **servord** and press **Enter**.

You will add a pilot number (UCD group DN) for the ports on the TIMG units by entering the following inputs at the prompts.

Table 11 *Inputs for Adding Agents to the UCD Group*

Prompt	Input	Description
SO:	NEW	
SONUMBER:	Press Enter .	When to invoke this service. Pressing Enter starts the service at the current date and time.
DN:		The directory number of the line. This is the DN that you enter in the SDN table.
LCC:	IBN	The line class code of service.
GROUP:		The name of the IBN customer group to which the line belongs.
SUBGRP:		The subgroup number.
NCOS:		The network class of service.
SNPA:		Serving NPA (area code) of the DN.
LEN:		Line equipment number of the line. For example, 4 0 1 0.
OPTION:	cfb	Call Forward Busy. Note This input is optional.
CFBCNTL:	N	Normal assignment for CFB. Note This input is optional.
CFBDN:		The primary UCD DN. Note This input is optional.
OPTION:	CFF	Call Forward Fixed.
CFFDN:		The Primary UCD DN.
OPTION:	CFU	Call Forward Universal.
OVRDACR:	N	Override Automatic Callback.
OPTION:	\$	The data you entered appears.
	Y	Enter Y to confirm the data.

27. Connect a phone to the line.
28. Pick up the handset.
29. Dial the call forward activation code followed by the pilot number (UCD DN). For example, dial *80 5551234.



Note If you do not know this code, look it up in Table IBNXLA. The code is in the CFWP field.

30. Confirm that you hear the confirmation tone, which indicates that the line has been forwarded.



Note If the phone system is restarted, you must repeat Step 27. through Step 30. for each line DN that CFUs to the UCD group.

31. Enter **table ibnxla** and press **Enter**.
You use Table IBNXLA to set up and message waiting indicators (MWIs).
32. Enter **add** and press **Enter**.
33. On Table IBNXLA, enter the following settings.

Table 12 Settings for Table IBNXLA

Field	Subfield	Setting
KEY	XLANAME	Enter the name of the translator, 1–8 characters, for the MWI feature.
	DGLIDX	Enter the access code for the MWI feature.
RESULT	TRSEL	Enter FEAT .
	ACR	Enter N .
	SMDR	Enter N .
	FEATURE	Enter the following features: CRA CRR CRDS CRDA UCDD UCDA CFWP CFWC

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 13 System Login Page Settings

Field	Setting
Username	Enter admin .
Password	Enter 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 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 14 *System Login Page Settings*

Field	Setting
Username	Enter admin .
Password	Enter 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 15 *Password Page Settings*

Field	Setting
Old Password	Enter IpodAdmin . (This setting is case sensitive.)

Table 15 Password Page Settings (continued)

Field	Setting
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 16 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 Yes . If you are not using DHCP, click No .

Step 27 Click **Apply Changes**.

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

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

Table 17 System Page Settings for the System and Telephony Groups

Field	Setting
Operating Mode	Click SIP .
PCM Coding	Click uLaw .

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 18 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 19 Gateway Routing Tab Settings

Field	Setting
Fault Tolerance Enabled	Click Yes .
Load Balancing Enabled	Enter No .
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 20 Gateway Advanced Tab Settings

Field	Settings
Advanced Call Routing	
Call Connect Mode	Click OnAnswer .
Send DNIS to VoIP Endpoint	Click No .

Table 20 Gateway Advanced Tab Settings (continued)



Field	Settings
Destination for Unroutable IP Calls	Leave this field blank.
Destination for Unroutable PBX Calls	Leave this field blank.
Monitor Call Connections	Click No .
Telephony	
Minimum Call Party Delay	Enter 0 .
Maximum Call Party Delay	Enter 2000 .
Dial Digit on Time	Enter 100 .
Dial Inter-Digit Time	Enter 100 .
Dial Pause Time	Enter 2000 .
Turn MWI On FAC	Leave this field blank.
Turn MWI Off FAC	Leave this field blank.
Dial Send Key	Click None .
Outbound Call Connect Timeout	Enter 10000 .
Wait for Ringback/Connect on Blind Transfer	Click Yes .
Hunt Group Extension	Enter the pilot number for the Cisco Unity voice messaging ports.
Audio	
Audio Compression	Click the preferred codec for audio compression: <ul style="list-style-type: none"> • G.711—The TIMG unit will use only the G.711 codec. • G.729—The TIMG unit will prefer the G.729a codec but can also use the G.711 codec.
RTP Digit Relay Mode	Click RFC2833 .
Signaling Digit Relay Mode	Click Off .
Voice Activity Detection	Click On .
Frame Size	Click the applicable setting: <ul style="list-style-type: none"> • G.711—20 • G.729a—10 <div style="border: 1px solid black; padding: 5px; margin-top: 10px;">  <p>Caution Failure to use the correct setting will result in recorded messages containing nothing but silence.</p> </div>

Table 20 Gateway Advanced Tab Settings (continued)

Field	Settings
Frames Per Packet	Click the applicable setting: <ul style="list-style-type: none"> • G.711—1 • G.729a—2  Caution Failure to use the correct setting will result in recorded messages containing nothing but silence.
Quality of Service	
Call Control QOS Byte	(TIMG units connect only to a LAN) Enter 0 (equivalent to DSCP AF31). (TIMG units connect to a WAN) Enter 104 (equivalent to DSCP AF31).
RTP QOS Byte	(TIMG units connect only to a LAN) Enter 184 (equivalent to DSCP EF). (TIMG units connect to a WAN) Enter 184 (equivalent to DSCP EF).
Traps and Alarms	
SNMP Traps Enabled	Click No .
E-mail Alarms Enabled	Click No .
HTTP Server Enabled	Click Yes .
HTTPs Server Enabled	Click No .

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 21 Gateway Capabilities Tab Settings

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

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 22 T1/E1 Mode Tab Settings

Field	Setting
Line Mode	Click T1 .

Table 22 *T1/E1 Mode Tab Settings (continued)*

Field	Setting
Signaling Mode	Click CAS .
Interface Mode	Click Terminal .

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 23 *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.
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 No .
CPID Type	Click TypeII_CPID .
Initial Wait for Inband CPID	Enter 100 .
Inband CPID Complete Timeout	Enter 300 .

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 24 Serial Protocol Page Settings

Field	Setting
Serial Mode (Master/Slave)	Click the applicable setting: <ul style="list-style-type: none"> • Master—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. • Slave—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.
Serial Interface Protocol	Click SMDI .
MWI Response Timeout	Enter 2000 .
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).
Serial Cpid Expiration	Enter 5000 .
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 25 SIP Page Settings

Field	Setting
Host and Domain Name	Enter the domain name of the TIMG unit.
Transport Type	Click UDP .
Call as Domain Name	Click No .
SIPS URI Scheme Enabled	Click No .
Invite Expiration	Enter 120 .
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 5060 . 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 3600 .

Table 25 SIP Page Settings (continued)

Field	Setting
UDP/TCP Transports Enabled	Click Yes .
TCP/UDP Server Port	Enter 5060 .
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 5060 . 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.
Backup Proxy Server Address	<i>(Cisco Unity without failover)</i> Not applicable. Leave the default setting. <i>(Cisco Unity with failover configured)</i> Enter 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. <i>(Cisco Unity with failover configured)</i> For the first TIMG unit, enter 5060 . 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 10 .
T1 Time	Enter 500 .
T2 Time	Enter 4000 .
T4 Time	Enter 5000 .

- 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.
- a. From a available phone, call a second phone.
 - b. Answer the second phone when it rings, and leave both handsets off so that both phones are busy.

- c. From a third phone, dial one of the busy phones.
 - d. Confirm that you hear a busy tone.
 - e. Hang up the third phone but leave the handsets for the other two phones off.
- Step 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.
- a. From an available phone, dial an extension that does not exist.
 - b. Confirm that you hear the reorder or error tone.
 - c. Hang up the phone.
- Step 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.
- a. From an available phone, dial an extension that does exist.
 - b. Confirm that you hear the ringback tone.
 - c. Hang up the phone.
- Step 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 Tones page, on the Learn tab, for the Dialtone event, confirm that the Acquire Tone check box is checked and enter ! in the Destination Address field.
- Step 68** Uncheck the Acquire Tone check boxes for all other tones.
- Step 69** Click **Learn**.
- Step 70** When the process is complete, check the check box for each newly learned tone and click **Apply**.
- Step 71** On the Configure menu, click **Restart**.
- Step 72** On the Restart page, click **Restart Unit Now**.
- Step 73** When the TIMG unit has restarted, in the View menu, click **Refresh**.
- Step 74** Repeat [Step 2](#) through [Step 73](#) on all remaining TIMG units.

To Configure the RS-232 Serial Cable

- Step 1** Use the applicable pinout to configure the RS-232 serial cable:

- For a DB-9 terminal adaptor connector, use the following pinout.

Table 26 *Pinout for a DB-9 Connector*

DB-9 Pin	Serial Port Pin Definition from the Phone System
1	Data Carrier Detect
2	Transmit Data (required)
3	Receive Data (required)
4	Data Terminal Ready
5	Signal Ground (required)
6	Data Set Ready
7	Clear to Send
8	Request to Send
9	Ring Indicator

- For a 34-pin to DB-25 terminal adaptor connector, use the following pinout.

Table 27 *Pinout for a 35-Pin to DB-25 Connector*

Pin for 34-Pin Connector	Pin for 34-Pin Connector	Serial Port Pin Definition from the Phone System
14	2	Transmit Data (required)
26	3	Receive Data (required)
23	4	Request to Send (required)
34	5	Clear to Send (required)
16	6	Data Set Ready (required)
11	7	Signal Ground (required)
31	8	Data Carrier Detect (required)
37	20	Data Terminal Ready (required)
16	17	Receive Clock
17	12	Secondary Data Carrier Detect
24	24	Secondary Clock Transmit External

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 28 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"> • TIMG (Single Span) • TIMG (Dual Span) • TIMG (Quad Span)
IP Address	Enter the IP address of this TIMG unit.
SIP Port	Enter 5060 .  Caution 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"> • 24 (for single-span TIMG units) • 48 (for dual-span TIMG units) • 96 (for quad-span TIMG units) <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 29 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 30 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> Caution 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 31 *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 32 *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

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 33](#) 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 33 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.
- 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

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.

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.

Requirements

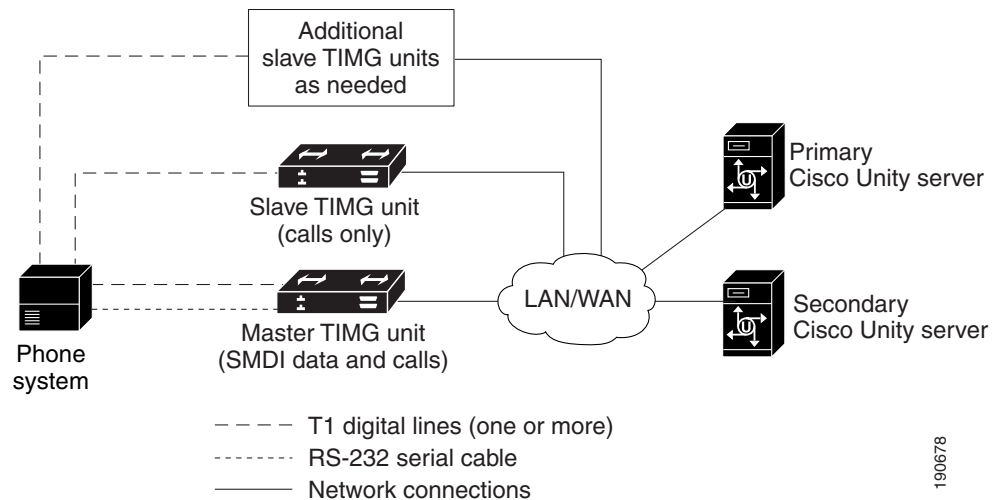
The following components are required to integrate a secondary server:

- 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 Nortel SL-100 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 *Nortel SL-100 Serial SMDI TIMG Integration Guide for Cisco Unity 4.2* uses the following conventions.

Table 34 *Nortel SL-100 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"> Key and button names. (Example: Click OK.) Information that you enter. (Example: Enter Administrator in the User Name box.)
< > (angle brackets)	Angle brackets are used around parameters for which you supply a value. (Example: In the Command Prompt window, enter ping <IP address> .)
- (hyphen)	Hyphens separate keys that must be pressed simultaneously. (Example: Press Ctrl-Alt-Delete .)
> (right angle bracket)	A right angle bracket is used to separate selections that you make: <ul style="list-style-type: none"> On menus. (Example: On the Windows Start menu, click Settings > Control Panel > Phone and Modem Options.) In the navigation bar of the Cisco Unity Administrator. (Example: Go to the System > Configuration > Settings page.)
[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 *Nortel SL-100 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.

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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