

Integration Guide

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Integration Guide

unityTM

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■ CENTREX/SMDI

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Integration overview

Use this section of the *Unity Integration Guide* with the *Unity Installation Guide*.

Integration steps

Follow these steps to set up this integration.

- 1 Review the system and equipment requirements to verify that all phone system and Unity™ server requirements have been met. See “Requirements,” below.
- 2 Configure Unity for the integration. See “Configuring Unity for the integration,” on page 6.
- 3 Program the phone system and extensions. See “Programming the phone system,” on page 7.
- 4 Test the connection to the central office. See “Testing the Centrex connection,” on page 8.

Requirements

The steps to set up the Centrex Simplified Message Desk Interface (SMDI) integration require the following:

Phone system

- ◆ Centrex 1AESS, 5ESS or DMS100 service.
- ◆ A Centrex service SMDI package with one SMDI 4-wire private data link connected to the external modem.
- ◆ A type 202T or 212T external modem set to 1200 baud.
- ◆ The data link of the external modem connected to a serial port (COM1 is the default) on the Unity server with an RS-232 serial cable.
- ◆ The central office’s voice messaging lines connected to the voice boards in the Unity server.

Unity server

- ◆ The Dialogic® D/41D™, D/41ESC™, D/41EPCI™, D/120JCT-LS™, or D/160SC-LS™ voice boards, installed.

NOTE: For the latest information on supported voice boards for Unity, contact your sales representative.

- ◆ Unity installed and ready for the integration as described in the *Unity Installation Guide*.

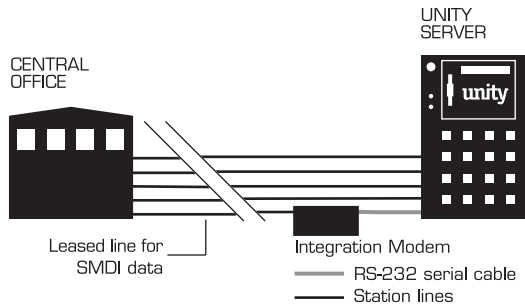
NOTE: To ensure that Windows does not mistake the serial connection for a serial mouse when rebooting the Unity server, confirm that the boot.ini file is set according to instructions. See "Installing Windows NT Server" in Chapter 3 of the Unity Installation Guide.

- ◆ A system key that enables the integration and the appropriate number of voice messaging ports.
- ◆ An available serial port (COM1 is the default).

Integration description

How the integration works

The Centrex/SMDI integration uses a data link, which consists of an RS-232 serial cable, a central office leased line, and an external integration modem. The integration modem provides the data link connection between the Centrex system at the central office and the Unity server. This modem, which is connected to the Unity server through the RS-232 serial cable, must be compatible with the type of modem at the central office (type 202T or 212T). The integration modem is separate from the modem in the Unity server, which is used for remote maintenance. The Centrex system's voice messaging lines connect to the analog voice boards in the Unity server. The following illustration shows the required connections.



The central office system sends the following information through the data link:

- ◆ The called party's extension
- ◆ The reason for the forward (the extension is busy, does not answer, or is set to forward all calls)
- ◆ The calling party's extension (for internal calls)

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

Integration features

The Centrex/SMDI integration with Unity provides the following features:

Call forward to personal greeting When an incoming call is routed to an unanswered or busy extension, the call is forwarded to the subscriber's voice mail. The caller then hears the subscriber's personal greeting and can leave a message.

Caller ID Unity receives caller ID information from the phone system (if available). This information displays in the message's subject line in Microsoft Outlook® (or other desktop messaging application).

Easy message access A subscriber can retrieve messages without entering an ID. Unity identifies a subscriber based on the extension from which the call originated. A password may be required.

Identified subscriber messaging Unity automatically identifies a subscriber who leaves a message during a forwarded internal call, based on the extension from which the call originated.

Message waiting indication (MWI) When a message is waiting for a subscriber, Unity notifies the phone system to activate the message waiting indicator on the subscriber's extension (if it is equipped with one) or to activate a stutter dial tone.

Configuring Unity for the integration

After ensuring that the Unity server is ready for the integration by completing the appropriate tasks in Chapters 1 through 4 of the *Unity Installation Guide*, perform the following procedure to confirm that the integration is enabled.

To confirm that the integration is enabled

Before beginning this procedure, complete all procedures in the section titled “Configuring Unity to work with the phone system,” in Chapter 4 of the *Unity Installation Guide*.

- 1 In Internet Explorer, go to the Unity Administrator (<http://<server name>/saweb>).
- 2 Go to System > Licensing > Licensed Features.
- 3 In the “Integration” box, confirm that the setting is “Serial.”
- 4 If the setting is not “Serial,” contact your sales representative for the necessary system key.
- 5 Go to System > Switch. Confirm that the settings match those in the following table.

If the settings are incorrect, integration features may not be enabled.

Switch Settings

Switch parameter	Required settings
Manufacturer	Centrex
Model	1AESS, 5ESS, DMS100
Switch PBX software version	All
Integration	Serial

Programming the phone system

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

To program the phone system

Instruct the Centrex service provider to set up the phone system in the following manner:

- ❶ Program the message desk lines as a multiline hunt group.
- ❷ Enable switchhook flash transfer capability on each message desk line.
- ❸ Enable SMDI caller ID on each subscriber extension.
- ❹ For each subscriber extension, set the call forwarding options to: unrestricted source; forward when the extension is not answered; and forward when the extension is busy.

Testing the Centrex connection

Perform the following procedures before testing the integration as instructed in Chapter 4 of the *Unity Installation Guide*.

To confirm that there is a data link connection to the central office

- ◆ Confirm that the carrier detect (CD) light on the integration modem is lit when the modem is connected to the data link. This indicates that there is a carrier tone from the central office.

To confirm that there is a connection between the integration modem and the Unity server

Confirm that the request-to-send (RTS) light is lit on the integration modem when Unity is running with the SMDI option enabled.

If the RTS light is not lit, perform one or more of the following corrective actions:

- ◆ In the Unity Administrator, go to System > Switch to confirm that the "Model" box shows a Centrex SMDI model.
- ◆ Confirm that the modem is connected to the correct COM port.
- ◆ Replace the RS-232 serial cable that connects the Unity server to the integration modem.
- ◆ Confirm that the integration modem is set up according to the manufacturer's requirements.

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Integration overview

The steps that you perform to integrate Unity™ with Cisco CallManager vary depending on the version of the Windows operating system that is installed on the Unity server.

Integration steps

Follow these steps to set up this integration.

- 1 Review the system and equipment requirements to verify that all Cisco CallManager and Unity system requirements have been met. See “Requirements,” on page 11.
- 2 Install the Cisco CallManager software. See “To install Cisco CallManager,” on page 15.
- 3 Disable the Cisco CallManager call waiting feature. See “To disable call waiting,” on page 15.
- 4 Add TAPI phone lines to Cisco CallManager. See “To add TAPI phone lines to Cisco CallManager,” on page 16.
- 5 Install the Cisco TAPI service provider on the Unity server. See “To install the Cisco TSP telephony driver,” on page 18.
- 6 Configure the Cisco TAPI service provider telephony driver on the Unity server.
 - ◆ For Microsoft Windows NT® Server, see “To configure the Cisco TSP telephony driver (Windows NT),” on page 19.
 - ◆ For Microsoft Windows® 2000 Server, see “To configure the Cisco TSP telephony driver (Windows 2000),” on page 21.
- 7 Install the Cisco wave driver on the Unity server.
 - ◆ For Windows NT, see “To install the Cisco wave driver (Windows NT),” on page 20.
 - ◆ For Windows 2000, see “To install the Cisco wave driver (Windows 2000),” on page 22.
- 8 Set the switch type in Unity. See “To confirm that the integration is enabled,” on page 23.

Requirements

- ◆ Unity installed and ready for the integration as described in the *Unity Installation Guide*.
- ◆ A system key with the integration type set to “None” and with the appropriate number of voice messaging ports enabled.

- ◆ A Cisco AT-8 analog gateway, a Cisco DT-24 digital gateway, or a Cisco DT-24+ digital gateway.

NOTE: For the latest information on supported Cisco gateways or routers for Unity, contact your sales representative.

- ◆ Cisco CallManager software, version 2.4(3) or later versions that also include TAPI support.
- ◆ Cisco licenses for all phone lines, IP phones, and other H.323-compliant devices or software (such as Cisco Virtual-Phone and Microsoft NetMeeting clients) that will be connected to the network, as well as one license for each Unity port.
- ◆ IP phones or other devices or software.

NOTE: For the current release, IP lines cannot be mixed with analog lines. For more information, see “Converting from analog lines to IP lines,” on page 32.

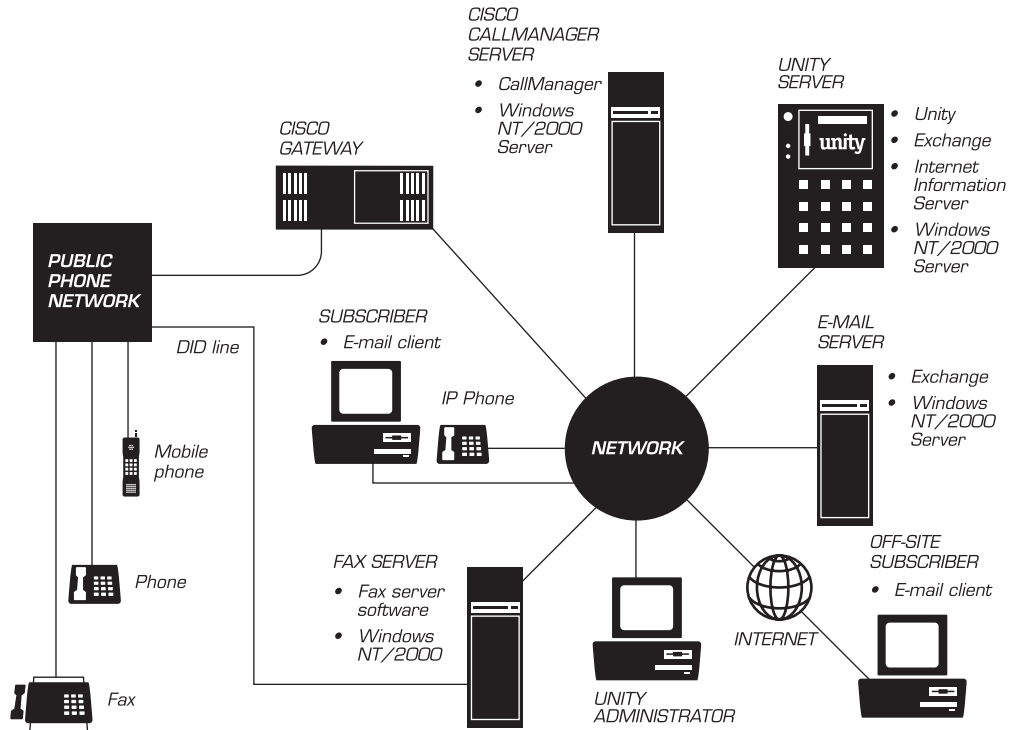
- ◆ A LAN connection in each location where you will plug an IP phone into the network.

Integration description

Components of a Unity installation with Cisco integration

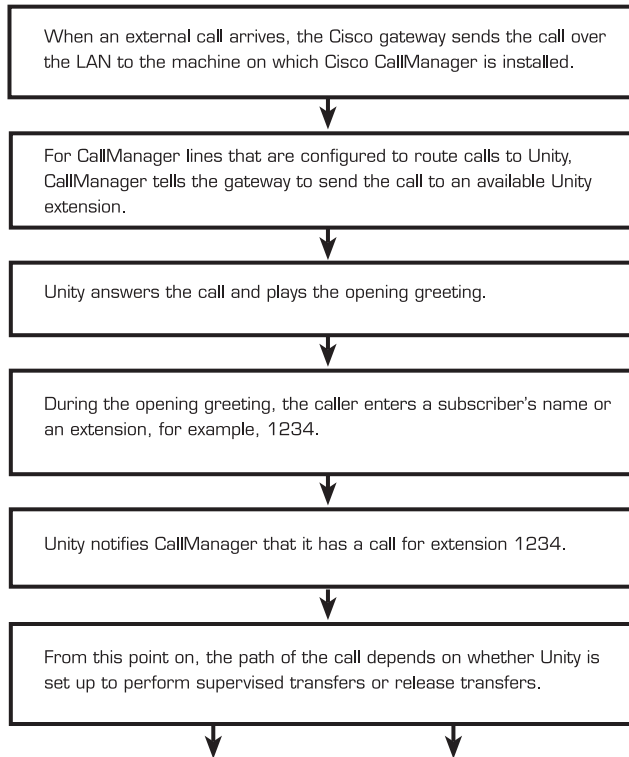
The following illustration shows a full-featured Unity installation integrating with the Cisco Architecture for Voice, Video and Integrated Data (AVVID) network.

NOTE: Some countries require a phone system between the public phone network and the gateway.



How the integration works

The following is an overview of the path an external call takes through the Cisco AVVID network. The call travels from the Cisco gateway or router and CallManager software, through Unity, and on to a subscriber's extension.



Supervised Transfers

While Unity holds the call, CallManager attempts to establish a connection with extension 1234:

- If the line is available, CallManager connects the call from Unity to extension 1234. CallManager and Unity drop out of the loop, and the call is connected directly from the Cisco gateway to extension 1234.
- If the line is busy or unanswered, CallManager gives that information to Unity, and Unity performs the specified operation, for example, it takes a message.

Release Transfers

Unity passes the call to CallManager and CallManager sends the call to extension 1234 without waiting to determine if the line is available. Then CallManager and Unity drop out of the loop. If Unity is set up to perform release transfers, and if you want Unity to take a message when a line is busy or unanswered, you must configure each phone to forward calls to Unity when the line is busy or unanswered.

Continued

Integration description

Continued

Integration features

The Cisco CallManager integration offers the following features:

Call forward to personal greeting When an incoming call is routed to an unanswered or busy extension, the call is forwarded to the subscriber's voice mail. The caller then hears the subscriber's personal greeting and can leave a message.

Caller ID Unity receives caller ID information from Cisco CallManager (if available). This information displays in the message's subject line in Microsoft Outlook® (or other desktop messaging application).

Easy message access A subscriber can retrieve messages without entering an ID. Unity identifies a subscriber based on the extension from which the call originated. A password may be required.

Identified subscriber messaging Unity automatically identifies a subscriber who leaves a message during a forwarded internal call, based on the extension from which the call originated.

Message waiting indication (MWI) When a message is waiting for a subscriber, Unity notifies Cisco CallManager to activate the message waiting indicator on the subscriber's extension. (For Cisco IP phones only.)

Installing and setting up Cisco CallManager

To install Cisco CallManager

- 1 Insert the Cisco CallManager Installation compact disc into the CD-ROM drive on the CallManager server.
- 2 If the startup program starts automatically, follow the screen prompts. Otherwise, run setup.exe from the Cisco CallManager Installation compact disc.
- 3 After the installation is complete, display the Cisco online Help. Click Start > Programs > Cisco CallManager 2.4 > Cisco Help > System Guide. The Cisco Help appears in Internet Explorer.
- 4 In the left frame, click Getting Started > Installing.
- 5 Follow the procedures for installing and configuring Cisco CallManager. Do not perform the last step, which instructs you to start CallManager and Cisco TFTP, until after you have performed the following procedures:
 - ◆ “To disable call waiting,” below.
 - ◆ “To add TAPI phone lines to Cisco CallManager,” on page 16.

CAUTION: Do not install the Cisco Messaging Interface. It can cause Unity to operate improperly.

To disable call waiting

- 1 On the Cisco CallManager server, click Start > Programs > Cisco CallManager 2.4 > CallManager Administration.
- 2 On the Cisco CallManager Administration toolbar, click “Configuration.” The System screen appears.
- 3 Scroll down to the System Parameters area, and click “Configure.”
- 4 Confirm that Call Waiting Enable is set to “Off.” If not, click “Off,” and then click “Update” to save the change.

NOTE: Call waiting must be disabled for Unity’s TAPI ports, but call waiting can be enabled for other devices, if required.
- 5 Go to the next procedure, “To add TAPI phone lines to Cisco CallManager,” on page 16.

Continued

Installing and setting up Cisco CallManager

Continued

To add TAPI phone lines to Cisco CallManager

- ❶ On the Cisco CallManager server, click Start > Programs > Cisco CallManager 2.4 > CallManager Administration.
- ❷ On the Cisco CallManager Administration toolbar, click “Configuration.” The System screen appears.
- ❸ Click “Device Wizard.” The Device Wizard–New screen appears.
- ❹ Follow the screen prompts, and enter the values shown in the table on the following page.

NOTE: If you expect to need more TAPI lines in the future, and if you have enough licenses available, you should add the additional lines now. Adding them now will not adversely affect the system, and the process of adding them later is difficult and time-consuming.

- ❺ When you have finished entering values, the Device Wizard–Confirmation screen appears. Click “Finish.” The Device Wizard–Finish screen appears.
- ❻ On the Device Wizard–Finish screen, click “OK, reset now.”
- ❼ Click “Continue” to confirm that you want to reset the TAPI port now.
- ❽ After receiving confirmation that the device was reset, click “Continue.”
- ❾ Repeat steps 4 through 8 for each TAPI line.
- ❿ On the Cisco CallManager Administration toolbar, click “Configuration,” and then click “Cisco IP Phone.” The Phone list shows the TAPI lines that you have added, as well as any IP phones and other devices that you have added.
- ⓫ Click the name of the TAPI line you added. A Device Information screen appears, showing the values you entered for that line. Use the Device Information screen to correct any errors.
- ⓬ Write down the number of TAPI lines that you added. You will need this information later when you configure the Cisco TAPI service provider (TSP) telephony driver on the Unity server.

- 13 Close Cisco CallManager Administration, then go to the next procedure, “To install the Cisco TSP telephony driver,” on page 18.

Field	Value
Device type	Click “TAPI port.”
Device name	Type a name for the TAPI port, for example, <code>tapi_01</code> .
Description	Optional. Type a brief description of the purpose, for example, <code>Customer Service</code> .
Device pool	Click “Default pool.”
Location	Click “Hub.”
How many lines do you want to assign to the device?	Type 1.
Directory number	Type the extension that will be connected to Unity.
Display	Type the number that appears in a phone’s display when an external call arrives on this line.
Forward all	Leave this field blank. <i>CAUTION: Do not set a value for this field. If information appears in this field, Unity may operate improperly.</i>
Forward busy	Type the extension of the next TAPI port in sequence. If this extension is the last TAPI port, then type the extension of the first TAPI port. If the directory number for this TAPI port is busy, calls are forwarded to the extension shown in this field.
Forward no answer	Type the extension of the next TAPI port in sequence. If this extension is the last TAPI port, then type the extension of the first TAPI port. If the directory number for this TAPI port is not answered, calls are forwarded to the extension shown in this field. Forward no answer is set in the event the port is disabled or out of service.

Configuring Unity for the integration

Perform the following procedures in the order given. The procedures for configuring the Cisco TSP and installing the Cisco wave driver are different for Windows NT and Windows 2000.

If upgrading to a new version of the Cisco TSP, you must prepare the Unity server before installing the new TSP. See “Preparing Unity for upgrading the Cisco TSP,” on page 24.

To install the Cisco TSP telephony driver

- ❶ On the Unity server, shut down Unity.
- ❷ Insert the Cisco CallManager Installation compact disc into the CD-ROM drive on the Unity server.
- ❸ In Windows Explorer, navigate to Cisco CM<version number>\Installs.
- ❹ Double-click “CiscoTsp.exe.”
- ❺ Follow the on-screen instructions, but do not restart the server when you are given that option.
- ❻ Remove the Cisco CallManager Installation compact disc from the CD-ROM drive.
- ❼ If the Unity server is using Windows NT, go to the next procedure, “To configure the Cisco TSP telephony driver (Windows NT),” on page 19.

If the Unity server is using Windows 2000, go to the procedure, “To configure the Cisco TSP telephony driver (Windows 2000),” on page 21.

To configure the Cisco TSP telephony driver (Windows NT)

- ❶ On the Unity server, click Start > Settings > Control Panel.
- ❷ Double-click “Telephony.” The Dialing Properties dialog box appears.
- ❸ Click the Telephony Drivers tab.
- ❹ Click “Cisco IP PBX Service Provider,” and click “Configure.”
- ❺ In the Cisco IP PBX Service Provider dialog box, enter the values as shown in the table below.
- ❻ Click “OK” to close the Cisco IP PBX Service Provider dialog box.
- ❼ Click “Close” to close the Dialing Properties dialog box. The following message appears:

The “Audio for Cisco Sound System” sound driver must be removed and reinstalled for the updated call requirement changes to take effect.
- ❽ Click “OK,” and go to the next procedure, “To install the Cisco wave driver (Windows NT),” on page 20.

Field	Value
Desired number of possible Automated Voice lines	Type the exact number of lines that you added to CallManager Administration in the procedure “To add TAPI phone lines to Cisco CallManager,” on page 16.
Enumerate only lines which support Automated Voice	Select this check box.
All phones and gateways perform silence suppression	If you are using both Cisco phones and the Cisco gateway, select this check box. Otherwise, clear this check box.
Desired computer name	Type the UNC name or IP address of the Cisco CallManager server.
Validate	Click the Validate button to ensure that the Cisco TAPI Service provider can communicate with the Cisco CallManager server. If the Unity server is able to contact Cisco CallManager, a confirmation will be displayed.

Continued

Configuring Unity for the integration

Continued

To install the Cisco wave driver (Windows NT)

- ❶ On the Unity server, click Start > Settings > Control Panel.
- ❷ Double-click “Multimedia.” The Multimedia Properties dialog box appears.
- ❸ Click the Devices tab.
- ❹ Under “Multimedia Drivers,” expand “Audio Devices.”
- ❺ Click “Add.” The Add dialog box appears.
- ❻ Click “Unlisted or updated driver,” and click “OK.” The Install Driver dialog box appears.
- ❼ Click “Browse.” On the drive that contains the Program Files folder, navigate to Program Files\Cisco\WaveDrivers.
- ❽ Click “OK.” The Install Driver dialog box reappears.
- ❾ Click “OK.” The Add Unlisted or Updated Driver dialog box appears. “Cisco TAPI Wave Driver” is the only item listed.
- ❿ Click “OK.” The following message appears:

The required Avaudio32.dll driver is already on the system. Do you want to use the current driver or install a new driver?
- ⓫ Click “New.”
- ⓬ If the message “Your driver settings have changed. You will need to exit and restart Windows NT so that the new settings can take effect” appears, click “Don’t restart now.”
- ⓭ Click “OK” to close the Multimedia Properties dialog box.
- ⓮ Go to the procedure, “To confirm that the integration is enabled,” on page 23.

To configure the Cisco TSP telephony driver (Windows 2000)

- ❶ On the Unity server, click Start > Settings > Control Panel.
- ❷ Double-click “Phone and Modem Options.” The Phone and Modem Options dialog box appears.
- ❸ Click the Advanced tab.
- ❹ Click “Cisco IP PBX Service Provider,” and click “Configure.”
- ❺ In the Cisco IP PBX Service Provider dialog box, enter the values as shown in the table below.
- ❻ Click “OK” to close the Cisco IP PBX Service Provider dialog box. The following message appears:

The “Audio for Cisco Sound System” sound driver must be removed and reinstalled for the updated call requirement changes to take effect.
- ❼ Click “Close” to close the Phone and Modem Options dialog box.
- ❽ Click “OK,” and go to the next procedure, “To install the Cisco wave driver (Windows 2000),” on page 22.

Field	Value
Desired number of possible Automated Voice lines	Type the exact number of lines that you added to CallManager Administration in the procedure “To add TAPI phone lines to Cisco CallManager,” on page 16.
Enumerate only lines which support Automated Voice	Select this check box.
All phones and gateways perform silence suppression	If you are using both Cisco phones and the Cisco gateway, select this check box. Otherwise, clear this check box.
Desired computer name	Type the UNC name or IP address of the Cisco CallManager server.
Validate	Click the Validate button to ensure that the Cisco TAPI Service provider can communicate with the Cisco CallManager server. If the Unity server is able to contact Cisco CallManager, a confirmation will be displayed.

Continued

Configuring Unity for the integration

Continued

To install the Cisco wave driver (Windows 2000)

- ❶ On the Unity server, click Start > Settings > Control Panel.
- ❷ Double-click “Add/Remove Hardware.” The Add/Remove Hardware Wizard Welcome window appears.
- ❸ Click “Next” to continue. The Choose a Hardware Task dialog box appears.
- ❹ Click “Add/Troubleshoot a device,” and then click “Next” to continue. The Choose a Hardware Device dialog box appears.
- ❺ Click “Add a new device,” and then click “Next” to continue. The Find New Hardware dialog box appears.
- ❻ Click “No, I want to select the hardware from a list,” and then click “Next” to continue. The Hardware Type dialog box appears.
- ❼ Click “Sound, video and game controllers,” and then click “Next” to continue. The Select a Device Driver dialog box appears.
- ❽ Click “Have Disk.” The Install from Disk dialog box appears.
- ❾ Click “Browse.” On the drive that contains the Program Files folder, navigate to Program Files\Cisco\Wavedrivers.
- ❿ Click “Oemsetup.inf,” and then click “Open.” The Install From Disk dialog box reappears.
- ⓫ In the Install From Disk dialog box, click “OK.”
- ⓬ The message “Digital signature not found” appears. Click “Yes” to continue the installation. The Select a Device Driver dialog box appears.
- ⓭ In the Device Driver list, click “Cisco TAPI Wave Driver,” and then click “Next” to continue. The Start Hardware Installation dialog box appears.
- ⓮ In the Start Hardware Installation dialog box, click “Next” to continue. The message “Digital signature not found” appears. Click “Yes” to continue the installation.
- ⓯ The message “The Cisco TAPI Wave Driver has been installed” appears. Click “Finish.”

- 16 If the message “You must restart your computer before the new settings will take effect. Do you want to restart your computer now?” appears, click “No.”
- 17 Go to the next procedure, “To confirm that the integration is enabled,” below.

To confirm that the integration is enabled

- 1 On the Unity server, start Unity.
- 2 In Internet Explorer, go to the Unity Administrator (<http://<server name>/saweb>).
- 3 Go to System > Licensing > Licensed Features.
- 4 In the “Integration” box, confirm that the setting is “None.”
If the setting is not “None,” contact your sales representative for the necessary system key.
- 5 Go to System > Switch. Confirm that the settings match those in the following table.

If the settings are incorrect, integration features may not be enabled.

Switch settings

Switch parameter	Required settings
Manufacturer	Cisco
Model	CallManager
Switch PBX software version	2.4.x
Integration	None

- 6 Close the Unity Administrator.
- 7 Shut down and restart the Unity server.

See also

Shutting down and starting
Unity..... 140

Preparing Unity for upgrading the Cisco TSP

Before upgrading the Cisco TSP on the Unity server, perform the appropriate procedures in this section. After the Cisco TSP and wave driver are removed from the Unity server, install and configure the new Cisco TSP and wave driver. See “Configuring Unity for the integration,” on page 18.

The steps that you must perform to remove the Cisco TSP and wave driver vary depending on the version of the Windows operating system that is installed on the Unity server.

- ◆ If the Unity server is using Windows NT, go to the next procedure, “To remove the Cisco TSP (Windows NT),” below.
- ◆ If the Unity server is using Windows 2000, go to the procedure, “To remove the Cisco TSP (Windows 2000),” on page 26.

To remove the Cisco TSP (Windows NT)

- ❶ On the Unity server, click Start > Settings > Control Panel.
- ❷ Double-click “Telephony.” The Dialing Properties dialog box appears.
- ❸ Click the Telephony Drivers tab.
- ❹ Click “Cisco IP PBX Service Provider,” and then click “Remove.”
- ❺ Click “Yes” to confirm that you want to remove the selected telephony driver.
- ❻ Click “Close” to close the Dialing Properties dialog box.
- ❼ Go to the next procedure, “To remove the Cisco wave driver (Windows NT),” below.

To remove the Cisco wave driver (Windows NT)

- ❶ In the Control Panel, double-click “Multimedia.” The Multimedia Properties dialog box appears.
- ❷ Click the Devices tab.
- ❸ Under “Multimedia Drivers,” expand “Audio Devices.”
- ❹ Click “Audio for Cisco TAPI Wave Driver,” and then click “Remove” to remove the Cisco wave driver.
- ❺ Click “Yes” to confirm.

- 6 Click “OK” to close the Multimedia Properties dialog box.
- 7 Shut down and restart the Unity server.
- 8 On the Unity server, install and configure the new Cisco TSP and wave driver. See “Configuring Unity for the integration,” on page 18.

Continued

Preparing Unity for upgrading the Cisco TSP

Continued

To remove the Cisco TSP (Windows 2000)

- 1 On the Unity server, click Start > Settings > Control Panel.
- 2 Double-click "Phone and Modem Options." The Phone and Modem Options dialog box appears.
- 3 Click the Advanced tab.
- 4 Click "Cisco IP PBX Service Provider," and then click "Remove."
- 5 Click "Yes" to confirm that you want to remove the selected telephony service provider.
- 6 Click "Close" to close the Phone and Modem Options dialog box.
- 7 Go to the next procedure, "To remove the Cisco wave driver (Windows 2000)," below.

To remove the Cisco wave driver (Windows 2000)

- 1 In the Control Panel, double-click "Add/Remove Hardware." The Add/Remove Hardware Wizard Welcome window appears.
- 2 Click "Next" to continue. The Choose a Hardware Task dialog box appears.
- 3 Click "Uninstall/Unplug a device," and then click "Next" to continue. The Choose a Removal Task dialog box appears.
- 4 Click "Uninstall a device," and then click "Next" to continue. The Installed Devices on Your Computer dialog box appears.
- 5 In the Device List, click "Cisco TAPI Wave Driver," and then click "Next" to continue. The Uninstall a Device dialog box appears.
- 6 Click "Yes, I want to uninstall this device," and then click "Next" to continue.
- 7 The message "The device has been removed" appears. Click "Finish."
- 8 Shut down and restart the Unity server.
- 9 On the Unity server, install and configure the new Cisco TSP and wave driver. See "Configuring Unity for the integration," on page 18.

See also

Shutting down and starting
Unity..... 140

Changing the number of TAPI lines

To change the number of TAPI lines after you have finished installing and setting up Cisco CallManager, perform all of the appropriate procedures on pages 27–31. The procedures performed on the Unity server are different for Windows NT and Windows 2000.

To change the number of TAPI lines in Cisco CallManager Administration

- 1 On the Cisco CallManager server, use Cisco CallManager Administration to add or remove TAPI lines as appropriate. When you have finished adding or removing lines, write down the total number of TAPI lines configured in Cisco CallManager Administration.

For information on adding TAPI lines, see “To add TAPI phone lines to Cisco CallManager,” on page 16. For information on removing TAPI lines, see the Cisco CallManager Administration online Help.

- 2 On the Unity server, shut down Unity.
- 3 If the Unity server is using Windows NT, go to the next procedure, “To change the number of lines specified for the Cisco telephony driver (Windows NT),” on page 28.

If the Unity server is using Windows 2000, go to the procedure, “To change the number of lines specified for the Cisco telephony driver (Windows 2000),” on page 30.

Continued

Changing the number of TAPI lines

Continued

To change the number of lines specified for the Cisco telephony driver (Windows NT)

- ❶ On the Unity server, click Start > Settings > Control Panel.
- ❷ Double-click "Telephony." The Dialing Properties dialog box appears.
- ❸ Click the Telephony Drivers tab.
- ❹ In the list of installed telephony drivers, click "Cisco IP PBX Service Provider," and click "Configure." The Cisco IP PBX Service Provider dialog box appears.
- ❺ Change "Desired number of possible Automated Voice lines" to the exact number of lines currently configured in Cisco CallManager Administration, as noted in step 1 of the procedure, "To change the number of TAPI lines in Cisco CallManager Administration," on page 27.
- ❻ Click "Close" to close the Dialing Properties dialog box. The following message appears:

The "Audio for Cisco Sound System" sound driver must be removed and reinstalled for the updated call requirement changes to take effect.
- ❼ Click "OK" to continue.
- ❽ Go to the next procedure, "To remove and reinstall the Cisco wave driver (Windows NT)," on page 29.

To remove and reinstall the Cisco wave driver (Windows NT)

- ❶ On the Unity server, if the Control Panel is no longer open, click Start > Settings > Control Panel.
- ❷ Double-click "Multimedia." The Multimedia Properties dialog box appears.
- ❸ Click the Devices tab.
- ❹ Under "Multimedia Drivers," expand "Audio Devices."
- ❺ Click "Audio for Cisco TAPI Wave Driver," and then click "Remove" to remove the Cisco wave driver.
- ❻ Click "Yes" to confirm.
- ❼ Click "OK" to close the Multimedia Properties dialog box.
- ❽ In some cases, this procedure fails to remove the Cisco wave driver. To confirm that the wave driver is gone, repeat steps 2 through 4. If "Audio for Cisco TAPI Wave Driver" still appears in the "Audio Devices" list, contact Technical Support.

CAUTION: The Cisco wave driver must be removed before you continue, or the integration will not work.

- ❾ On the Unity server, repeat the procedures "To configure the Cisco TSP telephony driver (Windows NT)," on page 19 and "To install the Cisco wave driver (Windows NT)," on page 20.
- ❿ Shut down and restart the Unity server.

Continued

Changing the number of TAPI lines

Continued

To change the number of lines specified for the Cisco telephony driver (Windows 2000)

- ❶ On the Unity server, click Start > Settings > Control Panel.
- ❷ Double-click "Phone and Modem Options." The Phone and Modem Options dialog box appears.
- ❸ Click the Advanced tab.
- ❹ In the list of installed telephony drivers, click "Cisco IP PBX Service Provider," and click "Configure." The Cisco IP PBX Service Provider dialog box appears.
- ❺ Change "Desired number of possible Automated Voice lines" to the exact number of lines currently configured in Cisco CallManager Administration, as noted in step 1 of the procedure, "To change the number of TAPI lines in Cisco CallManager Administration," on page 27.
- ❻ Click "OK" to close the Cisco IP PBX Service Provider dialog box. The following message appears:

The "Audio for Cisco Sound System" sound driver must be removed and reinstalled for the updated call requirement changes to take effect.
- ❼ Click "Close" to close the Phone and Modem Options dialog box.
- ❽ Go to the next procedure, "To remove and reinstall the Cisco wave driver (Windows 2000)," on page 31.

To remove and reinstall the Cisco wave driver (Windows 2000)

- ❶ On the Unity server, if the Control Panel is no longer open, click Start > Settings > Control Panel.
- ❷ Double-click "Add/Remove Hardware." The Add/Remove Hardware Wizard Welcome window appears.
- ❸ Click "Next" to continue. The Choose a Hardware Task dialog box appears.
- ❹ Click "Uninstall/Unplug a device," and then click "Next" to continue. The Choose a Removal Task dialog box appears.
- ❺ Click "Uninstall a device," and then click "Next" to continue. The Installed Devices on Your Computer dialog box appears.
- ❻ In the Device List, click "Cisco TAPI Wave Driver," and then click "Next" to continue. The Uninstall a Device dialog box appears.
- ❼ Click "Yes, I want to uninstall this device," and then click "Next" to continue.
- ❽ The message "The device has been removed" appears. Click "Finish."
- ❾ On the Unity server, repeat the procedures "To configure the Cisco TSP telephony driver (Windows 2000)," on page 21 and "To install the Cisco wave driver (Windows 2000)," on page 22.
- ❿ Shut down and restart the Unity server.

See also

Shutting down and starting
Unity..... 140

Converting from analog lines to IP lines

To convert an existing Unity system from analog lines to IP lines, it is necessary to upgrade the system key to enable the integration. Contact your sales representative for the necessary system key. To confirm that the integration is enabled, perform the procedure “To confirm that the integration is enabled,” on page 23.

After upgrading the system key, perform all the procedures under “Installing and setting up Cisco CallManager,” on pages 15–17, and “Configuring Unity for the integration,” on pages 18–23. Then remove the TSP telephony driver for the voice boards that are installed in the Unity server, by performing the procedure that is applicable for the version of Windows installed on the Unity server:

- ◆ For Windows NT, go to the procedure, “To remove the TSP telephony driver for voice boards (Windows NT),” on page 33.
- ◆ For Windows 2000, go to the procedure, “To remove the TSP for voice boards (Windows 2000),” on page 33.

If you want to remove the voice boards from the Unity server, also remove the wave driver for the voice boards. Perform the procedure that is applicable for the version of Windows installed on the Unity server:

- ◆ For Windows NT, go to the procedure, “To remove the wave driver for voice boards (Windows NT),” on page 34.
- ◆ For Windows 2000, go to the procedure, “To remove the wave driver for voice boards (Windows 2000),” on page 35.

If you remove Dialogic voice boards, set the Dialogic service to start manually rather than automatically. This setting will prevent Dialogic errors from appearing in the Windows event log. Go to the procedure, “To set the Dialogic service to start manually,” on page 36.

To remove the TSP telephony driver for voice boards (Windows NT)

- 1** On the Unity server, click Start > Settings > Control Panel.
- 2** Double-click "Telephony." The Dialing Properties dialog box appears.
- 3** Click the Telephony Drivers tab.
- 4** In the list of telephony drivers, click the driver for your voice boards, and then click "Remove."
- 5** Click "Yes" to confirm that you want to remove the selected telephony driver.
- 6** Click "Close" to close the Dialing Properties dialog box.

To remove the TSP for voice boards (Windows 2000)

- 1** On the Unity server, click Start > Settings > Control Panel.
- 2** Double-click "Phone and Modem Options." The Phone and Modem Options dialog box appears.
- 3** Click the Advanced tab.
- 4** In the list of telephony service providers, click the telephony service provider for your voice boards, and then click "Remove."
- 5** Click "Yes" to confirm that you want to remove the selected telephony service provider.
- 6** Click "Close" to close the Phone and Modem Options dialog box.

Continued

Converting from analog lines to IP lines

Continued

To remove the wave driver for voice boards (Windows NT)

- ➊ On the Unity server, click Start > Settings > Control Panel.
- ➋ Double-click “Multimedia.” The Multimedia Properties dialog box appears.
- ➌ Click the Devices tab.
- ➍ Under “Multimedia Drivers,” expand “Audio Devices.”
- ➎ Click the wave driver for your voice boards, and then click “Remove.”
- ➏ Click “Yes” to confirm the removal of the wave driver.
- ➐ Click “OK” to close the Multimedia Properties dialog box.
- ➑ Shut down and restart the Unity server.

To remove the wave driver for voice boards (Windows 2000)

- ❶ On the Unity server, click Start > Settings > Control Panel.
- ❷ Double-click "Sounds and Multimedia." The Sounds and Multimedia Properties dialog box appears.
- ❸ Click the Hardware tab.
- ❹ Double-click "Legacy Audio Drivers." The Legacy Audio Drivers Properties dialog box appears.
- ❺ Click the Properties tab.
- ❻ Expand "Audio Devices."
- ❼ Click the wave driver for your voice boards, and then click "Remove."
- ❽ Click "Yes" to confirm the removal of the wave driver.
- ❾ Click "OK" to close the Sounds and Multimedia Properties dialog box.
- ❿ Shut down and restart the Unity server.

Continued

Converting from analog lines to IP lines

Continued

To set the Dialogic service to start manually

- 1 On the Unity server, click Start > Programs > Dialogic System Software > Dialogic Configuration Manager-DCM.
- 2 On the Dialogic Configuration Manager menu, click Service > Startup Mode > Manual.
- 3 Close the Dialogic Configuration Manager.

See also

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■ FUJITSU 9600/SMDI

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Integration overview

Use this section of the *Unity Integration Guide* with the *Unity Installation Guide*.

Integration steps

Follow these steps to set up this integration.

- 1 Review the system and equipment requirements to verify that all phone system and Unity™ server requirements have been met. See “Requirements,” below.
- 2 Configure Unity for the integration. See “Configuring Unity for the integration,” on page 42.
- 3 Program the phone system and extensions. See “Programming the phone system,” on page 43.

Requirements

The steps to set up the Fujitsu Simplified Message Desk Interface (SMDI) integration require the following:

Phone system

- ◆ Fujitsu 9600 with an SMDI port installed.
- ◆ The phone system’s SMDI port connected to a serial port (COM1 is the default) on the Unity server with an RS-232 serial cable.
- ◆ The phone system’s voice messaging ports connected to the voice boards in the Unity server.
- ◆ The phone system ready for the integration as described in the phone system’s installation guide.

Unity server

- ◆ The Dialogic® D/41D™, D/41ESC™, D/41EPCI™, D/120JCT-LS™, or D/160SC-LS™ voice boards, installed.

NOTE: For the latest information on supported voice boards for Unity, contact your sales representative.

- ◆ Unity installed and ready for the integration as described in the *Unity Installation Guide*.

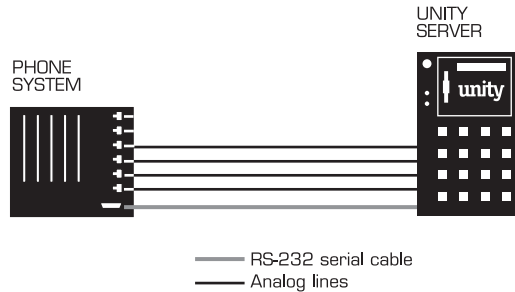
NOTE: To ensure that Windows does not mistake the serial connection for a serial mouse when rebooting the Unity server, confirm that the boot.ini file is set according to instructions. See "Installing Windows NT Server" in Chapter 3 of the Unity Installation Guide.

- ◆ A system key that enables the integration and the appropriate number of voice messaging ports.
- ◆ An available serial port (COM1 is the default).

Integration description

How the integration works

The Fujitsu 9600/SMDI integration uses a data link, which consists of an RS-232 serial cable connecting the phone system and the Unity server. The phone system's voice messaging lines connect to the analog voice boards in the Unity server. The following illustration shows the required connections.



NOTE: Serial cables longer than 50 feet require a modem.

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

- ◆ The called party's extension
- ◆ The reason for the forward (the extension is busy, does not answer, or is set to forward all calls)
- ◆ The calling party's extension (for internal calls)

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

Integration features

The Fujitsu 9600/SMDI integration with Unity provides the following features:

Call forward to personal greeting When an incoming call is routed to an unanswered or busy extension, the call is forwarded to the subscriber's voice mail. The caller then hears the subscriber's personal greeting and can leave a message.

Caller ID Unity receives caller ID information from the phone system (if available). This information displays in the message's subject line in Microsoft Outlook® (or other desktop messaging application).

Easy message access A subscriber can retrieve messages without entering an ID. Unity identifies a subscriber based on the extension from which the call originated. A password may be required.

Identified subscriber messaging Unity automatically identifies a subscriber who leaves a message during a forwarded internal call, based on the extension from which the call originated.

Message waiting indication (MWI) When a message is waiting for a subscriber, Unity notifies the phone system to activate the message waiting indicator on the subscriber's extension (if it is equipped with one) or to activate a stutter dial tone.

Configuring Unity for the integration

After ensuring that the Unity server is ready for the integration by completing the appropriate tasks in Chapters 1 through 4 of the *Unity Installation Guide*, perform the following procedure to confirm that the integration is enabled.

To confirm that the integration is enabled

Before beginning this procedure, complete all procedures in the section titled “Configuring Unity to work with the phone system,” in Chapter 4 of the *Unity Installation Guide*.

- 1 In Internet Explorer, go to the Unity Administrator (<http://<server name>/saweb>).
- 2 Go to System > Licensing > Licensed Features.
- 3 In the “Integration” box, confirm that the setting is “Serial.”
- 4 If the setting is not “Serial,” contact your sales representative for the necessary system key.
- 5 Go to System > Switch. Confirm that the settings match those in the following table.

If the settings are incorrect, integration features may not be enabled.

Switch settings

Switch parameter	Required settings
Manufacturer	Fujitsu
Model	9600
Switch PBX software version	All
Integration	Serial

Programming the phone system

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

To program the phone system

- ➊ Assign a DCC port for integration with Unity.
- ➋ On the phone system, in "Assign single line phone," set Type to "1" for the Fujitsu 9600 phone lines used as voice messaging ports.
- ➌ In "Change service parameter," set Type to "2" and ID:01 to "01" for the Fujitsu 9600 phone lines used as voice messaging ports.

NOTE: The settings in step 3 disable the immediate ring feature.

■ LUCENT/CALISTA AND NORTEL/CALISTA

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Integration overview

Use this section of the *Unity Integration Guide* with the *Unity Installation Guide*.

Integration steps

Follow these steps to set up this integration.

- 1 Review the system and equipment requirements to verify that all phone system and Unity™ server requirements have been met. See “Requirements,” below.
- 2 Configure Unity for the integration. See “Configuring Unity for the integration,” on page 53.
- 3 Program the phone system and extensions. See “Programming the phone system,” on page 54.
- 4 Install and configure the Calista™ PBXLink™ box. See “Setting up the PBXLink,” on page 55.

Requirements

The steps to set up the Lucent/Calista or Nortel/Calista integrations require the following:

Lucent phone systems

- ◆ One of the following phone systems with one or two digital lines connected to the PBX ports on the PBXLink box that are compatible with one of the following digital phones:

Phone system	Digital phone
Lucent Definity G1-G4	7434 or 8434
Lucent System 75/85	7405

- ◆ For each voice messaging port, one analog port connected to the voice boards in the Unity server.

See also

Calista PBXLink User Guide

- ◆ The PBXLink box's SMDI port connected to a serial port (COM1 is the default) on the Unity server with an RS-232 serial cable.

If multiple PBXLink boxes are installed, the SMDI port of the last PBXLink box is connected to the Management port of the first PBXLink box with an RS-232 serial cable. The SMDI port of the first PBXLink box is connected to a serial port (COM1 is the default) on the Unity server with another RS-232 serial cable.

NOTE: If multiple PBXLink boxes are installed, ensure that the serial protocol parameters for the Management port of the first PBXLink box are set to the same values as the SMDI port.

- ◆ The phone system ready for the integration as described in the phone system's installation guide.

Nortel phone system

- ◆ Nortel Meridian 1 with the following software option packages installed:

Option package	Package number
EES	10
MSB	17
DDSP	19
MWC	46
DSET	88
CPND	95
ARIE	170

- ◆ One or two digital lines that are compatible with the M2616 digital phone connected to the PBX ports on the PBXLink box.
- ◆ For each voice messaging port, one analog port connected to the voice boards in the Unity server.

Continued

Integration overview

Continued

- ◆ The PBXLink box's SMDI port connected to a serial port (COM1 is the default) on the Unity server with an RS-232 serial cable.

If multiple PBXLink boxes are installed, the SMDI port of the last PBXLink box is connected to the Management port of the first PBXLink box with an RS-232 serial cable. The SMDI port of the first PBXLink box is connected to a serial port (COM1 is the default) on the Unity server with another RS-232 serial cable.

NOTE: If multiple PBXLink boxes are installed, ensure that the serial protocol parameters for the Management port of the first PBXLink box are set to the same values as the SMDI port.

- ◆ The phone system ready for the integration as described in the phone system's installation guide.

Unity server

- ◆ The Dialogic® D/41D™, D/41ESC™, D/41EPCI™, D/120JCT-LS™, or D/160SC-LS™ voice boards, installed.

NOTE: For the latest information on supported voice boards for Unity, contact your sales representative.

- ◆ Unity installed and ready for the integration as described in the *Unity Installation Guide*.

NOTE: To ensure that Windows does not mistake the serial connection for a serial mouse when rebooting the Unity server, confirm that the boot.ini file is set according to instructions. See "Installing Windows NT Server" in Chapter 3 of the Unity Installation Guide.

- ◆ A system key that enables the integration and the appropriate number of voice messaging ports.
- ◆ An available serial port (COM1 is the default).

Integration description

How the integrations work

The Lucent Definity, System 75/85, and Nortel Meridian 1 integrations use one or two Calista PBXLink boxes that each emulate up to two Lucent 7405, 7434 or 8434 digital phones, or Nortel M2616 digital phones. The PBXLink is connected to the phone system with digital phone lines and connected to the Unity server with an RS-232 serial cable. The phone system's voice messaging lines connect to the analog voice boards in the Unity server.

The integrations work in either bridged mode or transfer mode. Transfer mode must be used for the Lucent System 75 phone system. Bridged mode should be used for the other integrations.

In either mode, the PBXLink box receives the following call information from the phone system:

- ◆ The called party's extension
- ◆ The reason for the forward (the extension is busy, does not answer, or is set to forward all calls)
- ◆ The calling party's extension (for internal calls)

The PBXLink box formats this information as a Simplified Message Desk Interface (SMDI) packet and sends the packet to Unity through the RS-232 serial cable.

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

The PBXLink box also activates or deactivates message waiting indicators after receiving a command from Unity.

Continued

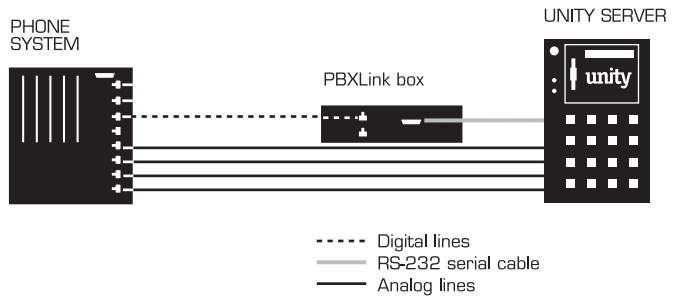
See also

Calista PBXLink User Guide: Introduction, in Chapter 1

Integration description

Continued

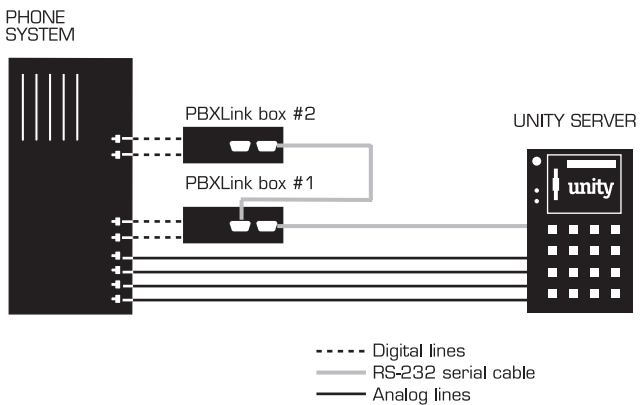
Small Lucent and Nortel systems Lucent integrations with 24 or fewer ports and Nortel integrations with 22 or fewer ports use a single PBXLink box. The PBXLink is connected to the phone system with a digital phone line and connected to the Unity server with an RS-232 serial cable. The phone system's voice messaging lines connect to the analog voice boards in the Unity server. The following illustration shows the required connections.



Large Lucent and Nortel systems Lucent integrations with 25 or more ports and Nortel integrations with 23 or more ports require two PBXLink boxes. Each PBXLink box is connected to the phone system with digital phone lines. The PBXLink boxes are connected to the Unity server by using an RS-232 cable to connect the SMDI port from the last PBXLink box to the Management port of the first PBXLink box, and then connecting the SMDI port of the first PBXLink box to the Unity server. The phone system's voice messaging lines connect to the analog voice boards in the Unity server.

NOTE: When connecting D/160SC-LS voice boards, use all 16 ports on the first and second voice boards and the first 12 ports on the third.

The following illustration shows the required connections.



Continued

Integration description

Continued

Integration features

The Lucent and Nortel integrations with Unity use the Calista PBXLink to provide the following features:

Call forward to personal greeting When an incoming call is routed to an unanswered or busy extension, the call is forwarded to the subscriber's voice mail. The caller then hears the subscriber's personal greeting and can leave a message.

Caller ID Unity receives caller ID information from the phone system (if available). This information displays in the message's subject line in Microsoft Outlook® (or other desktop messaging application).

Easy message access A subscriber can retrieve messages without entering an ID. Unity identifies a subscriber based on the extension from which the call originated. A password may be required.

Identified subscriber messaging Unity automatically identifies a subscriber who leaves a message during a forwarded internal call, based on the extension from which the call originated.

Message waiting indication (MWI) When a message is waiting for a subscriber, Unity notifies the phone system to activate the message waiting indicator on the subscriber's extension (if it is equipped with one) or to activate a stutter dial tone.

Configuring Unity for the integration

After ensuring that the Unity server is ready for the integration by completing the appropriate tasks in Chapters 1 through 4 of the *Unity Installation Guide*, perform the following procedure to confirm that the integration is enabled.

To confirm that the integration is enabled

Before beginning this procedure, complete all procedures in the section titled “Configuring Unity to work with the phone system,” in Chapter 4 of the *Unity Installation Guide*.

- 1 In Internet Explorer, go to the Unity Administrator (<http://<server name>/saweb>).
- 2 Go to System > Licensing > Licensed Features.
- 3 In the “Integration” box, confirm that the setting is “Serial.”
- 4 If the setting is not “Serial,” contact your sales representative for the necessary system key.
- 5 Go to System > Switch. Confirm that the settings match those for your phone system in the following tables.

If the settings are incorrect, integration features may not be enabled.

Switch settings: Lucent

Switch parameter	Required settings
Manufacturer	Lucent
Model	◆ Definity (G1-G4) ◆ System 75 ◆ System 85
Switch PBX software version	All or All (analog MWI)
Integration	Serial

Switch settings: Northern Telecom

Switch parameter	Required settings
Manufacturer	Northern Telecom
Model	Meridian-1
Switch PBX software version	All
Integration	Serial

Programming the phone system

Detailed instructions for programming the phone system are included in the *Calista PBXLink User Guide* that accompanies Unity, as specified below.

To program the Lucent phone system

- 1 Refer to Chapter 4 of the *PBXLink User Guide* to determine whether transfer mode or bridged mode is appropriate for the integration.

- 2 Set the appropriate mode:

Transfer mode Follow the instructions in Chapter 5 of the *PBXLink User Guide*.

Bridged mode Follow the instructions in Chapter 6 of the *PBXLink User Guide*.

To program the Nortel phone system

- ◆ Follow the instructions in Chapter 7 of the *PBXLink User Guide*.

Setting up the PBXLink

Detailed instructions for setting up the PBXLink box are included in the *PBXLink User Guide* that accompanies Unity, as specified below.

To set up the PBXLink box

- ◆ To set up the PBXLink box, follow the instructions in Chapters 2 and 8 of the *PBXLink User Guide*.

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Integration overview

Use this section of the *Unity Integration Guide* with the *Unity Installation Guide*.

Integration steps

Follow these steps to set up this integration.

- 1 Review the system and equipment requirements to verify that all phone system and Unity server requirements have been met. See “Requirements,” below.
- 2 Configure Unity for the integration. See “Configuring Unity for the integration,” on page 62.
- 3 Program the phone system and extensions. See “Programming the phone system,” on page 63.
- 4 Install and configure the Calista™ DMI-4. See “Installing and configuring the DMI-4,” on page 64.

Requirements

The steps to set up the Mitel/Calista DMI-4 integration require the following:

Phone system

One of the following phone systems with the required software installed:

Phone system	Software release
Mitel SX-200 Digital	Release 3.08 or later
Mitel SX-2000	Release 3.08 or later

- ◆ For each voice messaging port, one Digital Network Interface Circuit (DNIC) port, which emulates a Superset 430 digital phone, connected to a digital PBX port on the DMI-4 box.
- ◆ The DMI-4 box’s analog ports connected to the voice boards in the Unity server.

See also

Calista PBXLink ISA 4x4 and DMI-4 Installation Manual

- ◆ The DMI-4 box's SMDI port connected to a serial port (COM1 is the default) on the Unity server with an RS-232 serial cable.

If multiple DMI-4 boxes are installed, the SMDI port of the last DMI-4 box is connected to the Management port of the next DMI-4 box with an RS-232 serial cable. The SMDI port of the second DMI-4 box is connected to a serial port (COM1 is the default) on the Unity server with another RS-232 serial cable.

- ◆ The phone system ready for the integration as described in the phone system's installation guide.

Unity server

- ◆ The Dialogic® D/41D™, D/41ESC™, D/41EPCI™, D/120JCT-LS™, or D/160SC-LS™ voice boards, installed.

NOTE: For the latest information on supported voice boards for Unity, contact your sales representative.

- ◆ Unity installed and ready for the integration as described in the *Unity Installation Guide*.

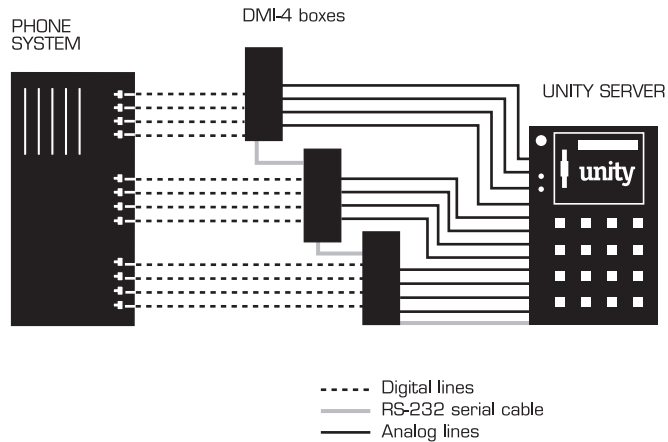
NOTE: To ensure that Windows does not mistake the serial connection for a serial mouse when rebooting the Unity server, confirm that the boot.ini file is set according to instructions. See "Installing Windows NT Server" in Chapter 3 of the Unity Installation Guide.

- ◆ A system key that enables the integration and the appropriate number of voice messaging ports.
- ◆ An available serial port (COM1 is the default).

Integration description

How the integration works

The Mitel SX-200D and SX-2000 integrations use one or more Calista DMI-4 boxes to emulate up to four Mitel Superset 430 digital phones for each DMI-4 box. The integration uses a data link which consists of an RS-232 serial cable connecting one or more Calista DMI-4 boxes and the Unity server. Each DMI-4 box connects to digital phone lines from the phone system, converts the voice signals to analog, then passes them to ports on the analog voice boards in the Unity server. The following illustration shows the required connections.



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

- ◆ The called party's extension
- ◆ The reason for the forward (the extension is busy, does not answer, or is set to forward all calls)
- ◆ The calling party's extension (for internal calls)

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

See also

Calista PBXLink ISA 4x4 and DMI-4 Installation Manual: Introduction, in Chapter 1

Integration features

The Mitel/Calista integration with Unity provides the following features:

Call forward to personal greeting When an incoming call is routed to an unanswered or busy extension, the call is forwarded to the subscriber's voice mail. The caller then hears the subscriber's personal greeting and can leave a message.

Caller ID Unity receives caller ID information from the phone system (if available). This information displays in the message's subject line in Microsoft Outlook® (or other desktop messaging application).

Easy message access A subscriber can retrieve messages without entering an ID. Unity identifies a subscriber based on the extension from which the call originated. A password may be required.

Identified subscriber messaging Unity automatically identifies a subscriber who leaves a message during a forwarded internal call, based on the extension from which the call originated.

Message waiting indication (MWI) When a message is waiting for a subscriber, Unity notifies the phone system to activate the message waiting indicator on the subscriber's extension (if it is equipped with one) or to activate a stutter dial tone.

Configuring Unity for the integration

After ensuring that the Unity server is ready for the integration by completing the appropriate tasks in Chapters 1 through 4 of the *Unity Installation Guide*, perform the following procedure to confirm that the integration is enabled.

To confirm that the integration is enabled

Before beginning this procedure, complete all procedures in the section titled “Configuring Unity to work with the phone system,” in Chapter 4 of the *Unity Installation Guide*.

- 1 In Internet Explorer, go to the Unity Administrator (<http://<server name>/saweb>).
- 2 Go to System > Licensing > Licensed Features.
- 3 In the “Integration” box, confirm that the setting is “Serial.”
- 4 If the setting is not “Serial,” contact your sales representative for the necessary system key.
- 5 Go to System > Switch. Confirm that the settings match those in the following table.

If the settings are incorrect, integration features may not be enabled.

Switch settings

Switch parameter	Required settings
Manufacturer	Mitel
Model	◆ SX-200D ◆ SX-200D ML/EL ◆ SX-2000
Switch PBX software version	All (Calista DMI-4)
Integration	Serial

Programming the phone system

If you use programming options other than those specified in the *Calista PBXLink ISA 4x4 and DMI-4 Installation Manual*, the integration's performance may be affected.

To program the phone system

- ◆ Follow the instructions in Chapter 4 of the *PBXLink ISA 4x4 and DMI-4 Installation Manual*.

Installing and configuring the DMI-4

Detailed instructions for installing and configuring the DMI-4 box are included in the *PBXLink ISA 4x4 and DMI-4 Installation Manual* that accompanies Unity, as specified below.

To install and configure the DMI-4

- ➊ To install the DMI-4 box, follow the instructions in Chapter 3 of the *PBXLink ISA 4x4 and DMI-4 Installation Manual*.
- ➋ To configure the DMI-4, follow the instructions in Chapters 8 and 9 of the *PBXLink ISA 4x4 and DMI-4 Installation Manual*.

■ MITEL SX-2000 T1

In this integration...

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Configuring Unity for the integration	70
Programming the phone system	74

Integration overview

Use this section of the *Unity Integration Guide* with the *Unity Installation Guide*.

Integration steps

Follow these steps to set up this integration.

- 1 Review the system and equipment requirements to verify that all phone system and Unity™ server requirements have been met. See “Requirements,” below.
- 2 Configure Unity for the integration. See “Configuring Unity for the integration,” on page 70.
- 3 Program the phone system and extensions. See “Programming the phone system,” on page 74.

Requirements

The steps to set up the Mitel SX-2000 T1 integration require the following:

Phone system

- ◆ Mitel SX-2000, with release N16 or later installed.
- ◆ Mitel Advanced Analog Networking installed.
- ◆ The phone system’s voice messaging ports connected to the voice boards in the Unity server.

CAUTION: The T1 lines from the phone system must be functioning and connected to the voice boards before starting Unity. Failure to do so may cause initialization problems between the Unity server and the T1 cards in the phone system.

- ◆ The phone system ready for the integration as described in the phone system’s installation guide.

Unity server

- ◆ The Natural MicroSystems™ AG-T1 or AG Dual T1 voice boards, installed.

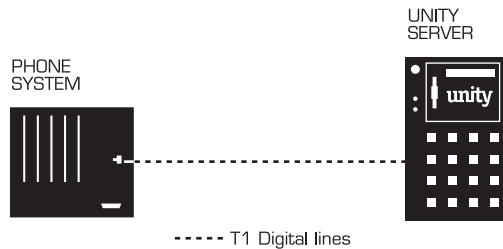
NOTE: For the latest information on supported voice boards for Unity, contact your sales representative.

- ◆ Unity installed and ready for the integration as described in the *Unity Installation Guide*.
- ◆ A system key that enables the integration and the appropriate number of voice messaging ports.

Integration description

How the integration works

The Mitel SX-2000 T1 integration uses a T1 line to connect the phone system and the Unity server. Each T1 line connects 24 voice messaging ports from the phone system's DS1 Formatter board to a T1 voice board in the Unity server. The following illustration shows the required connections.



The phone system sends the following information with forwarded calls:

- ◆ The called party's extension
- ◆ The calling party's extension (for internal calls) or the calling party's phone number (if it is an external call and the system uses caller ID)

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

Integration features

The Mitel SX-2000 T1 integration with Unity provides the following features:

Call forward to personal greeting When an incoming call is routed to an unanswered or busy extension, the call is forwarded to the subscriber's voice mail. The caller then hears the subscriber's personal greeting and can leave a message.

Caller ID Unity receives caller ID information from the phone system (if available). This information displays in the message's subject line in Microsoft Outlook® (or other desktop messaging application).

Easy message access A subscriber can retrieve messages without entering an ID. Unity identifies a subscriber based on the extension from which the call originated. A password may be required.

Identified subscriber messaging Unity automatically identifies a subscriber who leaves a message during a forwarded internal call, based on the extension from which the call originated.

Message waiting indication (MWI) When a message is waiting for a subscriber, Unity notifies the phone system to activate the message waiting indicator on the subscriber's extension (if it is equipped with one) or to activate a stutter dial tone.

Configuring Unity for the integration

After ensuring that the Unity server is ready for the integration by completing the appropriate tasks in Chapters 1 through 4 of the *Unity Installation Guide*, perform the following procedures to confirm that the integration is enabled and that the protocol setting for the T1 board is correct.

To confirm that the integration is enabled

Before beginning this procedure, complete all procedures in the section titled “Configuring Unity to work with the phone system,” in Chapter 4 of the *Unity Installation Guide*.

- 1 In Internet Explorer, go to the Unity Administrator (<http://<server name>/saweb>).
- 2 Go to System > Switch. Confirm that the settings match those in the following table.

If the settings are incorrect, integration features may not be enabled.

Switch settings

Switch parameter	Required settings
Manufacturer	Mitel
Model	SX-2000
Switch PBX software version	Release N16 or later with T1 ports
Integration	Analog

AG T1 or AG Dual T1 protocol setting

The steps that you must perform to change the protocol setting for the AG T1 or AG Dual T1 boards vary depending on the version of the Windows operating system that is installed on the Unity server.

- ◆ For Microsoft Windows NT® Server, see “To change the AG T1 or AG Dual T1 protocol setting (Windows NT),” on page 72.
- ◆ For Microsoft Windows® 2000 Server, see “To change the AG T1 or AG Dual T1 protocol setting (Windows 2000),” on page 73.

Continued

Configuring Unity for the integration

Continued

To change the AG T1 or AG Dual T1 protocol setting (Windows NT)

- ❶ On the Windows Start menu, click Settings > Control Panel.
- ❷ Double-click the Telephony icon. The Dialing Properties dialog box appears.
- ❸ Click the Telephony Drivers tab.
- ❹ Click “Natural MicroSystems service provider,” and click “Configure.”
- ❺ In the NMS Configurator dialog box, select the first board from the list of installed boards.
- ❻ Click “Configure Board.” The Configure Board dialog box appears.
- ❼ In the Protocol list, click “Digital/Analog Wink-Start.”
- ❽ Click “OK.” The NMS Configurator dialog box reappears.
- ❾ Repeat steps 5 through 8 for each additional AG T1 or AG Dual T1 voice board.
- ❿ Click “OK” to close the NMS Configurator dialog box.
- ⓫ Click “Close” to close the Dialog Properties dialog box.
- ⓬ Shut down and restart the Unity server.

To change the AG T1 or AG Dual T1 protocol setting (Windows 2000)

- ❶ On the Windows Start menu, click Settings > Control Panel.
- ❷ Double-click "Phone and Modem Options." The Phone and Modem Options dialog box appears.
- ❸ Click the Advanced tab.
- ❹ Click "Natural MicroSystems service provider," and click "Configure."
- ❺ In the NMS Configurator dialog box, select the first board from the list of installed boards.
- ❻ Click "Configure Board." The Configure Board dialog box appears.
- ❼ In the Protocol list, click "Digital/Analog Wink-Start."
- ❽ Click "OK." The NMS Configurator dialog box reappears.
- ❾ Repeat steps 5 through 8 for each additional AG T1 or AG Dual T1 voice board.
- ❿ Click "OK" to close the NMS Configurator dialog box.
- ⓫ Click "Close" to close the Phone and Modem Options dialog box.
- ⓬ Click "OK."
- ⓭ Shut down and restart the Unity server.

See also

Shutting down and starting
Unity..... 140

Programming the phone system

If you use programming options other than those supplied in the following procedures, the integration's performance may be affected.

After connecting the systems, change the default settings for the phone system, as shown in the following procedures. Refer to the phone manufacturer's documentation for more details regarding these steps.

The phone system displays these settings as a series of forms that are available after selecting "Customer data entry" on the console. The following pages show representations of the forms. Change only the settings listed in each form.

CAUTION: Change the settings for each form in the exact sequence shown in the following steps. Changing the settings for each form out of sequence may result in error messages.

Steps for changing the phone system settings

- 1 Change the settings for the class of service options assignment form. Go to "To change the class of service options assignment form," on page 76.
- 2 Change the settings for the link descriptor assignment form. Go to "To change the link descriptor assignment form," on page 78.
- 3 Change the settings for the digital E and M trunk circuit descriptor assignment form. Go to "To change the digital E and M trunk circuit descriptor assignment form," on page 79.
- 4 Change the settings for the digital link assignment form. Go to "To change the digital link assignment form," on page 80.
- 5 Change the settings for the trunk service assignment form. Go to "To change the trunk service assignment form," on page 81.
- 6 Change the settings for the trunk assignment form. Go to "To change the trunk assignment form," on page 82.
- 7 Change the settings for the trunk group assignment form. Go to "To change the trunk group assignment form," on page 83.

- 8 Change the settings for the route assignment form. Go to “To change the route assignment form,” on page 84.
- 9 Change the settings for the feature access code assignment form. Go to “To change the feature access code assignment form,” on page 85.
- 10 Change the settings for the call progress tone detection plan assignment form. Go to “To change the call progress tone detection plan assignment,” on page 86.
- 11 Change the settings for the digit modification assignment form. Go to “To change the digit modification assignment form,” on page 87.
- 12 Change the settings for the system speed call (speed dialing) assignment form. Go to “To change the system speed call (speed dialing) assignment form,” on page 88.
- 13 Change the settings for the automatic route selection assignment form. Go to “To change the automatic route selection assignment form,” on page 89.
- 14 Change the settings for the call rerouting first alternative assignment form. Go to “To change the call rerouting first alternative assignment form,” on page 90.

Continued

Programming the phone system

Continued

To change the class of service options assignment form

- 1 On the phone system's Form Groups menu, select "System forms," and then select "Class of service options assignment."
- 2 Change only those settings listed in the form below.

CLASS OF SERVICE OPTIONS ASSIGNMENT	
Class of service number : 15	
Option	Select
Auto Answer Allowed	No
Busy Override Security	Yes
Call Forwarding (External Destination)	Yes
Call Hold - Remote Retrieve	No
Call Pickup - Dialed: Accept	No
Call Pickup - Directed: Accept	No
Call Privacy	Yes
Camp-on Tone Security	Yes
Clear All Features - Remote	Yes
Conference Call	No
COV/ONS/E&M Voice Mail Port	Yes
Dialed Night Service	No
Group Call Forward Follow Me - Accept	Yes

Highlighted fields are user defined. Form shows only those settings that require changing.

CLASS OF SERVICE OPTIONS ASSIGNMENT	
Class of service number : 15	
Option	Select
Individual Trunk Access	No
Loudspeaker Pager Override	No
Loudspeaker Pager Override Security	No
Message Waiting – Deactivate On Off-Hook	No
Multiline Set On Hook Dialing	No
Multiline Set Phonebook Allowed	No
Non-Verified Account Code	No
ONS/OPS Internal Ring Cadence for External Callers	Yes
Pager Access – All Zones	No
Public Network Access via DPNSS	Yes
Public Network to Public Network Connection Allowed	Yes
Redial Facilities	No
SMDR – External	Yes
Third Party Call Forward Follow Me – Accept	Yes
Third Party Call Forward Follow Me – Allow	No
Attendant Busy-out Timer (1–1440 mins)	1440
Auto Camp-on Timer (0–30 secs, Blank for Off)	
Busy Tone Timer (10–120 secs)	10
Call Hold Timer (10–660 secs)	120
Camp-on Recall Timer (0–180 secs)	60
Dialing Conflict Timer (2–5 secs)	5
Lockout Timer (10–60 secs)	10

Highlighted fields are user defined. Form shows only those settings that require changing.

Continued

Programming the phone system

Continued

To change the link descriptor assignment form

- 1 On the phone system's Form Groups menu, select "Digital link forms," and then select "Link descriptor assignment."
- 2 Change only those settings listed in the form below.

LINK DESCRIPTOR ASSIGNMENT	
Digital Link Descriptor Number : 10	
Description	Value
Address for Message Control (A/B)	
BER - Maintenance Limit, 10**n, n=(3,4,5,6)	4
BER - Service Limit, 10**n, n=(3,4,5,6)	3
Data Call Alternate Digit Inversion (Yes/No)	Yes
Framing Losses in 24 hrs - Maintenance Limit (0-9000)	255
Framing Losses in 24 hrs - Service Limit (0-9000)	9000
Integrated Digital Access	T1D4
Satellite Link Delay (Yes/No)	No
Slip Rate - Maintenance Limit (0-9000)	255
Slip Rate - Service Limit (0-9000)	7000
DS1 Parameters:	
Alarm Debounce Timer - Service Limit (300-3200)	500
B8ZS Zero Code Suppression - (Yes/No)	No
Italian Parameters:	
Digital Link Fault Delay Timer (0-360 sec)	240

Highlighted fields are user defined. Form shows only those settings that require changing.

To change the digital E and M trunk circuit descriptor assignment form

- ❶ On the phone system's Form Groups menu, select "Trunk forms," and then select "Digital E and M trunk circuit descriptor assignment."
- ❷ Change only those settings listed in the form below.

DIGITAL E AND M TRUNK CIRCUIT DESCRIPTOR ASSIGNMENT													
Trunk Circuit Descriptor Number:	10												
Descriptor	Value												
<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="padding: 2px;">Call Collision Handling (AT&T/Normal)</td> <td style="text-align: right; padding: 2px;">Normal</td> </tr> <tr> <td style="padding: 2px;"> AT&T Call Collision Handling (Backoff/NoBackoff)</td> <td style="text-align: right; padding: 2px;">Backoff</td> </tr> <tr> <td style="padding: 2px;">Ignore Far End Disconnect (Yes/No)</td> <td style="text-align: right; padding: 2px;">No</td> </tr> <tr> <td style="padding: 2px;">Release Acknowledge Timer (40-2400 units)(unit = 50 msec)</td> <td style="text-align: right; padding: 2px;">80</td> </tr> <tr> <td style="padding: 2px;">Address Signalling (Loop/DTMF)</td> <td style="text-align: right; padding: 2px;">DTMF</td> </tr> <tr> <td style="padding: 2px;">Disconnect Timer (100-300 msec)</td> <td style="text-align: right; padding: 2px;">300</td> </tr> </table>		Call Collision Handling (AT&T/Normal)	Normal	AT&T Call Collision Handling (Backoff/NoBackoff)	Backoff	Ignore Far End Disconnect (Yes/No)	No	Release Acknowledge Timer (40-2400 units)(unit = 50 msec)	80	Address Signalling (Loop/DTMF)	DTMF	Disconnect Timer (100-300 msec)	300
Call Collision Handling (AT&T/Normal)	Normal												
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Disconnect Timer (100-300 msec)	300												
<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="padding: 2px;">Incoming Start Type (Immediate/Wink/Delay)</td> <td style="text-align: right; padding: 2px;">Wink</td> </tr> <tr> <td style="padding: 2px;">Dial Tone on Incoming Seizure (Yes/No)</td> <td style="text-align: right; padding: 2px;">Yes</td> </tr> <tr> <td style="padding: 2px;">Minimum Flash Timer (100-250 msec)</td> <td style="text-align: right; padding: 2px;">140</td> </tr> <tr> <td style="padding: 2px;">Maximum Flash Timer</td> <td style="text-align: right; padding: 2px;">280</td> </tr> </table>		Incoming Start Type (Immediate/Wink/Delay)	Wink	Dial Tone on Incoming Seizure (Yes/No)	Yes	Minimum Flash Timer (100-250 msec)	140	Maximum Flash Timer	280				
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<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="padding: 2px;">Outpulse Delay Timer (100-2000 msec)</td> <td style="text-align: right; padding: 2px;">800</td> </tr> <tr> <td style="padding: 2px;">Outgoing Start Type (Wink/Immediate/Delay/Delay Integ)</td> <td style="text-align: right; padding: 2px;">Wink</td> </tr> <tr> <td style="padding: 2px;">Supv Timer (200-12750 msec)</td> <td style="text-align: right; padding: 2px;">200</td> </tr> <tr> <td style="padding: 2px;">Maximum Wink Timer (300-4000 msec)</td> <td style="text-align: right; padding: 2px;">400</td> </tr> <tr> <td style="padding: 2px;">Minimum Wink Timer (50-150 msec)</td> <td style="text-align: right; padding: 2px;">100</td> </tr> <tr> <td style="padding: 2px;">Guard Timer (100-6000 msec)</td> <td style="text-align: right; padding: 2px;">2000</td> </tr> </table>		Outpulse Delay Timer (100-2000 msec)	800	Outgoing Start Type (Wink/Immediate/Delay/Delay Integ)	Wink	Supv Timer (200-12750 msec)	200	Maximum Wink Timer (300-4000 msec)	400	Minimum Wink Timer (50-150 msec)	100	Guard Timer (100-6000 msec)	2000
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Guard Timer (100-6000 msec)	2000												
<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="padding: 2px;">Fake Answer Supervision After Outputting (Yes/No)</td> <td style="text-align: right; padding: 2px;">Yes</td> </tr> <tr> <td style="padding: 2px;">Ignore Answer Supervision (Yes/No)</td> <td style="text-align: right; padding: 2px;">Yes</td> </tr> <tr> <td style="padding: 2px;">Release Supervision Expected (Yes/No)</td> <td style="text-align: right; padding: 2px;">Yes</td> </tr> <tr> <td style="padding: 2px;">Audio Inhibit Until Answer Supervision (Yes/No)</td> <td style="text-align: right; padding: 2px;">No</td> </tr> </table>		Fake Answer Supervision After Outputting (Yes/No)	Yes	Ignore Answer Supervision (Yes/No)	Yes	Release Supervision Expected (Yes/No)	Yes	Audio Inhibit Until Answer Supervision (Yes/No)	No				
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Far End Connection (Main PBX/Satellite PBX/...)	Satellite PBX												
Facility Type (Digital/Combination)	Digital												

Highlighted fields are user defined. Form shows only those settings that require changing.

Continued

Programming the phone system

Continued

To change the digital link assignment form

- 1 On the phone system's Form Groups menu, select "Digital link forms," and then select "Digital link assignment."
- 2 Change only those settings listed in the form below.

DIGITAL LINK ASSIGNMENT						
Cab	Shlf	Slot	Link	Card Type	Digital Link Descriptor Number	Text
3	1	2	1	DSI Formatter	10	Voice Mail
3	1	2	2	DSI Formatter	10	Voice Mail

Highlighted fields are user defined. Form shows only those settings that require changing.

To change the trunk service assignment form

- ❶ On the phone system's Form Groups menu, select "Trunk forms," and then select "Trunk service assignment."
- ❷ Change only those settings listed in the form below.

TRUNK SERVICE ASSIGNMENT										
Trunk Service No.	RLT	COS	COR	Baud Rate	Intercept No.	Non-dial Answer Day	In Trunks Points Night 1	Dial In Trunks Incoming Digit Modification Absorb	Insert	Trunk Label
10	No	15	1	300	1			0		Voice Mail

Highlighted fields are user defined. Form shows only those settings that require changing.

Continued

Programming the phone system

Continued

To change the trunk assignment form

- 1 On the phone system's Form Groups menu, select "Trunk forms," and then select "Trunk assignment."
- 2 Change only those settings listed in the form below.

TRUNK ASSIGNMENT									
Cab	Shlf	Slot	Circ	Card Type	Trunk Number	Trunk Service Number	DTS Service Number	Circuit Desc. Number	Inter-connect Number
3	1	2	1	DSI Formatter	101	10		10	1
3	1	2	2	DSI Formatter	102	10		10	1

Highlighted fields are user defined. Form shows only those settings that require changing.

To change the trunk group assignment form

- 1 On the phone system's Form Groups menu, select "Trunk forms," and then select "Trunk group assignment."
- 2 Change those settings listed in the form below, including the "Trunk number" field for all members.

TRUNK GROUP ASSIGNMENT	
Trunk Group Number:	10
Hunt Mode (Circular or Terminal):	Circular
Trunk Group Busy RAD:	
Max Network Hops:	
Member	Trunk Number
1	101
2	102
3	103
4	104
5	105
6	106
7	107
8	108
24	124

Highlighted fields are user defined. Form shows only those settings that require changing.

Continued

Programming the phone system

Continued

To change the route assignment form

- 1 On the phone system's Form Groups menu, select "Automatic route selection forms," and then select "Route assignment."
- 2 Change only those settings listed in the form below.

ROUTE ASSIGNMENT					
Route Number	Trunk Group Number	COR Group Number	Digit	Modification Number	Digits Before Outpulsing
10	10	1		10	
11	10	1		11	
12	10	1		12	

Highlighted fields are user defined. Form shows only those settings that require changing.

To change the feature access code assignment form

- 1 On the phone system's Form Groups menu, select "System forms," and then select "Feature access code assignment."
- 2 Change only those settings listed in the form below.

FEATURE ACCESS CODE ASSIGNMENT		
Feature Name	Primary Code	Alternative Code
Call Forwarding – Busy (External Source)	*62	
Call Forwarding – Busy (Internal Source)	*63	
Call Forwarding – Follow Me	*64	
Call Forwarding – No Answer (External Source)	*65	
Call Forwarding – No Answer (Internal Source)	*66	
Call Hold – Retrieve	*1	
Message Waiting – Activate	*20	
Message Waiting – Deactivate	#20	

Form shows only those settings that require changing.

Continued

Programming the phone system

Continued

To change the call progress tone detection plan assignment

- 1 On the phone system's Form Groups menu, select "Automatic route selection forms," and then select "Call progress tone detection plan assignment."
- 2 Assign "Tone plan number 1" a maximum wait for a tone of two seconds.

CALL PROGRESS TONE DETECTION PLAN ASSIGNMENT	
Tone Plan Number: 01	Maximum Wait For A Tone (0-99 secs): 2
Action on Time-Out:	
Tone To Detect	Action To Take

Form shows only those settings that require changing.

To change the digit modification assignment form

- ❶ On the phone system's Form Groups menu, select "Automatic route selection forms," and then select "Digit modification assignment."
- ❷ Change only those settings listed in the form below.

DIGIT MODIFICATION ASSIGNMENT			
Digit Modification Number	Number of Digits to Absorb	Digits to be Inserted	Final Tone Plan/Information Marker
10	4	<T01>*#50*<E>#	
11	4	<T01>*<R>*<E>*<F>#	
12	4	<T01>*#51*<E>	

Highlighted fields are user defined. Form shows only those settings that require changing.

Continued

Programming the phone system

Continued

To change the system speed call (speed dialing) assignment form

- 1 On the phone system's Form Groups menu, select "System forms," and then select "System speed call assignment."
- 2 Change only those settings listed in the form below.

SYSTEM SPEED CALL ASSIGNMENT		
Speed Call Number	Actual Number	Overrides Toll Control
6700	8700	No
6800	8800	No
6900	8900	No

Highlighted fields are user defined. Form shows only those settings that require changing.

To change the automatic route selection assignment form

- 1 On the phone system's Form Groups menu, select "Automatic route selection forms," and then select "Automatic route select assignment."
- 2 Change only those settings listed in the form below.

AUTOMATIC ROUTE SELECTION ASSIGNMENT			
Leading Digits: 8		COR Group Number:	
Second Dial Tone (Yes/No/Alternate): No			
Digits Dialed	Number of Digits to Follow	Type	Termination Number
7	2	Route	10
8	2	Route	11
9	2	Route	12

Form shows only those settings that require changing.

Continued

Programming the phone system

Continued

To change the call rerouting first alternative assignment form

- 1 On the phone system's Form Groups menu, select "Call rerouting forms," and then select "Call rerouting first alternative assignment."
- 2 Change only those settings listed in the form below.

CALL REROUTING FIRST ALTERNATIVE ASSIGNMENT									
First Alternative Number	Busy / DND Originating Device				No Answer Originating Device				Directory Number Routed To
	DID	TIE	CO	Int	DID	TIE	CO	Int	
1	Normal	Normal	Normal	Normal	This	This	This	This	6800

Highlighted fields are user defined. Form shows only those settings that require changing.

NOTE: In this example, with the form showing "6800" as a directory number, the following conditions are true:

- ◆ Calls forward to "6800."
- ◆ Users dial "6700" to retrieve messages.
- ◆ Trunks are directed to "6900" for automated attendant.

■ MITEL WITH VOICEBRIDGE 2000

In this integration...

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Integration overview

Use this section of the *Unity Integration Guide* with the *Unity Installation Guide*.

Integration steps

- 1 Review the system and equipment requirements to verify that all phone system and Unity™ server requirements have been met. See “Requirements,” below.
- 2 Configure Unity for the integration. See “Configuring Unity for the integration,” on page 96.
- 3 Program the phone system and extensions. See “Programming the phone system,” on page 97.

Requirements

The steps to set up the Mitel with VoiceBridge 2000™ integration require the following:

Phone system

- ◆ One of the following Mitel phone systems:

Phone system	Software version
SX-200 Digital	Liteware 15 or later
SX-2000	All M streams

- ◆ For each voice messaging port, one Digital Network Interface Circuit (DNIC) port, which emulates a Superset 430 digital phone, installed and set up as a voice messaging port.
- ◆ The phone system ready for the integration as described in the phone system’s installation guide.

- ◆ The phone system's voice messaging ports connected to the VoiceBridge 2000 boards installed in the Unity server with a 25-pair Amphenol cable and a Telco/VoiceBridge 2000 adapter cable for each group of eight voice messaging ports. See the table below for cabling requirements.

VoiceBridge 2000 cabling requirements

Phone line number	Type 66 Block pair	Pin number	Pair color	Lead designation
1	1	26	white-blue	tip
		1	blue-white	ring
2	4	29	white-brown	tip
		4	brown-white	ring
3	7	32	red-orange	tip
		7	orange-red	ring
4	10	35	red-slate	tip
		10	slate-red	ring
5	13	38	black-green	tip
		13	green-black	ring
6	16	41	yellow-blue	tip
		16	blue-yellow	ring
7	19	44	yellow-brown	tip
		19	brown-yellow	ring
8	22	47	violet-orange	tip
		22	orange-violet	ring

Unity server

- ◆ The Dialogic® D/160SC™ voice boards and Voice Technologies Group™ VoiceBridge 2000 network boards, installed.

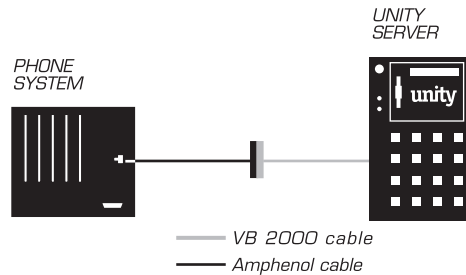
NOTE: For the latest information on supported voice boards for Unity, contact your sales representative.

- ◆ Unity installed and ready for the integration as described in the *Unity Installation Guide*.
- ◆ A system key that enables the integration and the appropriate number of voice messaging ports.

Integration description

How the integration works

The Mitel with VoiceBridge 2000 integration uses digital lines to connect the phone system and the Unity server. The phone system's digital voice messaging ports connect to Voice Technologies Group VoiceBridge 2000 network boards in the Unity server. Inside the Unity server, one or two VoiceBridge 2000 boards are connected to a Dialogic D/160SC voice board using an SCbus™ cable. The D/160SC voice board provides the voice processing for up to 16 ports while each VoiceBridge 2000 board emulates up to eight Mitel Superset 430 digital phones. The following illustration shows the required connections.



The phone system sends the following information with forwarded calls:

- ◆ The called party's extension
- ◆ The reason for the forward (the extension is busy, does not answer, or is set to forward all calls)
- ◆ The calling party's extension (for internal calls)

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

Integration features

The Mitel with VoiceBridge 2000 integration with Unity provides the following features:

Call forward to personal greeting When an incoming call is routed to an unanswered or busy extension, the call is forwarded to the subscriber's voice mail. The caller then hears the subscriber's personal greeting and can leave a message.

Caller ID Unity receives caller ID information from the phone system (if available). This information displays in the message's subject line in Microsoft Outlook® (or other desktop messaging application).

Easy message access A subscriber can retrieve messages without entering an ID. Unity identifies a subscriber based on the extension from which the call originated. A password may be required.

Identified subscriber messaging Unity automatically identifies a subscriber who leaves a message during a forwarded internal call, based on the extension from which the call originated.

Message waiting indication (MWI) When a message is waiting for a subscriber, Unity notifies the phone system to activate the message waiting indicator on the subscriber's extension (if it is equipped with one) or to activate a stutter dial tone.

Configuring Unity for the integration

After ensuring that the Unity server is ready for the integration by completing the appropriate tasks in Chapters 1 through 4 of the *Unity Installation Guide*, perform the following procedure to confirm that the integration is enabled.

To confirm that the integration is enabled

Before beginning this procedure, complete all procedures in the section titled “Configuring Unity to work with the phone system,” in Chapter 4 of the *Unity Installation Guide*.

- 1 In Internet Explorer, go to the Unity Administrator (<http://<server name>/saweb>).
- 2 Go to System > Licensing > Licensed Features.
- 3 In the “Integration” box, confirm that the setting is “Mitel Feature Set.”
- 4 If the setting is not “Mitel Feature Set,” contact your sales representative for the necessary system key.
- 5 Go to System > Switch. Confirm that the settings match those in the following table.

If the settings are incorrect, integration features may not be enabled.

Switch settings

Switch parameter	Required settings
Manufacturer	Mitel
Model	◆ SX-200D ◆ SX-2000
Switch PBX software version	◆ Liteware 15 and above ◆ All
Integration	Mitel Feature Set

- 6 If necessary, edit the codes that turn message waiting indication on and off. The MWI codes are located in the “Active Switch Setting” area at the top of the Switch Settings page.

Programming the phone system

After confirming that the integration is enabled, perform the procedures for your phone system model:

- ◆ For the Mitel SX-200D, go to the procedure, “To enable the Last Party Clear Dial Tone,” below, and then continue with the procedures on page 98.
- ◆ For the Mitel SX-2000, go to the procedure, “To change the class of service options assignment form on the Mitel SX-2000,” on page 100, and then continue with the procedures on page 102.

Mitel SX-200D

Perform the following procedures to set up the Mitel SX-200D for the integration. If you use programming options other than those supplied in the following procedures, the integration’s performance may be affected.

NOTE: The following procedures are written with the assumption that you will use an administrative terminal to program the phone system.

To enable the Last Party Clear Dial Tone

- ◆ On the phone system, enable the Last Party Clear Dial Tone option. Otherwise, the voice messaging system’s detection is delayed when internal calls are disconnected.

Continued

Programming the phone system

Continued

To set up the voice messaging extensions on the Mitel SX-200D

- ➊ On the phone system's Main menu, select option 9, Station/Supersets.
- ➋ Type 430 in each voice messaging extension's "TYP" field.
- ➌ Select a voice messaging extension field and then press ESC+3 to change the screen to expanded settings for the extension.
- ➍ Set the first key type to "Prime," and set the rest of the key types to "Speed Dial."
- ➎ Repeat steps 3 through 4 for each voice messaging extension.

To set up the hunt group on the Mitel SX-200D

- ➊ On the phone system's Main menu, select option 17, Hunt Groups.
- ➋ Create a hunt group with a pilot number, which is the published voice messaging number for all direct and forwarded calls. Include all of the voice messaging extensions in the hunt group.

Mitel SX-2000

Perform the following procedures to set up the Mitel SX-2000 for the integration. If you use programming options other than those supplied in the following procedures, the integration's performance may be affected.

NOTE: The following procedures are written with the assumption that you will use an administrative terminal to program the phone system.

Continued

Programming the phone system

Continued

To change the class of service options assignment form on the Mitel SX-2000

- 1 On the phone system's Form Groups menu, select "System forms," and then select "Class of service options assignment."
- 2 Change only those settings listed in the form below and on the following page.

CLASS OF SERVICE OPTIONS ASSIGNMENT	
Class of service number : 15	
Option	Select
Auto Answer Allowed	No
Busy Override Security	Yes
Call Forwarding (External Destination)	Yes
Call Hold - Remote Retrieve	No
Call Pickup - Dialed: Accept	No
Call Pickup - Directed: Accept	No
Call Privacy	Yes
Camp-on Tone Security	Yes
Clear All Features - Remote	Yes
Conference Call	No
COV/ONS/E&M Voice Mail Port	Yes
Dialed Night Service	No
Group Call Forward Follow Me - Accept	Yes

Highlighted field are user defined. Form shows only those settings that require changing.

CLASS OF SERVICE OPTIONS ASSIGNMENT	
Class of service number : 15	
Option	Select
Individual Trunk Access	No
Loudspeaker Pager Override	No
Loudspeaker Pager Override Security	No
Message Waiting – Deactivate On Off-Hook	No
Multiline Set On Hook Dialing	No
Multiline Set Phonebook Allowed	No
Non-Verified Account Code	No
ONS/OPS Internal Ring Cadence for External Callers	Yes
Pager Access – All Zones	No
Public Network Access via DPNSS	Yes
Public Network to Public Network Connection Allowed	Yes
Redial Facilities	No
SMDR – External	Yes
Third Party Call Forward Follow Me – Accept	Yes
Third Party Call Forward Follow Me – Allow	No
Attendant Busy-out Timer (1–1440 mins)	1440
Auto Camp-on Timer (0–30 secs, Blank for Off)	
Busy Tone Timer (10–120 secs)	10
Call Hold Timer (10–660 secs)	120
Camp-on Recall Timer (0–180 secs)	60
Dialing Conflict Timer (2–5 secs)	5
Lockout Timer (10–60 secs)	10

Highlighted field are user defined. Form shows only those settings that require changing.

Continued

Programming the phone system

Continued

To assign a class of service to the voice messaging extensions on the Mitel SX-2000

- 1 On the phone system's Form Groups menu, select "Station service forms," and then select "Station service assignment."
- 2 Assign to all voice messaging extensions the same class of service number that you assigned to the voice messaging port in the previous procedure.

To set up the voice messaging extensions on the Mitel SX-2000

- 1 On the phone system's Form Groups menu, select "System forms," and then select "DNI circuit assignment."
- 2 On the DNI Circuit Assignment form, set the "Device type channel #1" field to "Superset 430" for each voice messaging extension.

To assign the key numbers on the Mitel SX-2000

- 1 On the phone system's Form Groups menu, select "Multiline set forms," and then select "Multiline set key assignment."
- 2 On the Multiline Set Key Assignment form, type a voice messaging extension in the "Prime directory number" field.
- 3 Set the "Key numbers" field for the extension to "Not assigned."
- 4 Repeat steps 2 through 3 for each voice messaging system port extension number.

To set up the hunt group on the Mitel SX-2000

- ❶ On the phone system's Form Groups menu, select "Group forms," and then select "Hunt group assignment."
- ❷ On the Hunt Group Assignment form, in the "Pilot number" field, type the published voice messaging number for all direct and forwarded calls, and then press ESC+1.
- ❸ In the "Hunt mode" field, type `terminal` or `circular`.
- ❹ In the "Group type" field, type `voicemail`.
- ❺ In a "Directory number" field, type a voice messaging extension.
- ❻ Repeat step 5 for each voice messaging extension that must be set up for the integration.

■ NEC NEAX 2400 WITH SERIAL MCI

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Integration overview

Use this section of the *Unity Integration Guide* with the *Unity Installation Guide*.

Integration steps

Follow these steps to set up this integration.

- 1 Review the system and equipment requirements to verify that all phone system and Unity™ server requirements have been met. See “Requirements,” below.
- 2 Configure Unity for the integration. See “Configuring Unity for the integration,” on page 111.
- 3 If you are using T1 voice messaging ports, set up the digital trunk interface board. See “Setting up the trunk board,” on page 112.
- 4 Program the phone system and extensions. See “Programming the phone system,” on page 114.

Requirements

The steps to set up the NEC NEAX 2400 Message Center Interface (MCI) integration with a serial interface require the following:

Phone system

- ◆ One of the following NEAX 2400 phone systems with MCI:

Phone system	Software version
IMX	All software versions
IMG and SIM	5200 or later
MMG and UMG	4200 version 5 or later

NOTE: IMG and SIM phone systems may require upgrading to a high-density system to support MCI.

- ◆ One IOC serial port for the MCI data link connected to a serial port (COM1 is the default) on the Unity server with a PH-68 two-port cable and an RS-232C CA-1 cable.

See also

NEAX 2400 IMX Message Center Interface (MCI) Specification, ND-70428(E), Issue 1 (January 1999), NEC# 200892

- ◆ MCI feature II installed according to NEC document *ETI No. 121A (October 1998)* and the *NEAX 2400 IMX Message Center Interface (MCI) Specifications, ND-70428(E) Issue 1 (January 1999), NEC #200892.*
- ◆ One or more of the following station ports (analog) or trunk circuits (T1) set up as voice messaging ports and connected to the voice boards in the Unity server:

Port type	Description
Analog	One analog station port for each voice messaging port.
T1	One digital trunk interface board (card number PA-24DTR/DLI) with program version SP3298 3A 001 or later for each group of 24 voice messaging ports.

- ◆ Use of Unity's release-to-switch call transfer type requires one of the following versions of the NEAX 2400 phone system software:

Phone system	Software version
IMX	All versions, with blind transfer to the attendant and to the station
IMG and SIM	5200 or later, with blind transfer to the attendant and to the station
MMG and UMG	5200 or later, with blind transfer to the attendant and to the station

NOTE: *If the phone system does not support the release-to-switch transfer type, see "Transferring calls on phone systems without blind transfer," on page 118.*

Continued

Integration overview

Continued

Unity server

- ◆ One or more of the following voice boards, installed:

Line type	Voice board
Analog	Dialogic® D/41D™, D/41ESC™, D/41EPCI™, D/120JCT-LS™, or D/160SC-LS™
T1	Natural MicroSystems™ AG-T1 or AG Dual T1

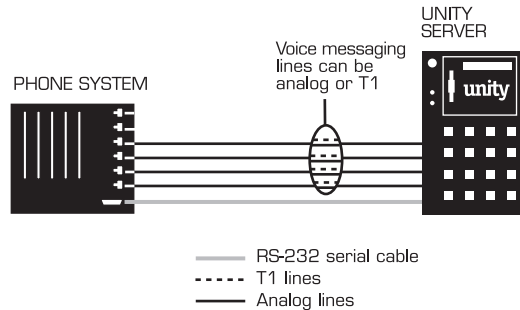
NOTE: For the latest information on supported voice boards for Unity, contact your sales representative.

- ◆ Unity installed and ready for the integration as described in the *Unity Installation Guide*.
- ◆ A system key that enables the integration and the appropriate number of voice messaging ports.
- ◆ An available serial port (COM1 is the default).

Integration description

How the integration works

The NEAX 2400 serial MCI integration uses a data link, which consists of an RS-232 serial cable connecting the phone system's IOC port to the Unity server. The phone system's voice messaging lines connect to the analog or T1 voice boards in the Unity server. The following illustration shows the required connections.



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

- ◆ The called party's extension
- ◆ The reason for the forward (the extension is busy, does not answer, or is set to forward all calls)
- ◆ The calling party's extension (for internal calls)

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

Continued

Integration description

Continued

Integration features

The NEC NEAX 2400 serial MCI integration with Unity provides the following features:

Call forward to personal greeting When an incoming call is routed to an unanswered or busy extension, the call is forwarded to the subscriber's voice mail. The caller then hears the subscriber's personal greeting and can leave a message.

Caller ID Unity receives caller ID information from the phone system (if available). This information displays in the message's subject line in Microsoft Outlook® (or other desktop messaging application).

Easy message access A subscriber can retrieve messages without entering an ID. Unity identifies a subscriber based on the extension from which the call originated. A password may be required.

Identified subscriber messaging Unity automatically identifies a subscriber who leaves a message during a forwarded internal call, based on the extension from which the call originated.

Message waiting indication (MWI) When a message is waiting for a subscriber, Unity notifies the phone system to activate the message waiting indicator on the subscriber's extension.

Configuring Unity for the integration

After ensuring that the Unity server is ready for the integration by completing the appropriate tasks in Chapters 1 through 4 of the *Unity Installation Guide*, perform the following procedure to confirm that the integration is enabled.

To confirm that the integration is enabled

Before beginning this procedure, complete all procedures in the section titled “Configuring Unity to work with the phone system,” in Chapter 4 of the *Unity Installation Guide*.

- 1 In Internet Explorer, go to the Unity Administrator (<http://<server name>/saweb>).
- 2 Go to System > Licensing > Licensed Features.
- 3 In the “Integration” box, confirm that the setting is “Serial.”
- 4 If the setting is not “Serial,” contact your sales representative for the necessary system key.
- 5 Go to System > Switch. Confirm that the settings match those in the following table.

If the settings are incorrect, integration features may not be enabled.

Switch settings

Switch parameter	Required settings
Manufacturer	NEC
Model	NEAX 2400
Switch PBX software version	◆ 4200 or later with analog ports ◆ 4200 or later with T1 ports
Integration	Serial

Setting up the trunk board

If the phone system uses a T1 line to connect to Unity, confirm that the trunk board's firmware version is correct, and set the DIP switches before you program the phone system.

To confirm the trunk board firmware version

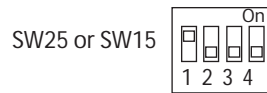
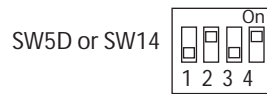
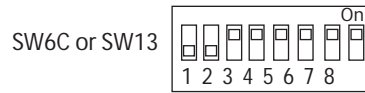
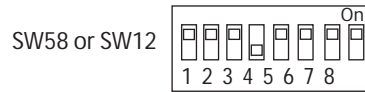
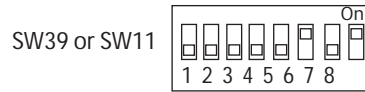
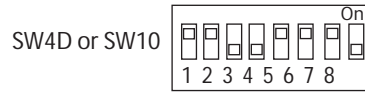
- 1 Remove the 24DTR/DLI trunk board from the phone system.
- 2 Locate the removable, integrated circuit (PROM) with a white label. This PROM contains the trunk board's program, or firmware. The relative location of the PROM may vary, but it is the only PROM with a white label on the trunk board.
- 3 Confirm that the firmware version printed on the label is "SP-3298 3A 001" or later. If the version is incorrect, update the firmware by replacing the PROM or the trunk board.

See also

NEAX 2400 IMX Circuit Card Manual

To set the trunk board switches

- ◆ Set the switches for the 24DTR/DLI trunk board as shown below.



Programming the phone system

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

NOTE: The following procedure is written with the assumption that you will use a Maintenance Administration Terminal (MAT) to program the phone system.

To program the phone system

- 1 Specify the Universal Call Distribution (UCD) group for the voice messaging system ports based on the type of voice messaging ports:

Analog Assign all voice messaging system ports to a UCD group.

T1 Assign each trunk circuit an extension number, then add each of the numbers to the UCD group.

If the number of voice messaging system ports exceeds the number of supported ports in a UCD group, specify additional UCD groups, and then link them together using the AUOG command. Software versions 7200 or later support up to 100 ports in a UCD group while software versions earlier than 7200 support up to 20 ports in a UCD group.

CAUTION: Do not use a phantom single line extension for the UCD pilot number.

See also

NEAX 2400 Office Data Specification

NEAX 2400 IMX Message Center Interface (MCI) Specifications, ND-70428(E), Issue 1 (January 1999), NEC# 200892

- 2 Program each phone to forward calls to the UCD pilot number assigned to the voice messaging system ports, based on the Unity call transfer type:

Release to switch transfer Program the phone to forward calls to the UCD pilot number when:

- ◆ The extension is busy
- ◆ The call is not answered

Supervised transfer Program the phone to forward calls to the UCD pilot number only when the call is not answered. Confirm that call forwarding is disabled when the extension is busy.

- 3 Set up the RS-232 serial data port for AIOC settings as follows:

- ◆ 9600 baud
- ◆ 8 data bits
- ◆ 1 stop bit
- ◆ No parity

- 4 If you use a remote maintenance modem, program the modem line for data line security.
- 5 Use the AUCD command to program the phone system to send UCD call information to MCI. Assign a value of "0" to the "MCI Data Transfer" field for the appropriate tenant and UCD pilot numbers.

Continued

Programming the phone system

Continued

- Use the programming system data table to program the ASYD settings. Each bit is part of a hexadecimal number displayed in the ASYD settings. Convert the hexadecimal number to binary to determine the individual settings.

Programming system data

System	Index	Bit	Value	Description
1	17	b4	1	Blind transfer to attendant console
	28	b0-4	0	Guard timer not required
		b5	1	MWI controlled by MCI
	29	b1-7	0/1	No/Yes: Assign I/O port for MCI output Port 1 = b1, port 2 = b2, and so on
	34	b1-4	0	Set output to no parity and 1 stop bit
	60	b3	0	UCD queuing required
	63	b0	1	Blind transfer for stations in service
	69	b0	1	No recall, execute call forwarding on no answer
	70	b0	1	Called number display, when forwarding to attendant console
		b1	1	Called station status display enabled
	77	b2	0	MWI refresh required
	78	b0	1	Calling number display enabled
		b1	1	Called station status display enabled
	238	b0-7	0	Lamp flash rate
	246	b3	0	MCI expansion set to normal
400	b2	1	Calling number information sent to MCI	
2	6	b0	1	MCI in service when terminating to a UCD group
	7	b1	0	MCI out of service when terminating to attendant console

See also

NEAX 2400 Office Data Specification

- 7 Use the programming system data local data table to program the ASYDL settings. Each bit is part of a hexadecimal number displayed in the ASYDL settings. Convert the hexadecimal number to binary to determine the individual settings.

Programming system data local data

System	Index	Bit	Value	Description
1	641	b1	0/1	0/1: MCI/IMX station number/phone number
	832	b0-7	00-FD	Assign the FPC of the node connected to MC
	833	b0	0	IOC serial interface enabled for MCI
		b1	0	0/1: ICS/IMX format

Transferring calls on phone systems without blind transfer

Because phone systems without blind transfer do not support Unity's release-to-switch call transfer type, the phone system does not forward calls to Unity when all of the following three conditions are met:

- ◆ The phone system does not support blind transfers to the attendant console and blind transfers to the stations.
- ◆ The phone system is set to forward unanswered calls.
- ◆ A call is transferred from an extension to another that does not answer.

When all of these conditions are met, the phone system returns the call to the extension originally called. (Calls that the operator transferred appear on the console as "Recall" when they return.)

■ NORSTAR VMI

In this integration...

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Programming the phones	130

Integration overview

Use this section of the *Unity Integration Guide* with the *Unity Installation Guide*.

Integration steps

Follow these steps to set up this integration.

- 1 Review the system and equipment requirements to verify that all phone system and Unity™ server requirements have been met. See “Requirements,” below.
- 2 Configure Unity for the integration. See “Configuring Unity for the integration,” on page 124.
- 3 Program the phone system. See “Programming the phone system,” on page 125.
- 4 Program the phone extensions. See “Programming the phones,” on page 130.

Requirements

The steps to set up the Norstar Voice Message Interface (VMI) integration require the following:

Phone system

- ◆ Norstar DR3-DR5A or Norstar-PLUS Modular ICS phone system.
- ◆ One Norstar VMI unit for every two voice messaging ports.
- ◆ A Norstar M7310 or M7324 phone with the appropriate overlay for programming the phone system.
- ◆ The phone system’s digital lines connected to the VMI unit, and the phone system’s analog voice messaging ports connected to the voice boards in the Unity server.
- ◆ The phone system ready for the integration as described in the phone system’s installation guide.

Unity server

- ◆ The Dialogic® D/41D™, D/41ESC™, D/41EPCI™, D/120JCT-LS™, or D/160SC-LS™ voice boards, installed.

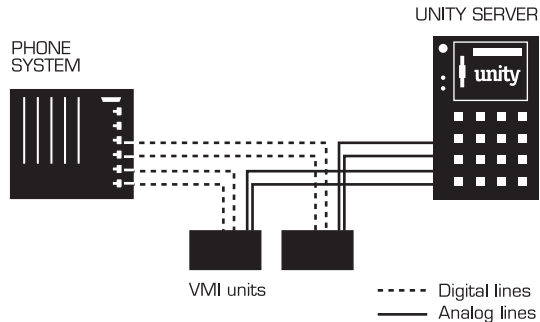
NOTE: For the latest information on supported voice boards for Unity, contact your sales representative.

- ◆ Unity installed and ready for the integration as described in the *Unity Installation Guide*.
- ◆ A system key that enables the integration and the appropriate number of voice messaging ports.

Integration description

How the integration works

The Norstar VMI unit enables the digital ports on a Meridian Norstar phone system to communicate with Unity's analog ports. The integration requires one VMI unit for every two voice messaging ports. The phone system's digital voice messaging lines connect to the VMI unit, and the VMI unit's analog lines connect to the voice boards in the Unity server. The following illustration shows the required connections.



The phone system sends the called party's extension through the VMI unit. Unity uses this information to answer the call appropriately. For example, a call forwarded to Unity is answered with the subscriber's personal greeting. If the phone system routes the call to Unity without this information, Unity answers with the opening greeting.

Integration features

The Norstar VMI integration with Unity provides the following features:

Call forward to personal greeting When an incoming call is routed to an unanswered or busy extension, the call is forwarded to the subscriber's voice mail. The caller then hears the subscriber's personal greeting and can leave a message.

Message waiting indication (MWI) When a message is waiting for a subscriber, Unity notifies the phone system to activate the message waiting indicator on the subscriber's extension (if it is equipped with one) or to activate a stutter dial tone.

Configuring Unity for the integration

After ensuring that the Unity server is ready for the integration by completing the appropriate tasks in Chapters 1 through 4 of the *Unity Installation Guide*, perform the following procedure to confirm that the integration is enabled.

To confirm that the integration is enabled

Before beginning this procedure, complete all procedures in the section titled “Configuring Unity to work with the phone system,” in Chapter 4 of the *Unity Installation Guide*.

- 1 In Internet Explorer, go to the Unity Administrator (<http://<server name>/saweb>).
- 2 Go to System > Switch. Confirm that the settings match those in the following table.

If the settings are incorrect, integration features may not be enabled.

Switch settings

Switch parameter	Required settings
Manufacturer	Northern Telecom
Model	Norstar (DR3–DR5)
Switch PBX software version	All
Integration	Analog

Programming the phone system

If you use programming options other than those supplied in the following procedures, the integration's performance may be affected. Perform the procedures in the order given.

When the VMI unit is installed properly, the phone system automatically assigns to it a programming feature code. You use this code to determine the intercom numbers, which are necessary for programming the phone system.

NOTE: Permit the VMI unit to warm up before proceeding.

To determine the VMI feature codes

- 1 On the key phone, press FEATURE 9*3. The phone's display screen shows "VMI Codes."
- 2 Press SHOW. The display screen shows the feature code for the first VMI unit. This code appears in the form F9XX, where XX is the feature code.
- 3 Write down the feature code.
- 4 If additional VMI units are installed, press NEXT to see the remaining feature codes. Then write down the codes.
- 5 Press RLS to end the programming session.

To determine the VMI intercom numbers

- 1 On the key phone, press FEATURE 9XX, where XX is the feature code of the first VMI unit.
- 2 Press NEXT. The display screen shows DN1:YYY, where YYY is the intercom number for the first port on the first VMI unit.
- 3 Write down the intercom number.
- 4 Press NEXT. The display screen shows DN2:YYY, where YYY is the intercom number for the second port on the first VMI unit.
- 5 Repeat steps 1 through 4 for each additional VMI unit.

Continued

Programming the phone system

Continued

To set line access for the voice messaging ports

- ➊ On the key phone, activate the configuration mode.
- ➋ Go to A. Configuration > 2. Line Access and press SHOW.
- ➌ At the “Show set” prompt, enter the intercom number of the first port on the first VMI unit (see step 3 in “To determine the VMI intercom numbers,” on page 125”).
- ➍ Set the menu items as shown in the following table.

Menu item	Setting
Line assignment	If Unity is used as an automated attendant, add the outside lines assigned to the VMI port. If Unity is not used as an automated attendant, choose “No lines assigned.”
Answer DNs	No answer DNs
Ringing	If Unity is used as an automated attendant, set each outside line assigned to the VMI port to “Ring.”
Line pool access	Line pool A:Y Set the remaining line pools to “N.”
Intercom keys	2
Prime line	I/C

- ➎ At the “Show set” prompt, copy the line access settings to the other port on the first VMI unit.
- ➏ Repeat the copy process as described in step 5 for all other VMI units.

To set abilities

- 1 On the key phone, go to B. General Admin > 5. Capabilities > Set Abilities and press SHOW.
- 2 At the “Show set” prompt, enter the intercom number of the first port on the first VMI unit.
- 3 Set the menu items as shown in the following table.

Menu item	Setting
Set lock	None
Full handsfree	No
HF answerback	No
Pickup group	No
Paging	No
Page zone	No
Aux ringer	No
Forward on busy	Set up a simulated circular hunt group where the first VMI port forwards to the second, the second forwards to the third, and so on. Forward the last port back to the first port.
Forward no answer	Set up the same simulated circular hunt group that you set up for “Forward on busy.”
Hotline	None
Priority call	No

- 4 At the “Show set” prompt, copy the abilities to the other VMI ports.

Continued

Programming the phone system

Continued

To enable delayed ringing and set the transfer callback delay

- ➊ On the key phone, go to A. Configuration > 3. Call Handling and press SHOW.
- ➋ Set the menu items as shown in the following table.

Menu item	Setting
Held reminder	Yes
DRT delay	2
Transfr callbk	6

NOTE: Setting "Held reminder" to "Yes" transfers external calls that are not answered by Unity to the operator.

To program the VMI unit to send a disconnect tone

- ❶ On the key phone, press FEATURE, then enter the feature code of the first VMI unit.
- ❷ Press NEXT until the display screen shows "Feature table."
- ❸ Press SHOW. The display screen shows "Send Msg L*1."
- ❹ Press NEXT until the display screen shows "Discon."
- ❺ Press CHANGE.
- ❻ Enter *8** on the phone keypad.
- ❼ Press NEXT.
- ❽ Repeat steps 1 through 7 for the remaining VMI units.
- ❾ Press RLS to end the programming session.

*NOTE: This procedure sets the VMI unit to send a disconnect tone of *8** when a caller hangs up prematurely or does not exit normally from the voice messaging system.*

Programming the phones

To assign call forwarding to individual subscriber phones

- 1 On the key phone, activate configuration mode.
- 2 Go to B. General Admin > 5. Capabilities > Set Abilities and press SHOW.
- 3 At the “Show set” prompt, enter the intercom number for the first phone.
- 4 Set the menu items as shown in the following table.

Menu item	Setting
Forward on busy	Set to forward to the first port on the first VMI unit.
Forward no answer	Set to forward to the first port on the first VMI unit. Forward delay: 6

NOTE: Setting the phone system to forward calls when an extension is busy disables Unity subscribers from delivering messages to an extension.

- 5 Repeat steps 2 through 4 for each remaining phone.

■ TOSHIBA DK280/424 SMDI

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Integration overview

Use this section of the *Unity Integration Guide* with the *Unity Installation Guide*.

Integration steps

Follow these steps to set up this integration.

- 1 Review the system and equipment requirements to verify that all phone system and Unity™ server requirements have been met. See “Requirements,” below.
- 2 Configure Unity for the integration. See “Configuring Unity for the integration,” on page 136.
- 3 Program the phone system and extensions. See “Programming the phone system,” on page 137.

Requirements

The steps to set up the Toshiba DK280/424 Simplified Message Desk Interface (SMDI) integration require the following:

Phone system

- ◆ Toshiba DK280 or DK424 with an SMDI port installed.
- ◆ The phone system’s SMDI port connected to a serial port (COM1 is the default) on the Unity server with an RS-232 serial cable.
- ◆ The phone system’s voice messaging ports connected to the voice boards in the Unity server.
- ◆ The phone system ready for the integration as described in the phone system’s installation guide.

Unity server

- ◆ The Dialogic® D/41D™, D/41ESC™, D/41EPCI™, D/120JCT-LS™, or D/160SC-LS™ voice boards, installed.

NOTE: For the latest information on supported voice boards for Unity, contact your sales representative.

- ◆ Unity installed and ready for the integration as described in the *Unity Installation Guide*.

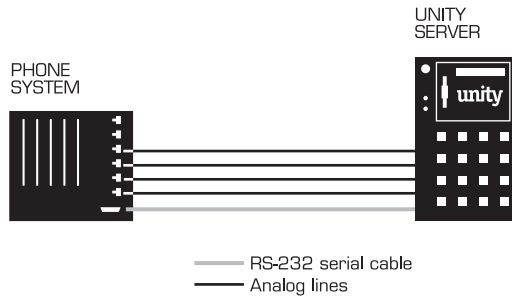
NOTE: To ensure that Windows does not mistake the serial connection for a serial mouse when rebooting the Unity server, confirm that the boot.ini file is set according to instructions. See "Installing Windows NT Server" in Chapter 3 of the Unity Installation Guide.

- ◆ A system key that enables the integration and the appropriate number of voice messaging ports.
- ◆ An available serial port (COM1 is the default).

Integration description

How the integration works

The Toshiba DK280/424 SMDI integration uses a data link, which consists of an RS-232 serial cable connecting the phone system and the Unity server. The phone system's voice messaging lines connect to the analog voice boards in the Unity server. The following illustration shows the required connections.



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

- ◆ The called party's extension
- ◆ The reason for the forward (the extension is busy, does not answer, or is set to forward all calls)
- ◆ The calling party's extension (for internal calls)

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

Integration features

The Toshiba DK280/424 SMDI integration with Unity provides the following features:

Call forward to personal greeting When an incoming call is routed to an unanswered or busy extension, the call is forwarded to the subscriber's voice mail. The caller then hears the subscriber's personal greeting and can leave a message.

Caller ID Unity receives caller ID information from the phone system (if available). This information displays in the message's subject line in Microsoft Outlook® (or other desktop messaging application).

Easy message access A subscriber can retrieve messages without entering an ID. Unity identifies a subscriber based on the extension from which the call originated. A password may be required.

Identified subscriber messaging Unity automatically identifies a subscriber who leaves a message during a forwarded internal call, based on the extension from which the call originated.

Message waiting indication (MWI) When a message is waiting for a subscriber, Unity notifies the phone system to activate the message waiting indicator on the subscriber's extension (if it is equipped with one) or to activate a stutter dial tone.

Configuring Unity for the integration

After ensuring that the Unity server is ready for the integration by completing the appropriate tasks in Chapters 1 through 4 of the *Unity Installation Guide*, perform the following procedure to confirm that the integration is enabled.

To confirm that the integration is enabled

Before beginning this procedure, complete all procedures in the section titled “Configuring Unity to work with the phone system,” in Chapter 4 of the *Unity Installation Guide*.

- 1 In Internet Explorer, go to the Unity Administrator (<http://<server name>/saweb>).
- 2 Go to System > Licensing > Licensed Features.
- 3 In the “Integration” box, confirm that the setting is “Serial.”
- 4 If the setting is not “Serial,” contact your sales representative for the necessary system key.
- 5 Go to System > Switch. Confirm that the settings match those in the following table.

If the settings are incorrect, integration features may not be enabled.

Switch settings

Switch parameter	Required settings
Manufacturer	Toshiba
Model	◆ DK 280 ◆ DK 424
Switch PBX software version	◆ CPU A, B, C/D ◆ All
Integration	Serial

Programming the phone system

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

To program the phone system

- ❶ On the phone system, activate the SMDI port.
- ❷ In "System assignments," set Station Number Digit Length to "10," whether caller ID is used or not.

NOTE: This setting does not affect the extension length on the phones.

- ❸ Confirm that the phones assigned to the voice messaging ports are designated as voice messaging ports on the phone system.

Shutting down and starting Unity

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Shutting down and starting Unity

Unity is a Windows NT/2000 service that is configured to start automatically when the server starts. You also can start and shut down Unity by using the Status Monitor, a separate application that also provides information about ports, licenses, and reports.

To shut down Unity

- 1 If you are at the Unity server, log on to Windows NT/2000 as a Unity administrator.

If you are at another computer, skip to step 2.

- 2 If you are at the Unity server, double-click the desktop shortcut to the Status Monitor.

If you are at another computer, start Internet Explorer and go to `http://<server name>/status`. If prompted, type the Unity Administrator name and password (by default, Administrator and 12345 on a Unity turnkey system).

- 3 Indicate a shutdown method: Unity shuts down after all calls are finished, or Unity interrupts calls in progress with a message, disconnects all calls, then shuts down.
- 4 Click "Shut down." On the Unity server, an "X" appears in the Unity icon in the status area of the taskbar when Unity has finished shutting down.

To shut down or restart the Unity server

- 1 Shut down Unity by using the previous procedure.
- 2 On the Windows Start menu, click "Shut down."
- 3 Click "Shut down the computer" or "Restart the computer."

During a restart, Unity starts automatically.

When Unity starts successfully, three tones play and a check mark appears in the Unity icon in the status area of the taskbar.

When Unity does not start successfully, two tones play and an "X" appears in the Unity icon in the status area of the taskbar.

To start Unity

Because Unity starts automatically when you turn on or restart the server, you do this procedure only if you shut down Unity but did not restart the server.

Exchange must be running before you start Unity. Like Unity, Exchange starts automatically when you turn on or restart the server. If you stopped Exchange manually but did not restart the server, start Exchange before doing this procedure.

- 1 If you are at the Unity server, log on to Windows NT/2000 as a Unity administrator.

If you are at another computer, skip to step 2.

- 2 If you are at the Unity server, double-click the desktop shortcut to the Status Monitor.

If you are at another computer, start Internet Explorer and go to `http://<server name>/status`. If prompted, type the Unity Administrator name and password (by default, Administrator and 12345 on a Unity turnkey system).

- 3 Click the System Status icon.

- 4 Click "Start."

When Unity starts successfully, three tones play and a check mark appears in the Unity icon in the status area of the taskbar.

When Unity does not start successfully, two tones play and an "X" appears in the Unity icon in the status area of the taskbar.

- 5 If you are at the Unity server, press CTRL+ALT+DELETE, then lock or log off of Windows NT/2000 to prevent access by unauthorized users.

TIP

From the Unity server, you also can start and shut down Unity by right-clicking the Unity icon in the status area of the taskbar, then clicking "Start Unity" or "Stop Unity."

