TIMG Integration Guide for Cisco Unity Connection Release 15

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CONTENTS

CHAPTER 1

Integration Description

Integration Description

Introduction

Cisco Unity Connection supports integrations through TIMG units (media gateways) with supported phone systems that provide in-band call information and MWI requests through the T1 digital lines.

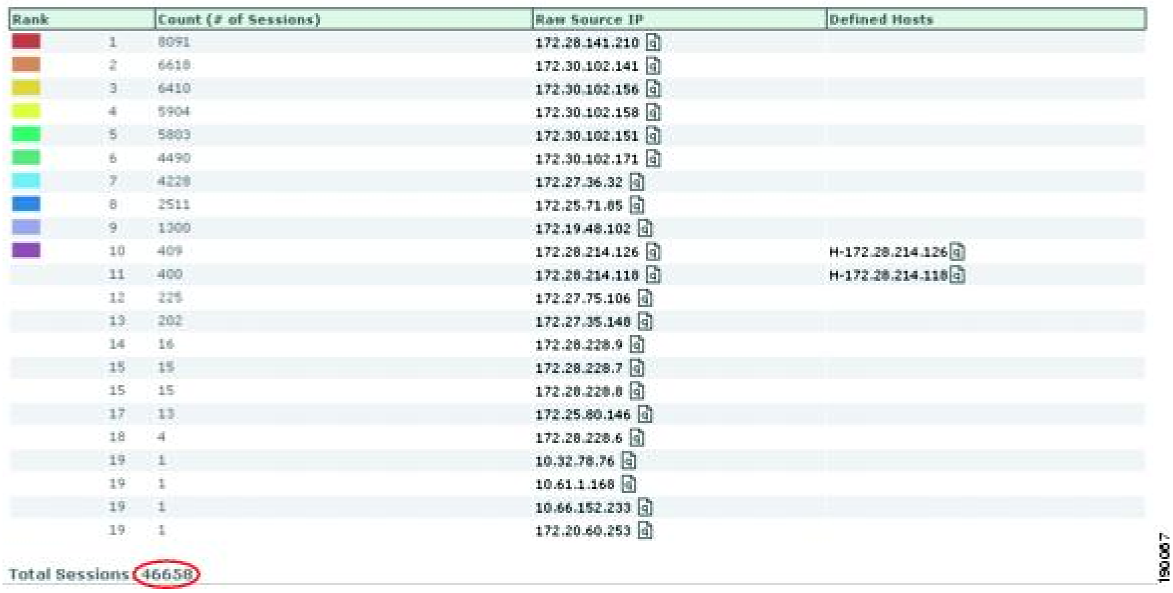
Unity Connection supports integrations through TIMG units (media gateways) with the following phone systems:

- Any phone system that provides call information and MWI requests through a serial data link (SMDI, MCI, or MD-110 protocol) to the master TIMG unit. For details, see the "Serial Integration with TIMG Units" section.
- A supported phone system that provides in-band call information and MWI requests—through the T1 digital lines. For details, see the "In-Band Integration with TIMG Units" section.

Serial Integration with TIMG Units

The phone system sends call information and MWI requests through the data link, which is an RS-232 serial cable that connects the phone system and the master TIMG unit. Voice connections are sent through the T1 digital lines between the phone system and the TIMG units. The TIMG units communicate with the Unity\\ Connection server through the LAN or WAN using Session Initialization Protocol (SIP). Figure 1-1 shows the required connections for a serial integration using TIMG units.

Figure 1: Connections for a Serial Integration Using TIMG Units



Rank	Count (# of Sessions)	Raw Source IP	Defined Hosts
1	8091	172.28.141.210	
2	6618	172.30.102.141	
3	6410	172.30.102.156	
4	5904	172.30.102.158	
5	5803	172.30.102.151	
6	4490	172.30.102.171	
7	4228	172.27.36.32	
8	2511	172.25.71.85	
9	1300	172.19.48.102	
10	409	172.28.214.126	H-172.28.214.126
11	400	172.28.214.118	H-172.28.214.118
12	225	172.27.75.106	
13	202	172.27.35.148	
14	16	172.28.228.9	
15	15	172.28.228.7	
15	15	172.28.228.8	
17	13	172.25.80.146	
18	4	172.28.228.6	
19	1	10.32.78.76	
19	1	10.61.1.168	
19	1	10.66.152.233	
19	1	172.20.60.253	

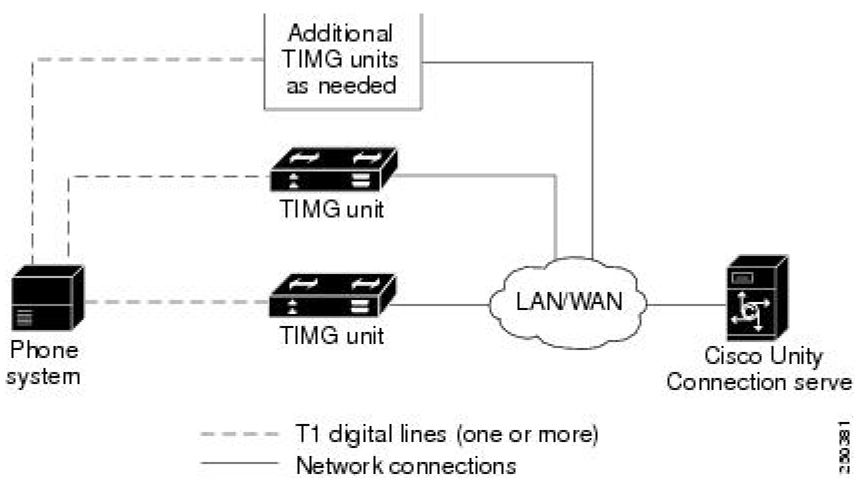
Total Sessions 46658

190067

In-Band Integration with TIMG Units

The phone system sends call information, MWI requests, and voice connections through the T1 digital lines, which connect the phone system and the TIMG units. The TIMG units communicate with the Unity Connection server through the LAN or WAN using Session Initialization Protocol (SIP). Figure 2: Connections for an In-Band Integration Using TIMG Units shows the required connections for an in-band integration using TIMG units.

Figure 2: Connections for an In-Band Integration Using TIMG Units



Call Information

The phone system sends the following information with forwarded calls:

- The extension of the called party

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

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

Note Serial integrations send requests to turn on and turn off MWIs through the data link.

Integration Functionality

The TIMG integration provides the following integration features:

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

Integrations with Multiple Phone Systems

Unity\\ Connection can be integrated with two or more phone systems at one time. For information on the maximum supported combinations and instructions for integrating Unity\\ Connection with multiple phone systems, see the *Multiple Phone System Integration Guide for Cisco Unity Connection Release 15*, available at

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/integration/multiple/b_cuc15intmultiple.html.

Call Information

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CHAPTER 2

Planning the Usage of Voice Messaging Ports in Cisco Unity Connection

Planning the Usage of Voice Messaging Ports in Cisco Unity Connection

Planning the Port Setup

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

- The number of voice messaging ports installed.

For a Unity Connection cluster, each server must have enough ports to handle all voice messaging traffic in case the other server stops functioning.

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

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

Table 1: Settings for the Voice Messaging Ports

Field	Considerations
Enabled	Check this check box to enable the port. The port is enabled during normal oper Uncheck this check box to disable the port. When the port is disabled, calls to th ringing tone but are not answered. Typically, the port is disabled only by the inst testing.

Field	Considerations
Extension	Enter the extension for the port as assigned on the phone system.
Answer Calls	Check this check box to designate the port for answering calls. These calls can be in calls from unidentified callers or from users.
Perform Message Notification	Check this check box to designate the port for notifying users of messages. Assign P Message Notification to the least busy ports.
Send MWI Requests *(not used by serial integrations)*	For serial integrations, uncheck this check box. Otherwise, the integration may not f correctly. For in-band integrations, check this check box to designate the port for turning MW off. Assign Send MWI Requests to the least busy ports.
Allow TRAP Connections	Check this check box so that users can use the phone as a recording and playback de Unity Connection web applications. Assign Allow TRAP Connections to the least b
Outgoing Hunt Order	Enter the priority order in which Unity Connection use the ports when dialing out (for if the Perform Message Notification, Send MWI Requests, or Allow TRAP Connecti box is checked). The highest numbers are used first. However, when multiple ports same Outgoing Hunt Order number, Unity Connection use the port that has been idle th

Determining the Number of Voice Messaging Ports

The following tasks describe the process for determining the number of voice messaging ports for Cisco Unity Connection to install, answer call and dial out calls:

- For determining the number of voice messaging ports to Install, see Voice Messaging Ports to Install.
- For determining the number of voice messaging ports to Answer Calls, see Voice Messaging Ports to Answer Calls.
- For determining the number of voice messaging ports to Dial Out, see Voice Messaging Ports that Dial Out.

Voice Messaging Ports to Install

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

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

- The number of calls use the automated attendant and call handlers when call traffic is at its peak.
- Whether a Unity Connection cluster is configured. For considerations, see the "Considerations for a Unity Connection Cluster" section.

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

Voice Messaging Ports to Answer Calls

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

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

If your system is configured for a Unity Connection cluster, see the "Considerations for a Unity Connection Cluster" section..

Voice Messaging Ports that Dial Out

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

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

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

If your system is configured for a Unity Connection cluster, see the "Considerations for a Unity Connection Cluster" section.

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

Considerations for a Unity Connection Cluster

If your system is configured for a Unity Connection cluster, consider how the voice messaging ports used in different scenarios.

When Both Unity Connection Servers are Functioning

- The number of ports provisioned on the phone system is the same as the number of voice messaging ports on each Unity Connection server.
- The TIMG units are configured to send incoming calls first to the subscriber server, then to the publisher server if no answering ports are available on the subscriber server.
- Both Unity Connection servers are active and handle voice messaging traffic for the system.
- The number of voice messaging ports on each Unity Connection server must be sufficient to handle all of the voice messaging traffic for the system (answering calls and dialing out) when the other Unity Connection server stops functioning.

If both Unity Connection servers must be functioning to handle the voice messaging traffic, the system do not have sufficient capacity when one of the servers stops functioning.

- Each Unity Connection server must have voice messaging ports that answer calls and that can dial out (for example, to set MWIs).

When Only One Unity Connection Server is Functioning

- TIMG units send all calls to the functioning Unity Connection server.
- The functioning Unity Connection server receives all voice messaging traffic for the system.
- The number of voice messaging ports that are assigned to the functioning Unity Connection server must be sufficient to handle all of the voice messaging traffic for the system (answering calls and dialing out).
- The functioning Unity Connection server must have voice messaging ports that answer calls and that can dial out (for example, to set MWIs).

If the functioning Unity Connection server does not have voice messaging ports for answering calls, the system is not able to answer incoming calls. Similarly, if the functioning Unity Connection server does not have voice messaging ports for dialing out, the system is not able to dial out (for example, to set MWIs).

CHAPTER 3

Setting Up an Avaya Definity G3 In-Band TIMG Integration with Cisco Unity Connection

For detailed instructions for setting up an Avaya Definity G3 in-band TIMG integration with Cisco Unity Connection, see the following sections in this chapter:

Setting Up an Avaya Definity G3 In-Band TIMG Integration with Cisco Unity Connection

For detailed instructions for setting up an Avaya Definity G3 in-band TIMG integration with Cisco Unity Connection, see the following sections in this chapter:

Task List to Create an Avaya Definity G3 In-Band TIMG Integration

Before doing the following tasks to integrate Unity Connection with the phone system using the T1 media gateway (TIMG), confirm that the Unity Connection server is ready for the integration after completing the installation following the steps as mentioned in the "Installing Cisco Unity Connection" chapter of the *Install, Upgrade, and Maintenance Guide for Cisco Unity Connection Release 15*, available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/unified_messaging/guide/b_15cucumgx.html.

1. Review the system and equipment requirements to confirm that all phone system and Unity Connection server requirements have been met. See the Requirements section.
2. Plan how the voice messaging ports used by Unity Connection. See the Planning the Usage of Voice Messaging Ports in Cisco Unity Connection chapter.
3. Program the phone system and extensions. See the Programming Phone System for In-Band TIMG Integration section.
4. Set up the TIMG units for an in-band integration. See the Setting Up the TIMG Units section.
5. Create the integration. See the Creating an Integration with the Phone System section.
6. Test the integration. See the Testing the Integration chapter.
7. If this integration is a second or subsequent integration, add the applicable new user templates for the new phone system. See the Adding New User Templates for Multiple Integrations chapter.

Requirements

The Avaya Definity G3 in-band TIMG integration supports configurations of the following components:

Phone System

- Avaya Definity G3 phone system.
- T1 digital trunk interface card.
- One or more TIMG units (media gateways).
- The voice messaging ports in the phone system connected by T1 digital lines (DS1 or “dry T1” digital lines only) to the ports on the TIMG units.

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

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

Unity Connection Server

- Unity Connection installed and ready for the integration after completing the installation following the steps as mentioned in the "Installing Cisco Unity Connection" chapter of the *Install, Upgrade, and Maintenance Guide for Cisco Unity Connection Release 14*, available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/14/unified_messaging/guide/b_14cucumgx.html.
- A license that enables the applicable number of voice messaging ports.

Centralized Voice Messaging

Unity Connection supports centralized voice messaging through the phone system, which supports various inter-phone system networking protocols including proprietary protocols such as Avaya DCS, Nortel MCDN, or Siemens CorNet, and standards-based protocols such as QSIG or DPNSS. Note that centralized voice messaging is a function of the phone system and its inter-phone system networking, not voicemail. Unity Connection supports centralized voice messaging as long as the phone system and its inter-phone system networking are properly configured.

Programming Phone System for In-Band TIMG Integration

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

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

Step 1 Create a coverage path which contains the TIMG unit hunt group number as the coverage point.

Step 2 Assign the coverage path that you created in Step 1 to the user stations that must forward to the voice messaging ports on the TIMG units when calls are not answered or when the user station is busy, based on one of the call transfer types shown in Table 3-1.

Table 2: Call Transfer Types

Transfer Type	Usage
Release transfer (blind transfer)	Program the user station to forward calls to the pilot number when: • The extension is busy. • The call is not answered.
Supervised transfer	Program the user station to forward calls to the pilot number only when the call is not answered (on the phone system, the number of rings before forwarding must be more than the number of rings to supervise the call). Confirm that call forwarding is disabled when the extension is busy.

Step 3 Assign an extension number for each voice messaging port, which is a digital line that connects to the TIMG unit, and set the voice messaging port options as shown in Table 3-2.

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

Table 3: Voice Messaging Port Options for All Lines

Option	Setting
Extension	Enter the extension number of the digital line.
Type	Enter the applicable setting: • **VMI** • **VMIFD**
Loss Group	Enter **4**.
Off Premises Station	Enter **y**.

Option	Setting
R Balance Network	Enter **n**.
Survivable COR	Enter **internal**.
Survivable Trunk Dest	Enter **y**.
LWC Activation	Enter **y**.
LWC Log External Calls	Enter **n**.
CDR Privacy	Enter **n**.
Redirect Notification	Enter **y**.
Per Button Ring Control	Enter **n**.
Bridged Call Alerting	Enter **n**.
Switch hook Flash	Enter **y**.
Ignore Rotary Digits	Enter **n**.
H.320 Conversion	Enter **n**.
Service Link Mode	Enter **as-needed**.
Multimedia Mode	Enter **basic**.
Coverage Message Retrieval	Enter **y**.
Auto Answer	Enter **none**.
Data Restriction	Enter **n**.
Distinctive Audible Alert	Enter **y**.
Adjunct Supervision	Enter **y**.
Audible Message Waiting	Enter **n**.
Coverage After Forwarding	Enter **s**.
Multimedia Early Answer	Enter **n**.
LWC Appearance	Enter **call-appr**.

Step 4 Set the DS1 Circuit Pack options as shown in Table 3-3.

Table 4: DS1 Circuit Pack Options

Option	Setting
Line Compensation	Enter **1**.

Option	Setting
Signaling Mode	Enter **robbed-bit**.
Line Coding	Enter **b8zs**.
Framing Mode	Enter **esf**.
Interface Companding	Enter **mulaw**.
Idle Code	Enter **11111111**.
Slip Detection	Enter **n**.
Echo Cancellation	Enter **n**.
Near-end CSU Type	Enter **other**.

Step 5 Assign a the voice messaging ports to a hunt group by setting the options shown in Table 3-4.

Table 5: Hunt Group Options

Option	Setting
LWC Reception	Enter **none**.
Message Center	Enter **none**.
Group Number	Enter the hunt group number.
Group Extension	Enter the pilot number for the hunt group.
Group Type	Enter **ucd-mia**.
Group Name	Enter the display name for the hunt group.
Queue?	Enter **n**.

Step 6 Enter the group member assignments for the voice messaging ports that answer calls and press **Enter**.

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

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

Setting Up the TIMG Units

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

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

- The phone system is connected to the TIMG units using T1 digital lines.
- The TIMG units are connected to a power source.

- The TIMG units are ready to be connected to the LAN or WAN.

Caution Because TIMG units have the same default IP address, you must set them up one at a time. Otherwise, you can experience IP address conflicts.

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

Step 1 On a Windows workstation that have access to the TIMG units, go to the following link: http://software.cisco.com/download/navigator.html?mdfid=280082558&i=rm.

Note To access the software download page, you must be signed in to Cisco.com as a registered user.

This procedure describes the steps when using Internet Explorer as your web browser. If you are using a different web browser, the steps may differ.

Step 2 In the tree control on the Downloads Home page, expand Unified Communications> Unified Communications Applications > Messaging > Cisco Unity and select Cisco Unity Telephony Integration.

Step 3 On the Log In page, enter your username and password, then select **Log In**.

Step 4 On the Select a Release page, under Latest Releases, select the most recent release.

Step 5 In the right column, select the version of the firmware for your TIMG units.

Step 6 On the Download Image page, select **Download**.

Step 7 On the Supporting Document(s) page, select **Agree**.

Step 8 In the File Download dialog box, select **Save**.

Step 9 In the Save As dialog box, browse to the Windows workstation that have access the TIMG units, browse to a directory where you want to save the file, and select **Save**.

Step 10 In the Download Complete dialog box, select **Open**. The window for extracting the TIMG firmware update files appears.

Step 11 Select **Extract**.

Step 12 In the Extract dialog box, browse to the directory where you want the extracted files, and select **Extract**.

Step 13 Close the window for the extracting application.

Setting Up the TIMG Units (Firmware Version 6.x)

Step 1 On the Windows workstation, add a temporary route to enable access to the TIMG units.

a) On the Windows Start menu, select **Run**.
b) Enter **cmd**, and press **Enter**. The Command Prompt window appears.
c) At the command prompt, enter **route add 10.12.13.74 <IP Address of Workstation>**, and press **Enter**.

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

d) Close the Command Prompt window.

Step 2 Connect a TIMG unit to the network.

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

Step 4 To sign in, enter the following case-sensitive settings.

Table 6: Sign-in Settings

Field	Setting
Username	Enter **admin**.
Password	Enter **IpodAdmin**.

Step 5 Select **OK**.

Step 6 On the System menu, select **Upgrade**.

Step 7 On the Upgrade page, under Browse for Upgrade File, select **Browse**.

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

Step 9 Select **T1E1_<xx > .app** (where <xx> is multiple digits), and select **Open**.

Step 10 On the Upgrade page, select **Install File**.

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

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

Step 12 Repeat Step 6 through Step 11 for the following files:

- T1E1_<xx>.fsh
- T1E1_<xx>.msd

Step 13 On the Configuration menu, select **Import/Export**.

Step 14 On the Import/Export page, select **Browse**.

Step 15 In the Choose File dialog box, browse to the file T1_LS_Cfg_LucentG3.ini.

Step 16 Select **T1_LS_Cfg_LucentG3.ini**, and select **Open**.

Step 17 On the Import/Export page, select **Import File**.

Step 18 After the file is imported, a message prompting you to restart the TIMG unit appears. Select **OK**.

Step 19 In the web browser, go to **http://10.12.13.74**.

Step 20 To sign in, enter the following case-sensitive settings.

Table 7: Sign-in Settings

Field	Setting
Username	Enter **admin**.
Password	Enter **IpodAdmin**.

Step 21 Select **OK**.

Step 22 On the System menu, select **Password**.

Step 23 On the Change Password page, enter the following settings.

Table 8: Change Password Page Settings

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

Step 24 Select **Change**.

Step 25 On the Configuration menu, select **Mgmt Protocols**.

Step 26 On the Management Protocols page, enter the following settings.

Table 9: Management Protocols Page Settings

Field	Settings
E-mail Alarms Enabled	Select **No**.
SNMP Traps Enabled	Select **No**.
HTTP Server Enabled	Select **Yes**.
HTTPs Server Enabled	Select **No**.

Step 27 Select **Submit**.

Step 28 On the Configuration menu, select **Routing Table**.

Step 29 On the Routing Table page, under Router Configuration, select **VoIP Host Groups**.

Step 30 Under VoIP Host Groups, enter the following settings for the first VoIP Host Group.

Table 10: First VoIP Host Group Settings

Field	Settings
Name	Accept the default or enter another descriptive name of the VoIP host group.
Load-Balanced	*(Unity Connection without a cluster)* Select **False**. *(Unity Connection with a cluster configured)* Select **False**.
Fault-Tolerant	*(Unity Connection without a cluster)* Select **False**. *(Unity Connection with a cluster configured)* Select **True**.

Step 31 For Unity Connection without a cluster, under Host List, enter the host name or IP address of the Unity Connection server and the server port in the format <host name or IP address>:5060.

For Unity Connection with a cluster configured, under Host List, enter the host name or IP address of the subscriber server (the second Unity Connection server that you installed) and the server port in the format <host name or IP address>:5060.

Step 32 For Unity Connection without a cluster, continue to Step 34. For Unity Connection with a cluster configured, select **Add Host**.

Step 33 In the second field, enter the host name or IP address of the publisher server (the first Unity Connection server that you installed) and the server port in the format <host name or IP address>:5060.

Note Do not add a third host under Host List or a second host group under VoIP Host Groups. Otherwise, the Unity Connection cluster may not function correctly.

Step 34 Select **Submit**.

Step 35 On the Configuration menu, select **TDM** > **T1/E1**.

Step 36 On the T1/E1 Configuration page, enter the following settings.

Table 11: T1/E1 Configuration Page Settings

Field	Settings
Line Settings	
Line Mode	Select **T1**.
Signaling Mode	Select **CAS**.
Interface Mode	Select **Terminal**.
T1 Line	
Line Encoding	Select **B8ZS**.
Framing	Select **EFS**.
Selects Transmit Pulse Waveform	Select **Short_Haul_110ft**.
T1 CAS Protocol	
T1 CAS Protocol	Select **Loop_Start**.
Flash Hook	Enter **550**.
Consult Call Dialtone Drop Code	Enter **!!**.
Consult Call Proceeding Drop Code	Enter **!!**.
Consult Call Busy Drop Code	Enter **!**.
Consult Call Error Drop Code	Enter **!!**.
Consult Call Connected Drop Code	Enter **,,,,**.

Field	Settings
Consult Call Disconnected Drop Code	Enter **!**.
MWI confirmation Tone	Select **No**.
CPID Type	Select **TypeII_CPID**.
Initial Wait for Inband CPID	Enter **5000**.
Inband CPID Complete Timeout	Enter **500**.
Failover Settings	
Enable Failover	Select **No**.

Step 37 Select **Submit**.

Step 38 On the Configuration menu, select **TDM** > **General**.

Step 39 On the TDM General Settings page, enter the following settings.

Table 12: TDM General Settings Page Settings

Field	Settings
PCM Coding	Select **uLaw**.
Minimum Call Party Delay (ms)	Enter **500**.
Maximum Call Party Delay (ms)	Enter **2000**.
Dial Digit on Time (ms)	Enter **100**.
Dial Inter-Digit Time (ms)	Enter **100**.
Dial Pause Time (ms)	Enter **2000**.
Turn MWI On FAC	Enter the code that the phone system uses to turn MWIs on.
Turn MWI Off FAC	Enter the code that the phone system uses to turn MWIs off.
Outbound Call Connect Timeout (ms)	Enter **10000**.
Wait for Ringback/Connect on Blind Transfer	Select **Yes**.
Hunt Group Extension	Enter the pilot number of the Unity Connection voice messaging ports.

Step 40 Select **Submit**.

Step 41 On the Configuration menu, select **TDM** > **Port Enable**.

Step 42 On the TDM Port Enabling page, select **No** for the ports that you want to disable on the TIMG unit.

Step 43 Confirm that **Yes** is selected for all other ports on the TIMG unit.

Step 44 Select **Submit**.

Step 45 On the Configuration menu, select **VoIP** > **General**.

Step 46 On the VoIP General Settings page, enter the following settings.

Table 13: VoIP General Settings Page Settings

Field	Setting
User-Agent	
Host and Domain Name	Enter the domain name of the TIMG unit.
Transport Type	Select **UDP**.
Call as Domain Name	Select **No**.
SIPS URI Scheme Enabled	Select **No**.
Invite Expiration (sec)	Enter **120**.
Server	
DNS Server Address	Enter the IP Address of the Domain Name Server that the TIMG unit uses.
Registration Server Address	Leave this field blank.
Registration Server Port	Enter **5060**.
Registration Expiration (sec)	Enter **3600**.
TCP/UDP	
UDP/TCP Transports Enabled	Select **Yes**.
TCP/UDP Server Port	Enter **5060**.
Proxy	
Primary Proxy Server Address	Leave this field blank.
Primary Proxy Server Port	Not applicable. Leave the default setting.
Backup Proxy Server Address	Not applicable. Leave the default setting.

Field	Setting
Backup Proxy Server Port	Not applicable. Leave the default setting.
Proxy Query Interval	Enter **10**.
Timing	
T1 Time	Enter **500**.
T2 Time	Enter **4000**.
T4 Time	Enter **5000**.
Monitoring	
Monitor Call Connections	Select **No**.

Step 47 Select **Submit**.

Step 48 On the Configuration menu, select **VoIP** > **Media**.

Step 49 On the VoIP Media Settings page, enter the following settings.

Table 14: VoIP Media Settings Page Settings

Field	Settings
Audio	
Audio Compression	Select the preferred codec for audio compression: • **G.711u**—The TIMG unit uses only the G.711 mu-law codec. • **G.729AB**—The TIMG unit prefers the G.729 codec but can also use the G.711 mu-law codec.
RTP Digit Relay Mode	Select **RFC2833**.
Signaling Digit Relay Mode	Select **Off**.
Voice Activity Detection	Select **On**.
Frame Size	Select the applicable setting: • G.711—**20** • G.729AB—**10** **Caution** Failure to use the correct setting results in recorded messages containing nothing but silence.

Field	Settings
Frames Per Packet	Select the applicable setting: • G.711—**1** • G.729AB—**2** **Caution** Failure to use the correct setting results in recorded messages containing nothing but silence.

Step 50 Select **Submit**.

Step 51 On the Configuration menu, select **VoIP** > **QOS**.

Step 52 On the VoIP QOS Configuration page, enter the following settings.

Table 15: VoIP QOS Configurative Page Settings

Field	Settings
Call Control QOS Byte	Enter **104** (equivalent to DSCP AF31).
RTP QOS Byte	Enter **184** (equivalent to DSCP EF).

Step 53 Select **Submit**.

Step 54 On the Configuration menu, select **IP**.

Step 55 On the IP Settings, LAN1 page, enter the following settings.

Table 16: IP Settings, LAN1 Page Settings

Field	Settings
Client IP Address	Enter the new IP address that you want to use for the TIMG unit. (This is the IP address that you enter in Cisco Unity Connection Administration when you create the integration.)
Client Subnet Mask	Enter the new subnet mask, if the subnet mask is different from the default IP address.
Default Network Gateway Address	Enter the IP address of the default network gateway router that the TIMG units use.
BOOTP Enabled	If you are using DHCP, select **Yes**. If you are not using DHCP, select **No**.

Step 56 Select **Submit**.

Step 57 On the Configuration menu, select **Tone Detection**.

Step 58 On the Tone Detection page, under Call Progress Tone - Learn, in the Learn Tone Event field, select **Busy** and do the following substeps to verify that the tone is correct.

a) From a available phone, call a second phone.

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

Step 59 Under Call Progress Tone - Learn, in the Dial String field, enter the extension that you dialed in Step 58c from the third phone.

Step 60 Select **Learn**.

Step 61 On the Tone Detection page, under Call Progress Tone - Learn, in the Learn field, select **Error** and do the following substeps to verify that the tone is correct.

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

Step 62 Under Call Progress Tone - Learn, in the Dial String field, enter the extension that you dialed in Step 61a

Step 63 Select **Learn**.

Step 64 On the Tone Detection page, under Call Progress Tone - Learn, in the Learn field, select **Ringback** and do the following substeps to verify that the tone is correct.

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

Step 65 Under Call Progress Tone - Learn, in the Dial String field, enter the extension that you dialed in Step 64a

Step 66 Select **Learn**.

Step 67 Select **Submit**.

Step 68 Hang up the phones that you used in Step 58.

Step 69 On the Configuration menu, select **Import/Export**.

Step 70 On the Import/Export page, under Export Files, select **Export All Settings**.

Step 71 In the File Download dialog box, select **Save**.

Step 72 In the Save As dialog box, browse to the Windows workstation that has access to the TIMG units, browse to a directory where you want to save the file, and select **Save**.

Step 73 In the Download Complete dialog box, select **Open**. Notepad opens the file Config.ini that you saved.

Step 74 Locate the line with the following parameter:

```
telautoanswer
```

Step 75 Confirm that the value of the parameter is **no** so that the line reads as follows:

```
telautoanswer = no
```

Step 76 Locate the line with the following parameter:

```
telFacCDropProc
```

Step 77 Confirm that the value of the parameter is **!!** so that the line reads as follows:

```
telFacCDropProc = !!
```

Caution The telFacCDropProc parameter must be set to !!. If the telFacCDropProc parameter is set to 1, supervised transfers fail, and the caller hears the called party standard greeting two times.

Step 78 Save the file, and exit Notepad.

Step 79 On the Configuration menu of the TIMG unit, select **Import/Export**.

Step 80 On the Import/Export page, under Browse for Import File, select **Browse**.

Step 81 In the Choose File dialog box, browse to the file Config.ini that you saved.

Step 82 Select **Config.ini**, and select **Open**.

Step 83 On the Import/Export page, select **Import File**.

Step 84 When prompted to restart the TIMG unit, select **OK**.

Step 85 Repeat Step 2 through Step 84 on all remaining TIMG units.

Creating an Integration with the Phone System

After ensuring that the phone system, the TIMG units and the Unity Connection server are ready for the integration, do the following procedures to set up the integration and to enter the port settings.

Step 1 In Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Phone System**.

Step 2 On the Search Phone Systems page, under Display Name, select the name of the default phone system.

Step 3 On the Phone System Basics page, in the Phone System Name field, enter the descriptive name that you want for the phone system.

Step 4 If you want to use this phone system as the default for TRaP connections so that administrators and users without voicemail boxes can record and playback through the phone in Unity Connection web applications, check the **Default TRAP Switch** check box. If you want to use another phone system as the default for TRaP connections, uncheck this check box.

Step 5 Select **Save**.

Step 6 On the Phone System Basics page, in the Related Links drop-down box, select **Add Port Group** and select **Go**.

Step 7 On the New Port Group page, enter the applicable settings and select **Save**.

Table 17: Settings for the New Port Group Page

Field	Setting
Phone System	Select the name of the phone system that you entered in Step 3.
Create From	Select **Port Group Template** and select **SIP to DMG/PIMG/TIMG** in the drop-down box.
Display Name	Enter a descriptive name for the port group. You can accept the default name or enter the name that you want.
SIP Security Profile	Select **5060**.
SIP Transport Protocol	Select the SIP transport protocol that Unity Connection uses.

Field	Setting
IPv4 Address or Host Name	Enter the IP address of the TIMG unit that you are integrating with Unity Connection.
IPv6 Address or Host Name	Do not enter a value in this field. IPv6 is not supported for TIMG integrations.
IP Address or Host Name	Enter the IP address of the TIMG unit that you are integrating with Unity Connection.
Port	Enter the SIP port of the TIMG unit that Unity Connection connects to. You should use the default setting. **Caution** This name must match the setting in the TCP/UDP Server Port field on the Configuration > VoIP > General page of the TIMG unit. Otherwise, the integration do not function correctly.

Step 8 On the Port Group Basics page, in the Related Links drop-down box, select **Add Ports** and select **Go**.

Step 9 On the New Port page, enter the following settings and select **Save**.

Table 18: Settings for the New Ports Page

Field	Considerations
Enabled	Check this check box.
Number of Ports	Enter the number of voice messaging ports that you want to create in this port group. **Note** For a Unity Connection cluster, the server must have the number of voice messaging ports that are set up on the phone system for the TIMG integration so that this server can handle all voice messaging traffic for the cluster if one of the servers stops functioning. For example, if the phone system is set up with 16 voice messaging ports, this server must have 16 voice messaging ports.
Phone System	Select the name of the phone system that you entered in Step 3.
Port Group	Select the name of the port group that you added in Step 7.

Step 10 On the Search Ports page, select the display name of the first voice messaging port that you created for this phone system integration.

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

Step 11 On the Port Basics page, set the voice messaging port settings as applicable. The fields in the following table are the ones that you can change.

Table 19: Settings for the Voice Messaging Ports

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

Step 12 Select **Save**.

Step 13 Select **Next**.

Step 14 Repeat Step 11 through Step 13 for all remaining voice messaging ports for the phone system.

Step 15 In Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Phone System**.

Step 16 On the Search Phone Systems page, under Display Name, select the name of the phone system that you entered in Step 3.

Step 17 Repeat Step 6 through Step 16 for each remaining TIMG unit integrated with Unity Connection.

Note Each TIMG unit is connected to one port group with the applicable voice messaging ports. For example, a system that uses two TIMG units requires two port groups, one port group for each TIMG unit.

Step 18 If another phone system integration exists, in Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Trunk**. Otherwise, skip to Step 22.

Step 19 On the Search Phone System Trunks page, on the Phone System Trunk menu, select **New Phone System Trunk**.

Step 20 On the New Phone System Trunk page, enter the following settings for the phone system trunk and select **Save**.

Table 20: Settings for the Phone System Trunk

Field	Setting
From Phone System	Enter the display name of the phone system that you are creating a trunk for.

Field	Setting
To Phone System	Enter the display name of the previously existing phone system that the trunk connects to.
Trunk Access Code	Enter the extra digits that Unity Connection must dial to transfer calls through the gateway to extensions on the previously existing phone system.

Step 21 Repeat Step 19 and Step 20 for all remaining phone system trunks that you want to create.

Step 22 In Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 23 On the Search Port Groups page, select the display name of the port group that you added in Step 7.

Step 24 On the Port Group Basics page, select **Reset**.

Step 25 When prompted that resetting terminates all call traffic, select **OK**.

Step 26 In the Related Links drop-down list, select **Check Telephony Configuration** and select **Go** to confirm the phone system integration settings.

If the test is not successful, the Task Execution Results displays one or more messages with troubleshooting steps. After correcting the problems, test the connection again.

Step 27 In the Task Execution Results window, select **Close**.

CHAPTER 4

Setting Up an Avaya S8500/S8700 In-Band TIMG Integration with Cisco Unity Connection

Introduction

This chapter contains detailed instructions for setting up an Avaya S8500/S8700 in-band TIMG integration with Cisco Unity Connection,

Note Caller ID is supported for internal callers only. The Avaya S8500/S8700 phone system does not support caller ID for external callers.

Task List to Create an Avaya S8500/S8700 In-Band TIMG Integration

Before doing the following tasks to integrate Unity Connection with the phone system using the T1 media gateway (TIMG), confirm that the Cisco Unity Connection server is ready for the integration after completing the installation following the steps as mentioned in the "Installing Cisco Unity Connection" chapter of the *Install, Upgrade, and Maintenance Guide for Cisco Unity Connection Release 15*, available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/unified_messaging/guide/b_15cucumgx.html.

Review the system and equipment requirements to confirm that all phone system and Unity Connection server requirements have been met. See the Requirements section.

1. Plan how the voice messaging ports used by Unity Connection. See the Planning the Usage of Voice Messaging Ports in Cisco Unity Connection chapter.
2. Program the phone system and extensions. See the Programming Phone System for In-Band TIMG Integration section.
3. Set up the TIMG units for an in-band integration. See the Setting Up the TIMG Units section.
4. Create the integration. See the Creating an Integration with the Phone System section.
5. Test the integration. See the Testing the Integration chapter.
6. If this integration is a second or subsequent integration, add the applicable new user templates for the new phone system. See the Adding New User Templates for Multiple Integrations chapter.

Requirements

The Avaya S8500/S8700 in-band TIMG integration supports configurations of the following components:

Phone System

- Avaya S8500/S8700 phone system.
- Software version Cisco Unified CallManager 2.0.
- T1 digital trunk interface card.
- One or more TIMG units (media gateways).
- The voice messaging ports in the phone system connected by T1 digital lines (DS1 or “dry T1” digital lines only) to the ports on the TIMG units.

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

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

Unity Connection Server

The Avaya S8500/S8700 in-band TIMG integration supports configurations of the following components:

- Unity Connection installed and ready for the integration after completing the installation following the steps as mentioned in the "Install" chapter of the *Install, Upgrade, and Maintenance Guide for Cisco Unity Connection Release 15*, available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/unified_messaging/guide/b_15cucumgx.html.
- A license that enables the applicable number of voice messaging ports.

Centralized Voice Messaging

Unity Connection supports centralized voice messaging through the phone system, which supports various inter-phone system networking protocols including proprietary protocols such as Avaya DCS, Nortel MCDN, or Siemens CorNet, and standards-based protocols such as QSIG or DPNSS. Note that centralized voice messaging is a function of the phone system and its inter-phone system networking, not voicemail. Unity Connection supports centralized voice messaging as long as the phone system and its inter-phone system networking are properly configured.

Programming Phone System for In-Band TIMG Integration

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

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

SUMMARY STEPS

1. Create a coverage path which contains the TIMG unit hunt group number as the coverage point.
2. Assign the coverage path that you created in Step 1 to the user stations that must forward to the voice messaging ports on the TIMG units when calls are not answered or when the user station is busy, based on one of the call transfer types shown in Table 4-1.
3. Assign an extension number for each voice messaging port, which is a digital line that connects to the TIMG unit, and set the voice messaging port options as shown in Table 4-2.
4. Set the DS1 Circuit Pack options as shown in Table 4-3.
5. Assign a the voice messaging ports to a hunt group by setting the options shown in Table 4-4.
6. Enter the group member assignments for the voice messaging ports that answer calls and press **Enter**.

DETAILED STEPS

Step 1 Create a coverage path which contains the TIMG unit hunt group number as the coverage point.

Step 2 Assign the coverage path that you created in Step 1 to the user stations that must forward to the voice messaging ports on the TIMG units when calls are not answered or when the user station is busy, based on one of the call transfer types shown in Table 4-1.

Table 21: Call Transfer Types

Transfer Type	Usage
Release transfer (blind transfer)	Program the user station to forward calls to the pilot number when: • The extension is busy. • The call is not answered.
Supervised transfer	Program the user station to forward calls to the pilot number only when the call is not answered (on the phone system, the number of rings before forwarding must be more than the number of rings to supervise the call). Confirm that call forwarding is disabled when the extension is busy.

Step 3 Assign an extension number for each voice messaging port, which is a digital line that connects to the TIMG unit, and set the voice messaging port options as shown in Table 4-2.

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

Table 22: Voice Messaging Port Options for All Lines

Option	Setting
Extension	Enter the extension number of the digital line.
Type	Enter the applicable setting: • **VMI** • **VMIFD**
Loss Group	Enter **4**.
Off Premises Station	Enter **y**.
R Balance Network	Enter **n**.
Survivable COR	Enter **internal**.
Survivable Trunk Dest	Enter **y**.
LWC Activation	Enter **y**.
LWC Log External Calls	Enter **n**.
CDR Privacy	Enter **n**.
Redirect Notification	Enter **y**.
Per Button Ring Control	Enter **n**.

Option	Setting
Bridged Call Alerting	Enter **n**.
Switch hook Flash	Enter **y**.
Ignore Rotary Digits	Enter **n**.
H.320 Conversion	Enter **n**.
Service Link Mode	Enter **as-needed**.
Multimedia Mode	Enter **basic**.
Coverage Message Retrieval	Enter **y**.
Auto Answer	Enter **none**.
Data Restriction	Enter **n**.
Distinctive Audible Alert	Enter **y**.
Adjunct Supervision	Enter **y**.
Audible Message Waiting	Enter **n**.
Coverage After Forwarding	Enter **s**.
Multimedia Early Answer	Enter **n**.
LWC Appearance	Enter **call-appr**.

Step 4 Set the DS1 Circuit Pack options as shown in Table 4-3.

Table 23: DS1 Circuit Pack Options

Option	Setting
Line Compensation	Enter **1**.
Signaling Mode	Enter **robbed-bit**.
Line Coding	Enter **b8zs**.
Framing Mode	Enter **esf**.
Interface Companding	Enter **mulaw**.
Idle Code	Enter **11111111**.
Slip Detection	Enter **n**.
Echo Cancellation	Enter **n**.
Near-end CSU Type	Enter **other**.

Step 5 Assign a the voice messaging ports to a hunt group by setting the options shown in Table 4-4.

Table 24: Hunt Group Options

Option	Setting
LWC Reception	Enter **none**.
Message Center	Enter **none**.
Group Number	Enter the hunt group number.
Group Extension	Enter the pilot number for the hunt group.
Group Type	Enter **ucd-mia**.
Group Name	Enter the display name for the hunt group.
Queue?	Enter **n**.

Step 6 Enter the group member assignments for the voice messaging ports that answer calls and press **Enter**.

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

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

Setting Up the TIMG Units

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

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

- The phone system is connected to the TIMG units using T1 digital lines.
- The TIMG units are connected to a power source.
- The TIMG units are ready to be connected to the LAN or WAN.

Note Because TIMG units have the same default IP address, you must set them up one at a time. Otherwise, you can experience IP address conflicts.

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

Downloading TIMG Firmware Update Files

Step 1 On a Windows workstation that have access to the TIMG units, go to the following link: http://software.cisco.com/download/navigator.html?mdfid=280082558&i=rm.

Note To access the software download page, you must be signed in to Cisco.com as a registered user.

This procedure describes the steps when using Internet Explorer as your web browser. If you are using a different web browser, the steps may differ.

Step 2 In the tree control on the Downloads Home page, expand Unified Communications> Unified Communications Applications > Messaging > Cisco Unity and select Cisco Unity Telephony Integration.

Step 3 On the Log In page, enter your username and password, then select **Log In**.

Step 4 On the Select a Release page, under Latest Releases, select the most recent release.

Step 5 In the right column, select the version of the firmware for your TIMG units.

Step 6 On the Download Image page, select **Download**.

Step 7 On the Supporting Document(s) page, select **Agree**.

Step 8 In the File Download dialog box, select **Save**.

Step 9 In the Save As dialog box, browse to the Windows workstation that have access the TIMG units, browse to a directory where you want to save the file, and select **Save**.

Step 10 In the Download Complete dialog box, select **Open**. The window for extracting the TIMG firmware update files appears.

Step 11 Select **Extract**.

Step 12 In the Extract dialog box, browse to the directory where you want the extracted files, and select **Extract**.

Step 13 Close the window for the extracting application.

Setting Up the TIMG Units (Firmware Version 6.x)

Step 1 On the Windows workstation, add a temporary route to enable access to the TIMG units.

a) On the Windows Start menu, select **Run**.
b) Enter **cmd**, and press **Enter**. The Command Prompt window appears.
c) At the command prompt, enter **route add 10.12.13.74 <IP Address of Workstation>**, and press **Enter**.

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

d) Close the Command Prompt window.

Step 2 Connect a TIMG unit to the network.

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

Step 4 To sign in, enter the following case-sensitive settings.

Table 25: Sign-in Settings

Field	Setting
Username	Enter **admin**.
Password	Enter **IpodAdmin**.

Step 5 Select **OK**.

Step 6 On the System menu, select **Upgrade**.

Step 7 On the Upgrade page, under Browse for Upgrade File, select **Browse**.

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

Step 9 Select **T1E1_<xx** > **.app** (where <xx> is multiple digits), and select **Open**.

Step 10 On the Upgrade page, select **Install File**.

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

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

Step 12 Repeat Step 6 through Step 11 for the following files:

- T1E1_<xx>.fsh
- T1E1_<xx>.msd

Step 13 On the Configuration menu, select **Import/Export**.

Step 14 On the Import/Export page, select **Browse**.

Step 15 In the Choose File dialog box, browse to the file T1_LS_Cfg_Avaya8500.ini.

Step 16 Select **T1_LS_Cfg_Avaya8500.ini**, and select **Open**.

Step 17 On the Import/Export page, select **Import File**.

Step 18 After the file is imported, a message prompting you to restart the TIMG unit appears. Select **OK**.

Step 19 In the web browser, go to **http://10.12.13.74**.

Step 20 To sign in, enter the following case-sensitive settings.

Table 26: Sign-in Settings

Field	Setting
Username	Enter **admin**.
Password	Enter **IpodAdmin**.

Step 21 Select **OK**.

Step 22 On the System menu, select **Password**.

Step 23 On the Change Password page, enter the following settings.

Table 27: Change Password Page Settings

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

Step 24 Select **Change**.

Step 25 On the Configuration menu, select **Mgmt Protocols**.

Step 26 On the Management Protocols page, enter the following settings.

Table 28: Management Protocols Page Settings

Field	Settings
E-mail Alarms Enabled	Select **No**.
SNMP Traps Enabled	Select **No**.
HTTP Server Enabled	Select **Yes**.
HTTPs Server Enabled	Select **No**.

Step 27 Select **Submit**.

Step 28 On the Configuration menu, select **Routing Table**.

Step 29 On the Routing Table page, under Router Configuration, select **VoIP Host Groups**.

Step 30 Under VoIP Host Groups, enter the following settings for the first VoIP Host Group.

Table 29: First VoIP Host Group Settings

Field	Settings
Name	Accept the default or enter another descriptive name of the VoIP host group.
Load-Balanced	*(Unity Connection without a cluster)* Select **False**. *(Unity Connection with a cluster configured)* Select **False**.
Fault-Tolerant	*(Unity Connection without a cluster)* Select **False**. *(Unity Connection with a cluster configured)* Select **True**.

Step 31 For Unity Connection without a cluster, under Host List, enter the host name or IP address of the Unity Connection server and the server port in the format <host name or IP address>:5060.

For Unity Connection with a cluster configured, under Host List, enter the host name or IP address of the subscriber server (the second Unity Connection server that you installed) and the server port in the format <host name or IP address>:5060.

Step 32 For Unity Connection without a cluster, continue to Step 34. For Unity Connection with a cluster configured, select **Add Host**.

Step 33 In the second field, enter the host name or IP address of the publisher server (the first Unity Connection server that you installed) and the server port in the format <host name or IP address>:5060.

Caution Do not add a third host under Host List or a second host group under VoIP Host Groups. Otherwise, the Unity Connection cluster may not function correctly.

Step 34 Select **Submit**.

Step 35 On the Configuration menu, select **TDM** > **T1/E1**.

Step 36 On the T1/E1 Configuration page, enter the following settings.

Table 30: T1/E1 Configuration Page Settings

Field	Settings
Line Settings	
Line Mode	Select **T1**.
Signaling Mode	Select **CAS**.
Interface Mode	Select **Terminal**.
T1 Line	
Line Encoding	Select **B8ZS**.
Framing	Select **EFS**.
Selects Transmit Pulse Waveform	Select **Short_Haul_110ft**.
T1 CAS Protocol	
T1 CAS Protocol	Select **Loop_Start**.
Flash Hook	Enter **550**.
Consult Call Dialtone Drop Code	Enter **!!**.
Consult Call Proceeding Drop Code	Enter **!!**.
Consult Call Busy Drop Code	Enter **!**.
Consult Call Error Drop Code	Enter **!!**.

Field	Settings
Consult Call Connected Drop Code	Enter **,,,,**.
Consult Call Disconnected Drop Code	Enter **!**.
MWI confirmation Tone	Select **No**.
CPID Type	Select **TypeII_CPID**.
Initial Wait for Inband CPID	Enter **5000**.
Inband CPID Complete Timeout	Enter **500**.
Failover Settings	
Enable Failover	Select **No**.

Step 37 Select **Submit**.

Step 38 On the Configuration menu, select **TDM** > **General**.

Step 39 On the TDM General Settings page, enter the following settings.

Table 31: TDM General Settings Page Settings

Field	Settings
PCM Coding	Select **uLaw**.
Minimum Call Party Delay (ms)	Enter **500**.
Maximum Call Party Delay (ms)	Enter **2000**.
Dial Digit on Time (ms)	Enter **100**.
Dial Inter-Digit Time (ms)	Enter **100**.
Dial Pause Time (ms)	Enter **2000**.
Turn MWI On FAC	Enter the code that the phone system uses to turn MWIs on.
Turn MWI Off FAC	Enter the code that the phone system uses to turn MWIs off.
Outbound Call Connect Timeout (ms)	Enter **10000**.
Wait for Ringback/Connect on Blind Transfer	Select **Yes**.

Field	Settings
Hunt Group Extension	Enter the pilot number of the Unity Connection voice messaging ports.

Step 40 Select **Submit**.

Step 41 On the Configuration menu, select **TDM** > **Port Enable**.

Step 42 On the TDM Port Enabling page, select **No** for the ports that you want to disable on the TIMG unit.

Step 43 Confirm that **Yes** is selected for all other ports on the TIMG unit.

Step 44 Select **Submit**.

Step 45 On the Configuration menu, select **VoIP** > **General**.

Step 46 On the VoIP General Settings page, enter the following settings.

Table 32: VoIP General Settings Page Settings

Field	Setting
User-Agent	
Host and Domain Name	Enter the domain name of the TIMG unit.
Transport Type	Select **UDP**.
Call as Domain Name	Select **No**.
SIPS URI Scheme Enabled	Select **No**.
Invite Expiration (sec)	Enter **120**.
Server	
DNS Server Address	Enter the IP Address of the Domain Name Server that the TIMG unit use.
Registration Server Address	Leave this field blank.
Registration Server Port	Enter **5060**.
Registration Expiration (sec)	Enter **3600**.
TCP/UDP	
UDP/TCP Transports Enabled	Select **Yes**.
TCP/UDP Server Port	Enter **5060**.
Proxy	
Primary Proxy Server Address	Leave this field blank.

Field	Setting
Primary Proxy Server Port	Not applicable. Leave the default setting.
Backup Proxy Server Address	Not applicable. Leave the default setting.
Backup Proxy Server Port	Not applicable. Leave the default setting.
Proxy Query Interval	Enter **10**.
Timing	
T1 Time	Enter **500**.
T2 Time	Enter **4000**.
T4 Time	Enter **5000**.
Monitoring	
Monitor Call Connections	Select **No**.

Step 47 Select **Submit**.

Step 48 On the Configuration menu, select **VoIP** > **Media**.

Step 49 On the VoIP Media Settings page, enter the following settings.

Table 33: VoIP Media Settings Page Settings

Field	Settings
Audio	
Audio Compression	Select the preferred codec for audio compression: • **G.711u**—The TIMG unit uses only the G.711 mu-law codec. • **G.729AB**—The TIMG unit prefers the G.729 codec but can also use the G.711 mu-law codec.
RTP Digit Relay Mode	Select **RFC2833**.
Signaling Digit Relay Mode	Select **Off**.
Voice Activity Detection	Select **On**.

<table>
<tr><th>Field</th><th>Settings</th></tr>
<tr><td>Frame Size</td><td>Select the applicable setting:
G.711—20
G.729AB—10
Caution Failure to use the correct setting results in recorded messages containing nothing but silence.</td></tr>
<tr><td>Frames Per Packet</td><td>Select the applicable setting:
G.711—1
G.729AB—2
Caution Failure to use the correct setting results in recorded messages containing nothing but silence.</td></tr>
</table>

Step 50 Select **Submit**.

Step 51 On the Configuration menu, select **VoIP** > **QOS**.

Step 52 On the VoIP QOS Configuration page, enter the following settings.

Table 34: VoIP QOS Configurative Page Settings

Field	Settings
Call Control QOS Byte	Enter **104** (equivalent to DSCP AF31).
RTP QOS Byte	Enter **184** (equivalent to DSCP EF).

Step 53 Select **Submit**.

Step 54 On the Configuration menu, select **IP**.

Step 55 On the IP Settings, LAN1 page, enter the following settings.

Table 35: IP Settings, LAN1 Page Settings

Field	Settings
Client IP Address	Enter the new IP address that you want to use for the TIMG unit. (This is the IP address that you enter in Cisco Unity Connection Administration when you create the integration.)
Client Subnet Mask	Enter the new subnet mask, if the subnet mask is different from the default IP address.
Default Network Gateway Address	Enter the IP address of the default network gateway router that the TIMG units use.

Field	Settings
BOOTP Enabled	If you are using DHCP, select **Yes**. If you are not using DHCP, select **No**.

Step 56 Select **Submit**.

Step 57 On the Configuration menu, select **Tone Detection**.

Step 58 On the Tone Detection page, under Call Progress Tone - Learn, in the Learn Tone Event field, select **Busy** and do the following substeps to verify that the tone is correct.

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

Step 59 Under Call Progress Tone - Learn, in the Dial String field, enter the extension that you dialed in Step 58c. from the third phone.

Step 60 Select **Learn**.

Step 61 On the Tone Detection page, under Call Progress Tone - Learn, in the Learn field, select **Error** and do the following substeps to verify that the tone is correct.

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

Step 62 Under Call Progress Tone - Learn, in the Dial String field, enter the extension that you dialed in Step 61a.

Step 63 Select **Learn**.

Step 64 On the Tone Detection page, under Call Progress Tone - Learn, in the Learn field, select **Ringback** and do the following substeps to verify that the tone is correct.

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

Step 65 Under Call Progress Tone - Learn, in the Dial String field, enter the extension that you dialed in Step 64a.

Step 66 Select **Learn**.

Step 67 Select **Submit**.

Step 68 Hang up the phones that you used in Step 58.

Step 69 On the Configuration menu, select **Import/Export**.

Step 70 On the Import/Export page, under Export Files, select **Export All Settings**.

Step 71 In the File Download dialog box, select **Save**.

Step 72 In the Save As dialog box, browse to the Windows workstation that has access to the TIMG units, browse to a directory where you want to save the file, and select **Save**.

Step 73 In the Download Complete dialog box, select **Open**. Notepad opens the file Config.ini that you saved.

Step 74 Locate the line with the following parameter:

```
telautoanswer
```

Step 75 Confirm that the value of the parameter is **no** so that the line reads as follows:

```
telautoanswer = no
```

Step 76 Locate the line with the following parameter:

```
telFacCDropProc
```

Step 77 Confirm that the value of the parameter is **!!** so that the line reads as follows:

```
telFacCDropProc = !!
```

Caution The telFacCDropProc parameter must be set to !!. If the telFacCDropProc parameter is set to 1, supervised transfers fail, and the caller hears the called party standard greeting two times.

Step 78 Save the file, and exit Notepad.

Step 79 On the Configuration menu of the TIMG unit, select **Import/Export**.

Step 80 On the Import/Export page, under Browse for Import File, select **Browse**.

Step 81 In the Choose File dialog box, browse to the file Config.ini that you saved.

Step 82 Select **Config.ini**, and select **Open**.

Step 83 On the Import/Export page, select **Import File**.

Step 84 When prompted to restart the TIMG unit, select **OK**.

Step 85 Repeat Step 2 through Step 84 on all remaining TIMG units.

Creating an Integration with the Phone System

After ensuring that the phone system, the TIMG units, and the Unity Connection server are ready for the integration, do the following procedures to set up the integration and to enter the port settings.

Step 1 In Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Phone System**.

Step 2 On the Search Phone Systems page, under Display Name, select the name of the default phone system.

Step 3 On the Phone System Basics page, in the Phone System Name field, enter the descriptive name that you want for the phone system.

Step 4 If you want to use this phone system as the default for TRaP connections so that administrators and users without voicemail boxes can record and playback through the phone in Unity Connection web applications, check the **Default TRAP Switch** check box. If you want to use another phone system as the default for TRaP connections, uncheck this check box.

Step 5 Select **Save**.

Step 6 On the Phone System Basics page, in the Related Links drop-down box, select **Add Port Group** and select **Go**.

Step 7 On the New Port Group page, enter the applicable settings and select **Save**.

Table 36: Settings for the New Port Group Page

Field	Setting
Phone System	Select the name of the phone system that you entered in Step 3.
Create From	Select **Port Group Template** and select **SIP to DMG/PIMG/TIMG** in the drop-down box.
Display Name	Enter a descriptive name for the port group. You can accept the default name or enter the name that you want.
SIP Security Profile	Select **5060**.
SIP Transport Protocol	Select the SIP transport protocol that Unity Connection uses.
IPv4 Address or Host Name	Enter the IP address of the TIMG unit that you are integrating with Unity Connection.
IPv6 Address or Host Name	Do not enter a value in this field. IPv6 is not supported for TIMG integrations.
IP Address or Host Name	Enter the IP address of the TIMG unit that you are integrating with Unity Connection.
Port	Enter the SIP port of the TIMG unit that Unity Connection connects to. We recommend that you use the default setting. **Caution** This name must match the setting in the TCP/UDP Server Port field on the Configuration > VoIP > General page of the TIMG unit. Otherwise, the integration do not function correctly.

Step 8 On the Port Group Basics page, in the Related Links drop-down box, select **Add Ports** and select **Go**.

Step 9 On the New Port page, enter the following settings and select **Save**.

Table 37: Settings for the New Ports Page

Field	Considerations
Enabled	Check this check box.
Number of Ports	Enter the number of voice messaging ports that you want to create in this port group. **Note** For a Unity Connection cluster, the server must have the number of voice messaging ports that are set up on the phone system for the TIMG integration so that this server can handle all voice messaging traffic for the cluster if one of the servers stops functioning. For example, if the phone system is set up with 16 voice messaging ports, this server must have 16 voice messaging ports.
Phone System	Select the name of the phone system that you entered in Step 3.
Port Group	Select the name of the port group that you added in Step 7.

Step 10 On the Search Ports page, select the display name of the first voice messaging port that you created for this phone system integration.

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

Step 11 On the Port Basics page, set the voice messaging port settings as applicable. The fields in the following table are the ones that you can change.

Table 38: Settings for the Voice Messaging Ports

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

Step 12 Select **Save**.

Step 13 Select **Next**.

Step 14 Repeat Step 11 through Step 13 for all remaining voice messaging ports for the phone system.

Step 15 In Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Phone System**.

Step 16 On the Search Phone Systems page, under Display Name, select the name of the phone system that you entered in Step 3.

Step 17 Repeat Step 6 through Step 16 for each remaining TIMG unit integrated with Unity Connection.

Note Each TIMG unit is connected to one port group with the applicable voice messaging ports. For example, a system that uses two TIMG units requires two port groups, one port group for each TIMG unit.

Step 18 If another phone system integration exists, in Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Trunk**. Otherwise, skip to Step 22.

Step 19 On the Search Phone System Trunks page, on the Phone System Trunk menu, select **New Phone System Trunk**.

Step 20 On the New Phone System Trunk page, enter the following settings for the phone system trunk and select **Save**.

Table 39: Settings for the Phone System Trunk

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

Step 21 Repeat Step 19 and Step 20 for all remaining phone system trunks that you want to create.

Step 22 In Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Port Group**.

Step 23 On the Search Port Groups page, select the display name of the port group that you added in Step 7.

Step 24 On the Port Group Basics page, select **Reset**.

Step 25 When prompted that resetting terminates all call traffic, select **OK**.

Step 26 In the Related Links drop-down list, select **Check Telephony Configuration** and select **Go** to confirm the phone system integration settings.

If the test is not successful, the Task Execution Results displays one or more messages with troubleshooting steps. After correcting the problems, test the connection again.

Step 27 In the Task Execution Results window, select **Close**.

CHAPTER 5

Setting Up a Serial (SMDI, MCI, or MD-110) TIMG Integration with Cisco Unity Connection

For detailed instructions for setting up a serial (SMDI, MCI, or MD-110) TIMG integration with Cisco Unity Connection, see the following sections in this chapter:

Task List to Create a Serial (SMDI, MCI, or MD-110) TIMG

Before doing the following tasks to integrate Unity Connection with the phone system using the T1 media gateway (TIMG), confirm that the Unity Connection server is ready for the integration after completing the installation following the steps as mentioned in the "Installing Cisco Unity Connection" chapter of the *Install, Upgrade, and Maintenance Guide for Cisco Unity Connection Release 15*, available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/unified_messaging/guide/b_15cucumgx.html.

1. Review the system and equipment requirements to confirm that all phone system and Unity Connection server requirements have been met. See the Requirements section.
2. Plan how the voice messaging ports are used by Unity Connection. See the Planning the Usage of Voice Messaging Ports in Cisco Unity Connection chapter.
3. Program the phone system and extensions. See the Programming Phone System for a Serial TIMG Integration section.
4. Set up the TIMG units. See the Setting Up the TIMG Units scetion.
5. Create the integration. See the Creating an Integration with the Phone System section.
6. Test the integration. See the Testing the Integration chapter.
7. If this integration is a second or subsequent integration, add the applicable new user templates for the new phone system. See the Adding New User Templates for Multiple Integrations chapter.

Requirements

The serial (SMDI, MCI, or MD-110) TIMG integration supports configurations of the following components:

Phone System

- A phone system that supports the SMDI, MCI, or MD-110 serial protocols.
- T1 digital trunk interface card.
- The firmware must be configured to support T1 line-side signaling.
- T1 CAS connections to the TIMG units using the FXS/FSO or E&M protocol.
- One or more TIMG units (media gateways).
- The serial data port on the phone system connected to the serial port on the master TIMG unit with an RS-232 serial cable (which is available from Cisco).

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

We recommend that the serial cable have the following construction:

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

- The voice messaging ports in the phone system connected by T1 digital lines (DS1 or "dry T1" digital lines only) to the ports on the TIMG units.

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

- The TIMG units connected to the same LAN or WAN that Unity Connection is connected to.
- If the TIMG units connect to a WAN, the requirements for the WAN network connections are:
    - For G.729a codec formatting, a minimum of 32.76 Kbps guaranteed bandwidth for each voice messaging port.
    - For G.711 codec formatting, a minimum of 91.56 Kbps guaranteed bandwidth for each voice messaging port.

- No network devices that implement network address translation (NAT).
- A maximum 200 ms one-way network latency.

- The phone system ready for the integration, as described in the documentation for the phone system.

Unity Connection Server

- Unity Connection installed and ready for the integration after completing the installation following the steps as mentioned in the "Installing Cisco Unity Connection" chapter of the *Install, Upgrade, and Maintenance Guide for Cisco Unity Connection Release 15*, available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/unified_messaging/guide/b_15cucumgx.html.
- A license that enables the applicable number of voice messaging ports.

Centralized Voice Messaging

Unity Connection supports centralized voice messaging through the phone system, which supports various inter-phone system networking protocols including proprietary protocols such as Avaya DCS, Nortel MCDN, or Siemens CorNet, and standards-based protocols such as QSIG or DPNSS. Note that centralized voice messaging is a function of the phone system and its inter-phone system networking, not voicemail. Unity Connection supports centralized voice messaging as long as the phone system and its inter-phone system networking are properly configured.

Programming Phone System for a Serial TIMG Integration

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

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

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

Programming the Phone System for a Serial Integration with Unity Connection

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

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

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

Step 3 Enable caller ID (via SMDI, MCI, or MD-110) on each subscriber extension.

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

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

Setting Up the TIMG Units

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

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

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

Caution Because TIMG units have the same default IP address, you must set them up one at a time. Otherwise, you can experience IP address conflicts.

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

Downloading TIMG Firmware Update Files for TIMG Units

SUMMARY STEPS

1. On a Windows workstation that have access to the TIMG units, go to the following link: http://software.cisco.com/download/navigator.html?mdfid=280082558&i=rm.
2. In the tree control on the Downloads Home page, expand **Unified Communications**> **Unified Communications Applications** > **Messaging** > **Cisco Unity and select Cisco Unity Telephony Integration**.
3. On the Log In page, enter your username and password, then select **Log In**.
4. On the Select a Release page, under **Latest Releases**, select the most recent release.
5. In the right column, select the version of the firmware for your TIMG units.
6. On the Download Image page, select **Download**.
7. On the Supporting Document(s) page, select **Agree**.
8. In the **File Download** dialog box, select **Save**.
9. In the **Save As** dialog box, browse to the Windows workstation that have access the TIMG units, browse to a directory where you want to save the file, and select **Save**.
10. In the Download Complete dialog box, select **Open**. The window for extracting the TIMG firmware update files appears.

11. Select **Extract**.
12. In the Extract dialog box, browse to the directory where you want the extracted files, and select **Extract**.
13. Close the window for the extracting application.

DETAILED STEPS

Step 1 On a Windows workstation that have access to the TIMG units, go to the following link: http://software.cisco.com/download/navigator.html?mdfid=280082558&i=rm.

Note To access the software download page, you must be signed in to Cisco.com as a registered user.

This procedure describes the steps when using Internet Explorer as your web browser. If you are using a different web browser, the steps may differ.

Step 2 In the tree control on the Downloads Home page, expand **Unified Communications**> **Unified Communications Applications** > **Messaging** > **Cisco Unity and select Cisco Unity Telephony Integration**.

Step 3 On the Log In page, enter your username and password, then select **Log In**.

Step 4 On the Select a Release page, under **Latest Releases**, select the most recent release.

Step 5 In the right column, select the version of the firmware for your TIMG units.

Step 6 On the Download Image page, select **Download**.

Step 7 On the Supporting Document(s) page, select **Agree**.

Step 8 In the **File Download** dialog box, select **Save**.

Step 9 In the **Save As** dialog box, browse to the Windows workstation that have access the TIMG units, browse to a directory where you want to save the file, and select **Save**.

Step 10 In the Download Complete dialog box, select **Open**. The window for extracting the TIMG firmware update files appears.

Step 11 Select **Extract**.

Step 12 In the Extract dialog box, browse to the directory where you want the extracted files, and select **Extract**.

Step 13 Close the window for the extracting application.

Setting Up the TIMG Units (Firmware Version 6.x)

Step 1 On the Windows workstation, add a temporary route to enable access to the TIMG units.

a) On the Windows Start menu, select **Run**.
b) Enter **cmd**, and press **Enter**. The Command Prompt window appears.
c) At the command prompt, enter **route add 10.12.13.74 <IP Address of Workstation>**, and press **Enter**.

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

d) Close the Command Prompt window.

Step 2 Connect a TIMG unit to the network.

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

Step 4 To sign in, enter the following case-sensitive settings.

Table 40: Sign-in Settings

Field	Setting
Username	Enter **admin**.
Password	Enter **IpodAdmin**.

Step 5 Select **OK**.

Step 6 On the System menu, select **Upgrade**.

Step 7 On the Upgrade page, under Browse for Upgrade File, select **Browse**.

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

Step 9 Select **T1E1_<xx** > **.app** (where <xx> is multiple digits), and select **Open**.

Step 10 On the Upgrade page, select **Install File**.

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

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

Step 12 Repeat Step 6 through Step 11 for the file T1E1_<xx>.fsh.

Step 13 On the System menu, select **Upgrade**.

Step 14 On the Upgrade page, under Browse for Upgrade File, select **Browse**.

Step 15 In the Choose File dialog box, browse to the file T1E1_<xx>.msd.

Step 16 Select **T1E1_<xx** > **.msd**, and select **Open**.

Step 17 On the Upgrade page, click **Install File**.

Step 18 After the file is installed, a message prompting you to restart the TIMG unit appears. Select **OK**.

Step 19 In the web browser, go to **http://10.12.13.74**.

Step 20 To sign in, enter the following case-sensitive settings.

Table 41: Sign-in Settings

Field	Setting
Username	Enter **admin**.
Password	Enter **IpodAdmin**.

Step 21 Select **OK**.

Step 22 On the Configure menu, select **Password**.

Step 23 On the Change Password page, enter the following settings.

Table 42: Change Password Page Settings

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

Field	Setting
New Password	Enter your new password. (This setting is case sensitive.)
Confirm Password	Enter your new password. (This setting is case sensitive.)

Step 24 Select **Change**.

Step 25 On the Configuration menu, select **Mgmt Protocols**.

Step 26 On the Management Protocols page, enter the following settings.

Table 43: Management Protocols Page Settings

Field	Settings
E-mail Alarms Enabled	Select **No**.
SNMP Traps Enabled	Select **No**.
HTTP Server Enabled	Select **Yes**.
HTTPs Server Enabled	Select **No**.

Step 27 Select **Submit**.

Step 28 On the Configuration menu, select **Routing Table**.

Step 29 On the Routing Table page, under Router Configuration, select **VoIP Host Groups**.

Step 30 Under VoIP Host Groups, enter the following settings for the first VoIP Host Group.

Table 44: First VoIP Host Group Settings

Field	Settings
Name	Accept the default or enter another descriptive name of the VoIP host group.
Load-Balanced	*(Unity Connection without a cluster)* Select **False**. *(Unity Connection with a cluster configured)* Select **False**.
Fault-Tolerant	*(Unity Connection without a cluster)* Select **False**. *(Unity Connection with a cluster configured)* Select **True**.

Step 31 For Unity Connection without a cluster, under Host List, enter the host name or IP address of the Unity Connection server and the server port in the format <host name or IP address>:5060.

For Unity Connection with a cluster configured, under Host List, enter the host name or IP address of the subscriber server (the second Unity Connection server that you installed) and the server port in the format <host name or IP address>:5060.

Step 32 For Unity Connection without a cluster, continue to Step 34. For Unity Connection with a cluster configured, select **Add Host**.

Step 33 In the second field, enter the host name or IP address of the publisher server (the first Unity Connection server that you installed) and the server port in the format <host name or IP address>:5060.

Caution Do not add a third host under Host List or a second host group under VoIP Host Groups. Otherwise, the Unity Connection cluster may not function correctly.

Step 34 Select **Submit**.

Step 35 On the Configuration menu, select **TDM** > **T1/E1**.

Step 36 On the T1/E1 Configuration page, enter the following settings.

Table 45: T1/E1 Configuration Page Settings

Field	Settings
Line Settings	
Line Mode	Select **T1**.
Signaling Mode	Select **CAS**.
Interface Mode	Select **Terminal**.
T1 Line	
Line Encoding	Enter the setting that matches the phone system programming.
Framing	Enter the setting that matches the phone system programming.
Selects Transmit Pulse Waveform	Enter the setting that matches the phone system programming.
T1 CAS Protocol	
T1 CAS Protocol	Enter the setting that matches the phone system programming.
Flash Hook	Enter the setting that matches the phone system programming.
Consult Call Dialtone Drop Code	Enter the setting that matches the phone system programming.
Consult Call Proceeding Drop Code	Enter the setting that matches the phone system programming.
Consult Call Busy Drop Code	Enter the setting that matches the phone system programming.
Consult Call Error Drop Code	Enter the setting that matches the phone system programming.
Consult Call Connected Drop Code	Enter the setting that matches the phone system programming.
Consult Call Disconnected Drop Code	Enter the setting that matches the phone system programming.
MWI confirmation Tone	Select **No**.
CPID Type	Select **TypeII_CPID**.

Field	Settings
Initial Wait for Inband CPID	Enter **100**.
Inband CPID Complete Timeout	Enter **300**.
Failover Settings	
Enable Failover	Select **No**.

Step 37 Select **Submit**.

Step 38 On the Configuration menu, select **TDM** > **General**.

Step 39 On the TDM General Settings page, enter the following settings.

Table 46: TDM General Settings Page Settings

Field	Settings
PCM Coding	Select **uLaw**.
Minimum Call Party Delay (ms)	Enter **500**.
Maximum Call Party Delay (ms)	Enter **2000**.
Dial Digit on Time (ms)	Enter **100**.
Dial Inter-Digit Time (ms)	Enter **100**.
Dial Pause Time (ms)	Enter **2000**.
Turn MWI On FAC	Leave this field blank.
Turn MWI Off FAC	Leave this field blank.
Outbound Call Connect Timeout (ms)	Enter **10000**.
Wait for Ringback/Connect on Blind Transfer	Select **Yes**.
Hunt Group Extension	Enter the pilot number of the Unity Connection voice messaging ports.

Step 40 Select **Submit**.

Step 41 On the Configuration menu, select **TDM** > **Port Enable**.

Step 42 On the TDM Port Enabling page, select **No** for the ports that you want to disable on the TIMG unit.

Step 43 Confirm that **Yes** is selected for all other ports on the TIMG unit.

Step 44 Select **Submit**.

Step 45 On the Configuration menu, select **VoIP** > **General**.

Step 46 On the VoIP General Settings page, enter the following settings.

Table 47: VoIP General Settings Page Settings

Field	Setting
User-Agent	
Host and Domain Name	Enter the domain name of the TIMG unit.
Transport Type	Select **UDP**.
Call as Domain Name	Select **No**.
SIPS URI Scheme Enabled	Select **No**.
Invite Expiration (sec)	Enter **120**.
Server	
DNS Server Address	Enter the IP Address of the Domain Name Server that the TIMG unit uses.
Registration Server Address	Leave this field blank.
Registration Server Port	Enter **5060**.
Registration Expiration (sec)	Enter **3600**.
TCP/UDP	
UDP/TCP Transports Enabled	Select **Yes**.
TCP/UDP Server Port	Enter **5060**.
Proxy	
Primary Proxy Server Address	Leave this field blank.
Primary Proxy Server Port	Not applicable. Leave the default setting.
Backup Proxy Server Address	Not applicable. Leave the default setting.
Backup Proxy Server Port	Not applicable. Leave the default setting.
Proxy Query Interval	Enter **10**.
Timing	

Field	Setting
T1 Time	Enter **500**.
T2 Time	Enter **4000**.
T4 Time	Enter **5000**.
Monitoring	
Monitor Call Connections	Select **No**.

Step 47 Select **Submit**.

Step 48 On the Configuration menu, select **VoIP** > **Media**.

Step 49 On the VoIP Media Settings page, enter the following settings.

Table 48: VoIP Media Settings Page Settings

Field	Settings
Audio	
Audio Compression	Select the preferred codec for audio compression: • **G.711u**—The TIMG unit uses only the G.711 mu-law codec. • **G.729AB**—The TIMG unit prefers the G.729 codec but can also use the G.711 mu-law codec.
RTP Digit Relay Mode	Select **RFC2833**.
Signaling Digit Relay Mode	Select **Off**.
Voice Activity Detection	Select **On**.
Frame Size	Select the applicable setting: • G.711—**20** • G.729AB—**10** **Caution** Failure to use the correct setting results in recorded messages containing nothing but silence.
Frames Per Packet	Select the applicable setting: • G.711—**1** • G.729AB—**2** **Caution** Failure to use the correct setting results in recorded messages containing nothing but silence.

Step 50 Select **Submit**.

Step 51 On the Configuration menu, select **VoIP** > **QOS**.

Step 52 On the VoIP QOS Configuration page, enter the following settings.

Table 49: VoIP QOS Configurative Page Settings

Field	Settings
Call Control QOS Byte	Enter **104** (equivalent to DSCP AF31).
RTP QOS Byte	Enter **184** (equivalent to DSCP EF).

Step 53 Select **Submit**.

Step 54 On the Configuration menu, select **Serial** > **General**.

Step 55 On the Serial Port, COM 1 page, enter the following settings.

Table 50: Serial Port, COM 1 Page Settings

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

Step 56 Select **Submit**.

Step 57 On the Configuration menu, select **Serial** > **Switch Protocol**.

Step 58 On the Switch Protocol page, enter the following settings.

Table 51: Switch Protocol Page Settings

Field	Settings
Serial Port, COM 1	

Field	Settings
Serial Mode (Master/Slave)	Select the applicable setting: • **Master**—Select this setting when this TIMG unit is connected to the data link serial cable from the phone system. There can be only one master TIMG unit in a phone system integration. • **Slave**—Select this setting when this TIMG unit is not connected to the data link serial cable from the phone system. There can be multiple slave TIMG units in a phone system integration.
Serial Interface Protocol	Select the serial protocol that your phone system uses: • **SMDI** • **MCI** • **MD110**
MCI Message Extension Length	*(For MCI protocol only)* Select the applicable number of extension digits.
MCI Message Type	*(For MCI protocol only)* Select the applicable message type.
Cpid Length	Select the applicable setting. Typically, the settings are 7 or 10.
Cpid Padding String	Enter the applicable string or leave this field blank. Typically, the setting is one of the following: • A string of zeros, where the number of zeros matches the setting of the Cpid Len field. • A prefix that is provided by the Centrex service.
Voice Mail Port Length	If the setting of the Serial Interface Protocol field is MD-110, enter **2**. Otherwise, accept the default of **7**.
System Number	Enter the applicable setting. Typically, the setting is 1.
MWI Response Timeout (ms)	Enter **2000**.
IP Address of Serial Server	If the TIMG unit is the master, leave this field blank. If the TIMG unit is a slave, enter the IP address of the master TIMG unit (the TIMG unit that is connected to the data link serial cable from the phone system).
Serial Cpid Expiration (ms)	Enter **5000**.
Logical Extension Number	

Field	Settings
<port number>	If the setting of the Serial Interface Protocol field is MCI or MD-110, enter the extension number for each port on the TIMG unit. If the setting of the Serial Interface Protocol field is SMDI, enter the logical port number. Typically, the setting is 1 for port 1, 2 for port 2, and so on beginning with the master TIMG unit and continuing through each of the slave TIMG units.

Step 59 Select **Submit**.

Step 60 On the Configuration menu, select **IP**.

Step 61 On the IP Settings, LAN1 page, enter the following settings.

Table 52: IP Settings, LAN1 Page Settings

Field	Settings
Client IP Address	Enter the new IP address that you want to use for the TIMG unit. (This is the IP address that you enter in Cisco Unity Connection Administration when you create the integration.)
Client Subnet Mask	Enter the new subnet mask, if the subnet mask is different from the default IP address.
Default Network Gateway Address	Enter the IP address of the default network gateway router that the TIMG units use.
BOOTP Enabled	If you are using DHCP, select **Yes**. If you are not using DHCP, select **No**.

Step 62 Select **Submit**.

Step 63 On the Configuration menu, select **Tone Detection**.

Step 64 On the Tone Detection page, under Call Progress Tone - Learn, in the Learn Tone Event field, select **Busy** and do the following substeps to verify that the tone is correct.

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

Step 65 Under Call Progress Tone - Learn, in the Dial String field, enter the extension that you dialed in Step 64c. from the third phone.

Step 66 Select **Learn**.

Step 67 On the Tone Detection page, under Call Progress Tone - Learn, in the Learn field, select **Error** and do the following substeps to verify that the tone is correct.

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

Step 68 Under Call Progress Tone - Learn, in the Dial String field, enter the extension that you dialed in Step 67a.

Step 69 Select **Learn**.

Step 70 On the Tone Detection page, under Call Progress Tone - Learn, in the Learn field, select **Ringback** and do the following substeps to verify that the tone is correct.

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

Step 71 Under Call Progress Tone - Learn, in the Dial String field, enter the extension that you dialed in Step 70a.

Step 72 Select **Learn**.

Step 73 Select **Submit**.

Step 74 Hang up the phones that you used in Step 64.

Step 75 On the System menu, select **Restart**.

Step 76 On the Restart page, select **Restart Unit Now**.

Step 77 Repeat Step 2 through Step 76 on all remaining TIMG units.

Creating an Integration with the Phone System

After ensuring that the phone system, the TIMG units, and the Unity Connection server are ready for the integration, do the following procedures to set up the integration and to enter the port settings.

Step 1 In Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Phone System**.

Step 2 On the Search Phone Systems page, under Display Name, select the name of the default phone system.

Step 3 On the Phone System Basics page, in the Phone System Name field, enter the descriptive name that you want for the phone system.

Step 4 If you want to use this phone system as the default for TRaP connections so that administrators and users without voicemail boxes can record and playback through the phone in Unity Connection web applications, check the **Default TRAP Switch** check box. If you want to use another phone system as the default for TRaP connections, uncheck this check box.

Step 5 Select **Save**.

Step 6 On the Phone System Basics page, in the Related Links drop-down box, select **Add Port Group** and select **Go**.

Step 7 On the New Port Group page, enter the applicable settings and select **Save**.

Table 53: Settings for the New Port Group Page

Field	Setting
Phone System	Select the name of the phone system that you entered in Step 3.
Create From	Select **Port Group Template** and select **SIP to DMG/PIMG/TIMG** in the drop-down box.
Display Name	Enter a descriptive name for the port group. You can accept the default name or enter the name that you want.

Field	Setting
SIP Security Profile	Select **5060**.
SIP Transport Protocol	Select the SIP transport protocol that Unity Connection uses.
IPv4 Address or Host Name	Enter the IP address of the TIMG unit that you are integrating with Unity Connection.
IPv6 Address or Host Name	Do not enter a value in this field. IPv6 is not supported for TIMG integrations.
IP Address or Host Name	Enter the IP address of the TIMG unit that you are integrating with Unity Connection.
Port	Enter the SIP port of the TIMG unit that Unity Connection connects to. We recommend that you use the default setting. **Caution** This name must match the setting in the TCP/UDP Server Port field on the Configuration > VoIP > General page of the TIMG unit. Otherwise, the integration do not function correctly.

Step 8 In the Related Links drop-down box, select **Add Ports** and select **Go**.

Step 9 On the New Port page, enter the following settings and select **Save**.

Table 54: Settings for the New Port Page

Field	Considerations
Enabled	Check this check box.
Number of Ports	Enter the number of voice messaging ports that you want to create in this port group. **Note** For a Unity Connection cluster, the server must have the number of voice messaging ports that are set up on the phone system for the TIMG integration so that this server can handle all voice messaging traffic for the cluster if one of the servers stops functioning. For example, if the phone system is set up with 16 voice messaging ports, this server must have 16 voice messaging ports.
Phone System	Select the name of the phone system that you entered in Step 3.
Port Group	Select the name of the port group that you added in Step 7.

Step 10 On the Search Ports page, select the display name of the first voice messaging port that you created for this phone system integration.

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

Step 11 On the Port Basics page, set the voice messaging port settings as applicable. The fields in the following table are the ones that you can change.

Table 55: Settings for the Voice Messaging Ports

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

Step 12 Select **Save**.

Step 13 Select **Next**.

Step 14 Repeat Step 11 through Step 13 for all remaining voice messaging ports for the phone system.

Step 15 In Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Phone System**.

Step 16 On the Search Phone Systems page, under Display Name, select the name of the phone system that you entered in Step 3.

Step 17 Repeat Step 6 through Step 16 for each remaining TIMG unit integrated with Unity Connection.

Note Each TIMG unit is connected to one port group with the applicable voice messaging ports. For example, a system that uses two TIMG units requires two port groups, one port group for each TIMG unit.

Step 18 If another phone system integration exists, in Cisco Unity Connection Administration, expand **Telephony Integrations**, then select **Trunk**. Otherwise, skip to Step 22.

Step 19 On the Search Phone System Trunks page, on the Phone System Trunk menu, select **New Phone System Trunk**.

Step 20 On the New Phone System Trunk page, enter the following settings for the phone system trunk and select **Save**.

Table 56: Settings for the Phone System Trunk

Field	Setting
From Phone System	Enter the display name of the phone system that you are creating a trunk for.

Field	Setting
To Phone System	Enter the display name of the previously existing phone system that the trunk connects to.
Trunk Access Code	Enter the extra digits that Unity Connection must dial to transfer calls through the gateway to extensions on the previously existing phone system.

Step 21 Repeat Step 19 and Step 20 for all remaining phone system trunks that you want to create.

Step 22 In the Related Links drop-down list, select **Check Telephony Configuration** and select **Go** to confirm the phone system integration settings.

If the test is not successful, the Task Execution Results displays one or more messages with troubleshooting steps. After correcting the problems, test the connection again.

Step 23 In the Task Execution Results window, select **Close**.

CHAPTER 6

Testing the Integration

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

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

- The installation guide for the phone system.
- *Troubleshooting Guide for Cisco Unity Connection*, Release 15, available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/troubleshooting/guide/b_15cuctsg.html .
- The setup information earlier in this guide.

Setting Up Test Configuration

Step 1 Set up two test extensions (Phone 1 and Phone 2) on the same phone system that Unity Connection is connected to.

Step 2 Set Phone 1 to forward calls to the Unity Connection pilot number when calls are not answered.

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

Step 3 In Cisco Unity Connection Administration, expand **Users**, then select **Users**.

Step 4 On the Search Users page, select the display name of a user to use for testing. The extension for this user must be the extension for Phone 1.

Step 5 On the Edit User Basics page, uncheck the **Set for Self-enrollment at Next Login** check box.

Step 6 In the Voice Name field, record a recorded name for the test user.

Step 7 Select **Save**.

Step 8 On the Edit menu, select **Message Waiting Indicators**.

Step 9 On the Message Waiting Indicators page, select the message waiting indicator. If no message waiting indication is in the table, select **Add New**.

Step 10 On the Edit Message Waiting Indicator page, enter the following settings.

Table 57: Settings for the Edit MWI Page

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

Step 11 Select **Save**.

Step 12 On the Edit menu, select **Transfer Rules**.

Step 13 On the Transfer Rules page, select the active option.

Step 14 On the Edit Transfer Rule page, under Transfer Action, select the **Extension** option and enter the extension of Phone 1.

Step 15 In the Transfer Type field, select **Release to Switch**.

Step 16 Select **Save**.

Step 17 Minimize the Cisco Unity Connection Administration window. Do not close the Cisco Unity Connection Administration window because you use it again in a later procedure.

Step 18 Sign in to the Cisco Unified Real-Time Monitoring Tool (RTMT).

Step 19 On the Cisco Unity Connection menu, select **Port Monitor**. The Port Monitor tool appears in the right pane.

Step 20 In the right pane, select **Start Polling**. The Port Monitor displays which port is handling the calls that you make.

Testing an External Call with Release Transfer

Step 1 From Phone 2, enter the access code necessary to get an outside line, then enter the number outside callers use to dial directly to Unity Connection.

Step 2 In the Port Monitor, note which port handles this call.

Step 3 When you hear the opening greeting, enter the extension for Phone 1. Hearing the opening greeting means that the port is configured correctly.

Step 4 Confirm that Phone 1 rings and that you hear a ringback tone on Phone 2. Hearing a ringback tone means that Unity Connection correctly released the call and transferred it to Phone 1.

Step 5 Leaving Phone 1 unanswered, confirm that the state of the port handling the call changes to "Idle." This state means that release transfer is successful.

Step 6 Confirm that, after the number of rings that the phone system is set to wait, the call is forwarded to Unity Connection and that you hear the greeting for the test user. Hearing the greeting means that the phone system forwarded the unanswered call and the call-forward information to Unity Connection, which correctly interpreted the information.

Step 7 On the Port Monitor, note which port handles this call.

Step 8 Leave a message for the test user and hang up Phone 2.

Step 9 In the Port Monitor, confirm that the state of the port handling the call changes to “Idle.” This state means that the port was successfully released when the call ended.

Step 10 Confirm that the MWI on Phone 1 is activated. The activated MWI means that the phone system and Unity Connection are successfully integrated for turning on MWIs.

Testing Listening to the Messages

Step 1 From Phone 1, enter the internal pilot number for Unity Connection.

Step 2 When asked for your password, enter the password for the test user. Hearing the request for your password means that the phone system sent the necessary call information to Unity Connection, which correctly interpreted the information.

Step 3 Confirm that you hear the recorded name for the test user (if you did not record a name for the test user, you hear the extension number for Phone 1). Hearing the recorded name means that Unity Connection correctly identified the user by the extension.

Step 4 Listen to the message.

Step 5 After listening to the message, delete the message.

Step 6 Confirm that the MWI on Phone 1 is deactivated. The deactivated MWI means that the phone system and Unity Connection are successfully integrated for turning off MWIs.

Step 7 Hang up Phone 1.

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

Setting Up Supervised Transfer

Step 1 In Cisco Unity Connection Administration, on the Edit Transfer Rule page for the test user, in the Transfer Type field, select **Supervise Transfer**.

Step 2 In the **Rings to Wait For** field, enter **3** and select **Save**.

Step 3 Minimize the Cisco Unity Connection Administration window. Do not close the Cisco Unity Connection Administration window because you use it again in a later procedure.

Testing Supervised Transfer

Step 1 From Phone 2, enter the access code necessary to get an outside line, then enter the number outside callers use to dial directly to Unity Connection.

Step 2 On the Port Monitor, note which port handles this call.

Step 3 When you hear the opening greeting, enter the extension for Phone 1. Hearing the opening greeting means that the port is configured correctly.

Step 4 Confirm that Phone 1 rings and that you do not hear a ringback tone on Phone 2. Instead, you should hear the indication your phone system uses to mean that the call is on hold (for example, music).

Step 5 Leaving Phone 1 unanswered, confirm that the state of the port handling the call remains "Busy." This state and hearing an indication that you are on hold mean that Unity Connection is supervising the transfer.

Step 6 Confirm that, after three rings, you hear the greeting for the test user. Hearing the greeting means that Unity Connection successfully recalled the supervised-transfer call.

Step 7 During the greeting, hang up Phone 2.

Step 8 On the Port Monitor, confirm that the state of the port handling the call changes to "Idle." This state means that the port was successfully released when the call ended.

Step 9 Select **Stop Polling**.

Step 10 Exit RTMT.

Deleting the Test User

If any of the above steps indicate failure, see the following documentation as applicable:

- The installation guide for the phone system.
- *Troubleshooting Guide for Cisco Unity Connection*, *Release 15*, available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/troubleshooting/guide/b_15cuctsg.html

Step 1 In Cisco Unity Connection Administration, expand **Users**, then select **Users**.

Step 2 On the Search Users page, check the check box to the left of the test user.

Step 3 Select **Delete Selected**.

CHAPTER 7

Adding New User Templates for Multiple Integrations

Adding New User Templates for Multiple Integrations

Introduction

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

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

For details on adding new user templates, or on selecting a user template when adding a new user, see the "User Templates" section of the "User Attributes" chapter of the System Administration Guide for Cisco Unity Connection, Release 15, available at .
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/15/administration/guide/b_15cucsag.html

CHAPTER 8

Application Note for the Intecom Pointspan 6880 TIMG Integration

Introduction

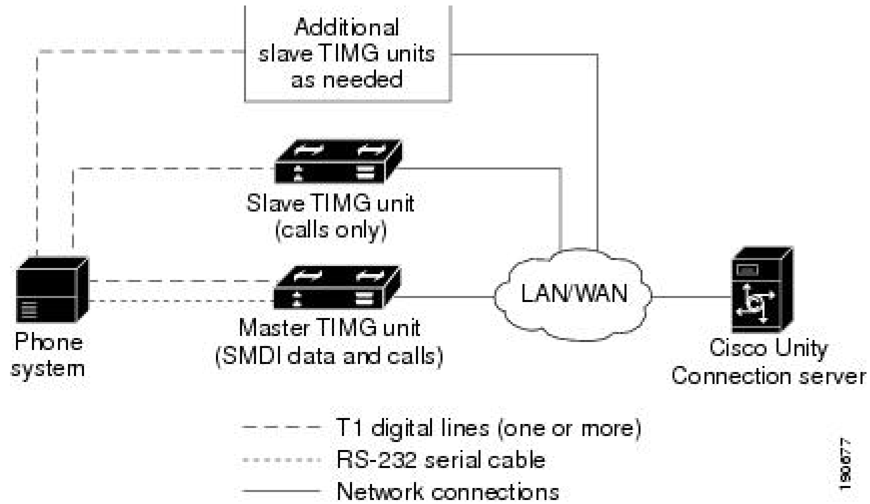
This is an application note for programming the Intecom Pointspan 6880 phone system for a serial SMDI integration with Cisco Unity Connection using TIMG units. For instructions on setting up the TIMG units (media gateways) and creating the integration in Unity Connection, see the Setting Up an Avaya Definity G3 In-Band TIMG Integration with Cisco Unity Connection chapter.

Network Topology

Figure 8-1 shows the required Cisco Unity Connection Administration for a serial SMDI integration using TIMG units.

Figure 3: Connections for a Serial SMDI TIMG Integration

For more information about this integration, see Integration Description chapter.



Requirements

The phone system met the following requirements:

- The Intecom Pointspan 6880 phone system.
- Software version 3.4K or later.
- PDI cards model 520-1000-004.
- T1 cards model 300-0289-001.

Programming Intecom Pointspan 6880 Phone System for TIMG Integration

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

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

Example of Programming for the Intecom Pointspan 6880 Phone System in a TIMG Integration

```
** TRUNK GROUP DEFINITION
***...TRUNK GROUP........................17
***...TRUNK GROUP TYPE...................UNIVERSAL
TCI...TRUNK CLASS IDENTIFICATION.........LOCAL CO
UGP...USER GROUP NUMBER..................11
VTT...VOICE TRUNK TRANSFER ENABLED.......YES
CDT...TRANSFER COLLECT DIGIT TABLE #.....NONE
```

```
DCS...DEFAULT CLASS OF SERVICE...........0
CNC...NATIONAL CALLING PARTY # CONTENTS..USER GROUP
CNI...USER GROUP CALLING PARTY #.........4 = 206-555-1212
DCP...DISPLAY CALLING PARTY NUMBER.......YES
PND...PRIVATE NETWORK....................NO
ITY...INTEMAIL TYPE......................INTEMAIL TYPE II
IST...DOES InteMail SUPERVISE TRANSFERS?.NO
IML...InteMail USER ID LENGTH............5
IIN...INTEMAIL INTERFACE NUMBER..........2
VNP...InteMail NUMBER FORMAT.............DIRN
OAM...OAI ASSOCIATED MEMBER..............NO
BTG...BROADCAST TRUNK GROUP..............NO
TCM...TRAVELING CLASSMARK................NO
FTH...FAILURE THRESHOLD..................3
RDT...RESEIZE DELAY TIME............MSEC:200
CHT...TRUNK MONITOR MINIMUM HOLD TIME....0
DET...DISTANT END RELEASE TIME.......SEC:55
DCT...DATA CALLS ALLOWED.................NO
SWM...SEIZE WHEN MOS.....................NO
TCH...TRUNK CALL HANDLING................INTERNAL
NDS...DISCONNECT SUPERVISION.............YES
IGG...IGNORE GLARE.......................NO
GDT...GLARE DETECT TIME.............MSEC:100
XFT...DISTANT IBX ALLOWS FEATURE TRANSP..NO
DPT...DTMF PASSTHROUGH TIMING INDEX......NONE
......TRUNK DIRECTION....................BOTH WAYS
 *** INCOMING PARAMETERS
STY...INCOMING CALL ORIGINATION TYPE.....T1 OFF PREM.....OPX (OFF PREMISE)
TYP...INCOMING TRUNK TYPE................DIALED
ICM...INCOMING CALL MESSAGE #............17
IDS...INCOMING DIGIT SEQUENCE............DESTINATION NUMBER ONLY
IRD...RESPONSE TO DESTINATION NUMBER.....NONE
IRC...RESPONSE TO CALLING PARTY NUMBER...NONE
IIT...INCOMING INFO DIGIT TYPE/LENGTH....NONE
WPR...WHISPER MESSAGE SOURCE GROUP.......NONE
APA...TRUNK GROUP AUTHORIZATION TYPE.....NONE
PVA...PRE-VALIDATE AUTHORIZATION CODE....NO
RSC...RESET COUNT........................1
LVL...PREDEFINED LEVEL CODE..............NONE
TNE...TONE TABLE ENTRY NUMBER............NONE
MOD...INCOMING DIAL MODE.................DTMF
RGF...DTMF RECEIVER GROUP................52
TOO...TIMEOUT TO ATTENDANT...............NO
MCL...MULTIPLE CALLING ALLOWED...........NO
RAC...REUSE AUTH FOR MULT. CALLS.........NO
GAC...GROUP AUTH REQUIRED FOR TRUNKS.....NO
SAC...SYSTEM ACCESS CODE.................NONE
CWR...CALLWAIT RINGBACK..................NO
UCT...TRUNK UPDATE CDR ON TRANSFER.......ALL
CPT...CALL PROGRESSING TONES:............IBX PROVIDED
RIO...RESPONSE TO INCOMING ORIGINATION...NONE
IUG...InteMail LAMP MESSAGE USER GROUPS..ALL
NUG...INTER-USER GROUP NNP USER GROUPS...ALL
TCT...STATION CALL RESTRICTION ENABLED...NO
8NC...800 TO 4D SPEED NUMBER CONVERSION..NO
NWT...CALL PARTY NAME WAIT TIME..........NONE
 *** OUTGOING PARAMETERS
MSG...MODEM SIGNALLING...................YES
TXA...DIRECT TGRP SELECT ALLOWED.........YES
ATG...ANNOUNCEMENT TRUNK GROUP...........NO
SLC...TRUNK SELECTION....................TOP DOWN
ICA...INTER-LATA CARRIER.................10XXX
OPS...OUTGOING OUTPULSING SEQUENCE.......DESTINATION NUMBER ONLY
DIAGNOSTIC PARAMETERS: Y or N..............N
```

Configuring the RS-232 Serial Cable

This integration followed the pinout below to configure the RS-232 serial cable using an RJ-45-to-DB-9 terminal adaptor connector.

Table 58: Pinout for the RS-232 Serial Cable

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

CHAPTER 9

Application Note for the NEC NEAX 2400 IMX TIMG Integration

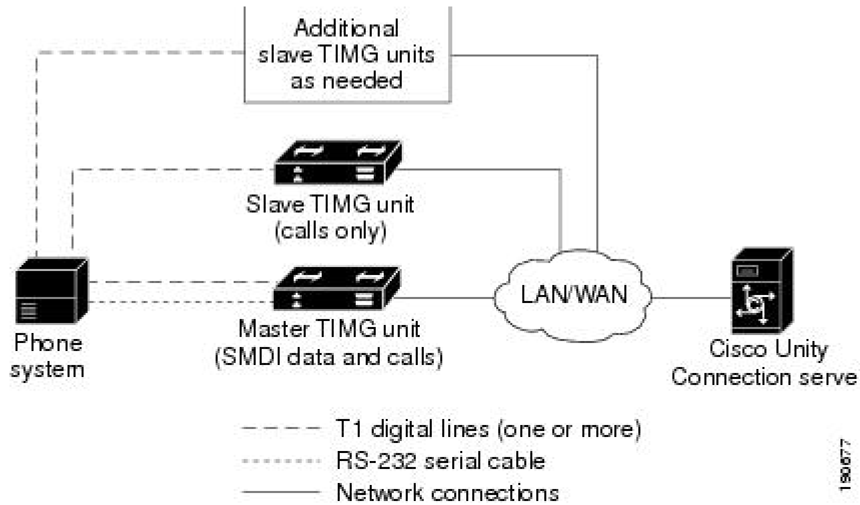
Introduction

This is an application note for programming the NEC NEAX 2400 IMX phone system for a serial SMDI integration with Cisco Unity Connection using TIMG units. For instructions on setting up the TIMG units (media gateways) and creating the integration in Unity Connection, see the Setting Up a Serial (SMDI, MCI, or MD-110) TIMG Integration with Cisco Unity Connection chapter.

Network Topology

Network Topology shows the required connections for a serial SMDI integration using TIMG units.

Figure 4: Connections for a Serial SMDI TIMG Integration



For more information about this integration, see Integration Description

Requirements

The phone system met the following requirements:

- The NEC NEAX 2400 IMX phone system.
- MCI feature II.
- One T1 digital trunk interface card (card number PA-24DTR/DLI) for each group of 24 voice messaging ports.

Note that the following requirements for the T1 digital trunk interface card before programming the phone system:

- The firmware must be configured to support T1 line-side signaling.
- The card must be validated.

Programming NEC NEAX 2400 IMX Phone System for TIMG Integration

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

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

Example of Programming for the NEC NEAX 2400 IMX Phone System in a TIMG Integration

1. 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 applicable tenant and UCD pilot numbers.
2. 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.

Table 59: Programming System Data

System	Index	Bit	Value	Description
1	17	b4	1	Release (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	Release 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
	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

3. 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.

Table 60: 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	MWI controlled by MCI

4. Use the ASDT command to add ports that connect to the first voice messaging port on the first TIMG unit by entering the following settings.

Table 61: ASDT Command Settings for Adding Ports

Field	Setting
TN	Enter the tenant number, which is typically 1.
STN	Enter the station number.
TEC	Enter 11 (which sets the port type as "voice mail"). Some sites may require setting this to 3 in order to complete transfers to Virtual Line Groups (AMNO).
RSC	Accept the default (all route options) or enter another route service class.
SFC	Accept the default (all options) or enter another service feature class.

5. In the WRT field, enter **Y** and press **Enter**.
6. Repeat Step 4. and Step 5. for all remaining ports that connect to the voice messaging ports on the TIMG unit.
7. Repeat Step 6. for all remaining TIMG units.
8. Use the ASHU command to add the UCD hunt group access number (a real or virtual extension number) by entering the following settings.

Table 62: ASHU Command Settings for Adding the Hunt Group Access Number

Field	Setting
TN	Enter the tenant number, which is typically 1.
STN	Enter the access number. for the hunt group.
Edit STN	Enter the extensions for each voice messaging port on the TIMG units, pressing Enter after each extension.

9. Select **Set**.

CHAPTER 10

Application Note for the Nortel SL-100 TIMG Integration

Introduction

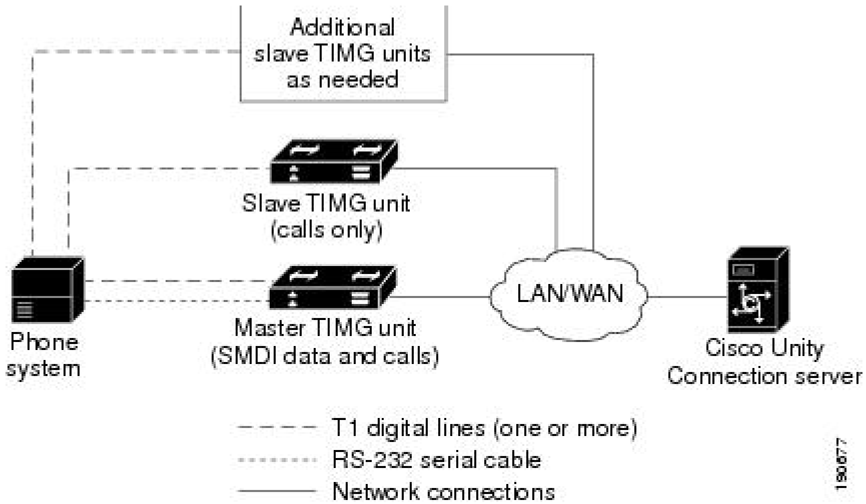
This is an application note for programming the Nortel SL-100 phone system for a serial SMDI integration with Cisco Unity Connection using TIMG units. For instructions on setting up the TIMG units (media gateways) and creating the integration in Unity Connection, see the Setting Up a Serial (SMDI, MCI, or MD-110) TIMG Integration with Cisco Unity Connection chapter.

Network Topology

Figure 10-1 shows the required connections for a serial SMDI integration using TIMG units.

Figure 5: Connections for a Serial SMDI TIMG Integration

For more information about this integration, see the Integration Description chapter.



Requirements

The phone system met the following requirements:

- The Nortel SL-100 phone system.
- Software version SE-06 or later.
- Line Side T1 Card (NT5D11 or equivalent) to terminate the T1 line.

Programming Nortel SL-100 Phone System for TIMG Integration

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

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

Example of Programming for the Nortel SL-100 Phone System in a TIMG Integration

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

Table 63: Settings for Table MPC

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

1. Enter **table mpclink** and press **Enter**.
2. Enter **add** and press **Enter**.
3. On Table MPCLINK, enter the following settings.

Table 64: Settings for Table MPCLINK

Field	Subfield	Setting
LINKKEY	MPCNO	Enter the MPC number used for SMDI (the same number entered in Table MPC).
	LINKNO	Enter the MPC link number for SMDI application with ASYNC protocol.
LINKALM		Enter **Y** to activate the MPCLINK alarm for system busy (SYSB) MPC links. Enter **N** if you do not want to activate the MPCLINK alarm for system busy (SYSB) MPC links. **Note** If you enter N, the system does not generate MPC908 (MPC link state transition) logs.
PROTOCOL		Enter **ASYNC**.
LINKNABL		Enter **765**.
PARM		Enter **APLDEFN**.
ADEFN		Enter **SMDI**.
PARM		Enter **BAUDRATE**.
RATE		Enter **B9600**.
PARM		Enter **PARITY**.
PRTY		Enter **EVEN**.

Field	Subfield	Setting
PARM		The following are among the optional parameters: L1IDLY, L2IDLY, LNKDOWN. If you enter a parameter, you are then prompted to enter a value for it. L1IDLY and L2IDLY timers can be used in offices with heavy SMDI/VMS traffic to shorten the amount of time the MPC can delay sending an MWI to the switch. (The default is 3 seconds.) The LNKDOWN timer adjusts the length of time the switch takes to recognize LINK failure and sets the LINK to SYSB. (The default is 2 seconds.)
CHARBITS		Enter **BIT7**.

1. Enter **table sllnkdev** and press **Enter**.
2. Enter **add** and press **Enter**.
3. On Table SLLNKDEV, enter the following settings.

Table 65: Settings for Table SLLNKDEV

Field	Setting
DEVNAME	Enter a unique device name.
DEVICE	Enter **1X89**.
MPCNO	Enter the value MPC number that was specified in Table MPC.
LINKNO	Enter the value for the MPC link number that was specified in Table MPCLINK.
XLATION	Enter **NONE**.
PROTOCOL	Enter **NONE**.
DIRECTION	Enter **INOUTLK**.
XFER	Enter **SMDIDATA**.
OPTION	Enter **NUMOFDIGS**.
NUMDIGS	The number of digits sent by the phone system to the voice messaging system through the SMDI link. **Note** The entry value selected should match the dialing plan configured on the phone system.
OPTION	Enter **CGNADDRDN**.
OPTION	Enter **$**.

1. Enter **table ofrt** and press **Enter**.

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

1. Enter **add** and press **Enter**.
2. On Table OFRT, enter the following settings.

Table 66: Settings for Table OFRT

Field	Setting
RTE	If the record is the first in the route list, enter the route reference number assigned to the route list. Otherwise, leave this field blank.
RTESEL	Enter the route selector.
SNPA	Enter the serving NPA (area code) of the DN.
TYPCALL	Enter the type of call: • DD • NP • OA
ORIGSCRE	Enter **LCL** (for local) or **NLCL** (for non-local).
REPLDIGS	Enter up to 11 replace digits.
CANCNORC	Enter **Y** or **N** to indicate whether to cancel normal change.
BILLCODE	Enter the billing code. If there is no billing number, enter **N**.

1. Enter **table digcol** and press **Enter**.

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

1. Enter **add** and press **Enter**.
2. On Table DIGCOL, enter the following settings.

Table 67: Settings for Table DIGCOL

Field	Subfield	Setting
DGKEY	DATNAME	Enter the character assigned to the block of data in Table DIGCOL.
	DIGIT	Enter a numeric value from 0–9, STAR (*), or OCT (#) to specify the digit that is applicable to the record.
DGDATA	DGCOLSEL	Enter **COL** for the collection of more digits.

Field	Subfield	Setting
COLDATA	TMODE	Enter **S** for short timing mode.
	NUMDIGS	Number of digits. If the TMODE value is S, specify the number of digits for which short timing is required after the receipt of each digit. The number of digits specified, which does not include the initial digit, must be no greater than three for short timing.

1. Enter **table ucdgroup** and press **Enter**.

You use Table UCDGRP to set up the UCD group.

Note The UCD group must have a unique primary DN.

1. Enter **add** and press **Enter**.
2. On Table UCDGRP, enter the following settings.

Table 68: Settings for Table UCDGRP

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

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

1. Enter **table dnroute** and press **Enter**.

You use Table DNROUTE to set up the UCD group.

Note The UCD group must have a unique primary DN.

1. Enter **add** and press **Enter**.
2. On Table DNROUTE, enter the following settings.

Table 69: Settings for Table DNROUTE

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

1. Enter **table lninv** and press **Enter**.

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

1. Enter **add** and press **Enter**.
2. On Table LNINV, enter the following settings.

Table 70: Settings for Table LNINV

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

1. Enter **servord** and press **Enter**.

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

Table 71: Inputs for Adding Agents to the UCD Group

Prompt	Input	Description
SO:	NEW	
SONUMBER:	Press **Enter**.	When to invoke this service. Pressing Enter starts the service at the current date and time.
DN:		The Directory Number of the line. Use ten-digit DNs.
LCC_ACC:	IBN	The line class code of service.
GROUP:		The name of the IBN customer group to which the line belongs. For example, covm.
SUBGRP:		The subgroup number. For example, 0.
NCOS:		The network class of service. For example, 1.
SNPA:		The serving NPA (area code) of the DN.
LEN_OR_LT D:		The line equipment number of the line. For example, 4 0 1 0 (separated by spaces).
OPTION:	COD	The cut-off on disconnect.

Prompt	Input	Description
OPTION:	UCD	Uniform call distribution.
OPTION:	DGT	Digitone.
OPTION:	3WC	Three-way calling.
OPTION:	CXR	Call Transfer.
CXFERTYP:	CTALL	Call Transfer Type. CTALL = transfer all calls.
CXRRCL:	N	Call Transfer Recall.
METHOD:	STD	Method of Call Transfer: Std = Std Call Transfer method.
OPTION:	SMDI	Simplified message desk interface.
LINENO:		The UCD terminal number. This is the line number associated with the SMDI channel. This parameter must be unique for each agent in the associated UCDGRP.
UCDGRP:		The UCDNAME from the UCDGRP table. This is the UCD group to which you are adding the agent.
AUTO_LOG:	Y	Autologon capability required.
OPTION:	$	The data you have entered appears.
	Y	Enter **Y**.

1. Enter **servord** and press **Enter**.

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

Table 72: Inputs for Adding Agents to the UCD Group

Prompt	Input	Description
SO:	NEW	
SONUMBER:	Press **Enter**.	When to invoke this service. Pressing Enter starts the service at the current date and time.
DN:		The directory number of the line. This is the DN that you enter in the SDN table.
LCC:	IBN	The line class code of service.
GROUP:		The name of the IBN customer group to which the line belongs.
SUBGRP:		The subgroup number.
NCOS:		The network class of service.

Prompt	Input	Description
SNPA:		Serving NPA (area code) of the DN.
LEN:		Line equipment number of the line. For example, 4 0 1 0.
OPTION:	cfb	Call Forward Busy. **Note** This input is optional.
CFBCNTL:	N	Normal assignment for CFB. **Note** This input is optional.
CFBDN:		The primary UCD DN. **Note** This input is optional.
OPTION:	CFF	Call Forward Fixed.
CFFDN:		The Primary UCD DN.
OPTION:	CFU	Call Forward Universal.
OVRDACR:	N	Override Automatic Callback.
OPTION:	$	The data you entered appears.
	Y	Enter **Y** to confirm the data.

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

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

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

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

5. Enter **table ibnxla** and press **Enter**.

You use Table IBNXLA to set up and message waiting indicators (MWIs).

1. Enter **add** and press **Enter**.
2. On Table IBNXLA, enter the following settings.

Table 73: Settings for Table IBNXLA

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

Configuring the RS-232 Serial Cable

This integration used a null modem cable. The pinout for a DB-9 terminal adaptor connector is below.

Table 74: Pinout for a DB-9 Connector

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

The pinout for a 34-pin to DB-25 terminal adaptor connector is below.

Table 75: Pinout for a 35-Pin to DB-25 Connector

Pin for 34-Pin Connector	Pin for DB-25 Connector	Serial Port Pin Definition from the Phone System
14	2	Transmit Data (required)

Pin for 34-Pin Connector	Pin for DB-25 Connector	Serial Port Pin Definition from the Phone System
26	3	Receive Data (required)
23	4	Request to Send (required)
34	5	Clear to Send (required)
16	6	Data Set Ready (required)
11	7	Signal Ground (required)
31	8	Data Carrier Detect (required)
37	20	Data Terminal Ready (required)
16	17	Receive Clock
17	12	Secondary Data Carrier Detect
24	24	Secondary Clock Transmit External

CHAPTER 11

Settings for TIMG Firmware Version 5.x

Introduction

This chapter provide the information for TIMG settings when firmware version 5.x is installed on the TIMG units. You must upgrade your TIMG units to the most recent version that is available at http://software.cisco.com/download/navigator.html?mdfid=280082558&i=rm. For instructions on downloading and installing the most recent TIMG firmware, see the chapter for your phone system integration in this guide.

TIMG Settings for a Serial Integration (Firmware Version 5.x)

Step 1 On a Windows workstation, sign in to a TIMG unit.

Step 2 On the Configure menu, select **IP**.

Step 3 On the IP page, enter the following settings for LAN1.

Table 76: IP Page Settings for LAN1

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

Step 4 Select **Apply Changes**.

Step 5 On the Configure menu, select **System**.

Step 6 On the System page, enter the following settings.

Table 77: System Page Settings for the System and Telephony Groups

Field	Setting
Operating Mode	Select **SIP**.
PCM Coding	Select **uLaw**.

Step 7 Determine which serial port on the TIMG unit you use to connect the data link serial cable from the phone system, then enter the following settings in the applicable group.

Table 78: System Page Settings for Serial Port Groups

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

Step 8 Select **Apply Changes**.

Step 9 On the Configure menu, select **Gateway**.

Step 10 On the Gateway page, select the **Gateway Routing** tab.

Step 11 On the Gateway Routing tab, Unity Connection without a cluster, skip to Step 12. If Unity Connection has a cluster configured, do the following substeps:

a) In the Fault Tolerance Enabled field, select **Yes**.

b) In the Load Balancing Enabled field, select **No**.

Step 12 Under VoIP Endpoint ID, enter the following settings.

Table 79: Gateway Routing Tab Settings

Field	Setting
VoIP Endpoint ID: 1	*(Unity Connection without a cluster)* Enter the server name of the Unity Connection server. *(Unity Connection with a cluster configured)* Enter the server name of the subscriber server.

Field	Setting
VoIP Endpoint ID: 2	*(Unity Connection without a cluster)* Enter the server name of the Unity Connection server. *(Unity Connection with a cluster configured)* Enter the server name of the publisher server.

Step 13 Select **Apply Changes**.

Step 14 On the Gateway page, select the **Gateway Advanced** tab.

Step 15 On the Gateway Advanced tab, enter the following settings.

Table 80: Gateway Advanced Tab Settings

Field	Settings
Advanced Call Routing	
Call Connect Mode	Select **OnAnswer**.
Send DNIS to VoIP Endpoint	Select **No**.
Destination for Unroutable IP Calls	Leave this field blank.
Destination for Unroutable PBX Calls	Leave this field blank.
Monitor Call Connections	Select **No**.
Telephony	
Minimum Call Party Delay	Enter **500**.
Maximum Call Party Delay	Enter **2000**.
Dial Digit on Time	Enter **100**.
Dial Inter-Digit Time	Enter **100**.
Dial Pause Time	Enter **2000**.
Turn MWI On FAC	Leave this field blank.
Turn MWI Off FAC	Leave this field blank.
Dial Send Key	Select **None**.
Outbound Call Connect Timeout	Enter **10000**.

Field	Settings
Wait for Ringback/Connect on Blind Transfer	Select **Yes**.
Hunt Group Extension	Enter the pilot number for the Unity Connection voice messaging ports.
Audio	
Audio Compression	Select the preferred codec for audio compression: • **G.711** • **G.729AB**
RTP Digit Relay Mode	Select **RFC2833**.
Signaling Digit Relay Mode	Select **Off**.
Voice Activity Detection	Select **On**.
Frame Size	Select the applicable setting: • G.711—**20** • G.729AB—**10** **Caution** Failure to use the correct setting results in recorded messages containing nothing but silence.
Frames Per Packet	Select the applicable setting: • G.711—**1** • G.729AB—**2** **Caution** Failure to use the correct setting results in recorded messages containing nothing but silence.
Quality of Service	
Call Control QOS Byte	Enter **104** (equivalent to DSCP AF31).
RTP QOS Byte	Enter **184** (equivalent to DSCP EF).
Traps and Alarms	
E-mail Alarms Enabled	Select **No**.
SNMP Traps Enabled	Select **No**.
HTTP Server Enabled	Select **Yes**.
HTTPs Server Enabled	Select **No**.

Step 16 Select **Apply Changes**.

Step 17 On the Gateway page, select the **Gateway Capabilities** tab.

Step 18 Enter the following settings for all ports that are used by voice messaging ports on Unity Connection.

Table 81: Gateway Capabilities Tab Settings

Field	Setting
Telephony Port Capability	Select **Both**.
Telephony Port Enabled	For ports that are used by voice messaging ports on Unity Connection, select **Yes**. For ports that are not used, select **No**.

Step 19 Select **Apply Changes**.

Step 20 On the Configure menu, select **T1E1**.

Step 21 On the T1E1 page, select the **T1/E1 Mode** tab.

Step 22 On the T1/E1 Mode tab, enter the following settings.

Table 82: T1/E1 Mode Tab Settings

Field	Setting
Line Mode	Select **T1**.
Signaling Mode	Select **CAS**.
Interface Mode	Select **Terminal**.

Step 23 Select **Apply Changes**.

Step 24 Select the **T1-CAS Protocol** tab.

Step 25 On the T1-CAS Protocol tab, enter the following settings.

Table 83: T1-CAS Protocol Tab Settings

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

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

Step 26 Select **Apply Changes**.

Step 27 On the Configure menu, select **Serial Protocol**.

Step 28 On the Serial Protocol page, enter the following settings.

Table 84: Serial Protocol Page Settings

Field	Setting
Serial Mode (Master/Slave)	Select the applicable setting: • **Master**—Select this setting when this TIMG unit is connected to the data link serial cable from the phone system. There can be only one master TIMG unit in a phone system integration. • **Slave**—Select this setting when this TIMG unit is not connected to the data link serial cable from the phone system. There can be multiple slave TIMG units in a phone system integration.
Serial Interface Protocol	Select the serial protocol that your phone system uses: • **SMDI** • **MCI** • **MD-110**
MWI Response Timeout	Enter **2000**.

Field	Setting
IP Address of Serial Server	If the TIMG unit is the master, leave this field blank. If the TIMG unit is a slave, enter the IP address of the master TIMG unit (the TIMG unit that is connected to the data link serial cable from the phone system).
Serial Cpid Expiration	Enter **5000**.
Logical Extension Number	If the TIMG unit is the master, leave this field blank. If the TIMG unit is a slave, enter the IP address of the master TIMG unit (the TIMG unit that is connected to the data link serial cable from the phone system).

Step 29 Select **Apply Changes**.

Step 30 On the Configure menu, select **SIP**.

Step 31 On the SIP page, enter the following settings.

Table 85: SIP Page Settings

Field	Setting
Host and Domain Name	Enter the domain name of the TIMG unit.
Transport Type	Select **UDP**.
Call as Domain Name	Select **No**.
SIPS URI Scheme Enabled	Select **No**.
Invite Expiration	Enter **120**.
DNS Server Address	Enter the IP address of the DNS server.
Registration Server Address	Leave this field blank.
Registration Server Port	Enter **5060**.
Registration Expiration	Enter **3600**.
UDP/TCP Transports Enabled	Select **Yes**.
TCP/UDP Server Port	Enter **5060**.
Primary Proxy Server Address	Leave this field blank.
Primary Proxy Server Port	Not applicable. Leave the default setting.
Backup Proxy Server Address	Not applicable. Leave the default setting.
Backup Proxy Server Port	Not applicable. Leave the default setting.
Proxy Query Interval	Enter **10**.
T1 Time	Enter **500**.

Field	Setting
T2 Time	Enter **4000**.
T4 Time	Enter **5000**.

Step 32 Select **Apply Changes**.

Step 33 On the Configure menu, select **Tones**.

Step 34 On the Tones page, select the **Learn** tab.

Caution Destination addresses cannot be duplicated in the same session. Otherwise, the process for learning tones do not succeed. If you do not have enough available phones to learn all the tones at one time, you can run multiple sessions to learn tones individually by checking or unchecking the applicable Acquire Tone check boxes.

Step 35 On the Tones page, for the Dialtone event, confirm that the Acquire Tone check box is checked and leave the Destination Address field blank.

Step 36 On the Tones page, for the Busy Tone event, confirm that the Acquire Tone check box is checked and do the following substeps to verify that the tone is correct.

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

Step 37 On the Tones page, in the Destination Address field for Busy Tone, enter the extension that you dialed in Step 36c. from the third phone.

Step 38 On the Tones page, for the Error/Reorder Tone event, confirm that the Acquire Tone check box is checked and do the following substeps to verify that the tone is correct.

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

Step 39 On the Tones page, in the Destination Address field for Error/Reorder Tone, enter the extension that you dialed in Step 38a.

Step 40 On the Tones page, for the Ringback Tone event, confirm that the Acquire Tone check box is checked and do the following substeps to verify that the tone is correct.

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

Step 41 On the Tones page, in the Destination Address field for Ringback Tone, enter the extension that you dialed in Step 40a.

Step 42 Select **Learn**.

Step 43 When the process is complete, check the check box for each newly learned tone and select **Apply**.

Step 44 Hang up the phones that you used in Step 36.

Step 45 If the TIMG unit is connected to a Nortel SL-100 phone system, do the following substeps to learn the stutter dial tone. Otherwise, continue to Step 46.

a) On the Tones page, on the Learn tab, for the Dialtone event, confirm that the Acquire Tone check box is checked and enter **!** in the Destination Address field.

b) Uncheck the Acquire Tone check boxes for all other tones.
c) Select **Learn**.
d) When the process is complete, check the check box for each newly learned tone and select **Apply**.

Step 46 On the Configure menu, select **Restart**.
Step 47 On the Restart page, select **Restart Unit Now**.
Step 48 When the TIMG unit has restarted, in the View menu, select **Refresh**.
Step 49 Repeat Step 1 through Step 48 on all remaining TIMG units.

TIMG Settings for an In-Band Integration (Firmware Version 5.x)

Step 1 On a Windows workstation, sign in to a TIMG unit.
Step 2 On the Configure menu, select **IP**.
Step 3 On the IP page, enter the following settings for LAN1.

Table 86: IP Page Settings for LAN1

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

Step 4 Select **Apply Changes**.
Step 5 On the Configure menu, select **System**.
Step 6 On the System page, enter the following settings.

Table 87: System Page Settings for the System and Telephony Groups

Field	Setting
Operating Mode	Select **SIP**.
PCM Coding	Select **uLaw**.

Step 7 Select **Apply Changes**.

Step 8 On the Configure menu, select **Gateway**.

Step 9 On the Gateway page, select the **Gateway Routing** tab.

Step 10 On the Gateway Routing tab, Unity Connection without a cluster, skip to Step 11. If Unity Connection has a cluster configured, do the following substeps:

a) In the Fault Tolerance Enabled field, select **Yes**.
b) In the Load Balancing Enabled field, select **No**.

Step 11 Under VoIP Endpoint ID, enter the following settings.

Table 88: Gateway Routing Tab Settings

Field	Setting
VoIP Endpoint ID: 1	*(Unity Connection without a cluster)* Enter the server name of the Unity Connection server. *(Unity Connection with a cluster configured)* Enter the server name of the subscriber server.
VoIP Endpoint ID: 2	*(Unity Connection without a cluster)* Enter the server name of the Unity Connection server. *(Unity Connection with a cluster configured)* Enter the server name of the publisher server.

Step 12 Select **Apply Changes**.

Step 13 On the Gateway page, select the **Gateway Advanced** tab.

Step 14 On the Gateway Advanced tab, enter the following settings.

Table 89: Gateway Advanced Tab Settings

Field	Settings
Advanced Call Routing	
Call Connect Mode	Select **OnAnswer**.
Send DNIS to VoIP Endpoint	Select **No**.
Destination for Unroutable IP Calls	Leave this field blank.
Destination for Unroutable PBX Calls	Leave this field blank.
Monitor Call Connections	Select **No**.
Telephony	
Minimum Call Party Delay	Enter **500**.

Field	Settings
Maximum Call Party Delay	Enter **2000**.
Dial Digit on Time	Enter **100**.
Dial Inter-Digit Time	Enter **100**.
Dial Pause Time	Enter **2000**.
Turn MWI On FAC	Enter the code that the phone system uses to turn MWIs on.
Turn MWI Off FAC	Enter the code that the phone system uses to turn MWIs off.
Dial Send Key	Select **None**.
Outbound Call Connect Timeout	Enter **10000**.
Wait for Ringback/Connect on Blind Transfer	Select **Yes**.
Hunt Group Extension	Enter the pilot number for the Unity Connection voice messaging ports.
Audio	
Audio Compression	Select the preferred codec for audio compression: • **G.711** • **G.729AB**
RTP Digit Relay Mode	Select **RFC2833**.
Signaling Digit Relay Mode	Select **Off**.
Voice Activity Detection	Select **On**.
Frame Size	Select the applicable setting: • G.711—**20** • G.729AB—**10** **Caution** Failure to use the correct setting results in recorded messages containing nothing but silence.

Field	Settings
Frames Per Packet	Select the applicable setting: • G.711—**1** • G.729AB—**2** **Caution** Failure to use the correct setting results in recorded messages containing nothing but silence.
Quality of Service	
Call Control QOS Byte	Enter **104** (equivalent to DSCP AF31).
RTP QOS Byte	Enter **184** (equivalent to DSCP EF).
Traps and Alarms	
E-mail Alarms Enabled	Select **No**.
SNMP Traps Enabled	Select **No**.
HTTP Server Enabled	Select **Yes**.
HTTPs Server Enabled	Select **No**.

Step 15 Select **Apply Changes**.

Step 16 On the Gateway page, select the **Gateway Capabilities** tab.

Step 17 Enter the following settings for all ports that are used by voice messaging ports on Unity Connection.

Table 90: Gateway Capabilities Tab Settings

Field	Setting
Telephony Port Capability	Select **Both**.
Telephony Port Enabled	For ports that are used by voice messaging ports on Unity Connection, select **Yes**. For ports that are not used, select **No**.

Step 18 Select **Apply Changes**.

Step 19 On the Configure menu, select **T1E1**.

Step 20 On the T1E1 page, select the **T1/E1 Mode** tab.

Step 21 On the T1/E1 Mode tab, enter the following settings.

Table 91: T1/E1 Mode Tab Settings

Field	Setting
Line Mode	Select **T1**.
Signaling Mode	Select **CAS**.

Field	Setting
Interface Mode	Select **Terminal**.

Step 22 Select **Apply Changes**.

Step 23 Select the **T1-CAS Protocol** tab.

Step 24 On the T1-CAS Protocol tab, enter the following settings.

Table 92: T1-CAS Protocol Tab Settings

Field	Setting
T1 CAS Protocol	Select **Loop_Start**.
Line Encoding	Select **B8ZS**.
Framing	Select **EFS**.
Selects Transmit Pulse Waveform	Select **Short_Haul_110ft**.
Flash Hook	Enter **550**.
Consult Call Dialtone Drop Code	Enter **!!**.
Consult Call Proceeding Drop Code	Enter **!!**.
Consult Call Busy Drop Code	Enter **!**.
Consult Call Error Drop Code	Enter **!!**.
Consult Call Connected Drop Code	Enter **,,,,**.
Consult Call Disconnected Drop Code	Enter **!**.
MWI confirmation Tone	Select **No**.
CPID Type	Select **TypeII_CPID**.
Initial Wait for Inband CPID	Enter **5000**.
Inband CPID Complete Timeout	Enter **500**.

Step 25 Select **Apply Changes**.

Step 26 On the Configure menu, select **SIP**.

Step 27 On the SIP page, enter the following settings.

Table 93: SIP Page Settings

Field	Setting
Host and Domain Name	Enter the domain name of the TIMG unit.
Transport Type	Select **UDP**.
Call as Domain Name	Select **No**.
SIPS URI Scheme Enabled	Select **No**.
Invite Expiration	Enter **120**.
DNS Server Address	Enter the IP address of the DNS server.
Registration Server Address	Leave this field blank.
Registration Server Port	Enter **5060**.
Registration Expiration	Enter **3600**.
UDP/TCP Transports Enabled	Select **Yes**.
TCP/UDP Server Port	Enter **5060**.
Primary Proxy Server Address	Leave this field blank.
Primary Proxy Server Port	Not applicable. Leave the default setting.
Backup Proxy Server Address	Not applicable. Leave the default setting.
Backup Proxy Server Port	Not applicable. Leave the default setting.
Proxy Query Interval	Enter **10**.
T1 Time	Enter **500**.
T2 Time	Enter **4000**.
T4 Time	Enter **5000**.

Step 28 Select **Apply Changes**.

Step 29 On the Configure menu, select **Tones**.

Step 30 On the Tones page, select the **Learn** tab.

Caution Destination addresses cannot be duplicated in the same session. Otherwise, the process for learning tones do not succeed. If you do not have enough available phones to learn all the tones at one time, you can run multiple sessions to learn tones individually by checking or unchecking the applicable Acquire Tone check boxes.

Step 31 On the Tones page, for the Dialtone event, confirm that the Acquire Tone check box is checked and leave the Destination Address field blank.

Step 32 On the Tones page, for the Busy Tone event, confirm that the Acquire Tone check box is checked and do the following substeps to verify that the tone is correct.

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

Step 33 On the Tones page, in the Destination Address field for Busy Tone, enter the extension that you dialed in Step 32c. from the third phone.

Step 34 On the Tones page, for the Error/Reorder Tone event, confirm that the Acquire Tone check box is checked and do the following substeps to verify that the tone is correct.

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

Step 35 On the Tones page, in the Destination Address field for Error/Reorder Tone, enter the extension that you dialed in Step 34a.

Step 36 On the Tones page, for the Ringback Tone event, confirm that the Acquire Tone check box is checked and do the following substeps to verify that the tone is correct.

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

Step 37 On the Tones page, in the Destination Address field for Ringback Tone, enter the extension that you dialed in Step 36a.

Step 38 Select **Learn**.

Step 39 When the process is complete, check the check box for each newly learned tone and select **Apply**.

Step 40 Hang up the phones that you used in Step 32.

Step 41 On the Configure menu, select **Import/Export**.

Step 42 On the Import/Export page, under Export Settings, select **Export Settings**.

Step 43 In the File Download dialog box, select **Save**.

Step 44 In the Save As dialog box, browse to the Windows workstation that has access to the TIMG units, browse to a directory where you want to save the file, and select **Save**.

Step 45 In the Download Complete dialog box, select **Open**. Notepad opens the file Config.ini that you saved.

Step 46 Locate the line with the following parameter:

```
telautoanswer
```

Step 47 Confirm that the value of the parameter is **no** so that the line reads as follows:

```
telautoanswer = no
```

Step 48 Locate the line with the following parameter:

```
telFacCDropProc
```

Step 49 Confirm that the value of the parameter is **!!** so that the line reads as follows:

```
telFacCDropProc = !!
```

Caution The telFacCDropProc parameter must be set to !!. If the telFacCDropProc parameter is set to 1, supervised transfers fail, and the caller hear the called party standard greeting two times.

Step 50 Save the file, and exit Notepad.

Step 51 On the Configure menu of the TIMG unit, select **Import/Export**.

Step 52 On the Import/Export page, under Import Settings, select **Browse**.

Step 53 In the Choose File dialog box, browse to the file Config.ini that you saved.

Step 54 Select **Config.ini**, and select **Open**.

Step 55 On the Import/Export page, select **Import Settings**.

Step 56 When prompted to restart the TIMG unit, select **OK**.

Step 57 When the TIMG unit has restarted, in the View menu, select **Refresh**.

Step 58 Repeat Step 1 through Step 57 on all remaining TIMG units.

INDEX