

NetVanta Unified Communications Technical Note

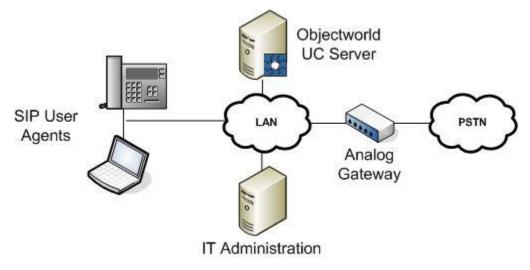
Installing and Configuring the Quintum Tenor AS Gateway

Introduction

The Quintum Tenor AS is a 2-4 port analog gateway used in NetVanta Unified Communications Server installations to provide a gateway between internal (SIP) phone calls and the outside phone network (PSTN). Voice communications from an internal phone have voice over IP (VoIP) signals converted into traditional analog voice, which is transmitted over the PSTN.

A gateway works in conjunction with the UC server's SIP Proxy and SIP. All telephony services are provided through the mutual cooperation of SIP gateways, SIP telephones, SIP proxy and the Core Application Service.

The following diagram illustrates the UC server's SIP architecture and its relationship with other components in a typical customer network.



Supported Features

Feature Name	Supported
Accept Incoming Calls	✓
Accept Outgoing Calls	✓
Trunk-to-trunk connect	✓
Calling Party Name	✓
Calling Party Number	✓
Answer Supervision	✓
Disconnect detection	✓
DTMF Tone Support (RFC2833 Compliant)	√
Conferencing with SIP Endpoints	✓
Direct Inward Dialing	✓
System Music on Hold Support	✓
Outgoing Fax Support	✓
Incoming Fax Support	✓
Unified Communication Features	Supported by Gateway
Active Message Delivery	✓
Paging Notification	 ✓
Transfer—Assisted/Supervised	✓
Transfer—Blind	✓
Multiple SIP Proxy Support	 ✓ *Available with survivability option

Interoperability Software Versions

The following gateway version was tested for interoperability:

- System Description: Quintum Tenor AS
- Hardware Version: P106-02-00
- Firmware Version: P106-12-00

Overview of Procedure

To provide its functionality, the **Quintum Tenor AS** must be connected to the internal LAN (a 100 Mbps connection is recommended) and from 1-4 PSTN analog phone lines.

The **Quintum Tenor AS** is primarily configured using a java configuration program. The program must be installed to configure and manage the gateway.

The basic steps for installation and configuration are:

- 1. Unpack the Quintum Tenor AS.
- 2. Mount the **Quintum Tenor AS**.
- 3. Connect cables.
- 4. Power up the **Quintum Tenor AS**.
- 5. Set a DHCP IP address reservation for the **Quintum Tenor AS** based on its MAC address.
- 6. Run the initial configuration wizard.
- 7. Configure the UC server to use the **Quintum Tenor AS**.

Note: Please see the instructions provided by Quintum for steps 1 to 4, and for information about running and configuring the gateway.

The rest of this document provides instructions for steps 5 to 7, which allow you to configure the **Quintum Tenor AS** for operation with the UC server.

Address Reservation

By default, the gateway is configured to use an IP address assigned by DHCP. The gateway can also be configured to use a static IP address. For routing calls out from the UC server, the **Quintum Tenor AS** must have an IP address that does not change.

Initial Configuration

Installing Tenor Configuration Manager

To begin configuration of the Quintum gateway, you must first install the Tenor Configuration Manager. You can either get it from the CD included with the gateway or at the Quintum support website (<u>http://www.quintum.com/support</u>).

After you have installed and run the Tenor Configuration Manager, the following screen appears.

Addres	s Book	-	_	_	_	
We	Icome to Tenor Config	juration Manag	er! Please specify/se	lect a Tenor DX/BX/AX/A	S/AF/CMS.	
	Discover	Cancel			Add Delete	Edit
	Tenor IP Address	Server Port	Description	Serial Number	Software Version	Login
		Connec	Close	Export	mport	

Adding the Gateway

If your PC is running on the same subnet as the gateway, the gateway can be added automatically. If your PC is running on a different subnet than the gateway, the gateway must be manually added.

To add the gateway automatically

- 1. Select **Discover** to automatically detect the gateway.
- 2. When the wizard finds the gateway, select **Connect**.

To add the gateway manually

1. Select Add.

The following screen appears.

Add Address	X	
Tenor IP ADDRESS:	192 . 168 . 8 . 29	
Tenor Server Port:	8080	
Description:		
Serial Number:		
Software Version:		
Login:	admin	
Password:	•••••	
Confirm Password:	•••••	
Remember Password		
OK Cancel		

- 2. Enter the IP Address of the gateway.
- 3. Enter **admin** as the username and password.
- 4. Select **OK**.
- 5. Select **Connect** on the *Address Book* screen.

Running the Configuration Wizard

After you connect, a wizard opens to set up the initial configuration of the gateway.

1. Select Next.

Tenor Configura	ition Wizard				×
Welcome to	o the Quintum Configuration Wiz	ard			
TENC	DRCONFIGURATION	WIZARD	CUINTUM The perfect fit.	Tell Me More About Configuration Task The Tenor Configuration Wizard	^
	Task IP Address Configuration Dial Plan Configuration Phone Port Configuration Multi Path Configuration Line Port Configuration VoIP Routing Configuration Idle Channel Configuration Configuration Summary	Remarks		approaches the preliminary setup of your Tenor as a series of Tasks. When you first start your configuration, the Status of every Task will appear in the Settings Table as "new." Once you have entered a configuration for the Task, the Status changes to "done," and the Remarks column of the table will reflect your change. You must advance through these Tasks in the order they are listed in the Settings Table. As you complete each Task, click Next at the bottom right of the Tenor Configuration Wizard window. Go Back to Finished Tasks If at any point you wish to return to a previously finished Task to make a	
	Tenor has the latest s	oftware version (P105-	.19-12).	< Exit <back next<="" th=""><th>></th></back>	>

2. On the screen below, you have the option to choose how your gateway obtains its IP Address and network settings. A static IP address is recommended for a gateway.

Tenor Configuration Wizard	X
IP Address Configuration	
Select "Specify a static IP", if your LAN administrator has assigned an IP for the Tenor AF. Select "Use DHCP", if your LAN uses DHCP to assign IP addresses. Select "Use PPPoE", if you are connecting directly to a DSL modem.	Tell Me More About
Please specify how your Tenor AF will obtain an IP Address: Use DHCP Specify a static IP Use PPPoE 	Click on this option to enable DHCP (Dynamic Host Configuration Protocol). DHCP provides a framework for passing configuration information to hosts on a TCP/IP network. You should select this if your LAN administrator tells you that it is how your LAN assigns IP addresses.
	Specify a static IP Click on this option if you have a specific static IP address that you want to assign to your Tenor. This is a fixed address, not a dynamic one that changes each time you connect to your Internet provider. Use PPPoE
	Exit <back next<="" td=""></back>

3. You can specify whether you want to obtain DNS server addresses automatically or if you want to manually configure them. If you are using DHCP, you can automatically obtain the DNS server addresses; otherwise you must manually configure them. Select **Next** to continue.

Tenor Configuration Wizard	\mathbf{X}
IP Address Configuration	
Select "Obtain DNS Server addresses automatically" if the DHCP server supplies DNS Server IP addresses; select "Manually Configure DNS Server addresses" if you know your DNS Server IP addresses; otherwise, select "Manually Configure DNS Static DNS Host" to configure a static DNS Host name and IP address	Tell Me More About Obtain DNS Server addresses automatically
 Please specify how your Tenor AF will obtain a DNS Server IP Address: Obtain DNS Server addresses automatically Manually Configure DNS Server addresses Manually Configure Static DNS Host 	Select this option for your Tenor to perform a search on your LAN to find a DNS (Domain Name Service) server. This server contains information on how to route calls, resolving domain names into IP addresses. Manually Configure DNS Server addresses If you have a specific IP address for a DNS server, select this option. You will be allowed to configure both a primary and secondary DNS server IP address. Manually Configure Static DNS Host
	Configure a static DNS host name and IP address.

4. The first task is complete. Select **Next** to continue.

			Configuration Task
Status	Task	Remarks	The Tenor Configuration Wizard approaches the preliminary setup o your Tenor as a series of Tasks.
V	IP Address Configuration	DHCP + Auto DNS Server IP	When you first start your
	Dial Plan Configuration		configuration, the Status of every
	Line Port Configuration		Task will appear in the Settings
	VoIP Routing Configuration		Table as "new." Once you have
	Idle Channel Configuration		entered a configuration for the Task
	Configuration Summary		the Status changes to "done," and
			the Remarks column of the table wi reflect your change. You must
			advance through these Tasks in the
			order they are listed in the Settings
			Table. As you complete each Task.
			click Next at the bottom right of the
			Tenor Configuration Wizard window
			Tenor Configuration Wizard windov Go Back to Finished Tasks
			click Next at the bottom right

5. The **Dial Plan Configuration** screen allows you to set up the dialing plan. Choose **None** from the **Dial Plan Country** list. Currently the dial plan rules do not work and will result in outgoing calls not working. When finished, select **Next** to continue.

Tenor Configuration Wizard		
Dial Plan Configuration		
Please select a Dial Plan Co	ountry for your Tenor AS.	Tell Me More About Dial Plan Country
Dial Plan Country:	None	Dial Plan Country is used to select the country (or set of dial plan rules) where the Tenor is located. This automatically configures the Tenor with the international dial plan for the specified country.
		Options are as follows: Generic CCITT (default) US/NANPA (North American Numbering Plan Area) 7-digit local numbers US/NANPA 10-digit local numbers China Taiwan UK Australia No dial plan/None
		Exit <back next=""></back>

 On the Phone Port Configuration screen, choose the method for Disconnect Generation and Caller ID Generation. These settings depend on your carrier and location. When finished, select Next to continue.

Tenor Configuration Wizard		×
Phone Port Configuration	Disconnect Generation and Caller ID Generation for the phone port Configuration Loop Start, Forward Disconnect FSK	Tell Me More About Disconnect Generation This setting allows you to configure the Loop Start signaling for your Tenor to determine how a call is disconnected on the phone ports of your Tenor. The options are as follows: • Disabled. Signaling is set to
		Loop Start. • Loop Start, Reverse Battery. Signaling is set to Loop Start Reverse Battery. • Loop Start Reverse Battery. • Loop Start Forward Disconnect. • Tone Based Disconnect Supervision. Signaling is set to Loop Start and Tone Based Disconnect Supervision is enabled.

7. On the **Phone Port Configuration** screen, you can map DID numbers to individual channels on the gateway. When finished, select **Next** to continue.

Tenor Configuration Wizard	$\mathbf{\overline{\mathbf{N}}}$
Phone Port Configuration	
Please add a phone number/extension and select a corresponding channel for your Tenor AS. You can add a shared phone number for all channels, or an individual number for each channel.	Tell Me More About Phone Number/Extension
Add Phone Number/Extension Phone Number/Extension: 16135999698 Channel: 1	Enter a phone number or PBX extension to be associated with a specific channel on your Tenor. You may specify an individual number for each channel, or one number for all channels. If you select "All" to assign a single number for all channels, this will create a hunt group so that when a call comes in and the first phone is busy, the second answers, and so on.
Save/OK Cancel Done Please add a phone number/extension.	The format of the number will depend on whether the number is public or private. If you wish to assign extensions from a private PBX, enter as many digits as are used for that installation (e.g., 7705).
	Exit <back next=""></back>

8. On the **Multipath Configuration** screen, you can choose for all calls to automatically pass through between the phone side and lines. You do not need to enable Pass Through calls, so you can select **No**.

Tenor Configuration Wizard	X
Multi Path Configuration	
"Pass Through Calls" is a feature that allows calls to unconditionally "pass through" between the phone side and line side of your Tenor AS.	Tell Me More About Pass Through Calls
Do you want to enable Pass Through Calls? ○ Yes ④ No	Sets whether the line-side configuration you are creating will allow calls to "pass through" to an opposite phone side. With this option set to "yes," calls coming from the channels on the line side will automatically "pass through" to the phone side.
	<
	Exit <back next=""></back>

9. On the Add Bypass Number screen, select Done, and then select Next.

Iulti Path Configuration A bypass call is one that is automatically sent from the phone-side to the line-side (PSTN); it will not be routed by VoIP. Some examples of Bypass Numbers include toll-free calls, high security calls, or emergency calls (911).	Tell Me More About Bypass Concept
Add Bypass Number	A bypass number is a telephone number that is automatically sent to th Public Switched Telephone Network (PSTN), and is not routed via VoIP. Some examples of bypass numbers include 1800 toll-free calls, and emergency calls such as 911. Bypass Number Pattern A bypass number is specified in the format in which it is dialed from the phone side (e.g., from a PBX, you may have a 3- or 4-digit number). It is permissible to use a "** as a wildcard digit (e.g., 1800*).
Save/OK Cancel Done Click the "Done" button if no bypass number to add.	

10. On the Line Port Configuration screen, choose the method for Disconnect Detection and Caller ID Detection. These settings depend on your carrier and location. Select Next to continue.

ine Port Configuration Please select the desired options fo Answer Detection for the line ports (r Disconnect Detection, Caller ID Detection, of your Tenor AS.	and Tone Based	Tell Me More About
PSTN/Line-Side Lopp Start Config			Disconnect Detection This setting allows you to configure the following options for the detection of a disconnect on the line ports of
Disconnect Detection: Caller ID Detection:	Loop Start, Forward Disconnect FSK or DTMF	× ×	your Tenor: • Disabled. Set Signaling Type to Loop Start.
Tone Based Answer Detection:	Disabled		 Loop Start, Reverse Battery. Set Signaling Type to Loop Start Forward Disconnect. Loop Start, Forward Disconnect. Set Signaling Type to Loop Start Reverse Battery. Tone Based Disconnect Supervision. Set Signaling Type to Loop Start and enable Tone Based Disconnect Supervision.

11. On the **VoIP Routing Configuration** screen, for integration with the UC server, choose **SIP only**. Select **Next** to continue.

Tenor Configuration Wizard	X
VoIP Routing Configuration	
Your Tenor AS supports two different protocols for outgoing VoIP calls: H.323 and SIP (Session Initiation Protocol).	Tell Me More About
Which Outgoing IP Routing protocol should be used? ○ H.323 only ④ SIP only	H.323 is an International Telecommunications Union (ITU) standard that provides specification for computers, equipment, and services for multimedia communication. H.323 defines how audio and video information is formatted and packaged for transmission over the network.
	If you select H.323 you must choose to route calls via defined endpoints (static routes) or via a Gatekeeper (routing table). SIP only
	SIP (Session Initiation Protocol) is an alternative method of signaling and routing calls that is used to establish a session on an IP network. The
	Exit <back next=""></back>

12. In the **SIP Server Information** section, change the Primary SIP Server to the IP address of your UC server. When finished, select **Next** to continue.

Tenor Configuration Wizard	X
VolP Routing Configuration	
Your Tenor AS requires a SIP Server, or another SIP endpoint, to make outgoing SIP calls. Please specify a Primary SIP Server (or the other SIP endpoint's) IP Address or URL (only if you configured a DNS server).	Tell Me More About Primary SIP Server IP/URL
SIP Server Information Primary SIP Server IP/Domain Name: 10.10.8.155	The IP address or domain name of the primary server used to make outgoing SIP calls. This server may be used for both Proxy and Registrar services.
Primary SIP Server Port: 5060	Primary SIP Server Port
Register Expiry Time (in sec.): 300	The default port is 5060. Enter the port number of the primary server used to make outgoing SIP calls.
	Register Expiry Time
	Enter the number of seconds between SIP registration messages. The default is 300 seconds. If the registration attempt does not include an expiration value, this time will be used. If this is set to 0, the Tenor will not attempt to register.
	<
	Exit <back next=""></back>

13. In the Add SIP User Information section, the wizard requires you to enter a User ID and Password. Enter **9999** for both fields. Select **Done** and then **Next** to continue.

Tenor Configuration Wizard		X
VolP Routing Configuration	1	
	nd a password (if required by your service provider). Then, select the er from those which you previously configured for your Tenor AS. Add SIP User Information	Tell Me More About User ID Enter the username assigned by the Primary SIP Registrar Server's
User ID: Password:	9999	administrator for authentication. It is also used by the Proxy to determine where to send a phone call. Password
Phone Number:	16135999698	Enter the password assigned by the Primary SIP Registrar Server's administrator for authentication. Phone Number
	Save/OK Cancel Done	Select from a drop-down list of phone numbers you previously entered in the Phone Port Configuration.
		Exit <back next=""></back>

14. In the **Idle Channel Configuration** screen, select **Yes** if you are not using the maximum number of phone ports on your gateway. Select **Next** to continue.

Tenor Configuration Wizard	$\overline{\mathbf{X}}$		
Idle Channel Configuration			
Your Tenor AS is equipped with 4 Phone/PBX-side channels, which are all enabled by default. You can disable those channels which are not assigned with an individual number.	Tell Me More About Disable Phone-Side Channels		
Do you want to disable Phone/PBX-side channels? ⊙ Yes ○ No	Phone-side channels on the Analog Tenor are enabled by default Select "Yes" if you want to disable one or more channels, and select "No" if you want to leave the channels enabled.		
	<		
	Exit <back next=""></back>		

15. If you selected **Yes** on the previous screen, you will be presented with the screen below. Clear the checkboxes of the ports that are not used. Select **Next** to continue.

Tenor Configuration Wizard	X
Idle Channel Configuration	
Your Tenor AS is equipped with 4 Phone/PBX-side channels, which are all enabled by default. You can disable those channels which are not assigned with an individual number. Please uncheck those channels which you want to disable.	Tell Me More About How to Disable Phone-Side Channels
Enable/Disable Phone/PBX-Side Channels Channel 1 Channel 2 Channel 3 Channel 4	Phone-side channels are enabled by default. Uncheck the box in front of a Phone-side channel that you want to disable.
	<
	Exit <back next=""></back>

16. In the **Idle Channel Configuration** screen, select **Yes** if you are not using the maximum number of PSTN ports on your gateway. Select **Next** to continue.

Tenor Configuration Wizard	X
Idle Channel Configuration	
Your Tenor AS is equipped with 4 Trunk/PSTN-side channels, which are all enabled by default. You can disable those channels which are not physically connected to the PSTN.	Tell Me More About Disable Line-Side Channels
Do you want to disable Trunk/PSTN-side channels? ⊙ Yes	Line-side channels on the Analog Tenor are enabled by default. Select "Yes" if you want to disable one or more channels, and select "No" if you want to leave the channels enabled.
O No	
	<
	Exit <back next=""></back>

17. If you selected **Yes** on the previous screen, you will be presented with the screen below. Clear the checkboxes of the ports that are not used. Select **Next** to continue.

Tenor Configuration Wizard	X
Idle Channel Configuration	
Your Tenor AS is equipped with 4 Trunk/PSTN-side channels, which are all enabled by default. You can disable those channels which are not physically connected to the PSTN. Please uncheck those channels which you want to disable.	Tell Me More About How to Disable Line-Side Channels
Enable/Disable Trunk/PSTN-Side Channels Channel 1 Channel 2 Channel 3 Channel 4	Line-side channels are enabled by default. Uncheck the box in front of a Line-side channel that you want to disable.
	<
	Exit <back next=""></back>

18. On the **Configuration Summary** screen, check the settings to make sure everything is correct. You can go back and make changes if necessary. When finished, select **Accept** to continue.

Т	enor Configuration Wizard	$\overline{\mathbf{X}}$
	Configuration Summary	
	Below is a summary of your Tenor AS configuration. Please verify the configuration, and click on the "Accept" button to accept the configuration or click on the "Back" button to make a change.	Tell Me More About Configuration Summary
	IP Address Configuration: Use DHCP DNS Server Configuration: Obtain DNS Server addresses automatically	When you complete the Tenor Configuration Wizard, a summary of your configuration is displayed for review. This summary is in plain text format and can be copied and pasted to save. Click
	Time Server Configuration:	Accept at the bottom right of the window to accept this configuration as defined;
	UTC Offset: O hour	click Back to return to one of the
	Primary Time Server IP Address: 192.43.244.18	configuration windows to make a
	Secondary Time Server IP Address: 128.138.140.44	change; click Exit to abort the Configuration Wizard.
	Dial Plan Configuration:	
	Dial Plan Country: None	
	Progress Tone Country: USA/Canada	
	Minimum Dial Digit Length: 1	
	Maximum Dial Digit Length: 30	
	Country Code:	
		<
	,	Exit <back accept<="" td=""></back>

19. After the initial configuration is done, reboot the Quintum gateway and navigate to the Line Port Configuration tab.On that tab, you can set up numbers that are allowed go through to the PSTN from the SIP side. A hopoff number would usually contain the first few digits of a PSTN number based on your location. For example, to allow local calls you would add an entry with 613 as the number pattern and replacement number where 613 is your local area code. If you want to allow international numbers for North America then you would use 011 as the number pattern and replacement number. When finished, select Confirm/OK.

Tenor Configuration Manager (Connected to T File View Tools Help	enor AS IP=192.168.8.29 SI	N=A012-2004DE S₩=P105-19-12		_	_	_	_ 🗆 🔀
Advanced Explore Basic Config	(—Line Port Configur	ation			
IP Address Configuration Time Server Configuration Dial Plan Configuration Phone Port Configuration Multi Path Configuration Line Port Configuration	Disconnect/Caller ID Con Disconnect Detection: Caller ID Detection: Hopoff Number Configure	Loop Start, Forward Disconne	ect 💌	Tone B	ased Answer Dete	ection: Disabled	
	Thepoin Warnser Comigan				Add	Delete Edit	
Port Configuration	Number Pattern	Replacement Number	Description	Туре	TON	NPI	
	613 555	613 555		Public Public	Unknown Unknown	Unknown Unknown	_
	1	1		Public	Unknown	Unknown	
	800	800		Public	Unknown	Unknown	
		Confirm/OK	Cancel	Help			

20. Select the Advanced Explore tab, and navigate to VoIP Configuration > Voice Codecs > Voice Codec-1. Set Voice Codec to G.711 Mu-law 64 kb. Select Confirm/OK.

🖬 Tenor Configuration Manager (Connected to Tenor AS IP=192.168.8.29 SN=A012-2004DE SW=P105-19-12)					
File View Tools Help					
🗢 🖻 📕 🔍					
Advanced Explore Basic Config	Voice Codec-1				
🗐 🖳 📴 System-Wide Configuration					
Ethernet Configuration VoIP Configuration	Description:				
Gatekeeper/Border Element	Voice Codec: G.711 Mu-law 64 Kbps 💌				
······H323 Signaling Group ⊞-···SIP Signaling Groups	Codec Payload Size: 20 ms				
DN Channel Map					
Gateway					
Fax Profile					
End Point Address Directory					
Voice Codecs Voice Codec-1 Voice Codec-2					
E-W Codec Profiles					
🚛 🔟 IP Dial Plans					
IP Routing Groups					
Image: Image and the second secon					
Grcuit Configuration					
Phone (FXS)/Line (FXO) Configuration					
In the second s					
	Confirm/OK Cancel Refresh Help				
	ок				

21. Navigate to Circuit Configuration > Trunk Circuit Routing Groups > Trunk Circuit Routing Group-line. Under the Advanced tab and in the Forced Routing Number box, enter the auto-attendant identity. Typically, this is set to 10000.

	Tenor AS IP=192.168.8.29 SN=A012-2004DE SW=P105-19-12)	_ 🗆 🔀
File View Tools Help		
도 🗈 📕 🖂		
Advanced Explore Basic Config	Trunk Circuit Routing Group-line	
	General Trunk ID/Caller ID IVR Call Services Hopoff Advanced Interface	
🗊 🐺 Ethernet Configuration		
VoIP Configuration	Forced Routing Number Type: Public	
🖃 🕎 Circuit Configuration		
Gignaling Configuration	Forced Routing Number: 10000	
Auto Switch Confi		
Caller ID Translation Directories	Two Stage Dialing	
Index International Internation Directories		
Hopoff Number Directories		
Hoporr Number Directories		
Trunk Circuit Routing Group-line		
	Modern Bypass: Disabled V Play 170	00 Promot
Phone (FXS)/Line (FXO) Configuration		
■	Stop/Radius Account ID: IP Address 💌 🗌 Provide	Auto Switch Progress Tone
	Auto Switch Number Type: DID received 🔽 Enable.	Auto Switch
	Auto Switch Number (E.164):	
	Confirm/OK Cancel Refresh Help	
	OK	

22. Under the **Trunk ID/Caller ID** tab and in the **Trunk ID Delivery** list, choose **Calling Party Number**.

🔤 Tenor Configuration Hanager (Connected to Tenor AS – IP=192.168.8.29 – SH=A012-2004DE – SW=P105-19-12) 📃 🔀					
File View Tools Help					
Advanced Explore Basic Config	Add End Of Dial Digit End Of D Answer Disconnect Tone: Disabled	nd ANI Number Digits: 0 Dial Digit: # 💌			
	Call Termination Indication: Off	D Delivery: Calling Party Number 💌			
	Caller ID Type: Use obtained caller ID Caller ID Translation Directory: -Not Set-	-Not Set-			
	Inbound DNIS Translation Dir: -Not Set-	-Not Set-			
	Confirm/OK Cancel Refresh Help				
ок					

23. Navigate to **VoIP Configuration > SIP Signaling Groups > SIP Signaling Group-1**. Under the **Advanced** tab, make sure that **Nortel** is selected in the **SIP Info Fo***rmat* list.

🔁 Tenor Configuration Manager (Connected to Tenor AS IP=192.168.8.29 SH=A012-2004DE SW=P105-19-12)					
File View Tools Help					
Advanced Explore Basic Config	SIP Signaling Group-1				
🖅 🔄 System-Wide Configuration	General MWI & Session Timer Advanced User Agent				
🕀 🤠 Ethernet Configuration					
VoIP Configuration	Request Retransmit Count: 11	Maximum Forwards: 70			
Gatekeeper/Border Element					
H323 Signaling Group	User Agent Header: Quintum/1.0.0	SIP No Connect Timeout (in sec.): 180			
SIP Signaling Groups	Proxy Fail-Over Behavior: O No Fail-Over (Always try the 1st Proxy) ③ Fail-Over on Ei	rror Response			
SIP Signaling Group-1					
DN Channel Map					
Gateway					
Fax Profile	SDP in 180 Ringing	Send 180 Ringing			
End Point Address Directory	SDP in 183 Progress	Send 183 Progress			
Voice Codecs					
Voice Codec-2	SIP Server in From Header	SIP Telephone Events			
Codec Profiles	SIP-PSTN Interworking				
B - Dial Plans					
IP Routing Groups	PRACK Method: Supported	SIP Info Format: Nortel 💌			
E Gruit Configuration	Send Remote Party ID				
🗊 🚛 Signaling Configuration	SIP Use DN In Register From/To Header				
Auto Switch Confi					
Caller ID Translation Directories					
Inbound DNIS Translation Directories					
Configuration					
+Hopoff Number Directories	Confirm/OK Cancel Refresh Help				
ОК					

24. Navigate to VoIP Configuration > IP Routing Groups > IP Routing Group-default. Under the **ANI** tab and in the **Relay Calling Name** list, choose **Relay CNAM in Invite**. Select Confirm/OK.

🖬 Tenor Configuration Manager (Connected to Tenor AS IP=192.168.8.29 SN=A012-2004DE SW=P106-12-07)					
File View Tools Help					
Advanced Explore Basic Config	ſ	IP Routing Group-default			
E System-Wide Configuration	General Advanced ANI Fax/QOS				
Ethernet Configuration					
VoIP Configuration					
Gatekeeper/Border Element	Relay ANI:	Relay ANI			
H323 Signaling Group	Default ANI:				
	Dolaan Val.				
DN Channel Map	Default ANI Screen Indicator:	Pass-through			
Gateway		Relay ANI			
Fax Profile	Default ANI Presentation Indicator:				
End Point Address Directory	Relay Calling Name:	Relay CNAM in INVITE			
Voice Codecs					
Codec Profiles IP Dial Plans					
IP Dial Plans					
IP Routing Groups					
E S Circuit Configuration					
Phone (FXS)/Line (FXO) Configuratio	Reject No-ANI Calls:	Disabled			
DSP Configuration					
<		Confirm/OK Cancel Refresh Help			
ОК					

25. To complete the changes, select Confirm/OK and then select the submit changes button.

Enabling CNG Tone Detection for Faxing

By default, a Quintum gateway will not detect CNG tones used for faxing unless the call is directed at a fax service. In order to receive faxes when a call is answered by a standard service (not a fax service), you must create a file and upload it to the gateway via FTP.

To enable CNG detection

- 1. Open notepad or another text editor.
- 2. Put in the following line: **enableCNGdetection 1**
- 3. Save the file as **var_config.cfg**.
- 4. From your Windows PC select **Start > All Programs > Accessories > Command Prompt**. The *Command Prompt* window is displayed.
- 5. Use the **CD** command to change to the directory on your PC in which you saved the **var_config.cfg** file.
- 6. Type **ftp** followed by the IP address of the unit. Press **Enter**.
- 7. Login with the username and password. The default for both is **admin**.
- 8. Use the **CD** command to change to the cfg directory (this is the directory on the Tenor into which you will copy the **var_config.cfg** file). Depending upon the product type and software revision, the directory structure you see in your Tenor VoIP device may be different.
- 9. Type **bin <Enter>**.
- 10. Type **put var_config.cfg <Enter>**
- 11. Restart the gateway from the **Tenor Configuration Manager** in **Tools > Reboot Tenor**.

Configuring the UC Server

After you add the gateway to your network, the UC server must be configured to handle incoming and outgoing phone calls. For outgoing calls you must add: a SIP gateway, a dial plan entry to route calls out through the gateway, and a toll restriction entry to allow those calls. For incoming calls you must add a UC server identity that can answer incoming calls from the gateway. These instructions are for release 4.1 of the UC server.

Adding a Trunk Identity

- 1. Go to **Identities**.
- 2. Right-click the right panel and select **New Identity**.
- 3. In the first page of the Wizard, select an **Attendant** identity. Make sure that the Identity is associated with the Admin profile.

4. On the following page, enter a descriptive name and enter **10000** for the address (assuming a standard configuration). Make sure that **Default Trunk Service** is the selected service.

Adding a SIP Gateway

- 1. Select Gateways.
- 2. Right-click the right panel and select **New Gateway**.
- 3. Choose Public Switched Telephone Network (PSTN) from the gateway list.
- 4. In the **Host** name field, enter the IP address of the gateway.
- 5. Enter a descriptive name for the gateway.
- 6. Save.

Configuring the Dial Plan

Incoming calls from the PSTN are already configured by having incoming calls routed to the 10000 Trunk identity. An entry or entries must be entered in the Dial Plan for outgoing calls to the PSTN.

- 1. Go to Communication Service > UC Server > Routing.
- 2. There are many possibilities here. If regular PSTN calls are to be routed out the gateway, add or modify an entry where the **Original Digits** are [0-9]{7,} and select the gateway. For example:

Dial Plan Entry 🛛 🔀				
Routi	ing rule			
Å	Original digits: Description: Priority:	8[0-9]{7,} PSTN calls through gateway 30		
Destination				
	Gateway:	Quintum		
	◯ Host:			
	Call next member after 0 📚 seconds			
Digit	Digit manipulation			
	Digits to skip:	1		
		vialed number		
	Prefix to add	Suffix to add		
Optio	ons			
	Transport:	udp 🗸		
	Source pattern:	*		
		OK Cancel Help		

Configuring Toll Restrictions

Configure the toll restrictions to match the requirements of your organization. Consult the *NetVanta Unified Communications Server Administrator Guide*, available online at <u>http://kb.adtran.com</u>, for the correct use of regular expressions in the toll restrictions to enforce corporate dialing policy. It is explained in detail in the "Routing and Restricting Calls > Allowing and Restricting Long Distance and Other Calls > Restricting long distance calls" section.

Glossary of Features

Accept Incoming Calls

This feature allows the gateway to answer an incoming call from the PSTN. The gateway then makes a SIP call to extension 10000.

Accept Outgoing Calls

An outgoing SIP call from the UC server results in an outgoing PSTN call.

Active Message Delivery

The gateway must support the UC server calling out to the PSTN to deliver voice messages.

Answer Supervision

The gateway must detect that a call has been answered. There are a number of techniques used for this, including loop start, battery reversal and voice detection.

Calling Party Name

The gateway detects the calling party name on an incoming PSTN call and provides that name to the UC server.

Conferencing with SIP Endpoints

The gateway needs to support conferencing between itself and other SIP endpoints.

Direct Inward Dialing

Calls incoming from the PSTN must be automatically routed to the UC server for auto attendant functionality.

Disconnect Detection

The gateway must detect that a call has been dropped. There are a number of techniques used for this, including loop start, battery reversal and no voice detection.

DTMF Tone Support (RFC2833 Compliant)

Calls incoming from the PSTN to the UC server are usually handled by an auto attendant. Feature operation is implemented using DTMF tones from telephones. These tones must be sent to the UC server as SIP packets via RFC2833.

Incoming Fax Support

The UC server supports the transmission of faxes to standard fax machines. The gateway must support T.38 fax transport to provide this capability. Additionally, the gateway should support CNG tones so that an incoming PSTN fax call can be distinguished from a voice call and handled appropriately.

Multiple SIP Proxy Support

In high reliability applications, if the main UC server is not available the gateway routes incoming PSTN calls to an alternative SIP Proxy.

Outgoing Fax Support

The UC server supports the transmission of faxes to standard fax machines. The gateway must support T.38 fax transport to provide this capability.

Paging Notification

The gateway must support the UC server calling out to the PSTN to deliver pages.

System Music on Hold Support

The UC server supports music on hold. When PSTN callers are on hold they hear music, if that feature is enabled on the system.

Transfer—Assisted/Supervised

After a call is established between an outside PSTN call and an internal SIP device, the gateway must allow a supervised transfer to another SIP device.

Transfer—Blind

After a call is established between an outside PSTN call and an internal SIP device, the gateway must allow a blind transfer to another SIP device.

Trunk-to-trunk connect

This feature allows an established call through the gateway, which can be extended back out the gateway on another PSTN trunk.