NetVanta Unified Communications Technical Note

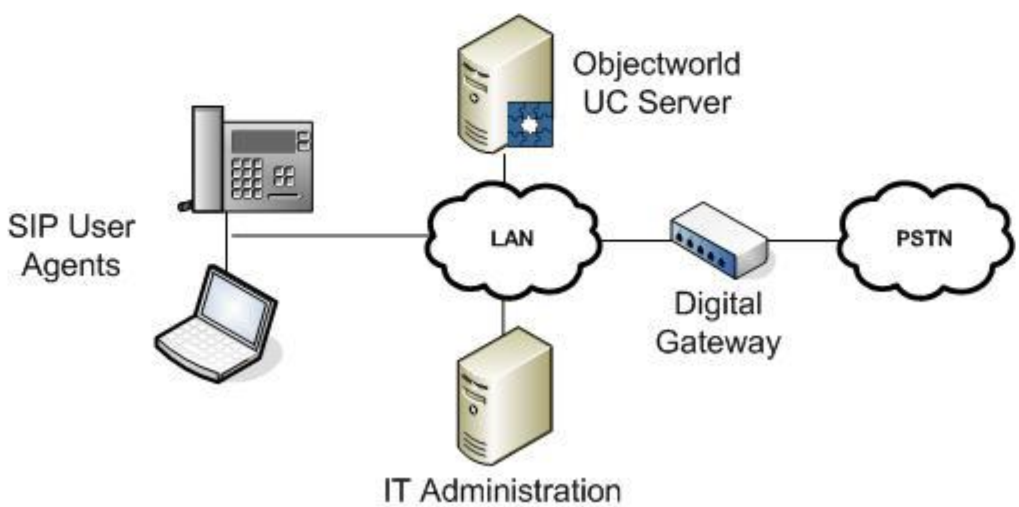
Installing and Configuring the Quintum Tenor DX Gateway

Introduction

The **Tenor DX** is a modular T1 digital 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 digital TDM voice, which are 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 SIP architecture and its relationship with other components in a typical customer network.



Supported Features

Feature Name	Supported	Notes
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	✓	
Outgoing Caller ID Creation	✓	
Unified Communication Features Supported by Gateway		
Active Message Delivery	✓	
Paging Notification	✓	
Transfer—Assisted/Supervised	✓	
Transfer—Blind	✓	
Unified Communication Features Supported by Gateway		
Multiple SIP Proxy Support	✓	Available with survivability option

Interoperability Software Versions

The following gateway version was tested for interoperability:

- **System Description**: Quintum Tenor DX
- **Boot Version**: P106-02-00
- **Firmware Version**: P106-12-00

Overview of Configuration Procedure

To provide its functionality the Tenor DX must be connected to the internal LAN (a 100 Mbps connection is recommended) and a T1/ISDN digital line.

The **Quintum Tenor DX** 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

1. Unpack the **Tenor DX**.
2. Mount the **Tenor DX**.
3. Connect cables.
4. Set a DHCP IP address reservation for the **Tenor DX** based on its MAC address.
5. Connect to console port and set IP address.
6. Run the initial configuration wizard.
7. Configure the UC server to use the **Tenor DX**.

NOTE: *Please see the instructions provided by Quintum for information about running and configuring the gateway*.

The rest of this document provides instructions about configuring the **Tenor DX** for operation with the UC server.

Address Reservation

The **Tenor DX** can be configured either with a reserved IP address from DHCP or a static IP address. For routing calls out from the UC server, the **Tenor DX** must have an IP address that does not change.

Set IP Address and Subnet Mask

Connect a serial cable to the RS-232 port of the **Tenor DX**. Configure a terminal program, such as HyperTerminal, with the following: 38,400 bits per second, 8 data bits, 1 stop bit, No parity, No flow control.

A login prompt appears. Log in using username: **admin**, and password: **admin**.

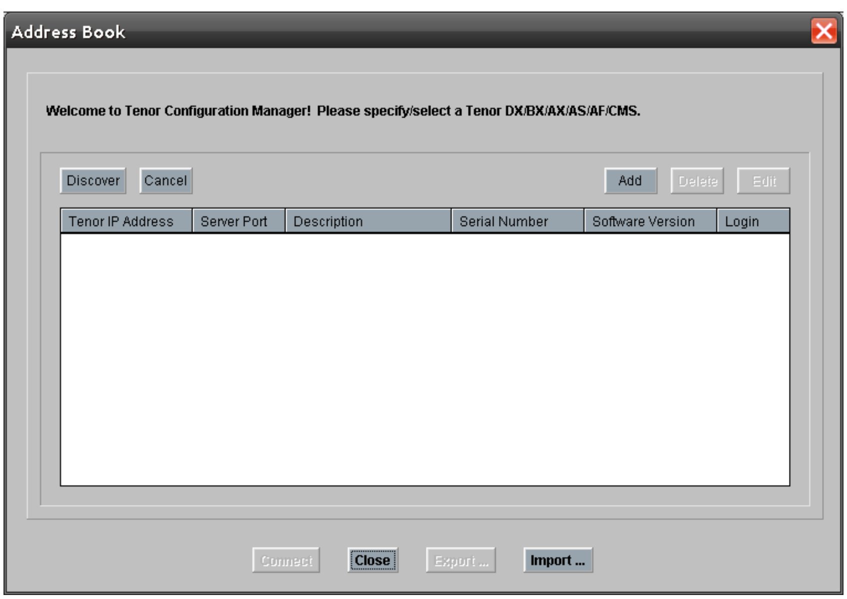
It is important that the **Tenor DX** have a LAN IP address that does change. This can be set using a static IP address and subnet mask compatible with the onsite LAN. To configure the IP address, do the following:

- At the Quintum prompt, enter **ei** to reach the Ethernet prompt, and then enter **config** to change to the configuration mode.
- To set the IP address, enter **set ipa** followed by the IP address.
- To set the Subnet Mask, enter **set subnetmask**, followed by the subnet mask.
- Enter **siprd** to change to the Static IP Route Directory.
- To set the Default Gateway IP, enter **change 1 g** followed by the IP address for the default gateway IP.
- Enter **submit**.
- Enter **maint** to reach the maintenance mode and then **mc**. Enter **reset**.
- A confirmation message asks if you want to reset the unit. Enter **yes** to reset the unit and incorporate the new settings.

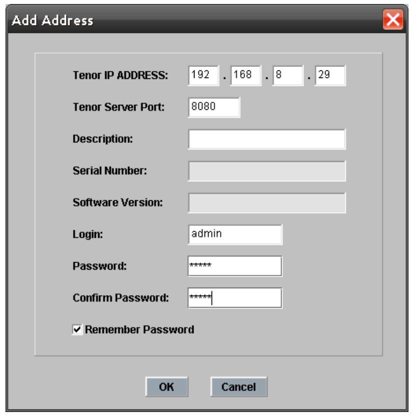
Initial Configuration Wizard

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 (*http://www.quintum.com/support*)

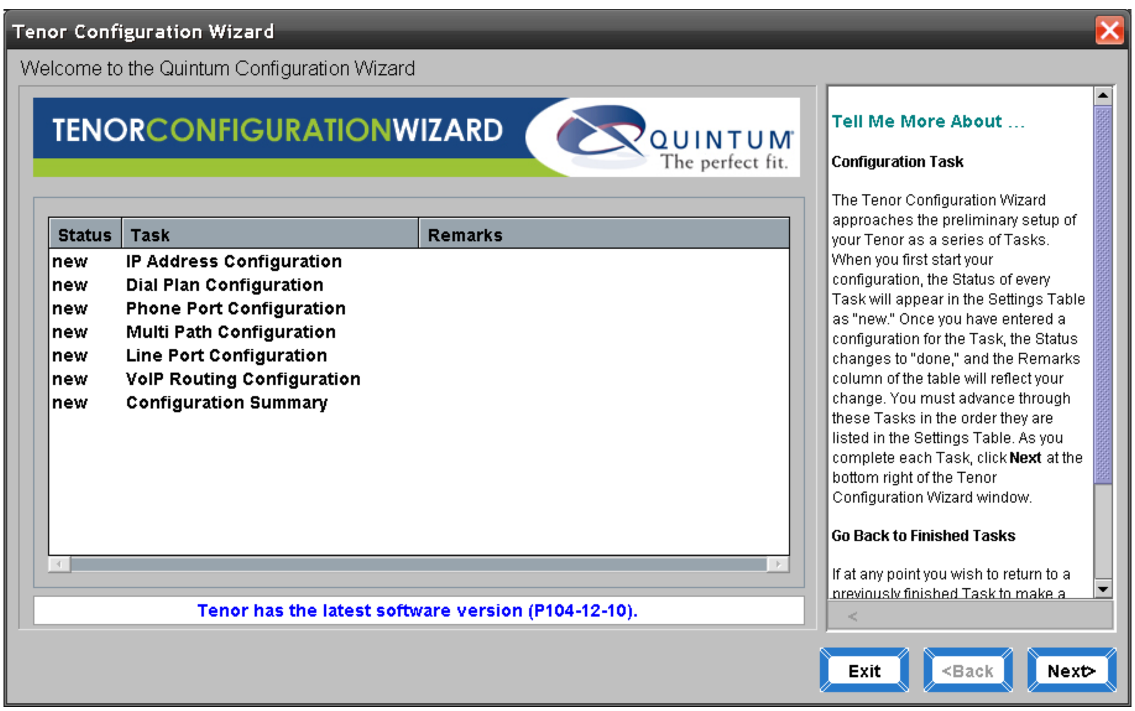
After you install and run the Tenor Configuration Manager, the following screen appears.



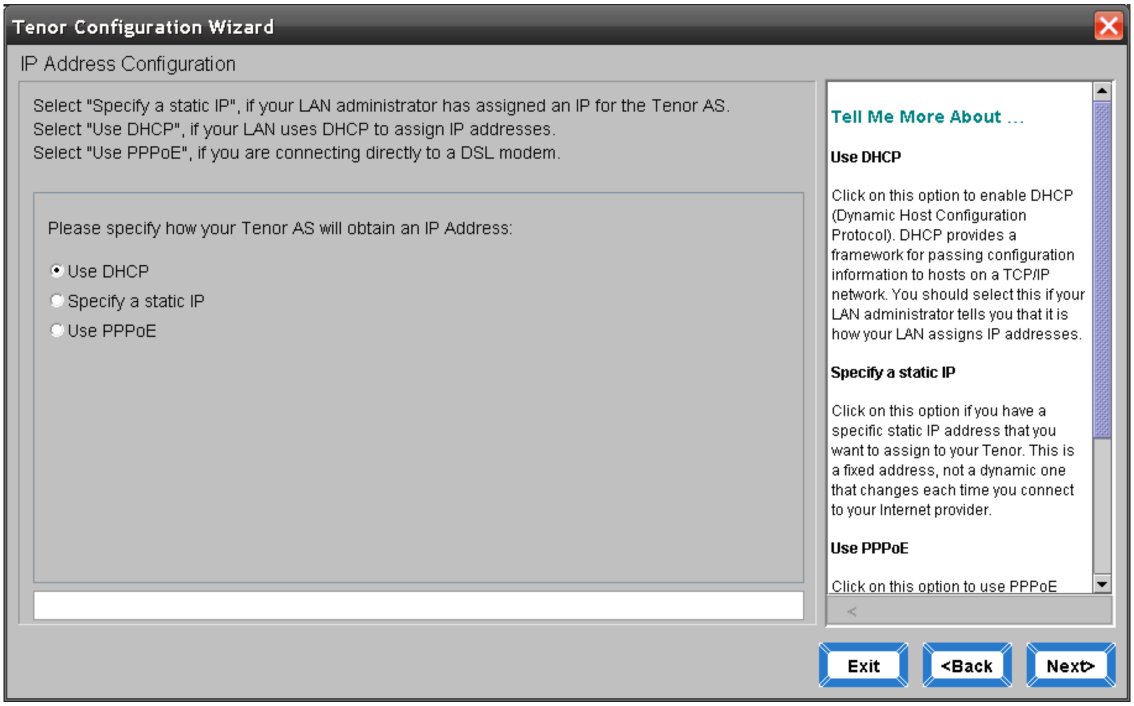
1. In most cases, by selecting **Discover** the program automatically detects the gateway. Select **Connect**. If the program does not detect the gateway, you must select **Add** and manually enter the IP address of the Quintum gateway.
2. After you select **Add** the following screen appears. Enter the IP address of the gateway. The login and password is **admin**. After you add an address, select **OK** and then select **Connect**.



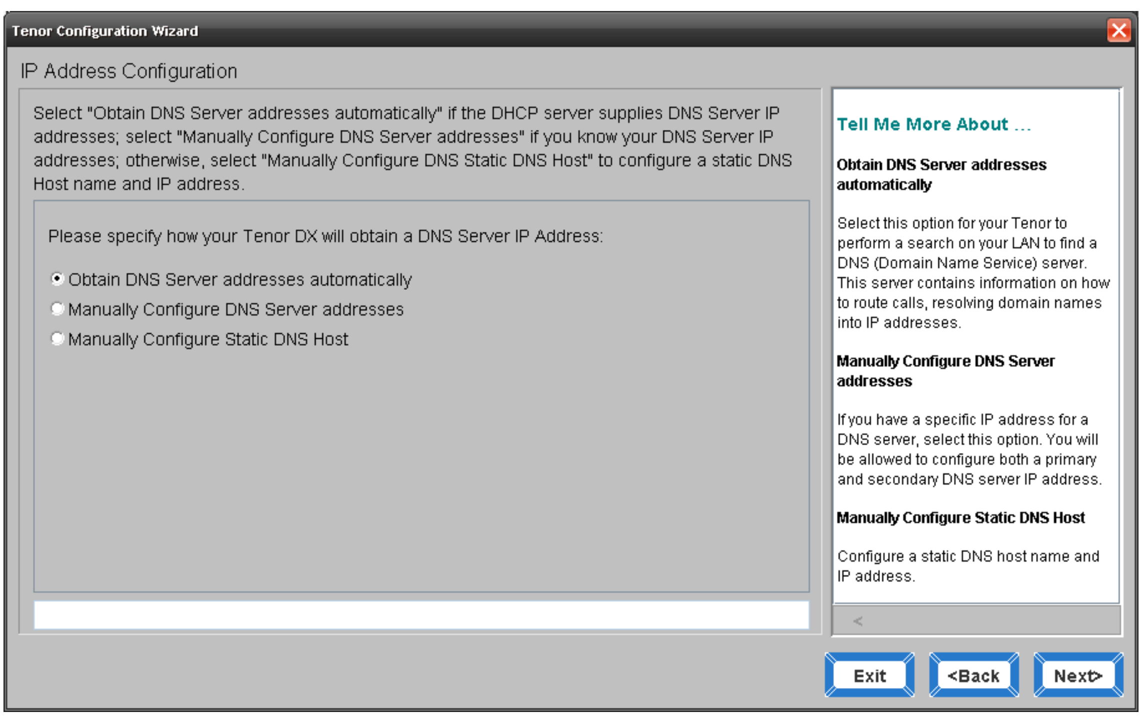
3. After you connect, a wizard opens to set up the initial configuration of the gateway. Select **Next**.



4. On the IP Address Configuration screen, you can choose how your gateway obtains its IP Address and network settings. A static IP address is recommended for a gateway. Select **Next>** to continue.

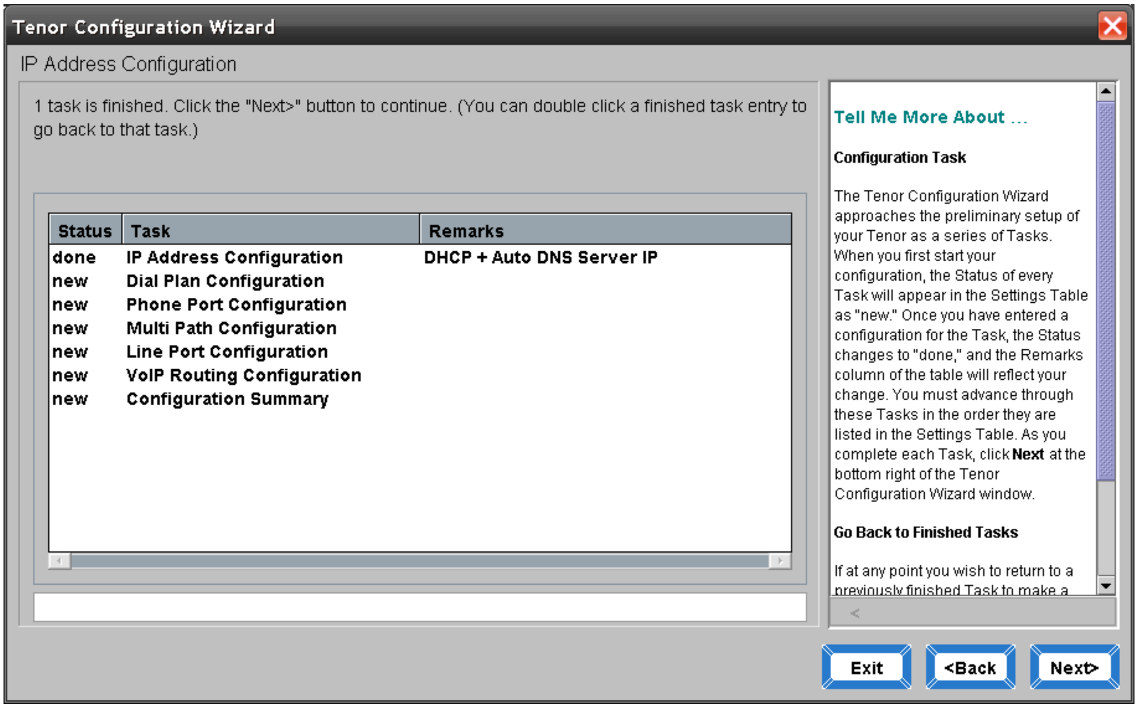


5. You can specify whether you want to obtain DNS server addresses automatically, manually configure DNS Server addresses, or Manually Configure Static DNS Host. If you are using DHCP, you can automatically obtain the DNS server addresses; otherwise you must manually configure them. Select **Next>** to continue.

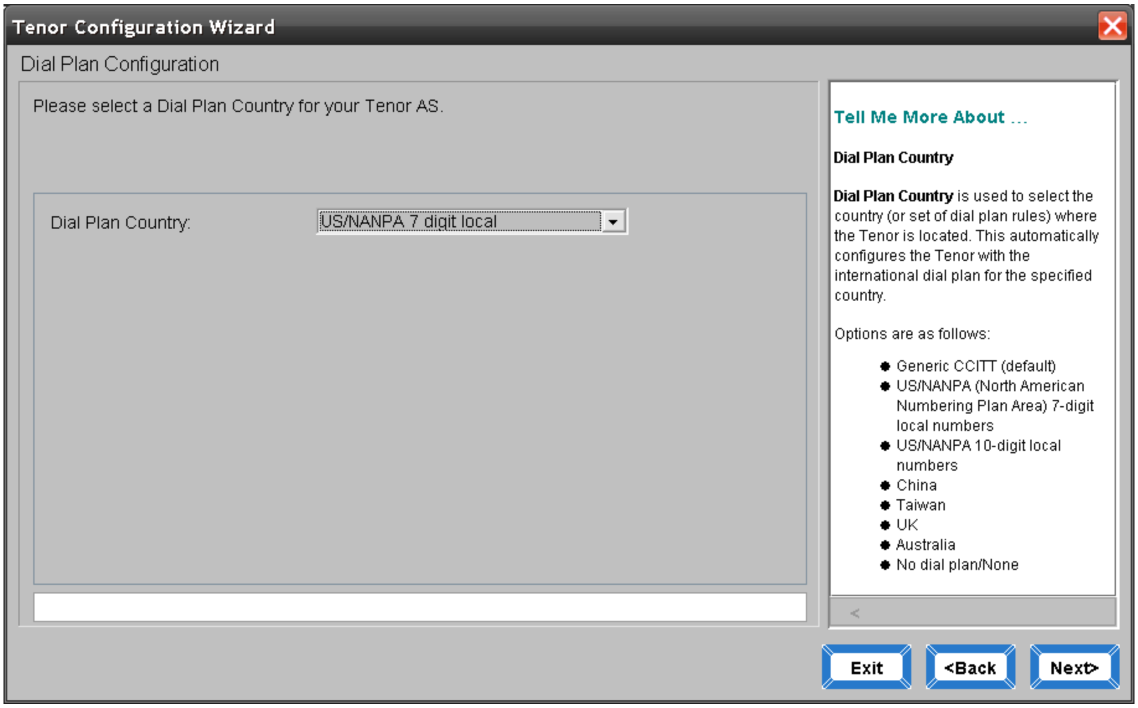


6. The first task is complete. Select **Next>** to continue.

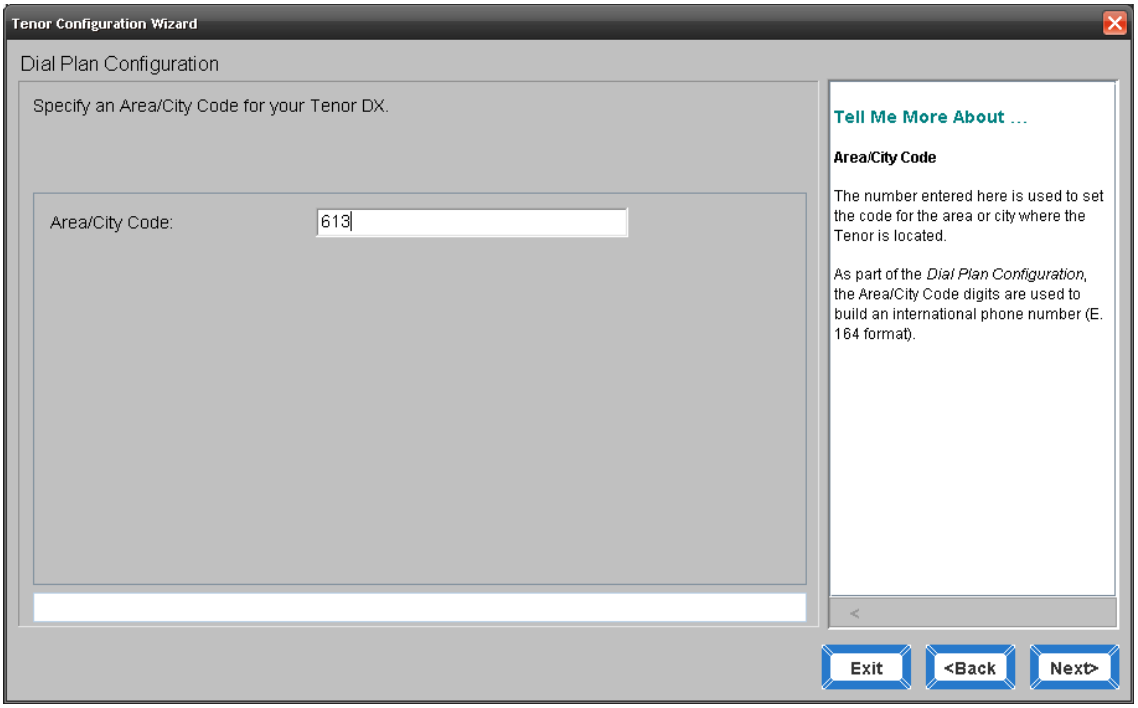
NOTE: *If you want to change any of the configuration parameters, you can navigate back through the wizard or make changes after the initial configuration wizard.*



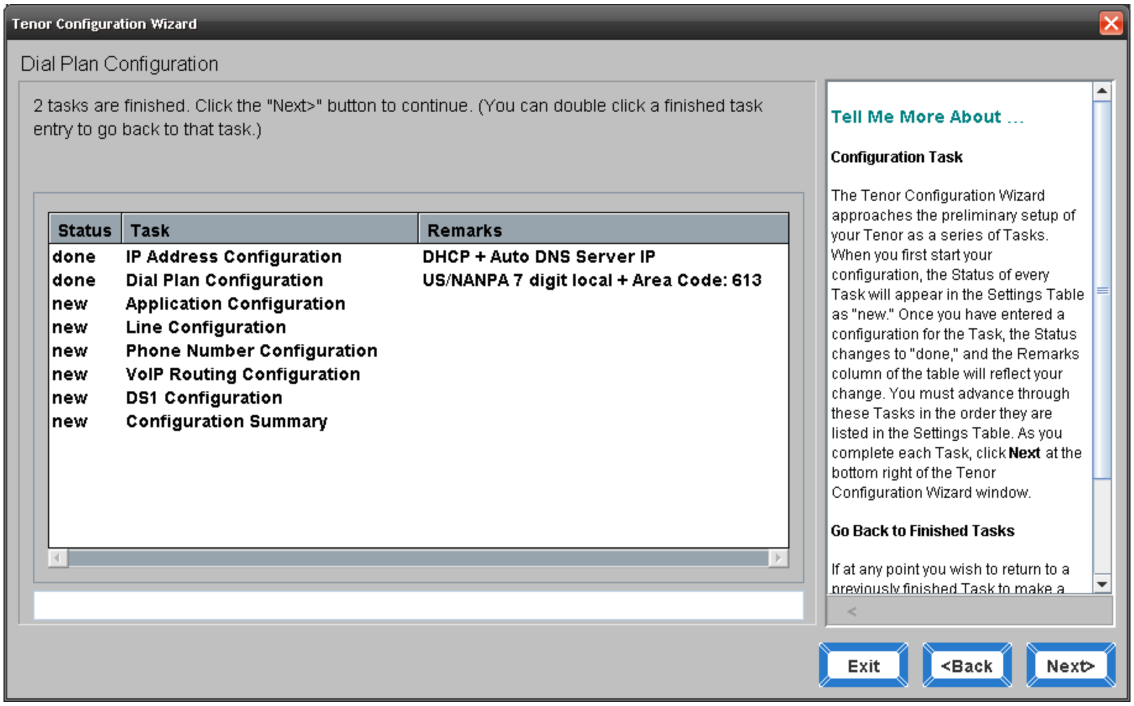
7. The Dial Plan Configuration screen allows you to set up the dialing plan. Depending on your region, you can choose the dial plan suitable for your country. If you are in **North America**, choose either **US/NANPA 7 digit local** or **US/NANPA 10 digit local**. Select **Next>** to continue.



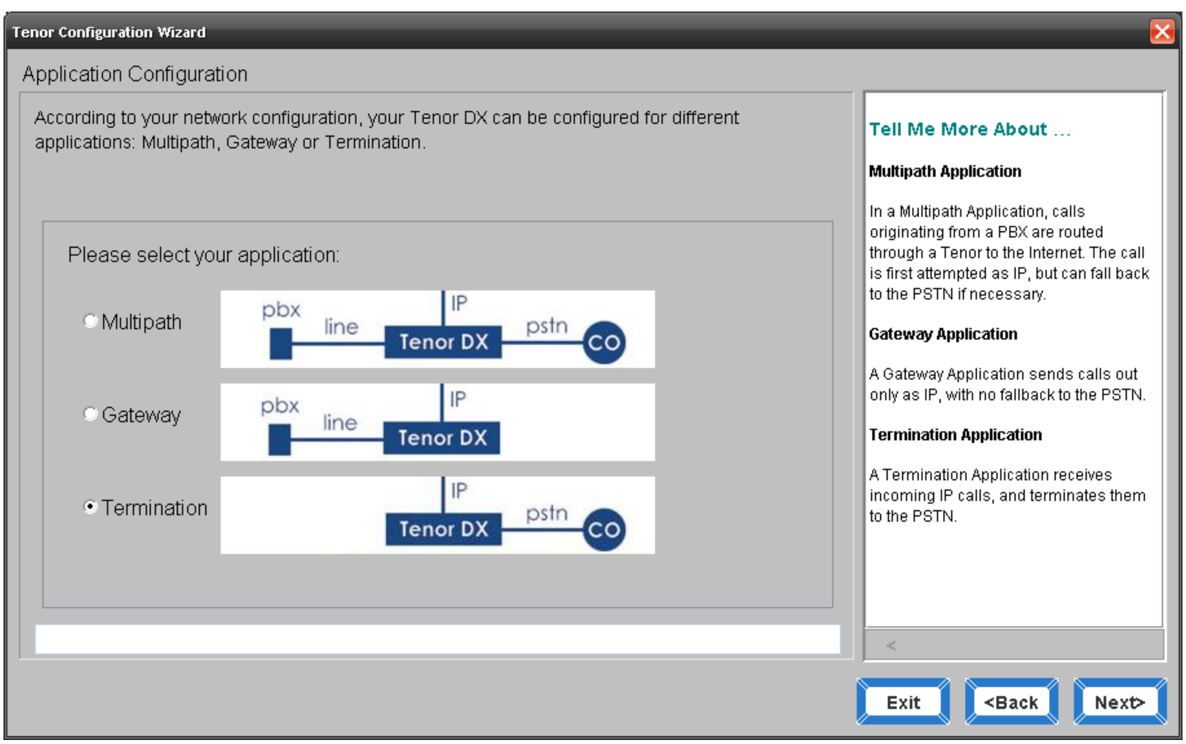
8. On the next Dial Plan Configuration screen, enter your local area code, which the Tenor uses to correctly route calls through the PSTN. Select **Next>** to continue.



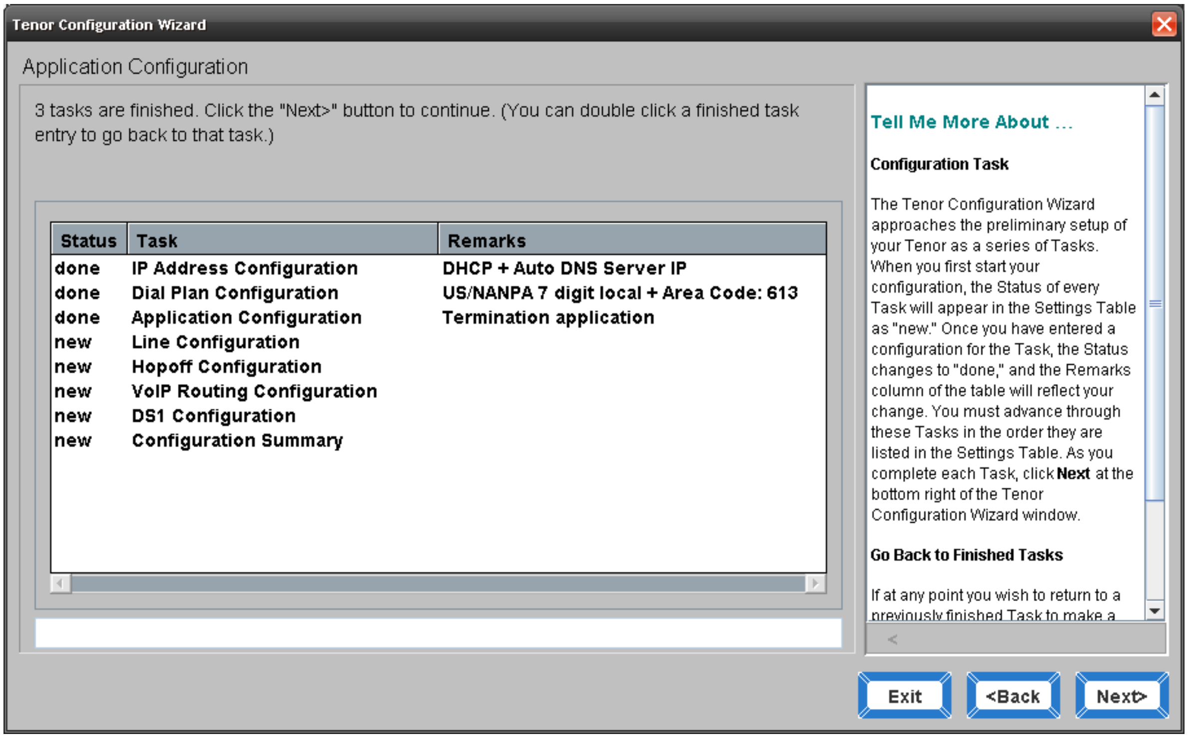
9. You have completed the dial plan configuration. Select **Next>** to continue to the next step.



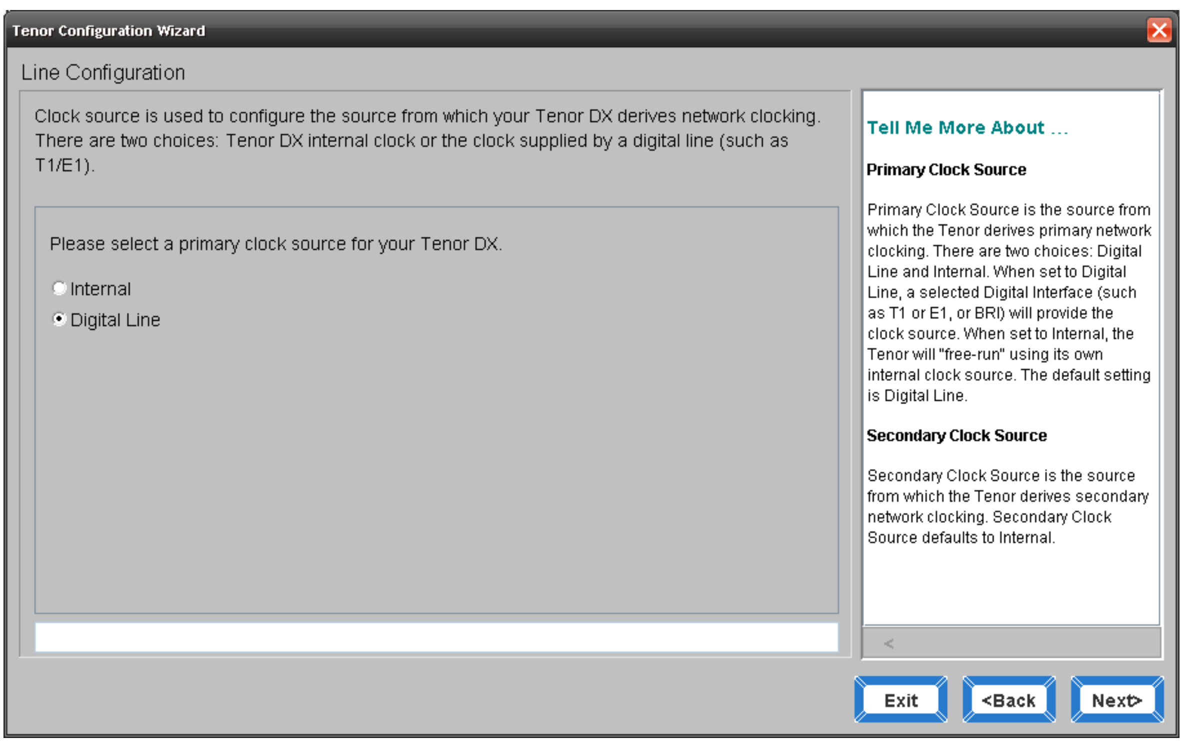
10. You must now choose the application configuration, which specifies the gateway functions. To make and receive calls through the PSTN, choose **Termination**. Select **Next>** to continue.



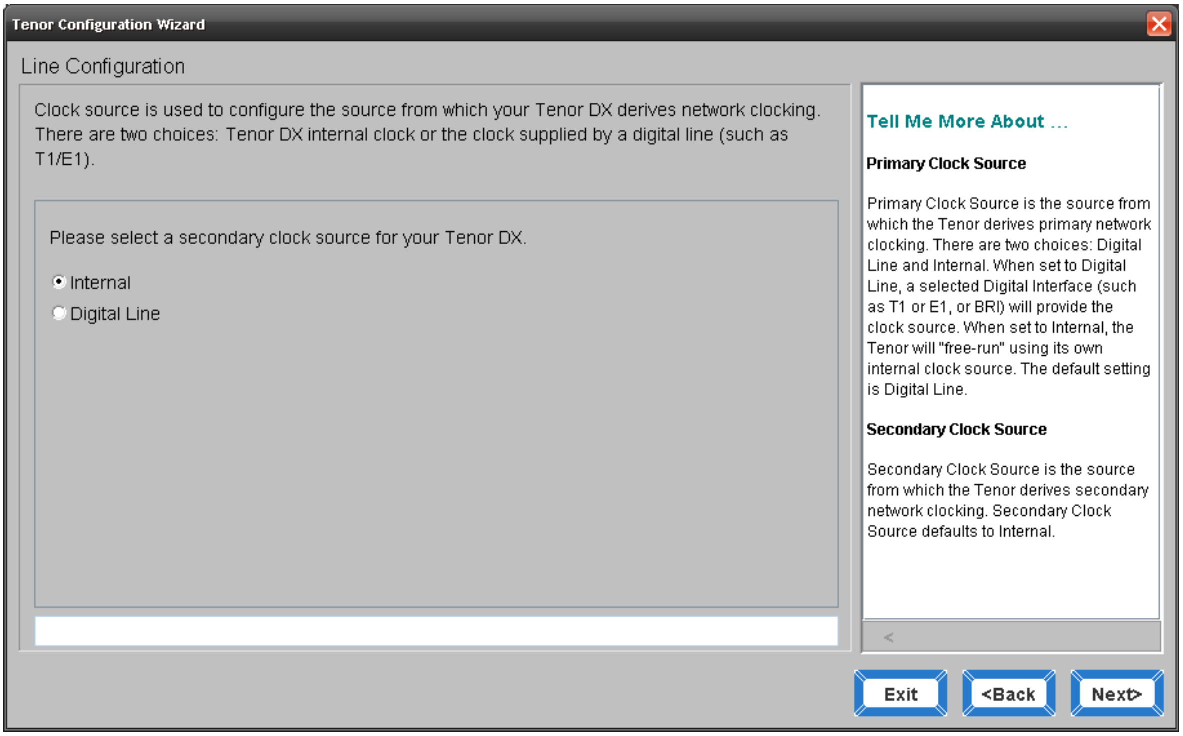
11. The next screen allows you to specify the Line Configuration. Select **Next>** to continue.



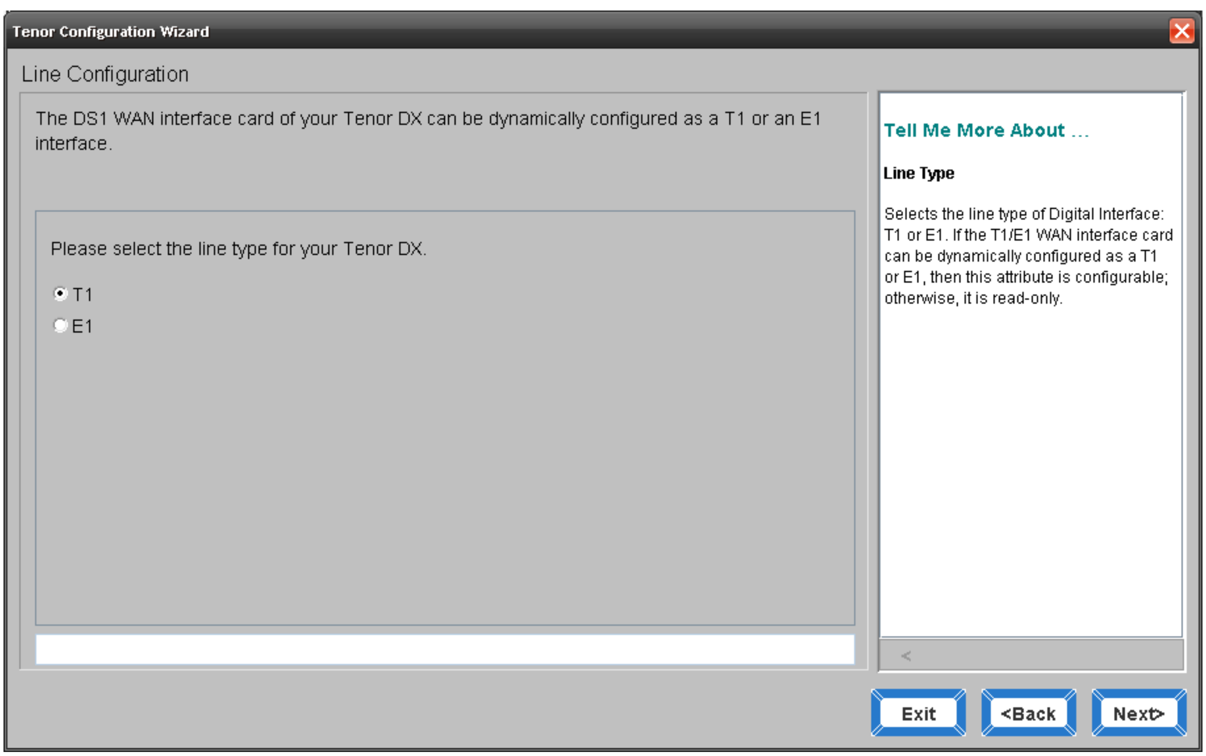
12. On the next Line Configuration screen, choose the primary clock source for the PRI link. A typical configuration requires that you choose **Digital Line**. Select **Next>** to continue.



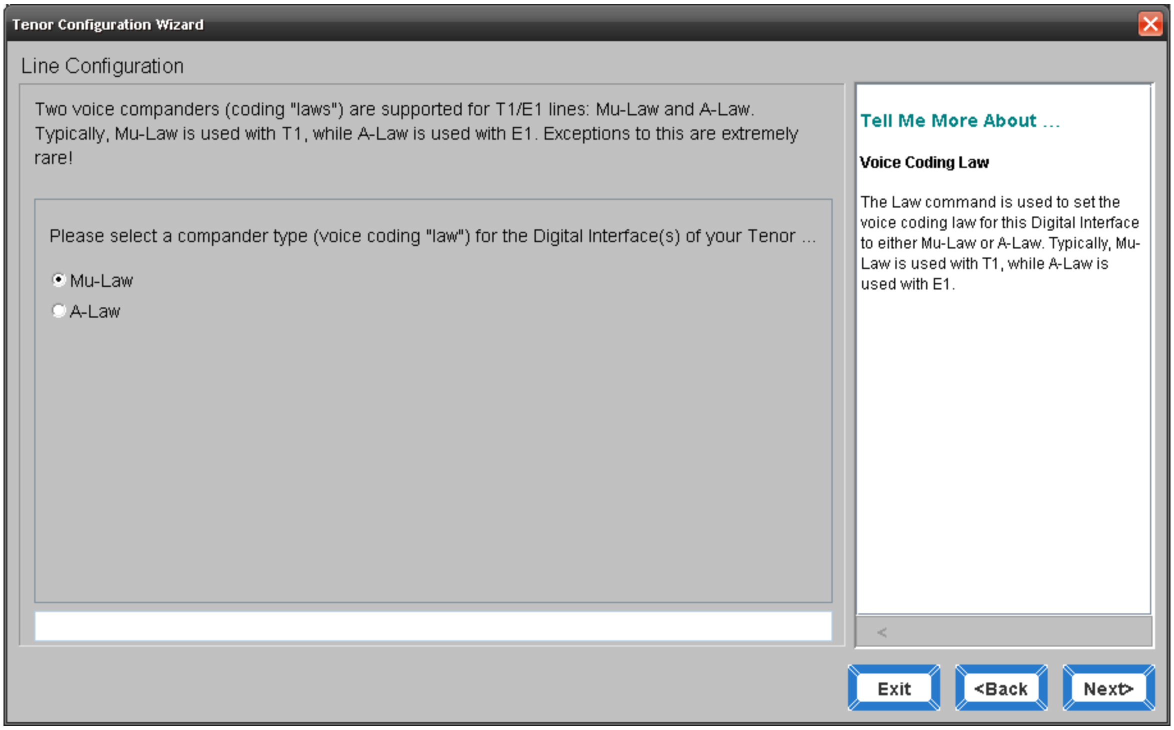
13. On the next Line Configuration screen, choose the secondary clock source for the PRI link. A typical configuration requires that you choose **Internal**. Select **Next>** to continue.



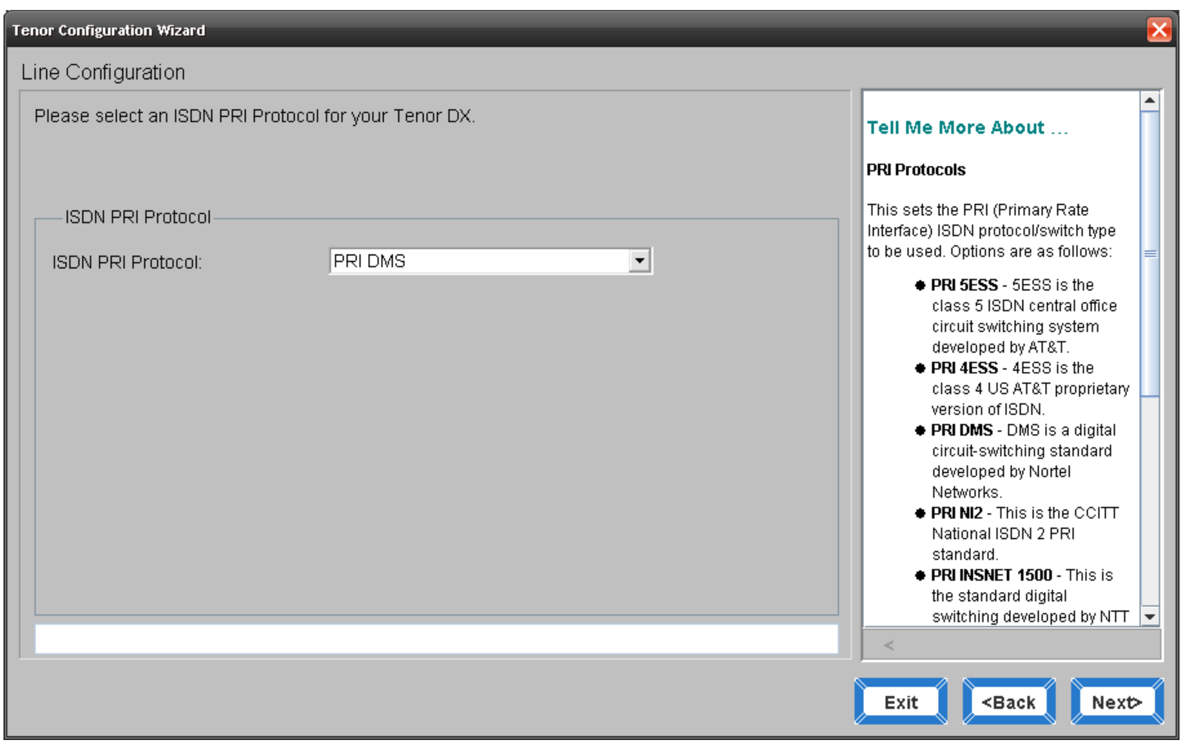
14. On the next Line Configuration screen, choose the line type for the PRI link. Select **Next>** to continue.



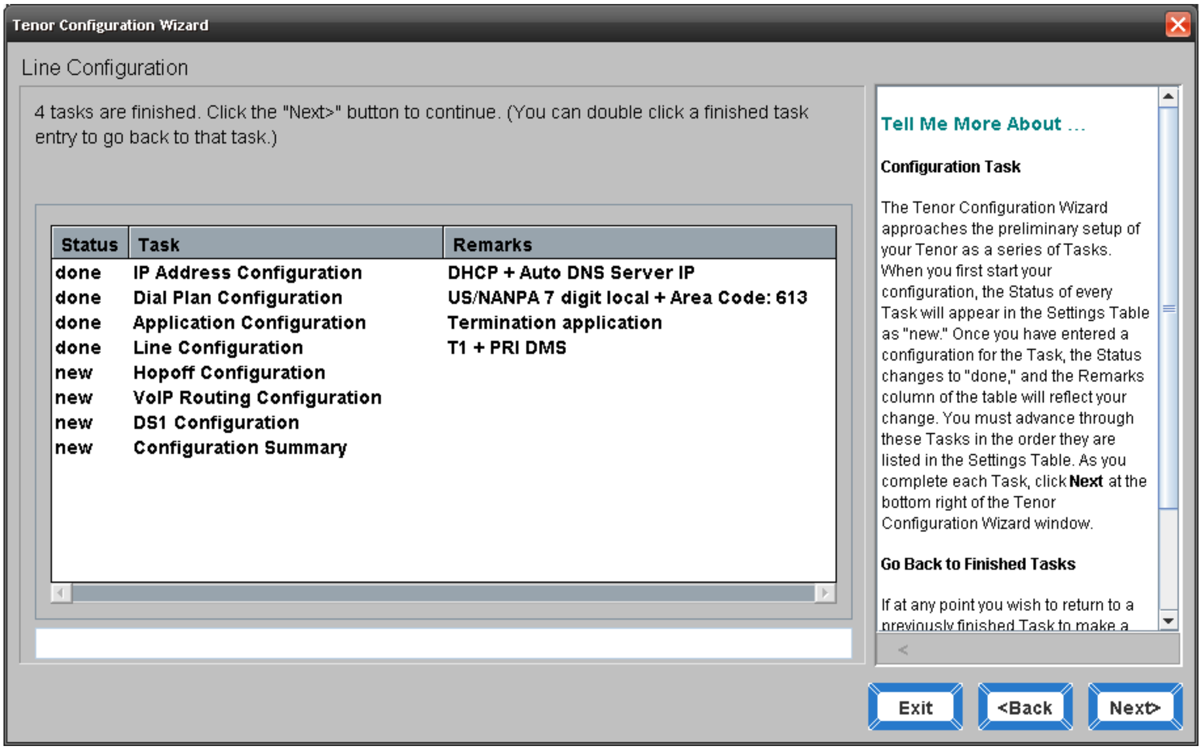
15. On the next Line Configuration screen, choose the voice coding law for the digital link. Typically, **Mu-Law** is used for **T1 links** and **A-Law** is used for **E1 links**. Select **Next>** to continue.



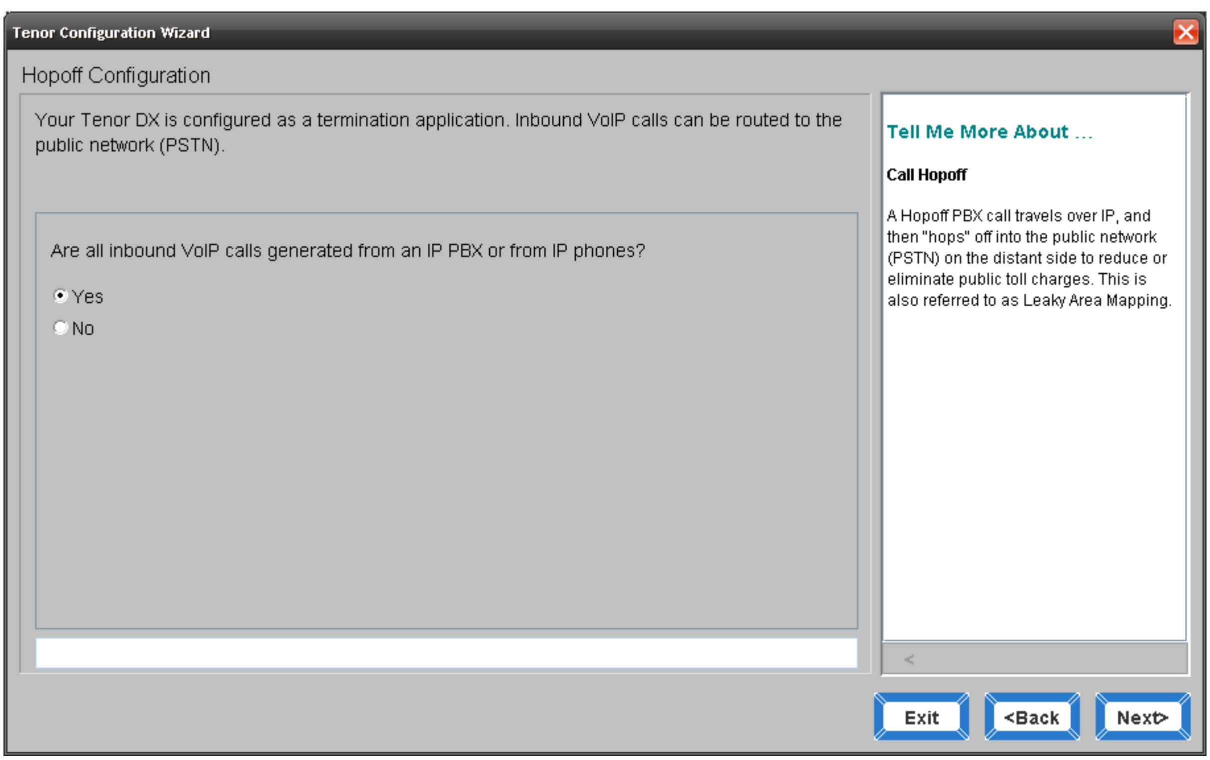
16. On the next Line Configuration screen, choose the PRI protocol for the gateway. This setting varies depending on the configuration of the link. Select **Next>** to continue.



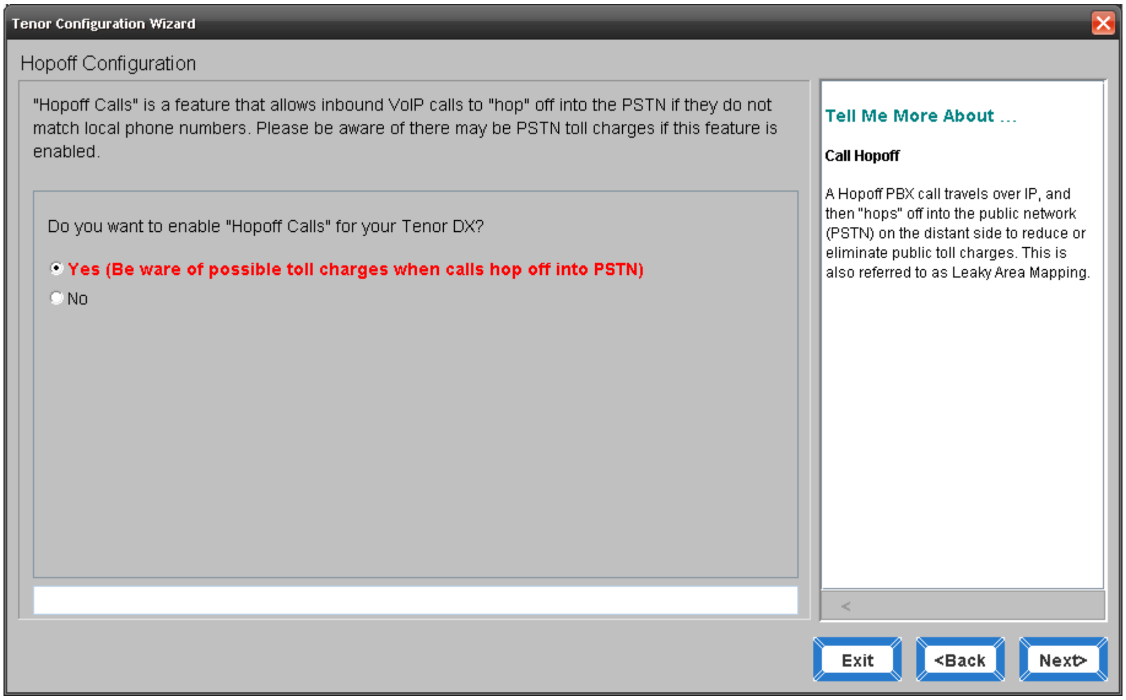
17. Review the Line configuration summary screen. Select **Next>** to continue.



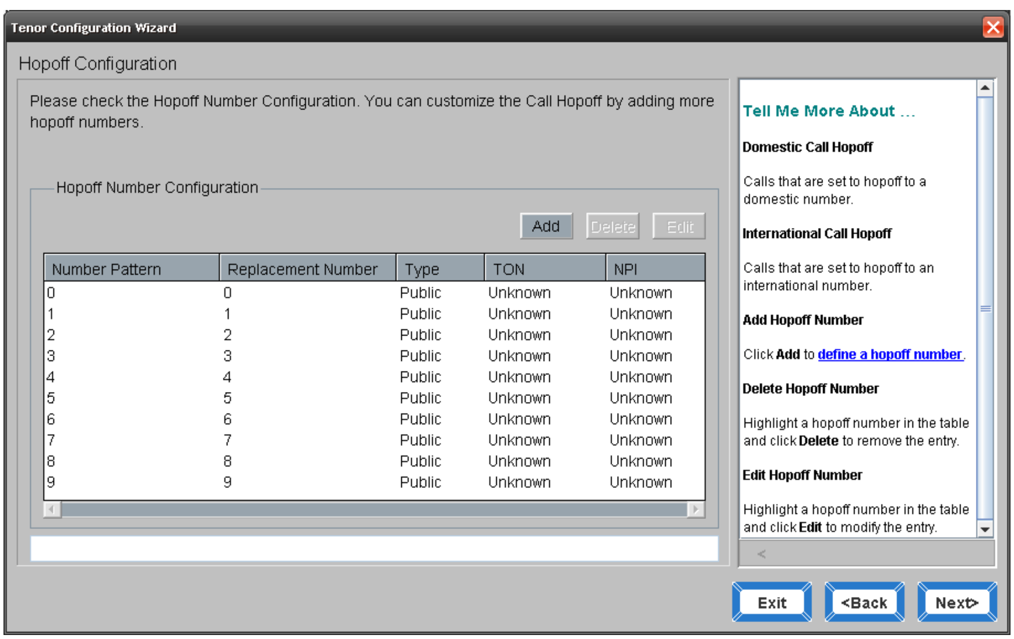
18. On the Hopoff Configuration screen, choose **Yes** to make sure all the calls go through the UC server. Select **Next>** to continue.



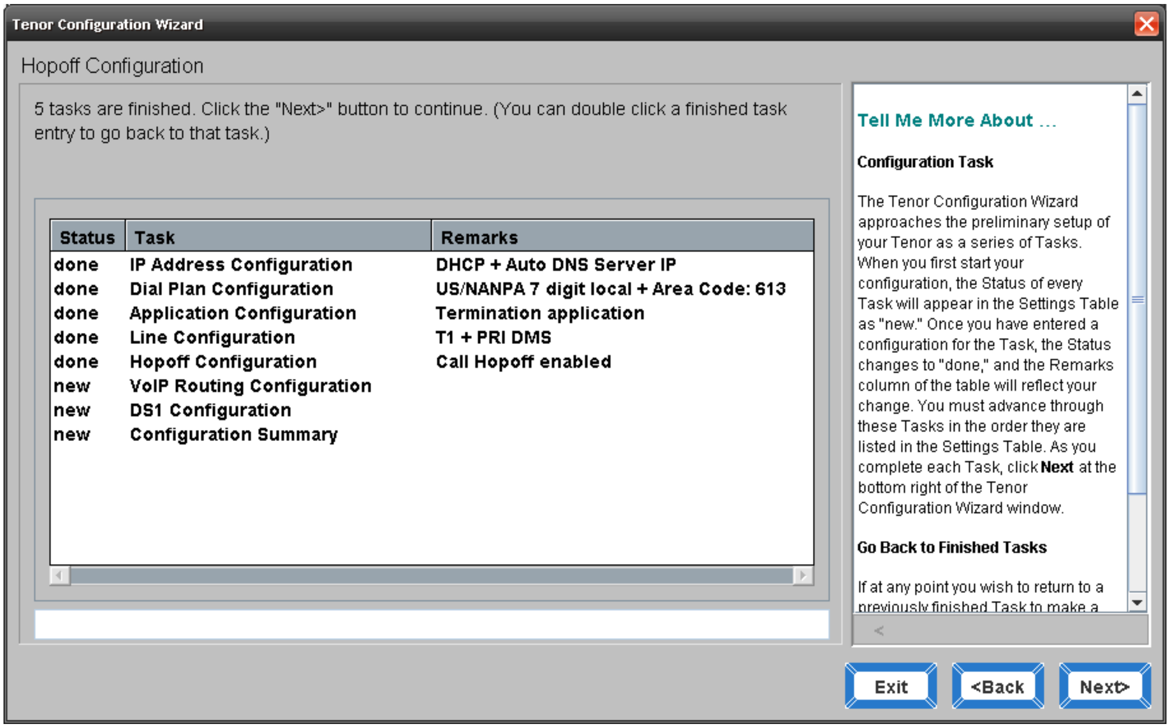
19. On the next Hopoff Configuration screen, choose **Yes** to enable Hopoff Calls. Select **Next>** to continue.



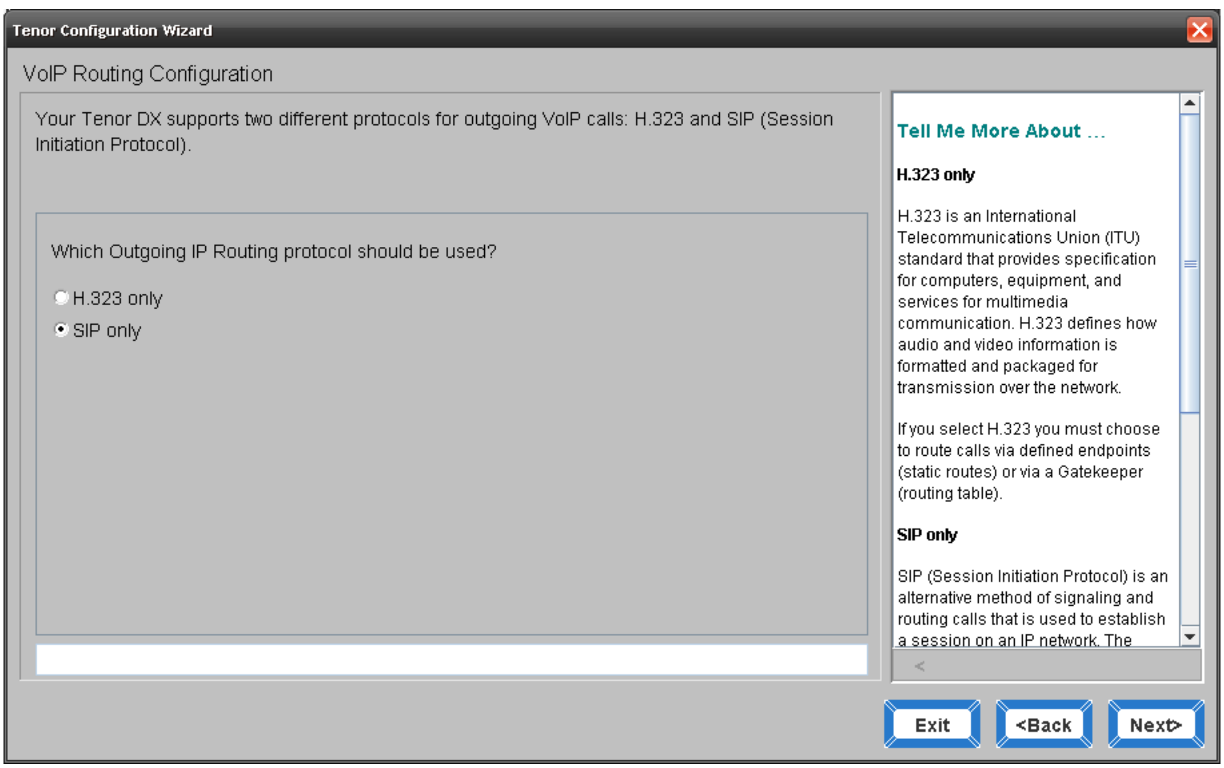
20. On the next Hopoff Configuration screen, configure the hopoff numbers that you want. You can set up numbers that are allowed go through to the PSTN line. Configure these numbers based on your location and requirements. A hopoff number would usually contain the first few digits of a PSTN number based 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 then you would use **011** as the **number pattern** and **replacement number.**



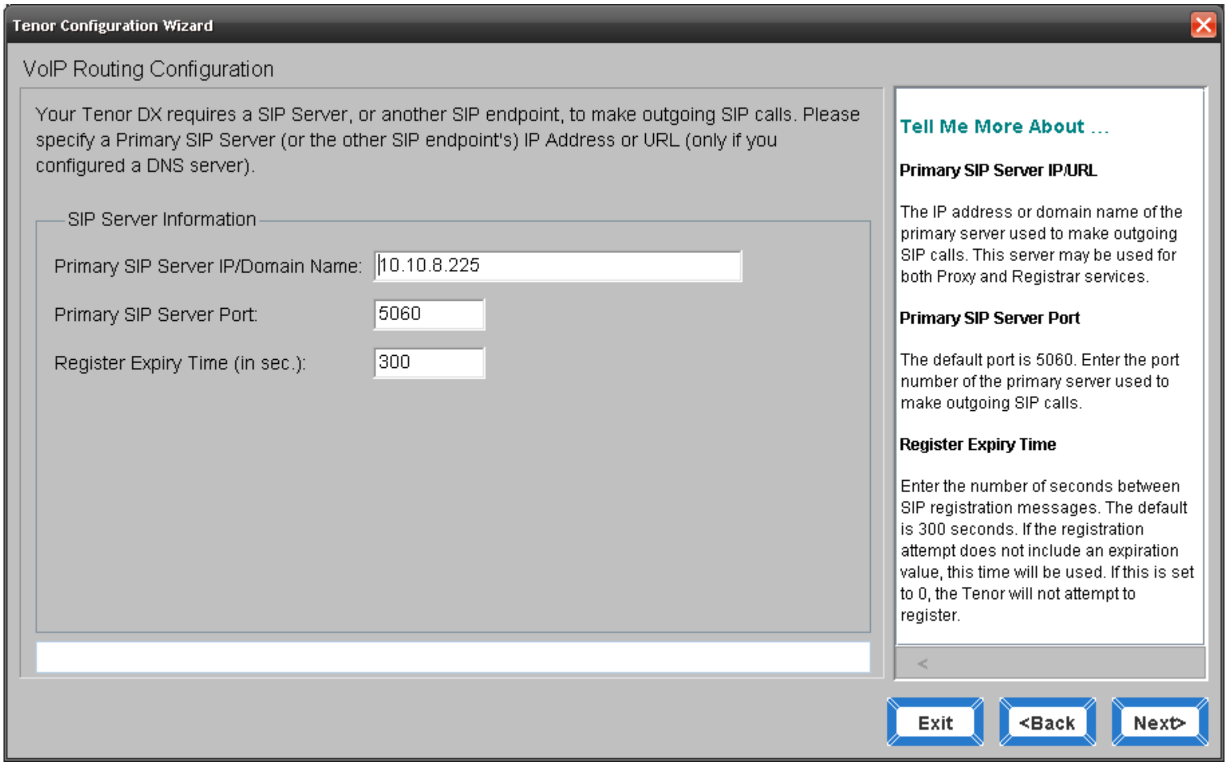
21. Review the Hopoff Configuration summary screen. Select **Next>** to continue.



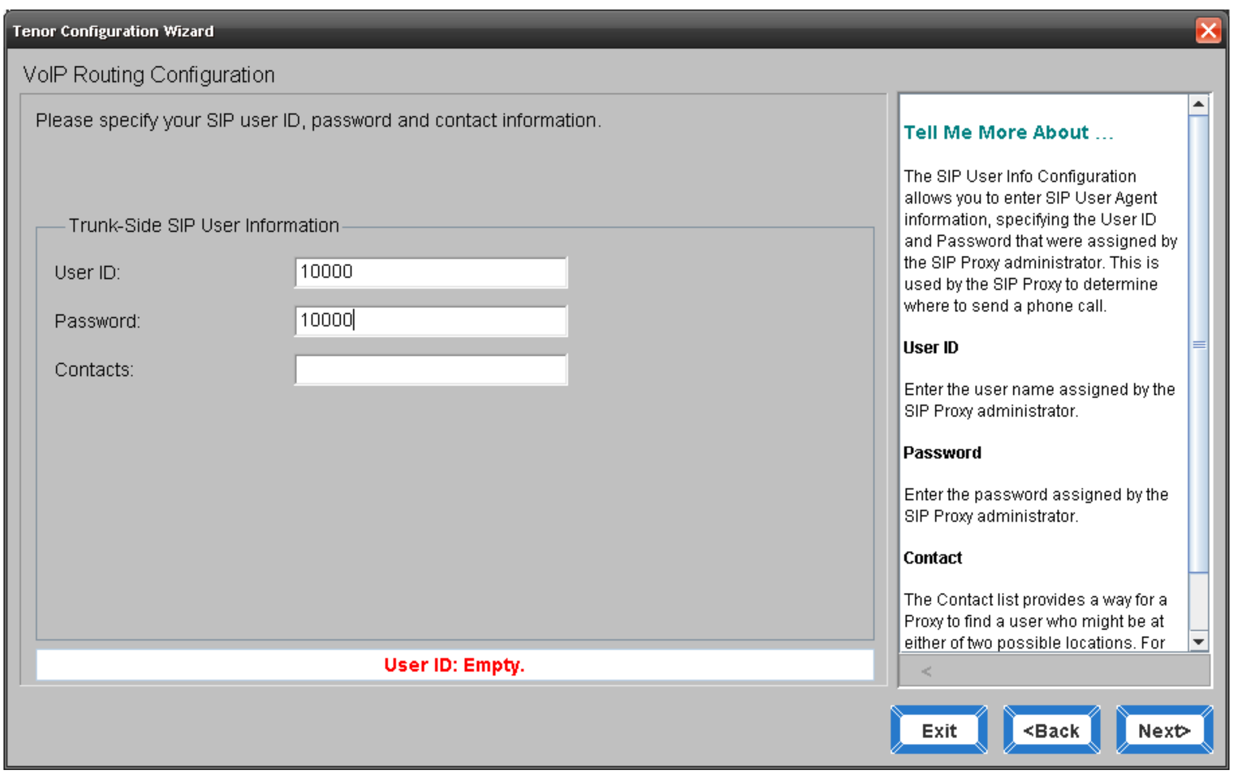
22. On the VoIP Routing Configuration screen, for integration with the UC server, choose **SIP only**. Select **Next>** to continue.



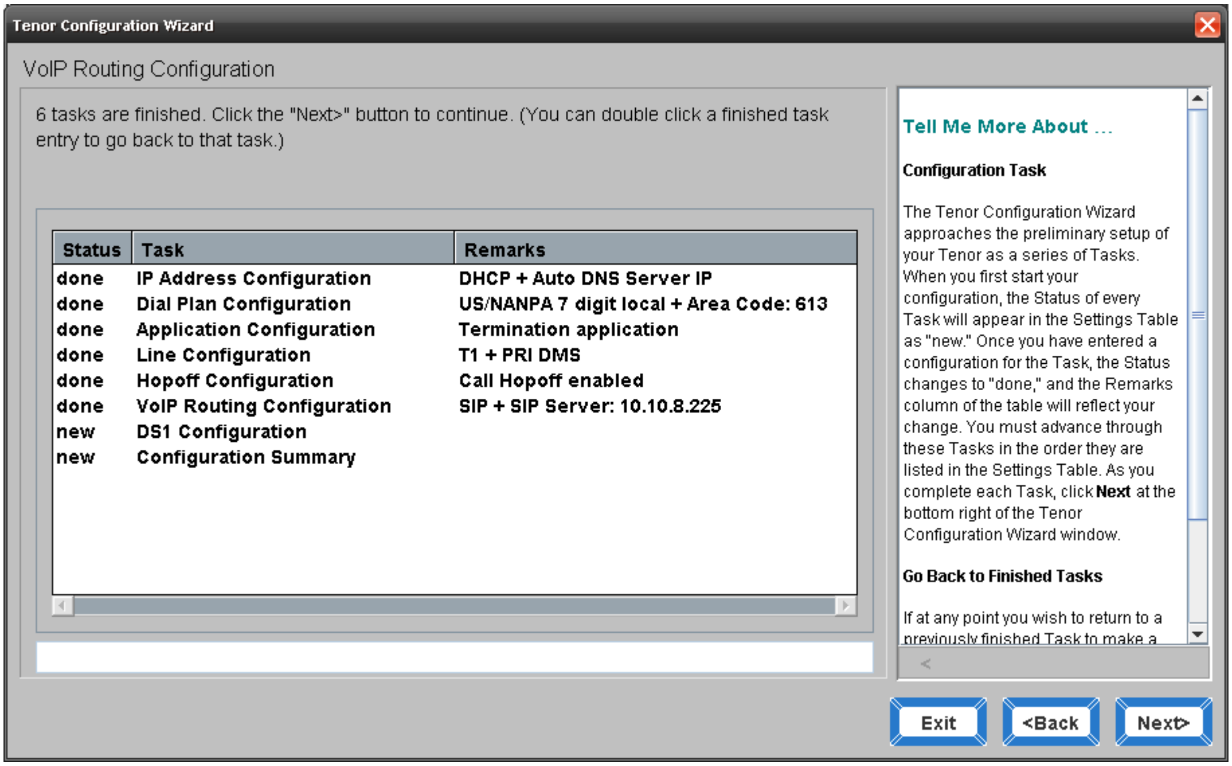
23. On the next VoIP Routing Configuration screen, change the Primary SIP Server to the IP address where your UC server is hosted. Select **Next>** to continue.



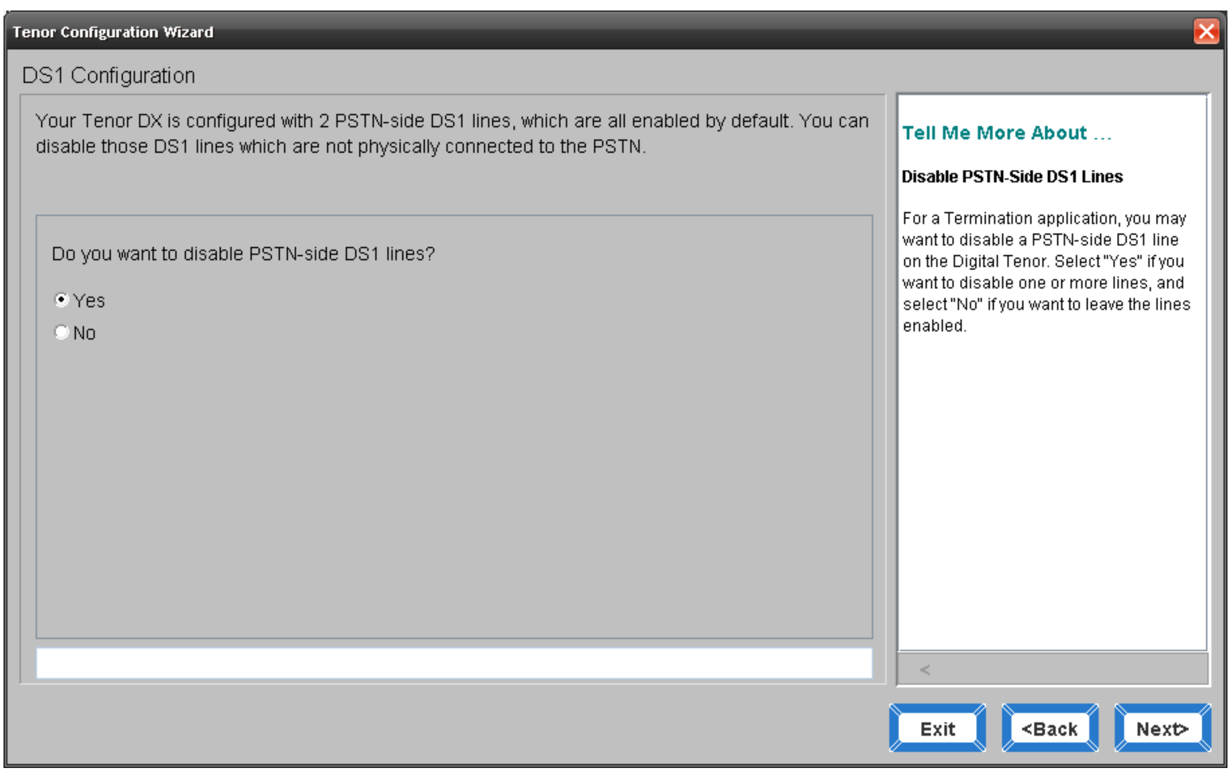
24. On the VoIP Routing Configuration screen, enter **10000** for **User ID** and **Password**. Select **Next>** to continue.



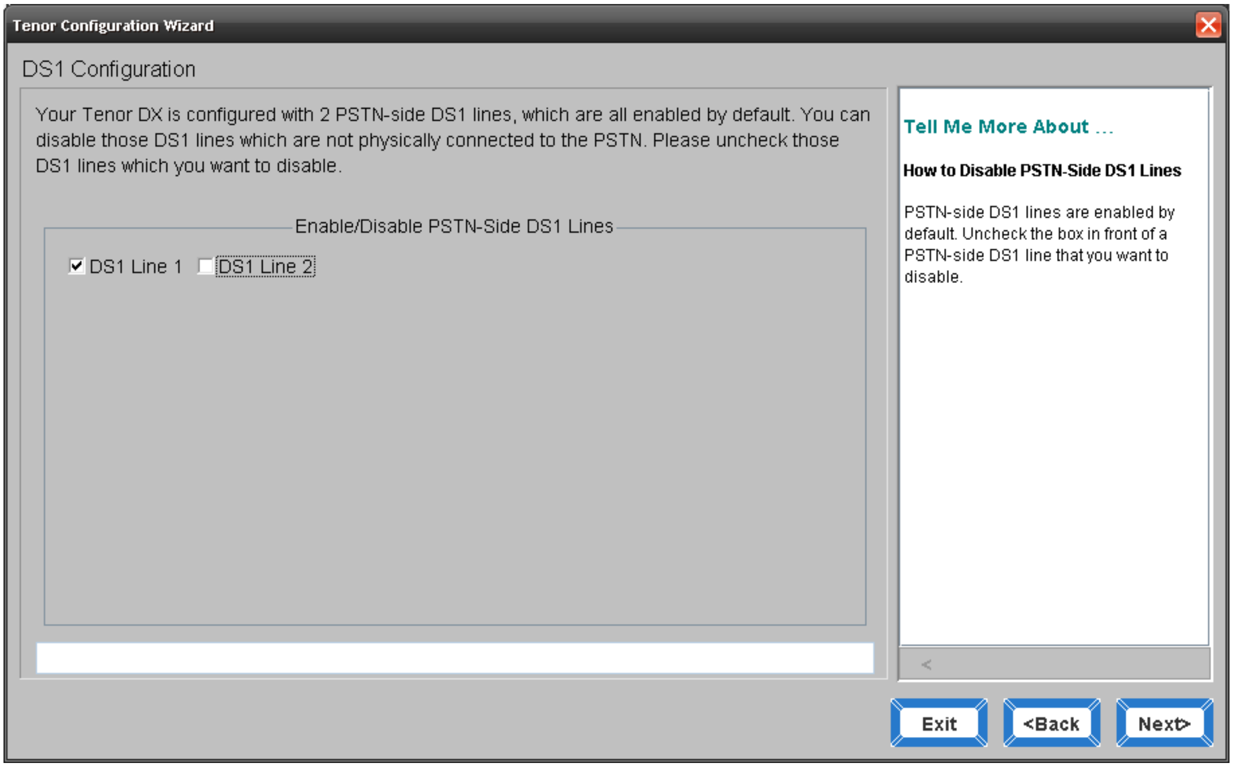
25. Review the VoIP Routing Configuration summary screen. Select **Next>** to continue.



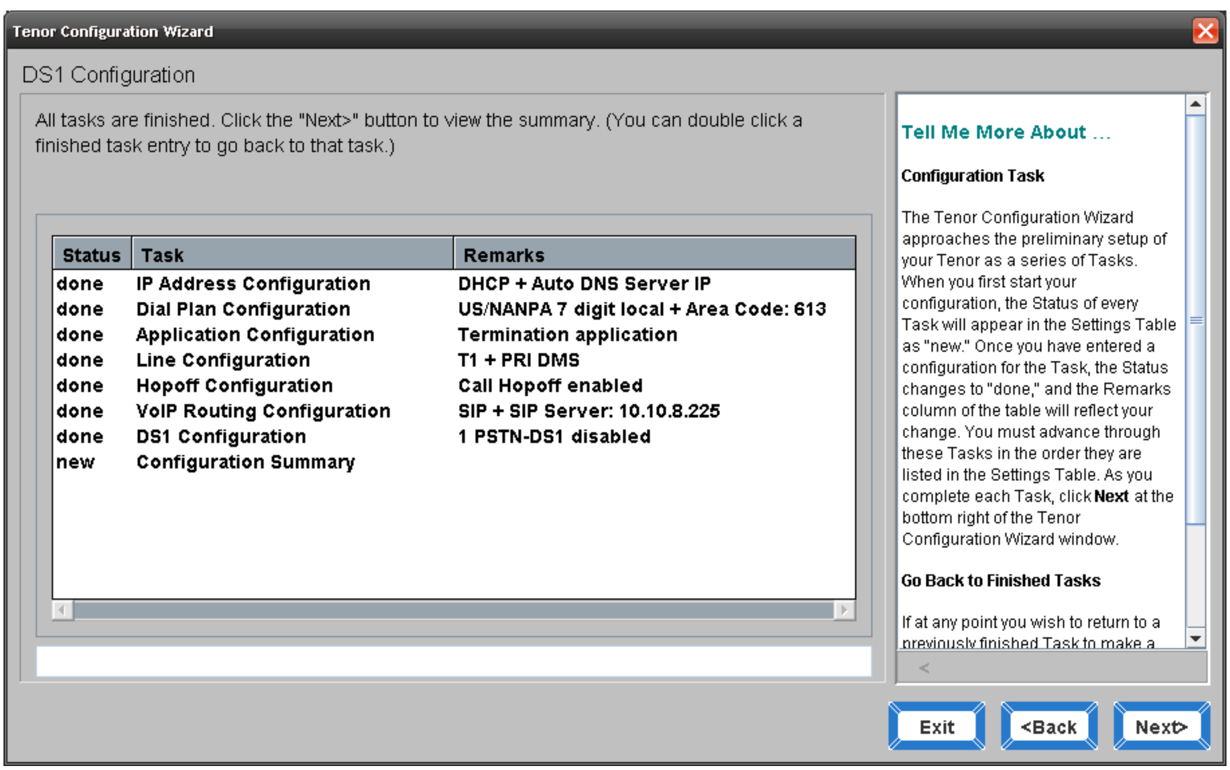
26. On the DS1 Configuration screen, if you only have one PRI link, choose **Yes**. Select **Next>** to continue.



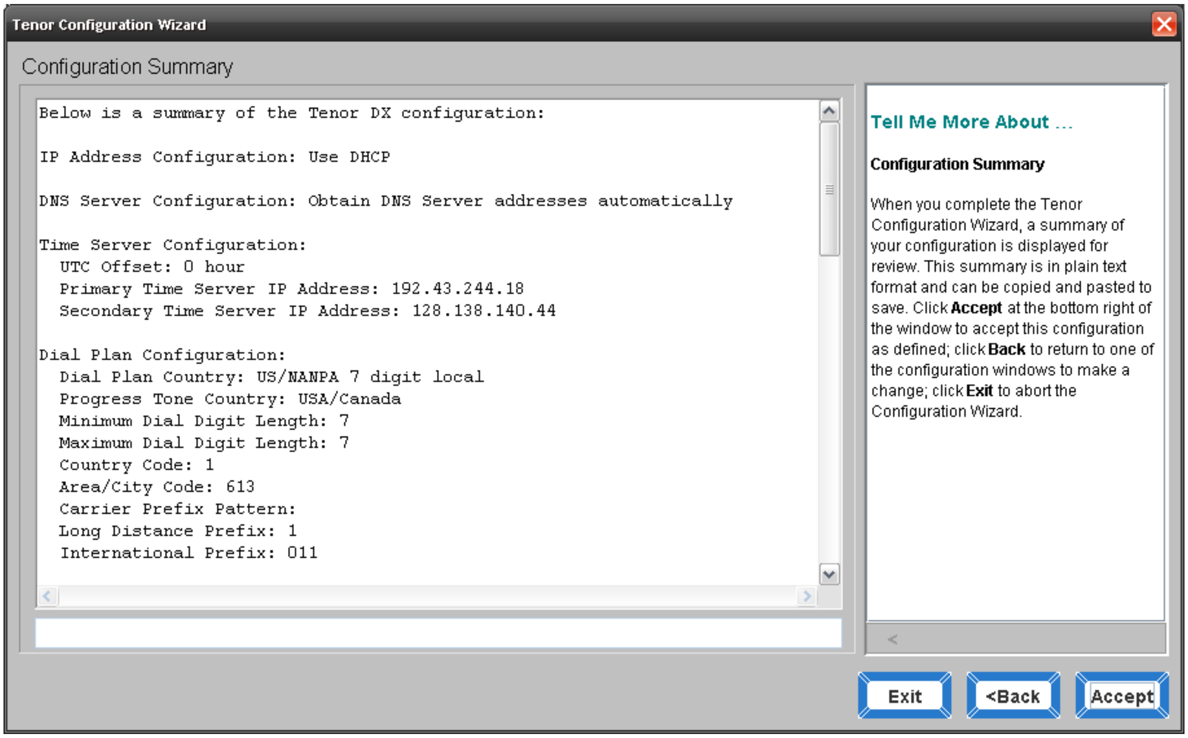
27. If you selected **Yes** on the previous step, the following DS1 Configuration screen appears. On this screen, clear unused DS1 Line check boxes. Select **Next>** to continue.



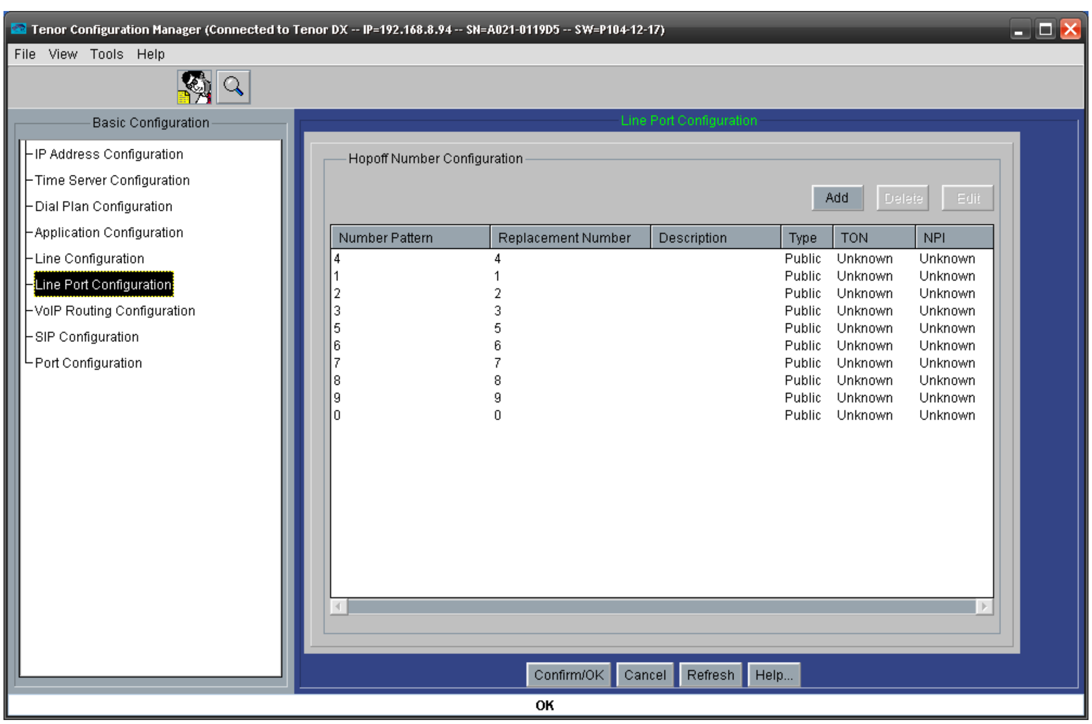
28. Review the DS1 Configuration summary screen. Select **Next>** to continue.



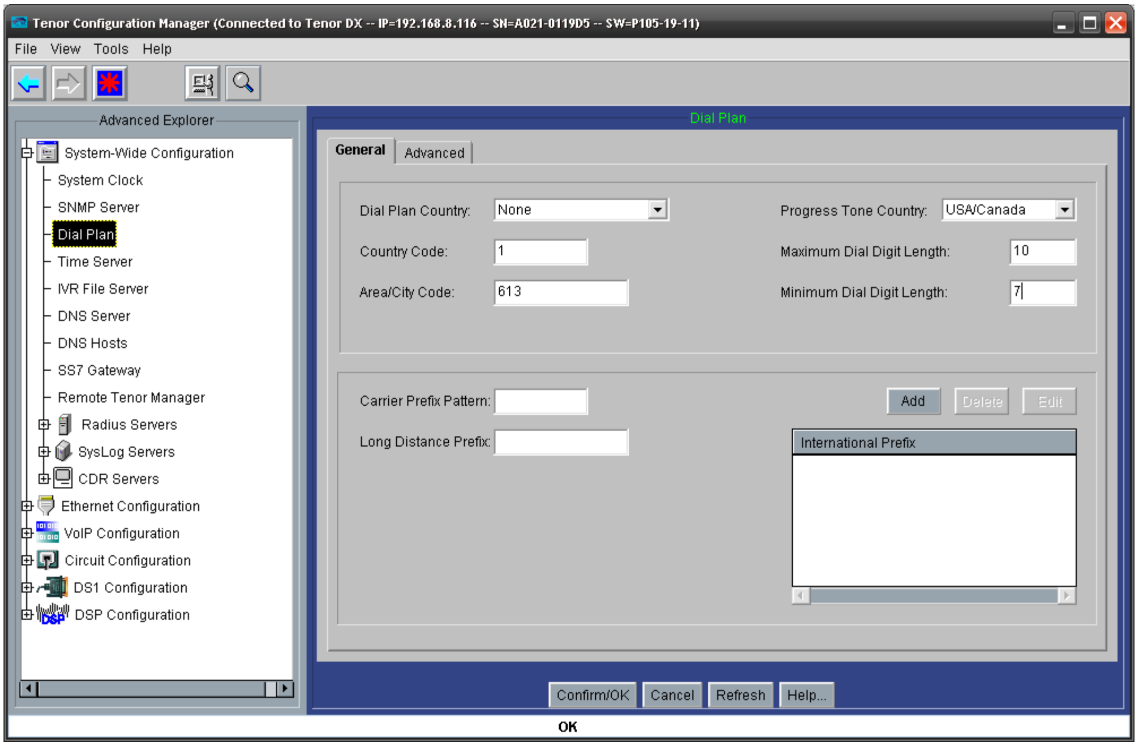
29. Check the settings to make sure they are correct. You can go back and make changes if necessary. When finished, select **Accept** to continue.



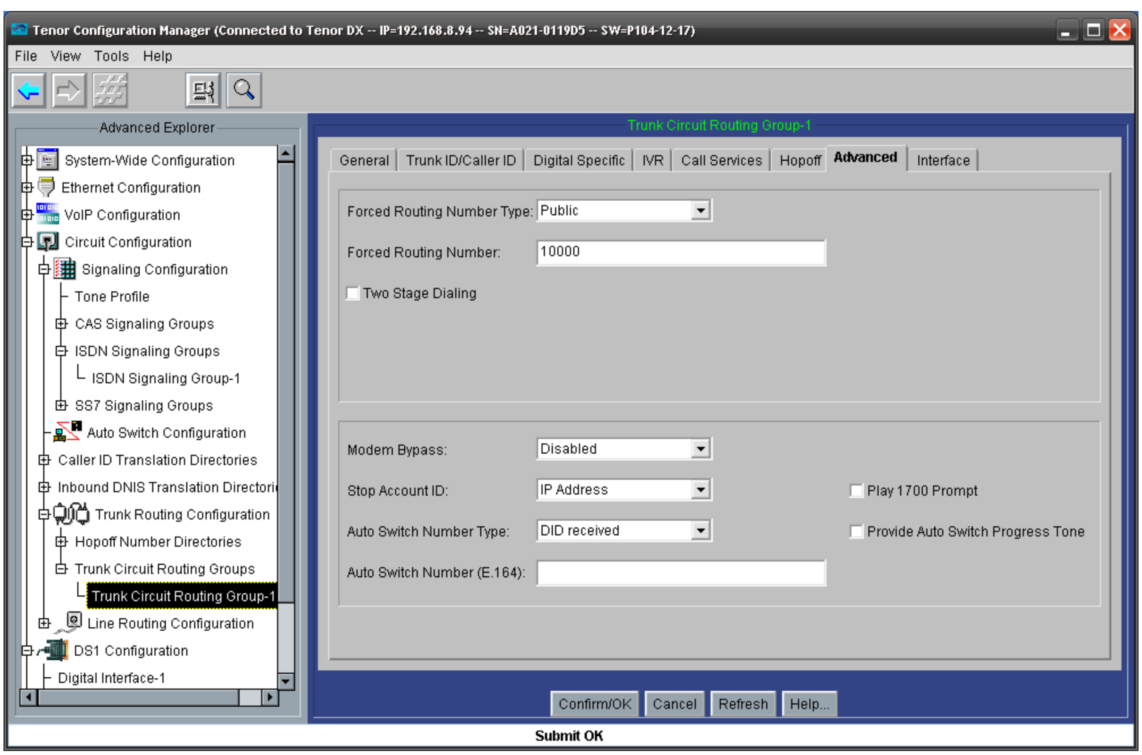
30. After you complete the initial configuration and reboot the Quintum gateway, navigate to the **Line Port Configuration** tab. You can set up numbers that are allowed go through to the PSTN line. Configure these numbers based on your location and requirements, and, when finished, select **Confirm/OK**. Select the first icon on the toolbar to continue to the next step. 



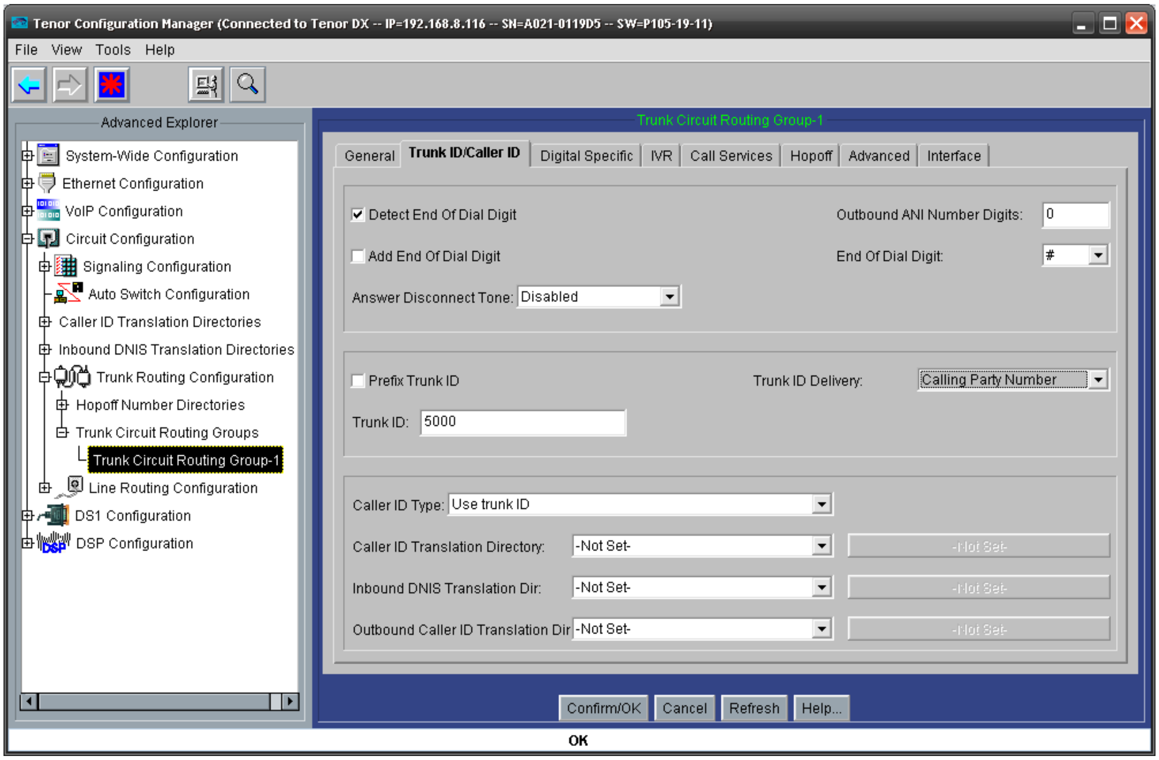
31. Navigate to **System-Wide Configuration > Dial Plan**. On the **General** tab, change Maximum Dial Digit Length to 10 and Minimum Dial Digit Length to 7. You can also change other settings like Dial Plan Country and Progress Tone Country. Select **Confirm/OK**.



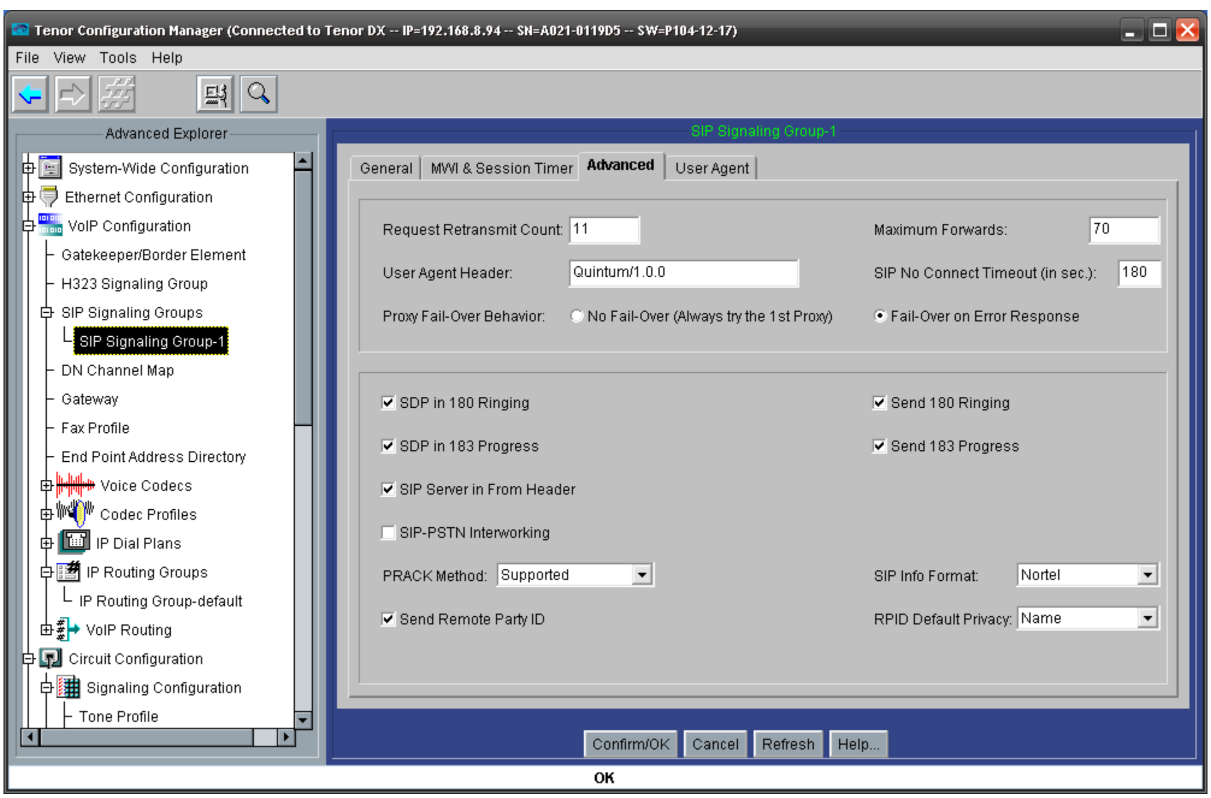
32. Navigate to **Circuit Configuration > Trunk Routing Configuration > Trunk Circuit Routing Groups > Trunk Circuit Routing Group-1**. In the **Forced Routing Number** box, enter the auto-attendant identity. Typically, this is set to **10000**. Select **Confirm/OK**.



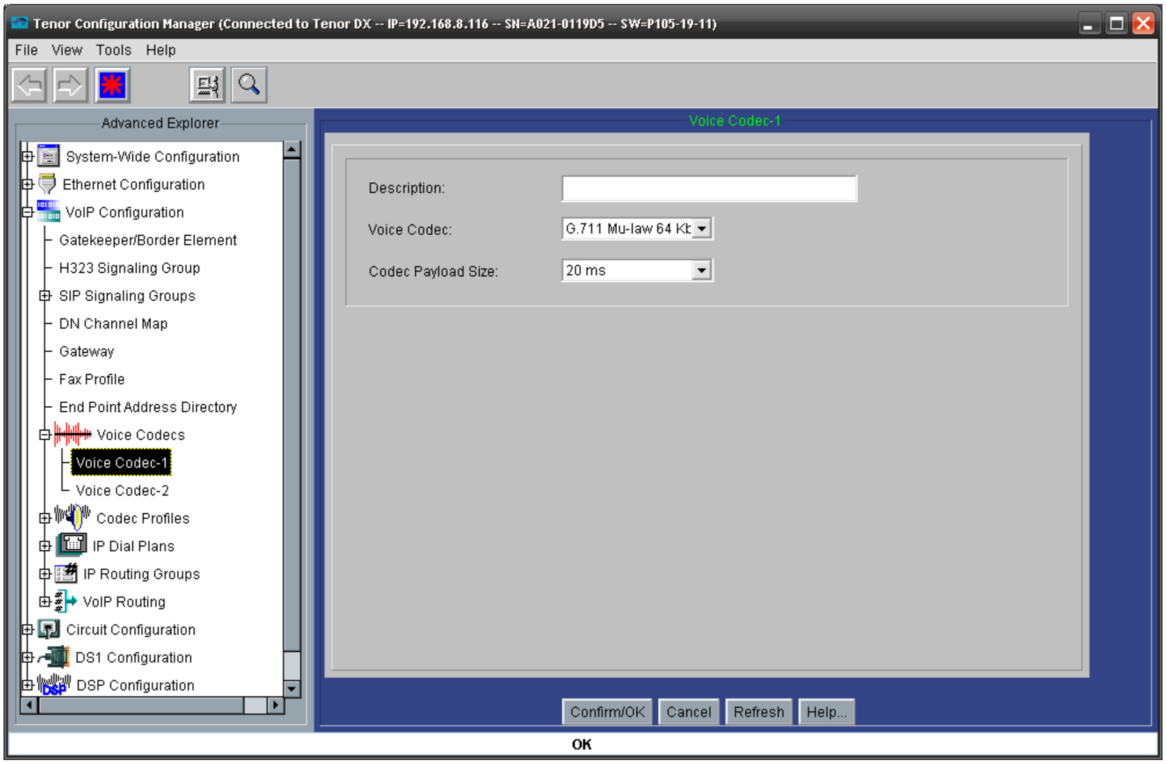
33. In **Trunk Circuit Routing Group-1**, Navigate to the **Trunk ID/Caller ID** tab. Under **Trunk ID Delivery**, choose **Calling Party Number**. Select **Confirm/OK**.



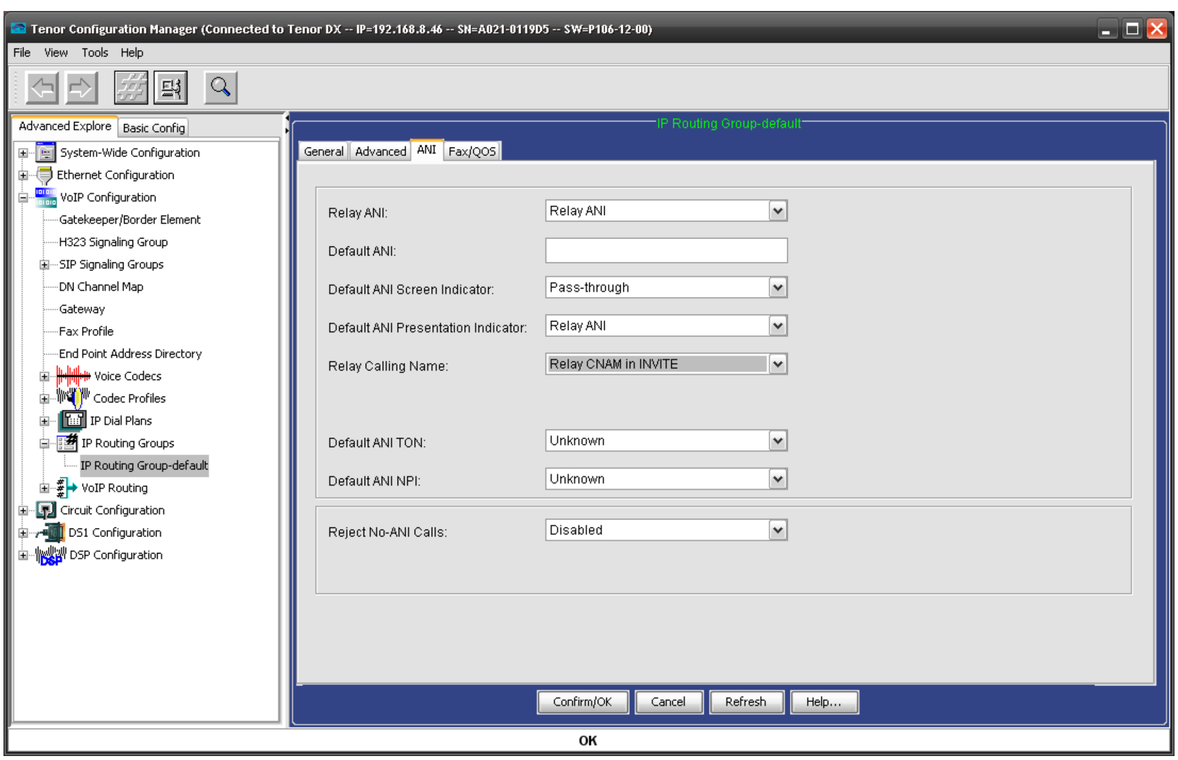
34. Navigate to VoIP **Configuration > SIP Signaling Groups > SIP Signaling Group-1**. Make sure that **Nortel** is selected from the **SIP Info Format** list box.



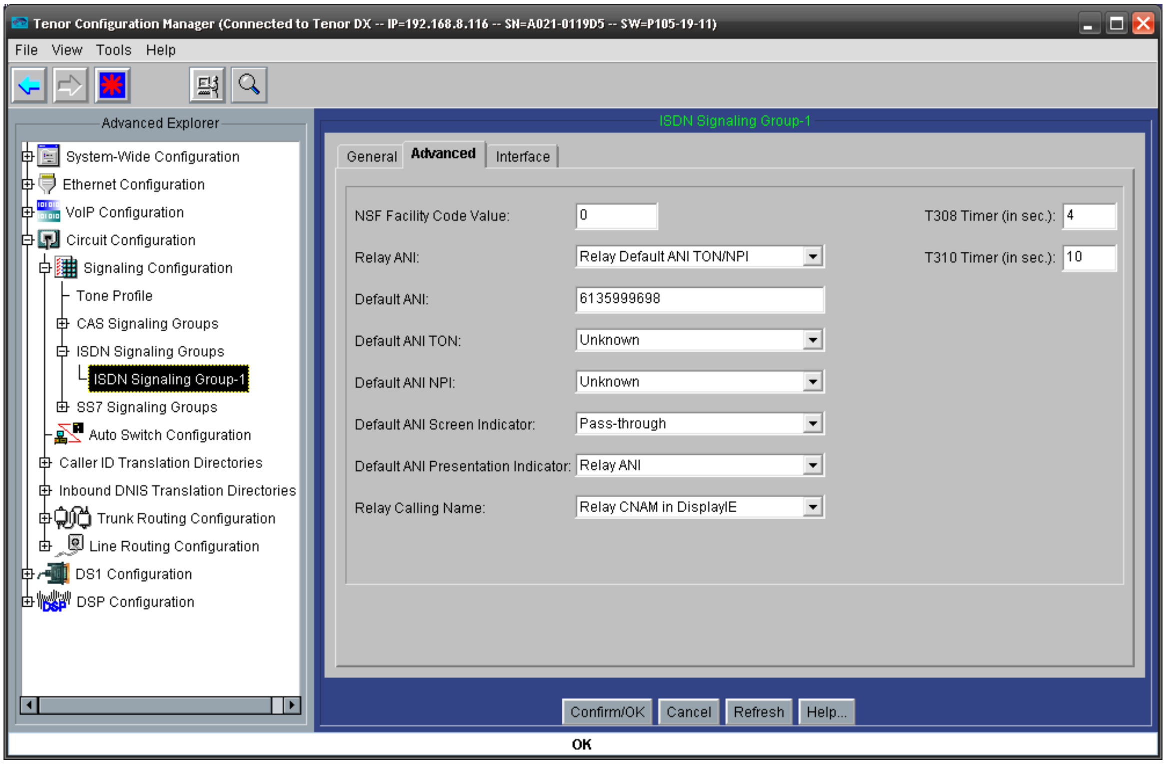
35. Navigate to **VoIP Configuration > Voice Codec-1**. Set **Voice Codec** to **G.711 Mu-law 64 kb**. Select **Confirm/OK**.



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



37. If you wish to modify the outgoing caller ID, navigate to **Circuit Configuration > ISDN Signaling Groups > ISDN Signaling Group-1** and the **Advanced** tab. Under **Relay ANI**, choose **Relay Default ANI TON/NPI**. Under **Default ANI** enter the outgoing number you wish to use. This can be any telephone number, for example, **6135999698.**



38. To complete the changes, select **Confirm/OK** and then 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. Enter **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. Enter **bin <Enter>**.
10. Enter **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 Call Attendant 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. Ensure 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

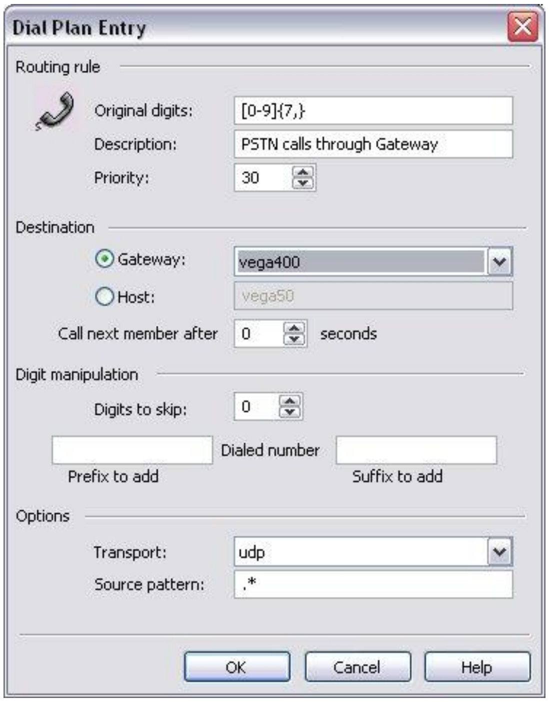
1. Select **Gateways**.
2. Right-click the right panel and select **New Gateway**.
3. Choose **Public Switched Telephone Network (PSTN)** from the gateway list.
4. In the **Host** name field, enter the IP address of the gateway.
5. Enter a descriptive name for the gateway.
6. Save.

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 Vega gateway. For example:



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 *http://kb.adtran.com*, for the correct use of regular expressions in the toll restrictions to enforce corporate dialing policy. It is explained in detail in *Managing PBX Configuration Categories > Routing—Toll Restrictions*.

Glossary of Features

Accept Incoming Calls

This feature allows an incoming call from the PSTN to be answered by the gateway and a SIP call is then made to extension 10000.

Accept Outgoing Calls

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

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.

Calling Party Name

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

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.

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.

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.

System Music on Hold Support

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

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.

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.

Active Message Delivery

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

Paging Notification

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

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.

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.