NetVanta Unified Communications Technical Note

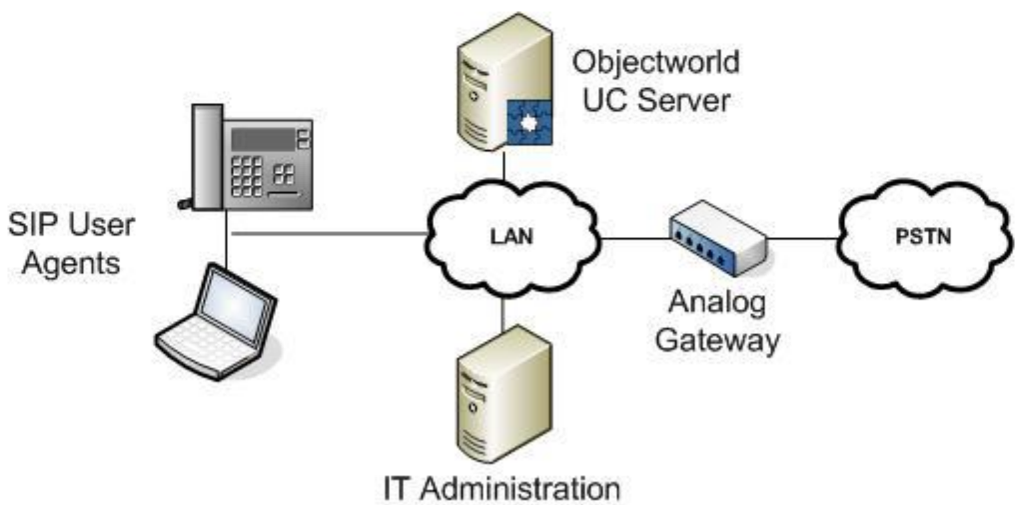
Installing and Configuring the Quintum Tenor AS Gateway

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

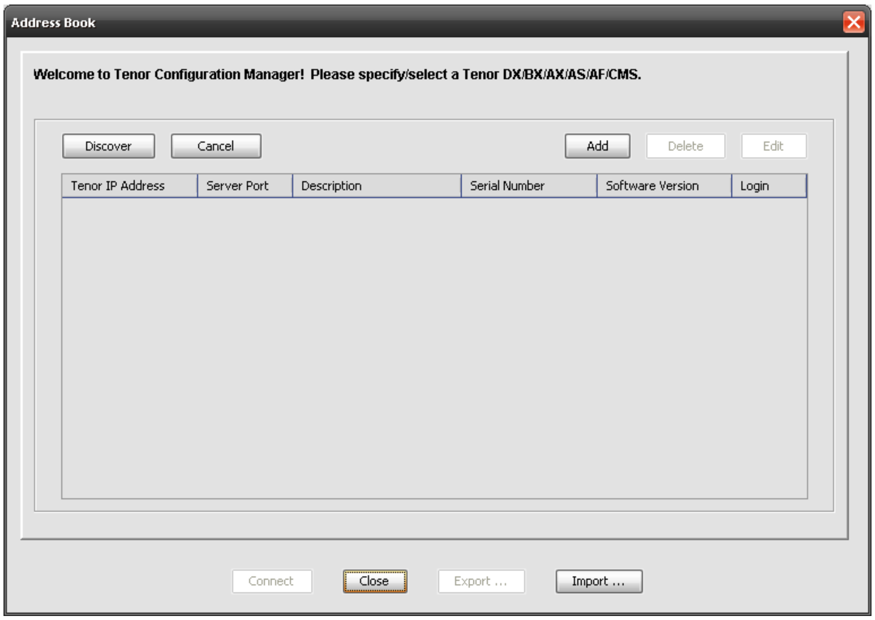
By default, the gateway is configured to use an IP address assigned by DHCP. The gateway can also be configured to use a static IP address. For routing calls out from the UC server, the **Quintum Tenor AS** must have an IP address that does not change.

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

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



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.

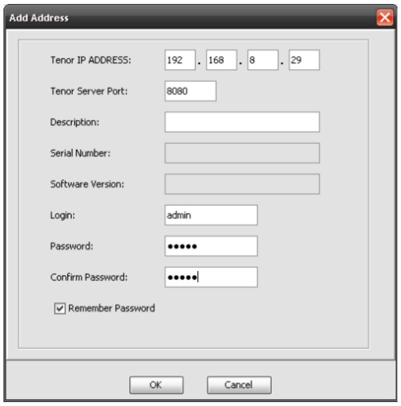
To add the gateway automatically

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.

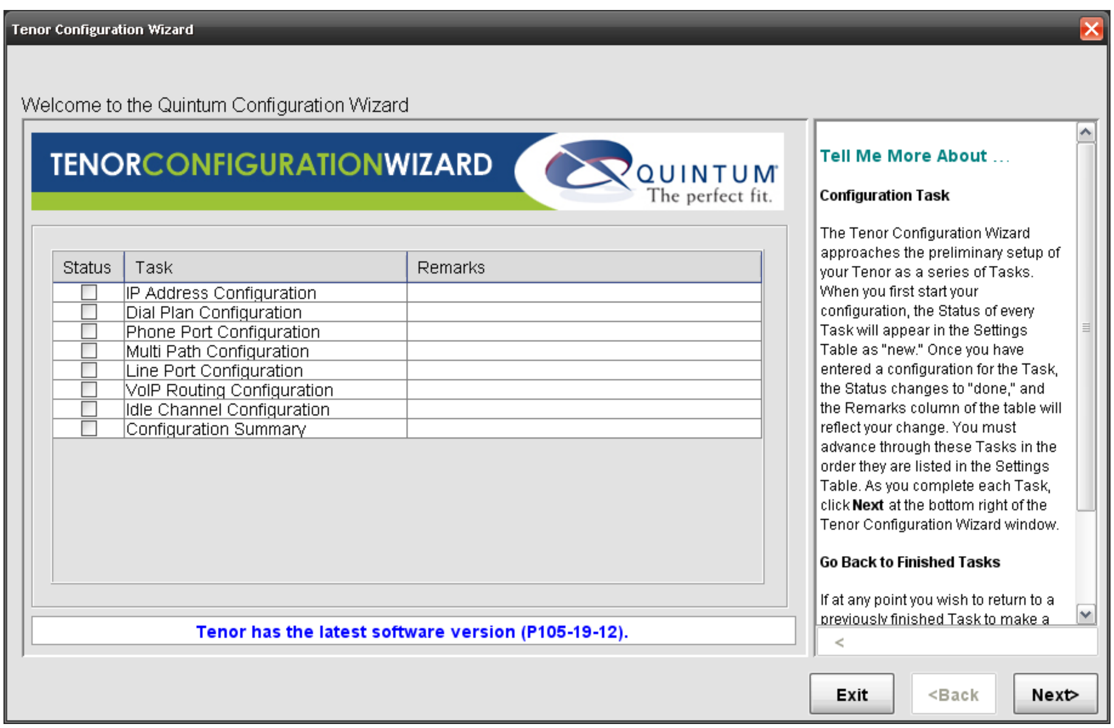


2. Enter the IP Address of the gateway.
3. Enter **admin** as the username and password.
4. Select **OK**.
5. Select **Connect** on the *Address Book* screen.

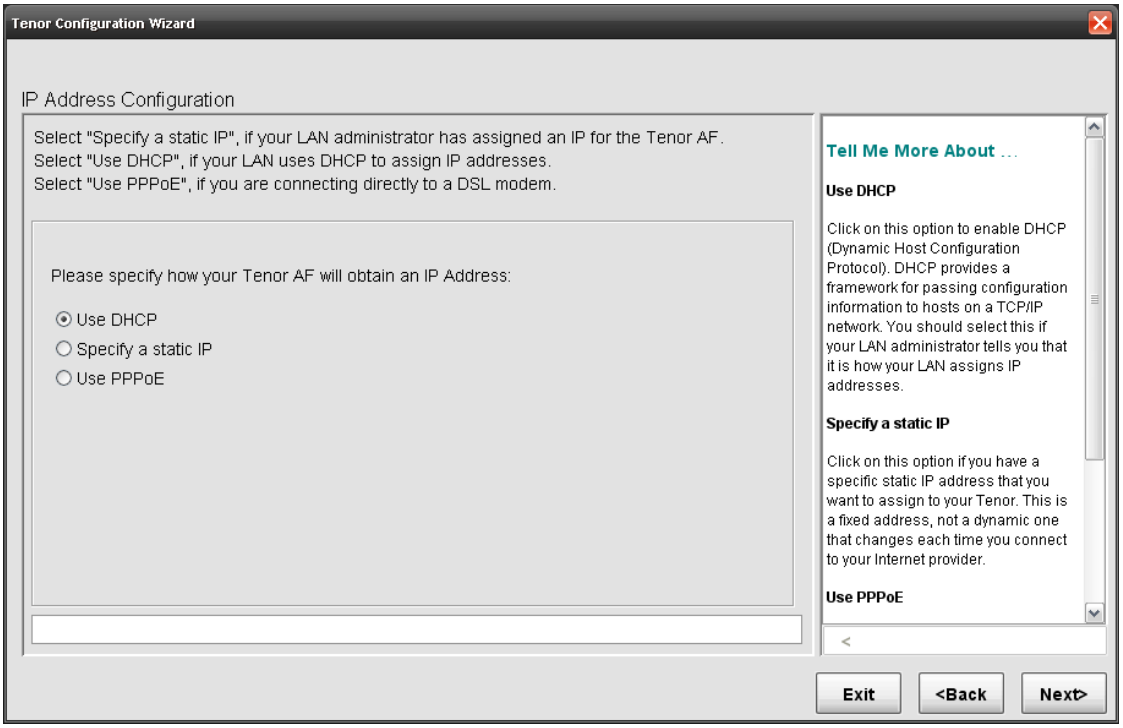
Running the Configuration Wizard

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

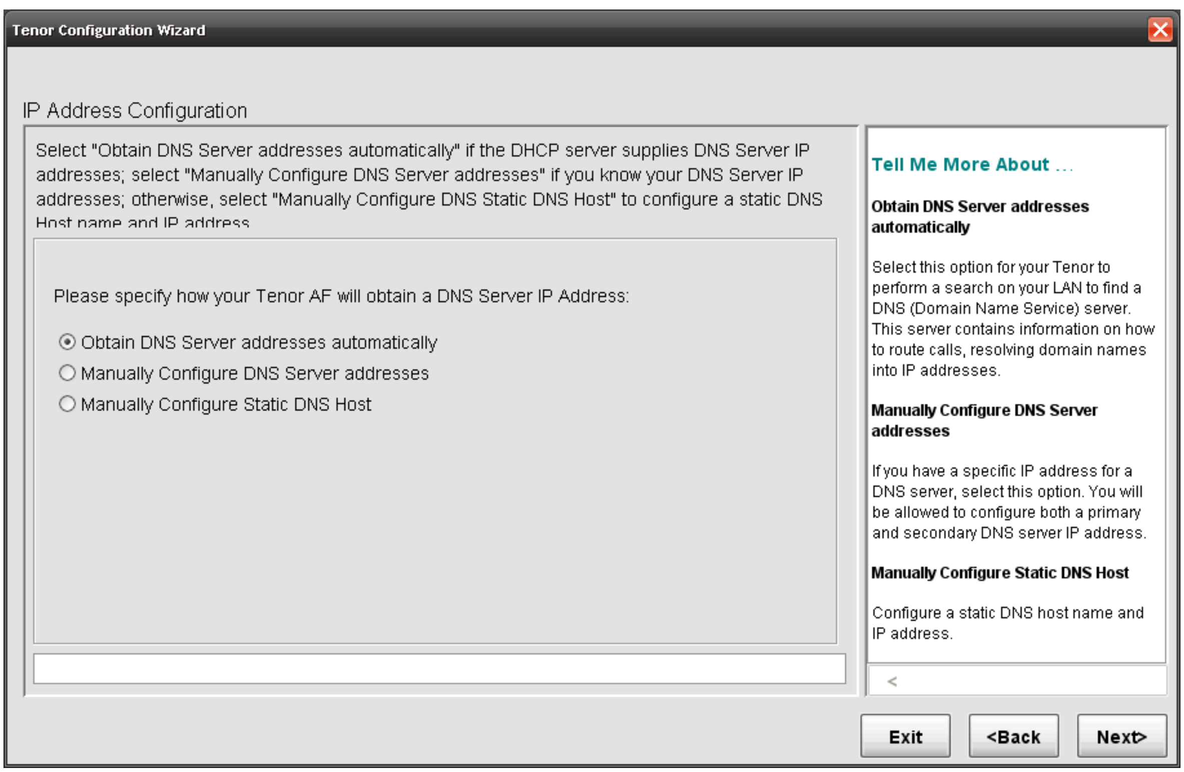
1. Select **Next**.



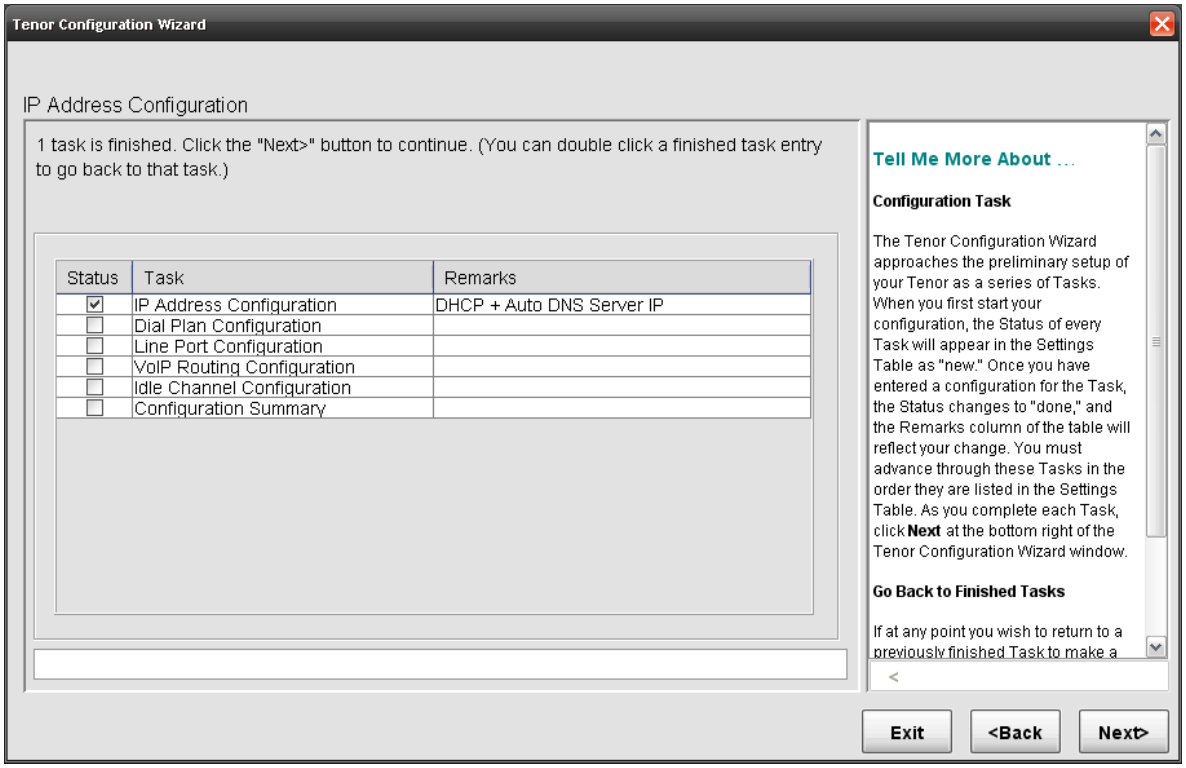
2. On the screen below, you have the option to choose how your gateway obtains its IP Address and network settings. A static IP address is recommended for a gateway.



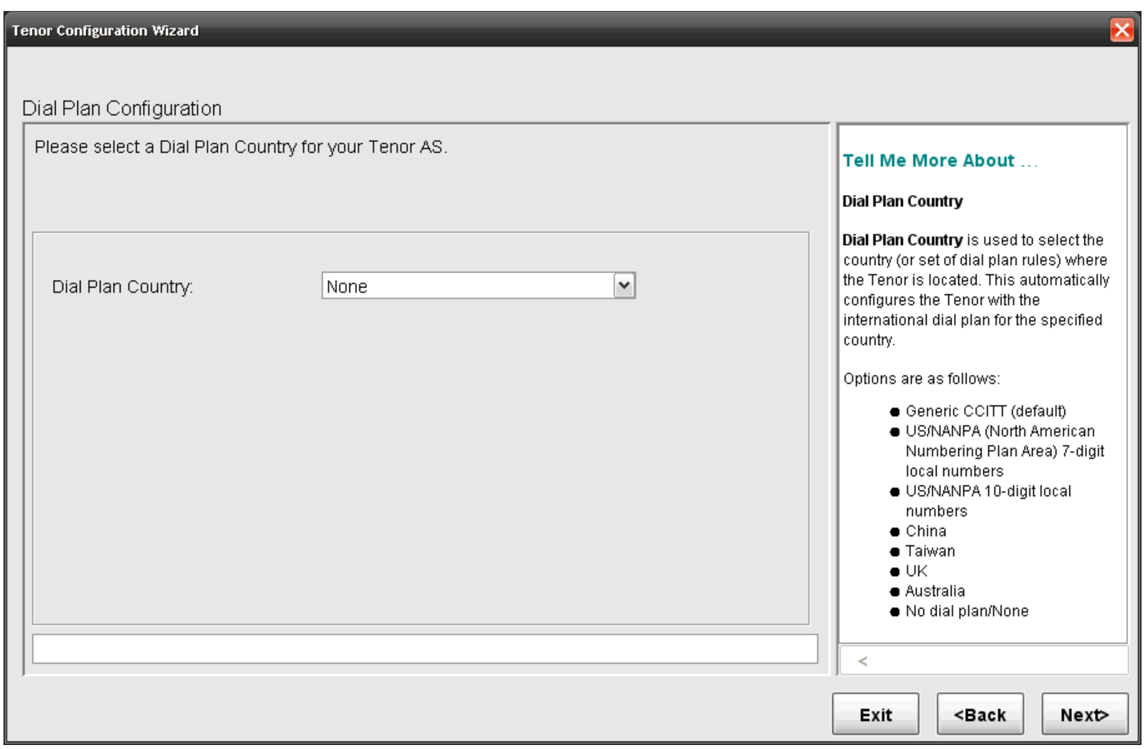
3. You can specify whether you want to obtain DNS server addresses automatically or if you want to manually configure them. If you are using DHCP, you can automatically obtain the DNS server addresses; otherwise you must manually configure them. Select **Next** to continue.



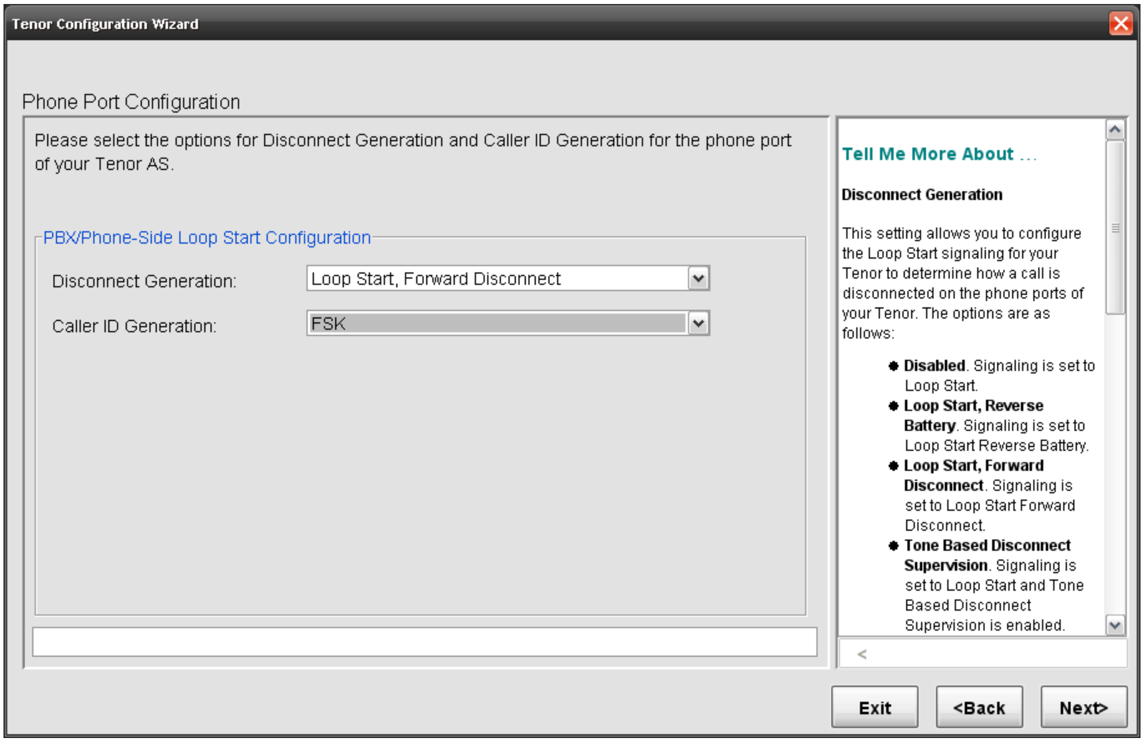
4. The first task is complete. Select **Next** to continue.



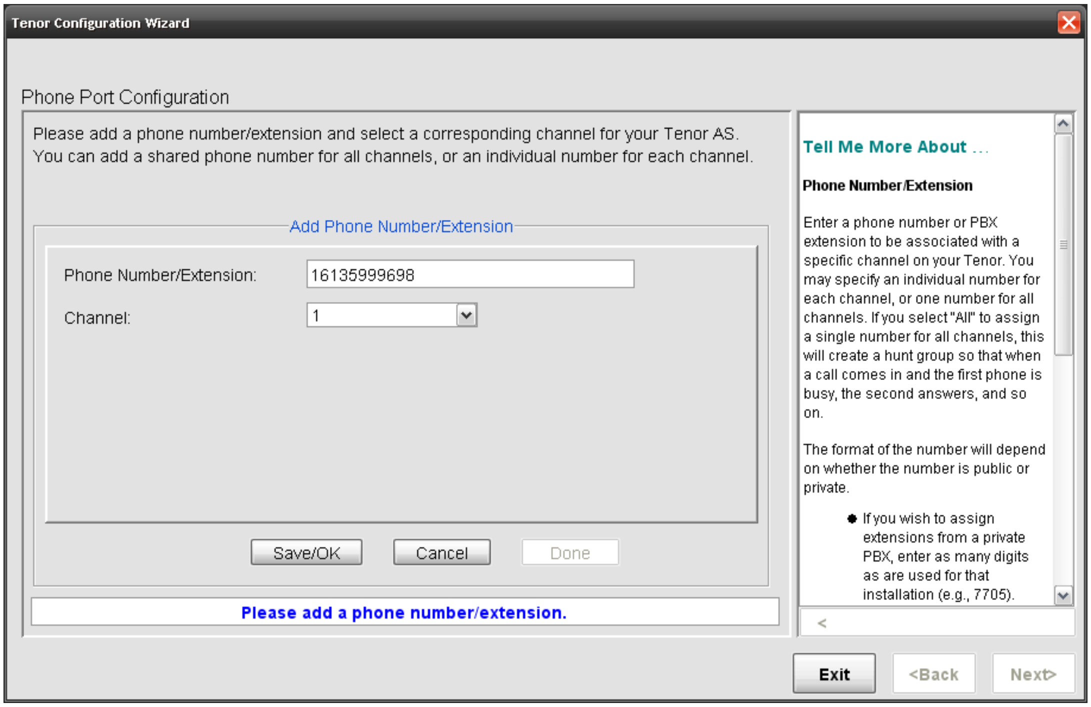
5. The **Dial Plan Configuration** screen allows you to set up the dialing plan. Choose **None** from the **Dial Plan Country** list. Currently the dial plan rules do not work and will result in outgoing calls not working. When finished, select **Next** to continue.



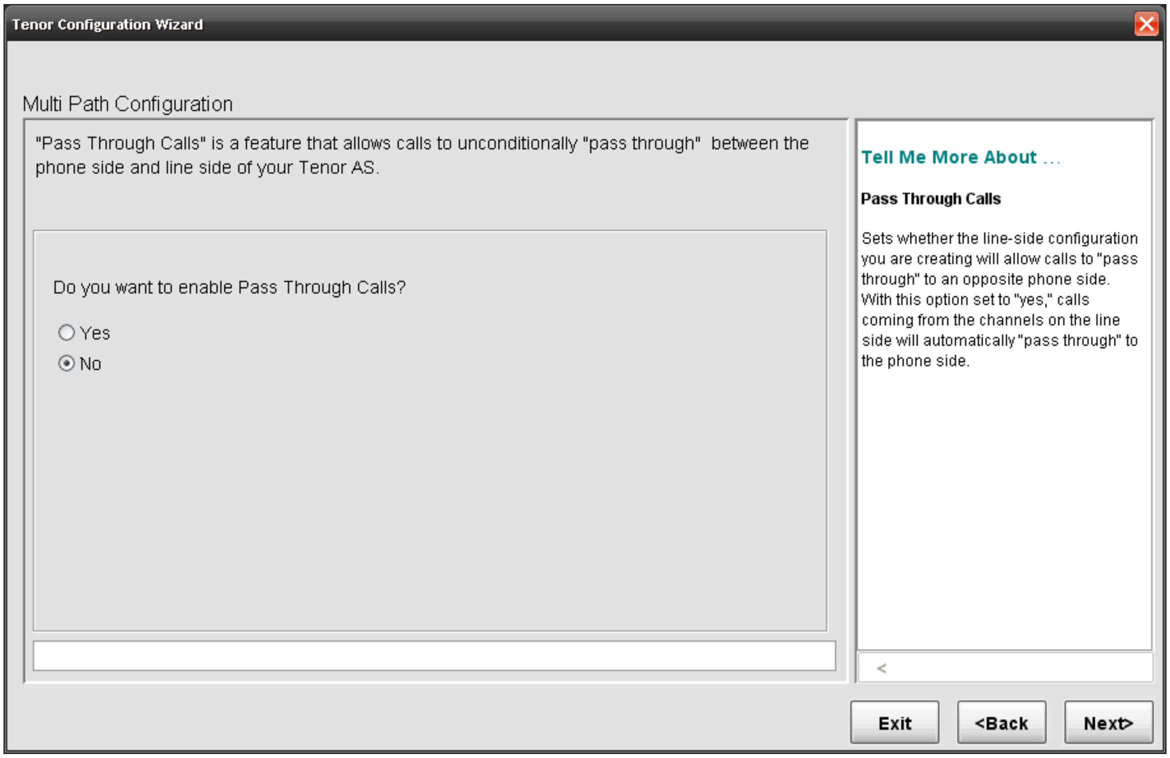
6. On the **Phone Port Configuration** screen, choose the method for **Disconnect Generation** and **Caller ID Generation**. These settings depend on your carrier and location. When finished, select **Next** to continue.



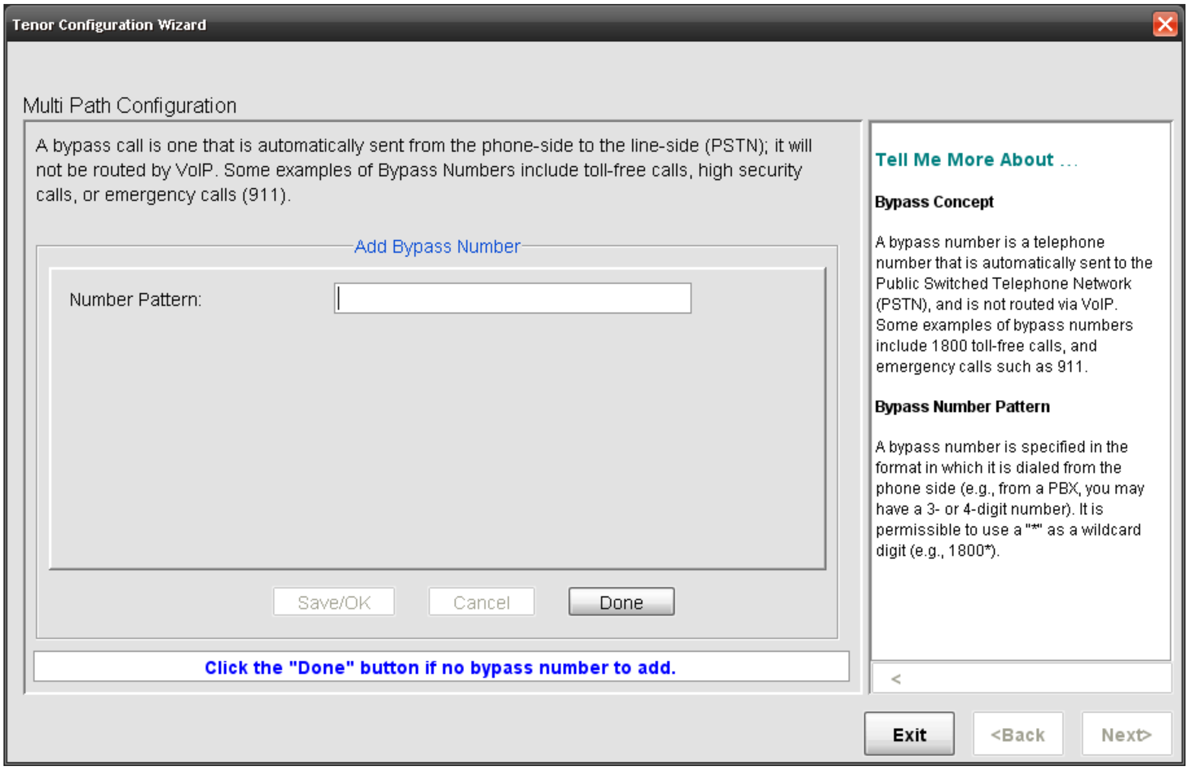
7. On the **Phone Port Configuration** screen, you can map DID numbers to individual channels on the gateway. When finished, select **Next** to continue.



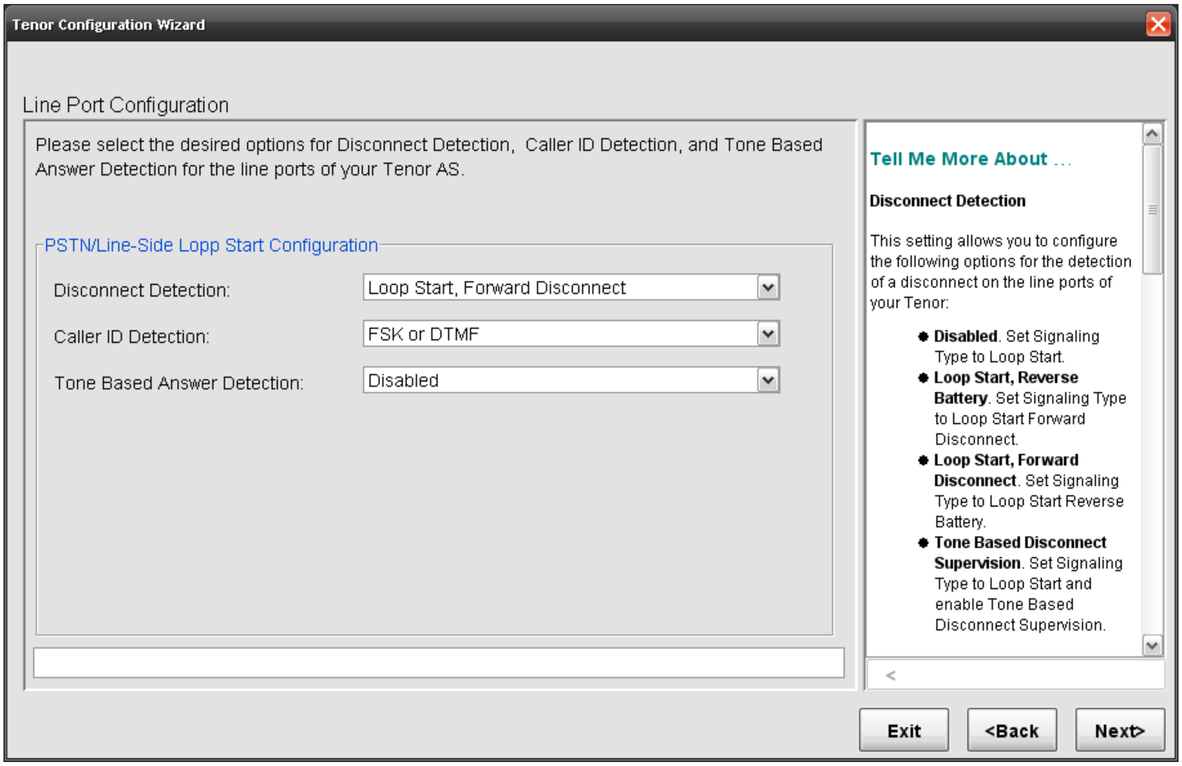
8. On the **Multipath Configuration** screen, you can choose for all calls to automatically pass through between the phone side and lines. You do not need to enable Pass Through calls, so you can select **No**.



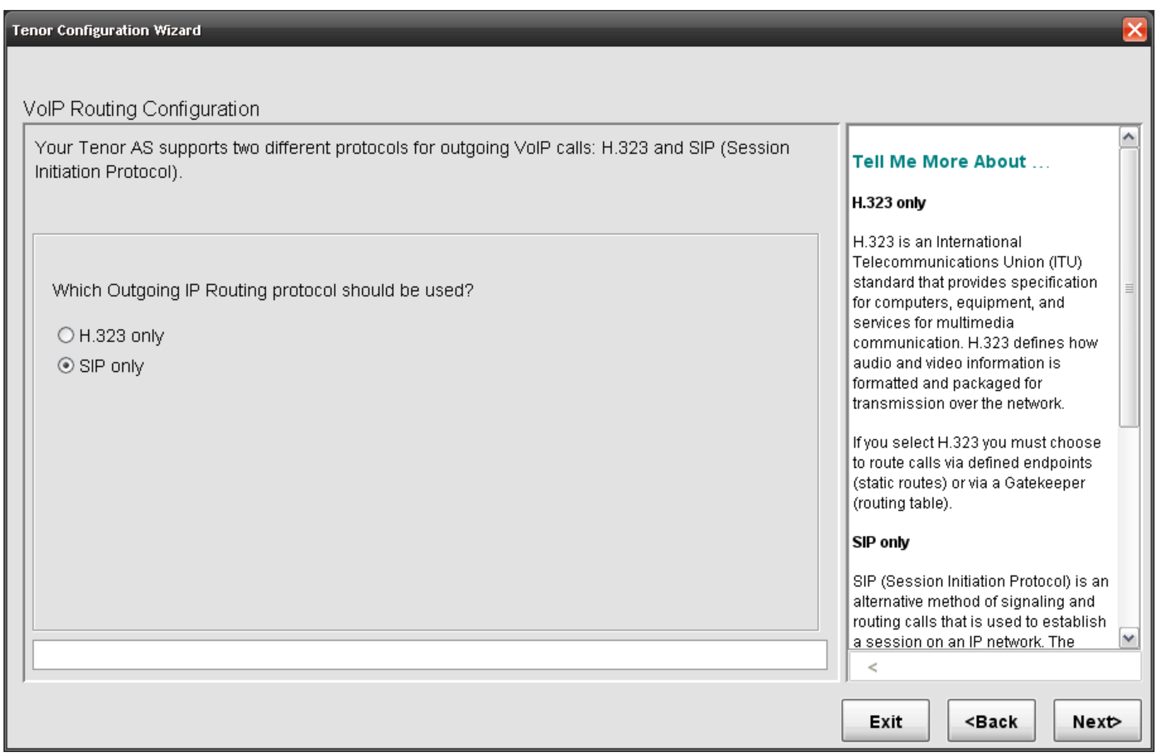
9. On the **Add Bypass Number** screen, select **Done**, and then select **Next.**



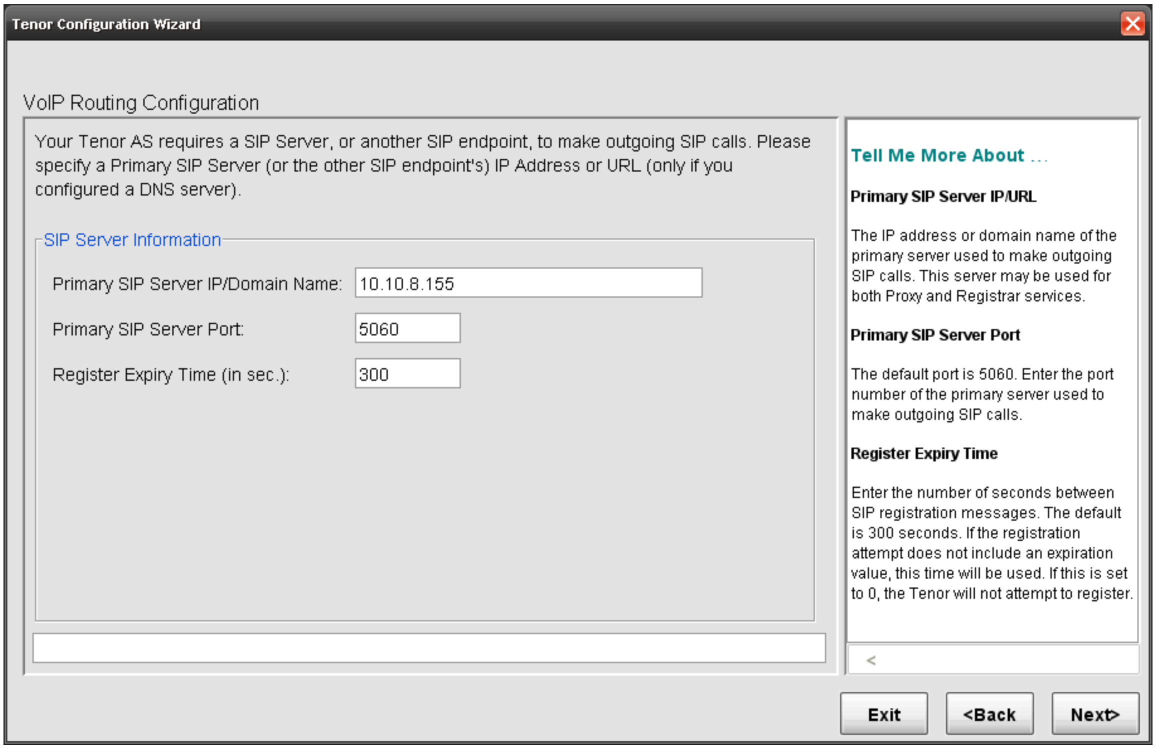
10. On the **Line Port Configuration** screen, choose the method for **Disconnect Detection** and **Caller ID Detection**. These settings depend on your carrier and location. Select **Next** to continue.



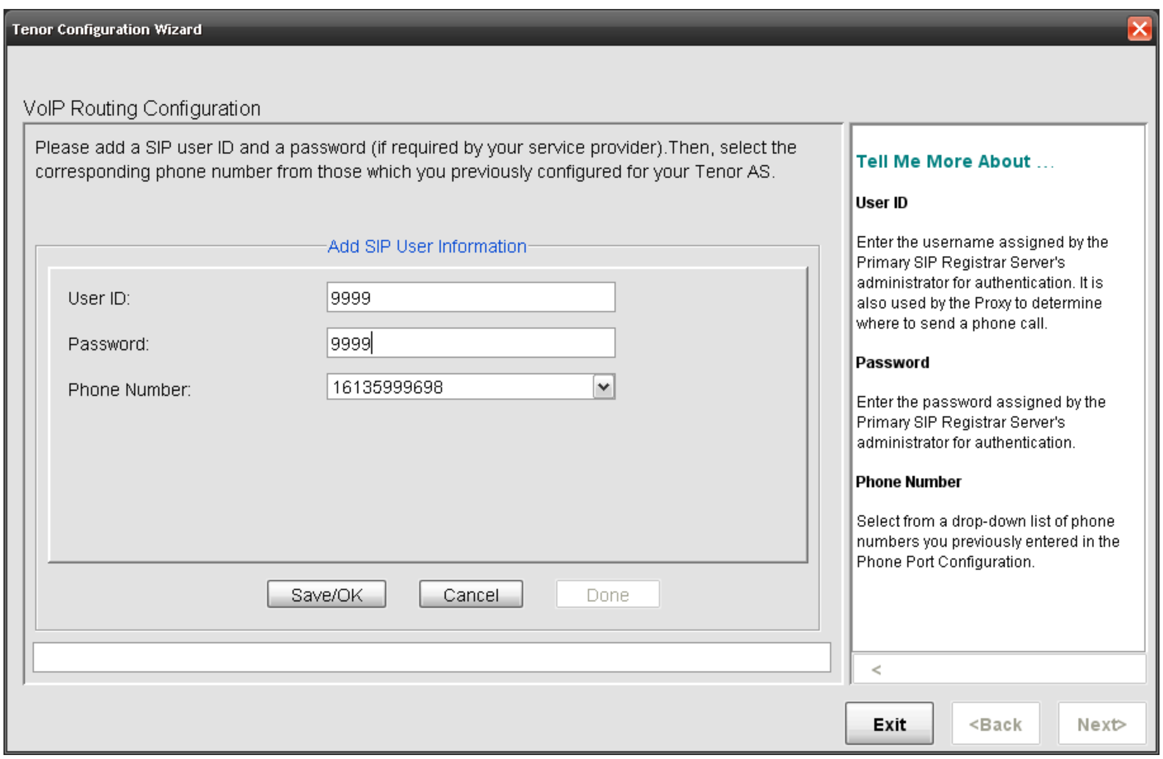
11. On the **VoIP Routing Configuration** screen, for integration with the UC server, choose **SIP only**. Select **Next** to continue.



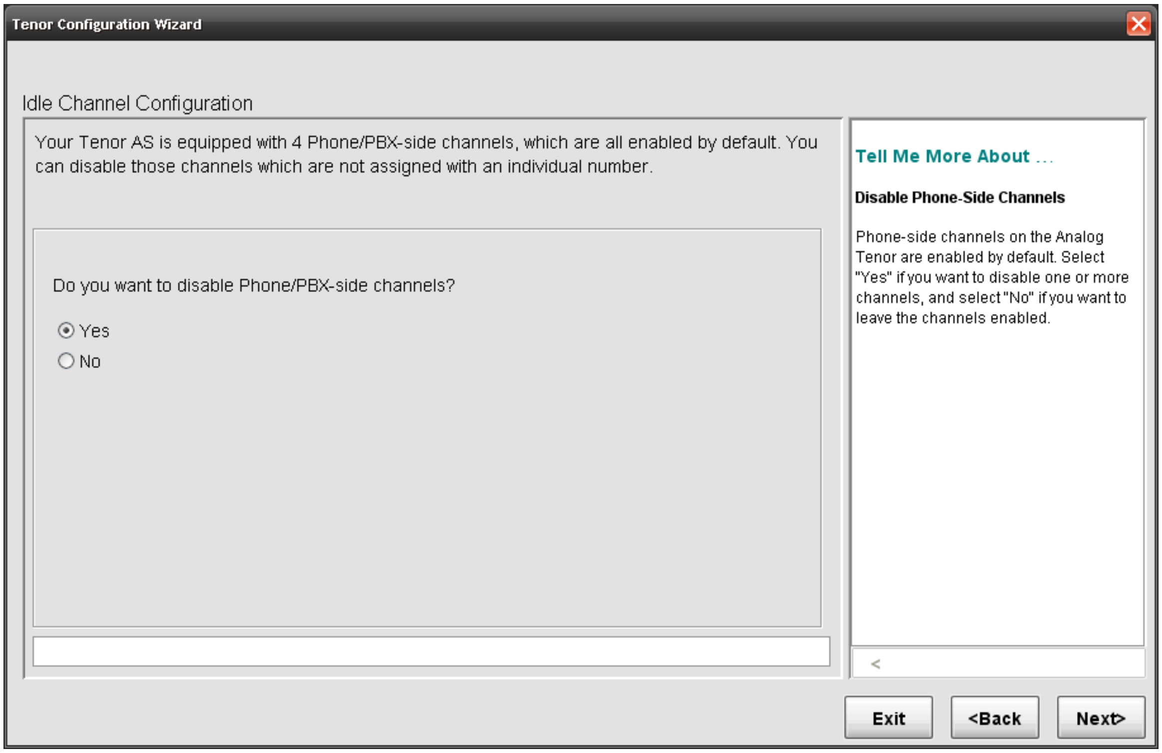
12. In the **SIP Server Information** section, change the Primary SIP Server to the IP address of your UC server. When finished, select **Next** to continue.



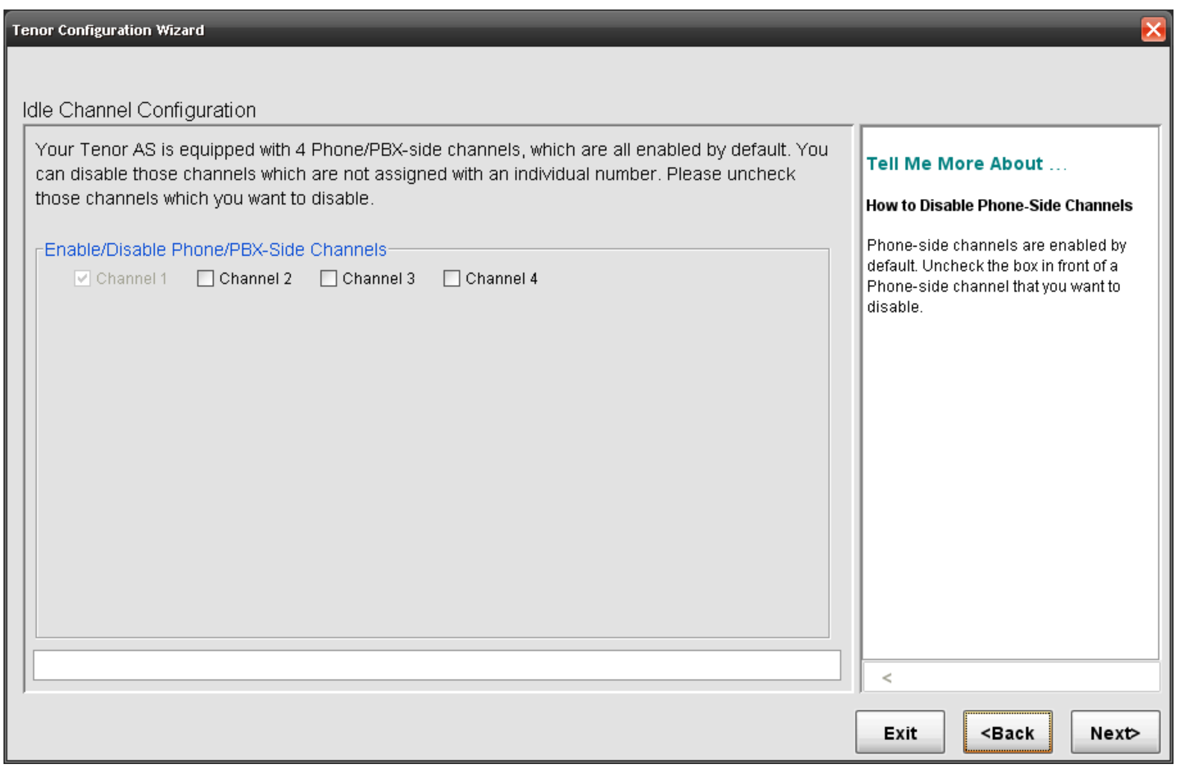
13. In the **Add SIP User Information** section, the wizard requires you to enter a User ID and Password. Enter **9999** for both fields. Select **Done** and then **Next** to continue.



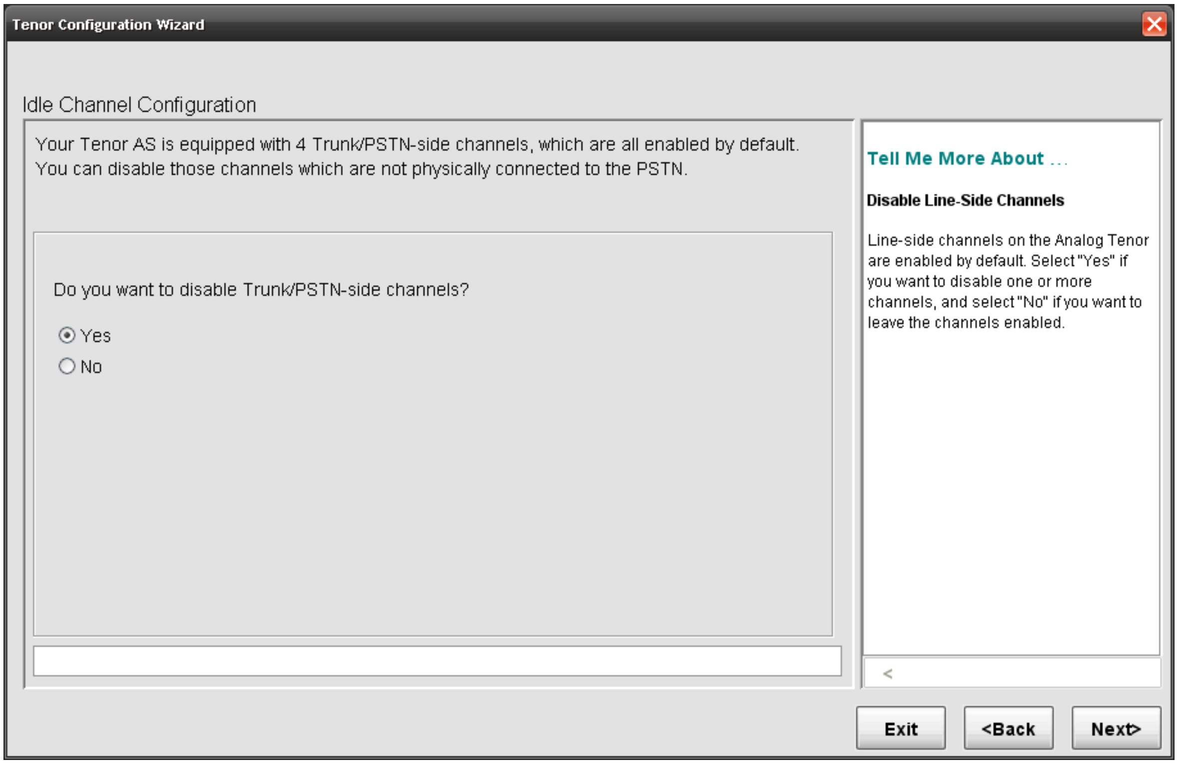
14. In the **Idle Channel Configuration** screen, select **Yes** if you are not using the maximum number of phone ports on your gateway. Select **Next** to continue.



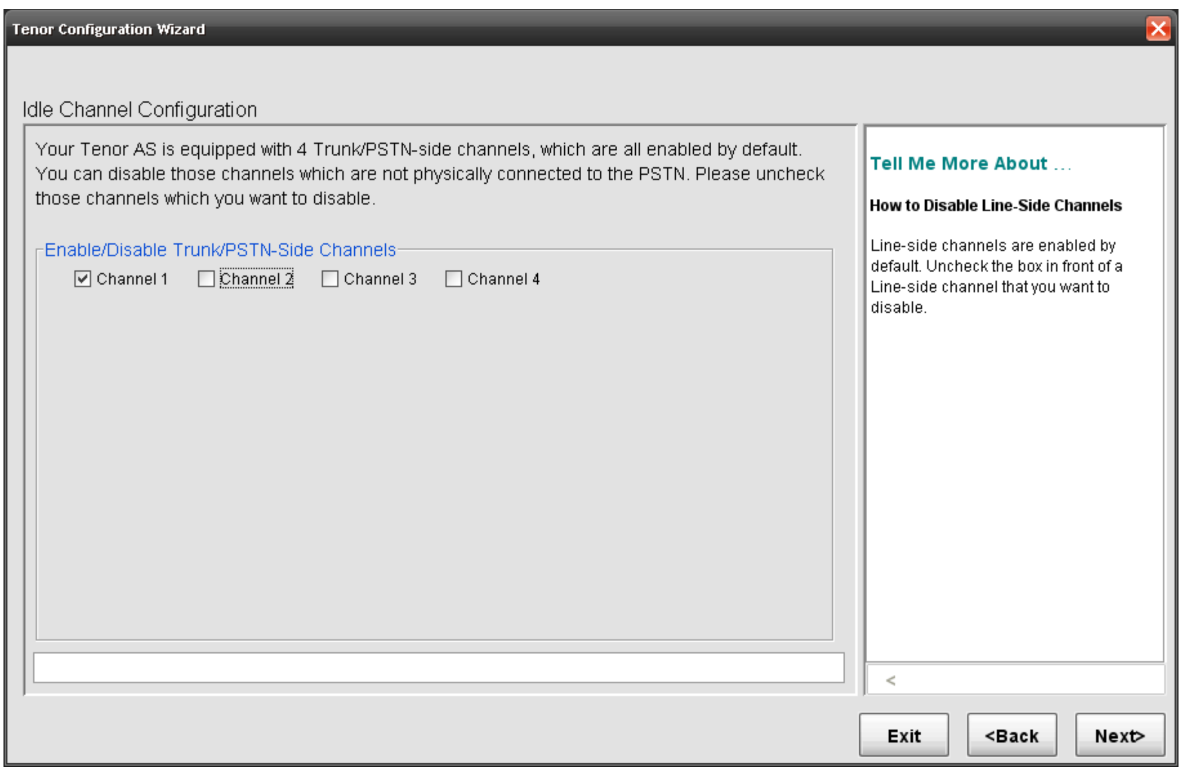
15. If you selected **Yes** on the previous screen, you will be presented with the screen below. Clear the checkboxes of the ports that are not used. Select **Next** to continue.



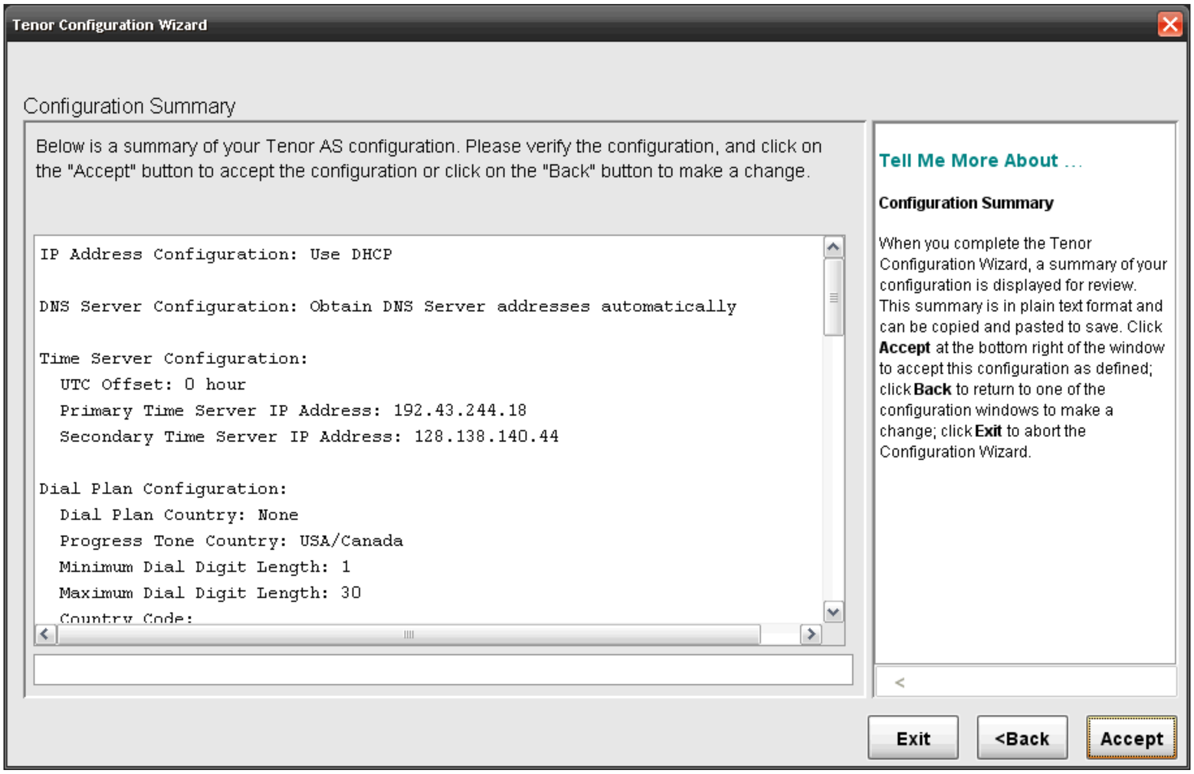
16. In the **Idle Channel Configuration** screen, select **Yes** if you are not using the maximum number of PSTN ports on your gateway. Select **Next** to continue.



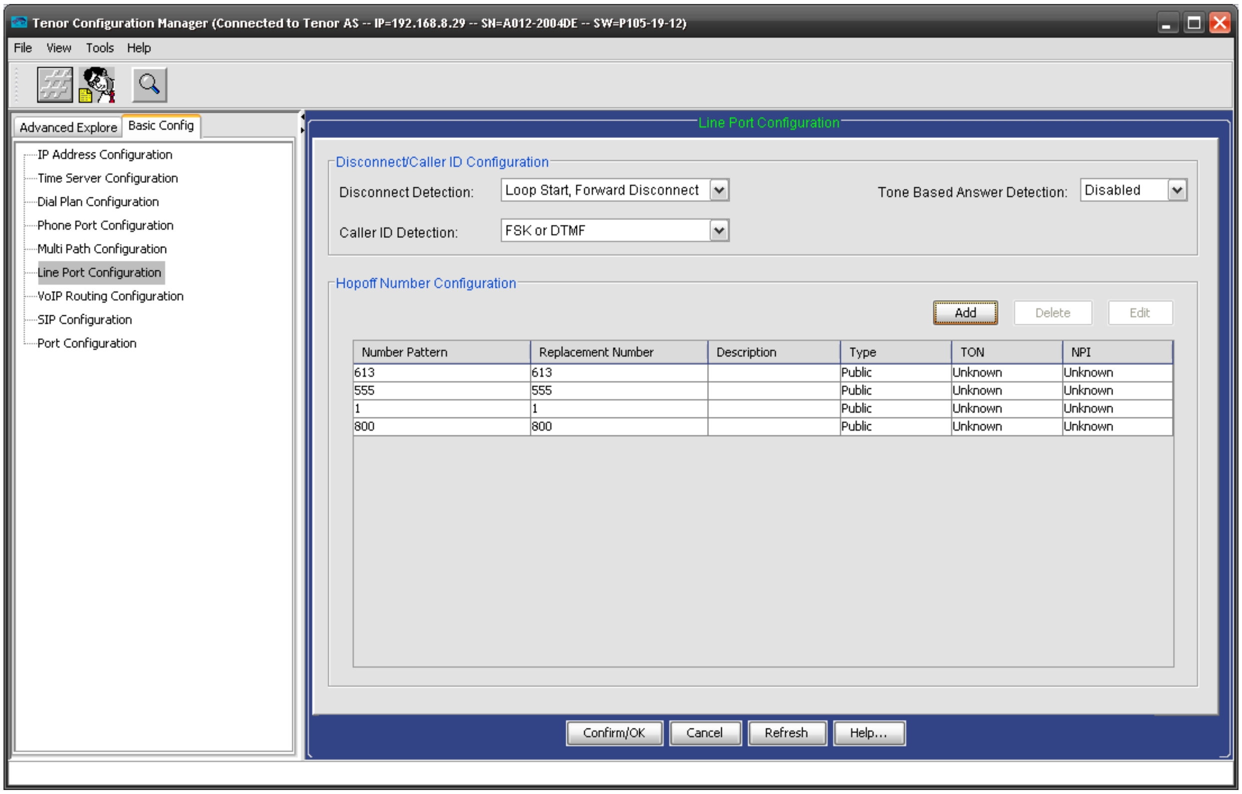
17. If you selected **Yes** on the previous screen, you will be presented with the screen below. Clear the checkboxes of the ports that are not used. Select **Next** to continue.



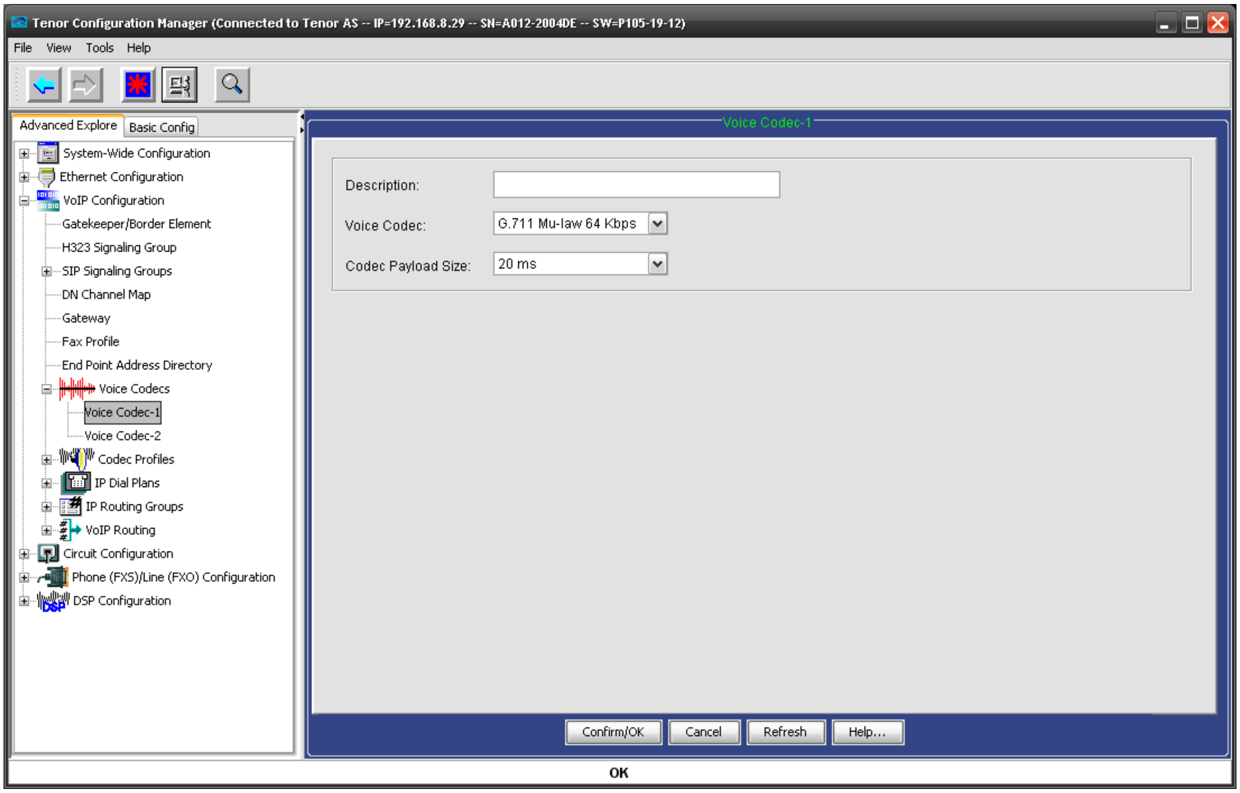
18. On the **Configuration Summary** screen, check the settings to make sure everything is correct. You can go back and make changes if necessary. When finished, select **Accept** to continue.



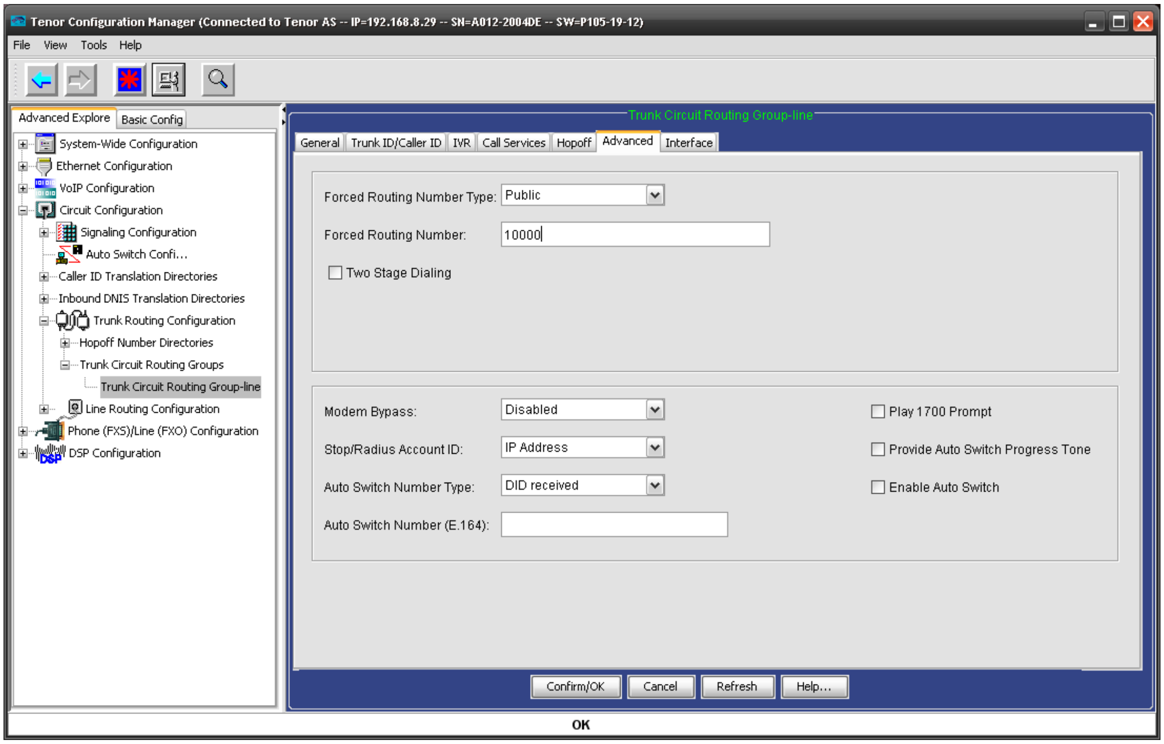
19. After the initial configuration is done, reboot the Quintum gateway and navigate to the **Line Port Configuration** tab.On that tab, you can set up numbers that are allowed go through to the PSTN from the SIP side. A hopoff number would usually contain the first few digits of a PSTN number based 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**.



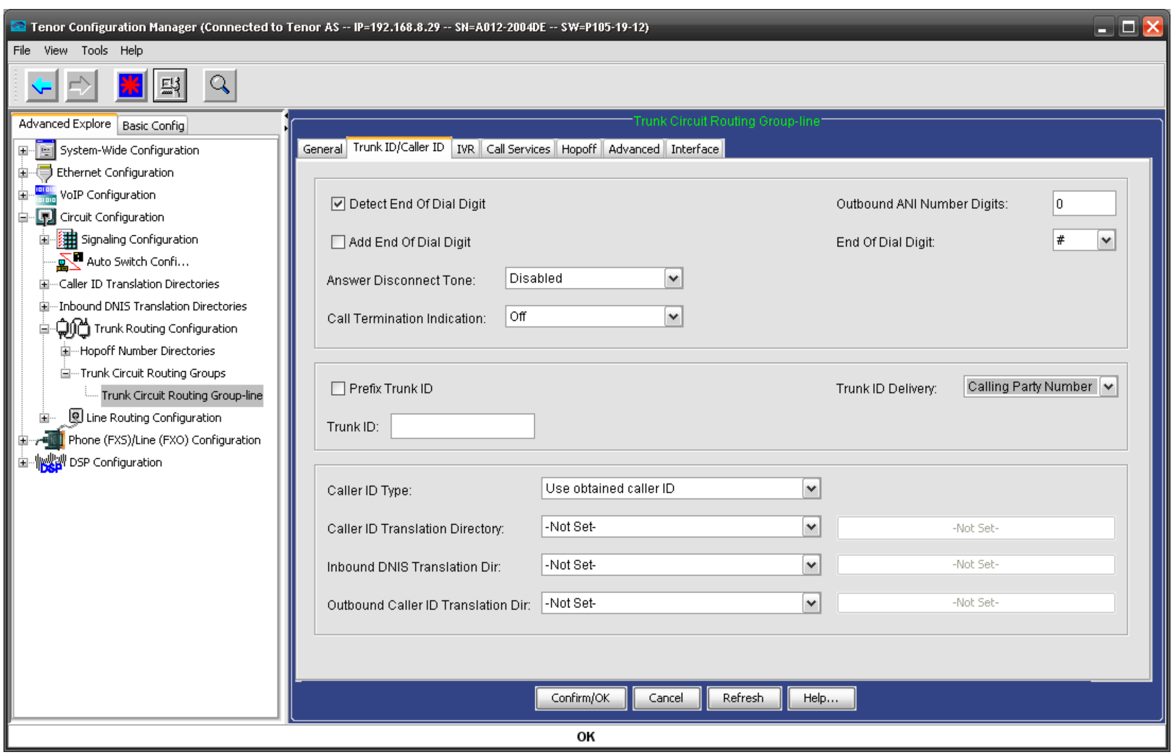
20. Select the **Advanced Explore** tab, and navigate to **VoIP Configuration > Voice Codecs > Voice Codec-1**. Set **Voice Codec** to **G.711 Mu-law 64 kb**. Select **Confirm/OK**.



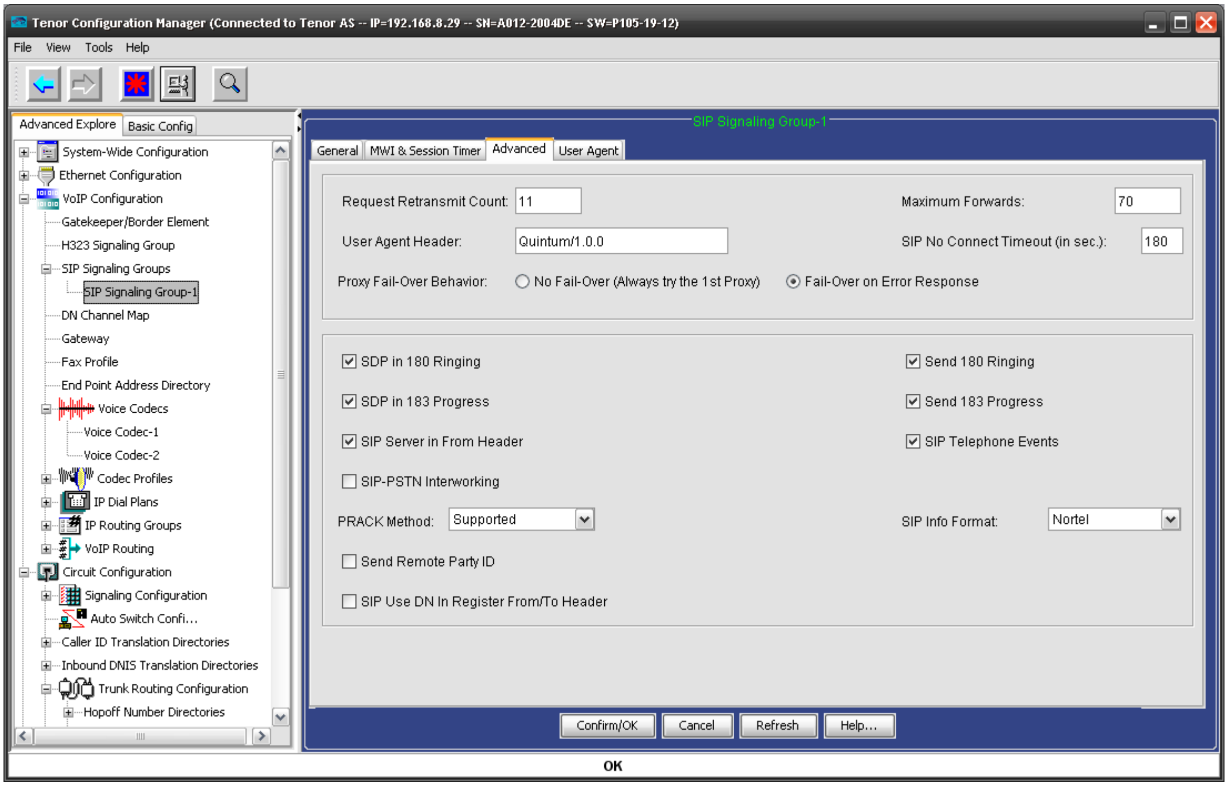
21. Navigate to **Circuit Configuration > Trunk Circuit Routing Groups > Trunk Circuit Routing Group-line**. Under the **Advanced** tab and in the **Forced Routing Number** box, enter the auto-attendant identity. Typically, this is set to **10000**.



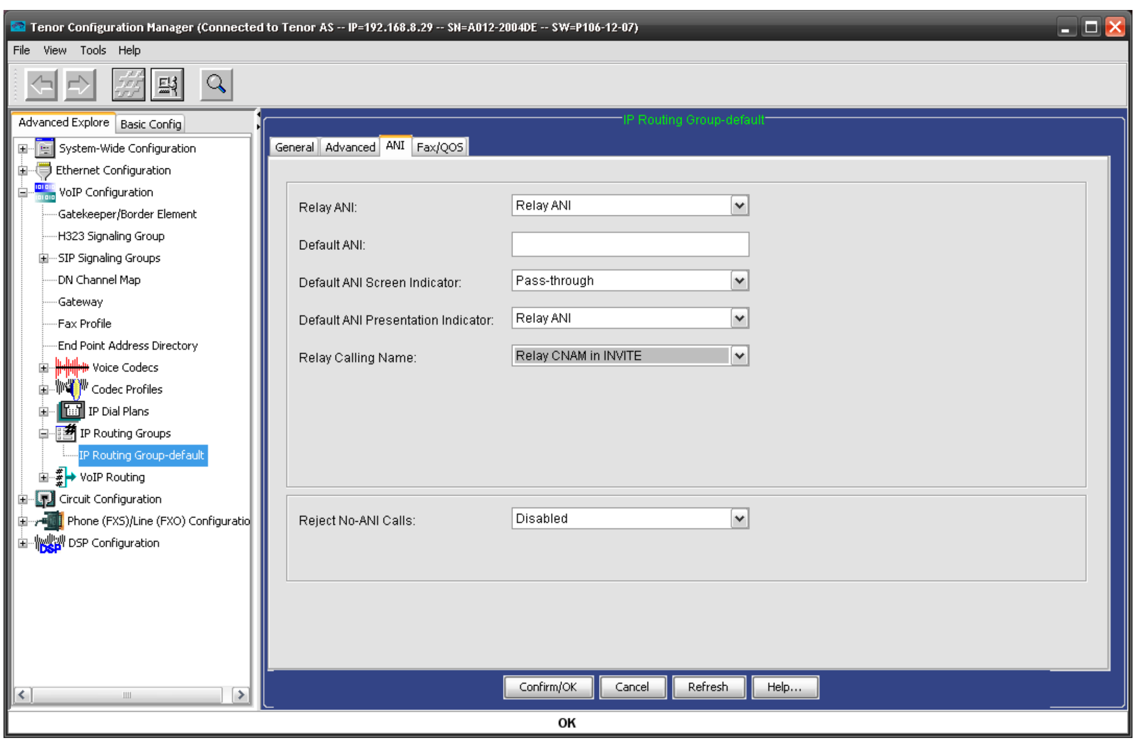
22. Under the **Trunk ID/Caller ID** tab and in the **Trunk ID Delivery** list, choose **Calling Party Number**.



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.



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**.



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.

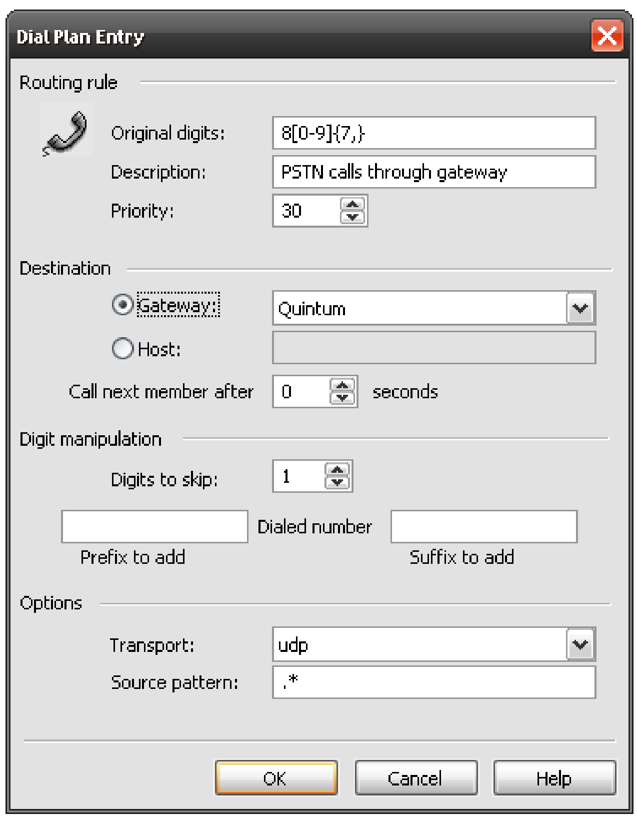
Adding a SIP Gateway

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:



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 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.