Quintum Technologies Tenor
Configuration Guide
For Use with AT&T's
IP Flexible Reach Service

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1 Introduction

This Guide describes the steps for configuring Quintum Analog Tenors, Tenor AF, or Tenor AX to work with AT&T's Flexible Reach Service. The Quintum Tenor AF/AX provides VoIP capability to Analog Telephones, Analog Key Systems and Analog PBXs. Tenor Software release P104.12.02 was tested with the AT&T IP Flexible Reach Service.

The Tenor AF/AX is a gateway device that converts Analog Telephones, Analog Key Systems and Analog PBXs to Voice Over IP (VoIP). The Tenor AF/X uses SIP signaling to complete VoIP telephone calls over the AT&T IP Flexible Reach network. Under normal conditions, calls from Analog devices (Telephones, FAX Machines, Key Systems, PBXs) are converted into VoIP packets by the Tenor AF/AX and sent through the AT&T IP network. The Tenor AF/AX can also use a PSTN connection to bypass the IP data network (AT&T and Customer) if there is an outage.

References in this Service Guide to "Tenor" refer to the Tenor AF or Tenor AX.

Tenor AX Series	**MultiPath AXM Series**	**Station AXG Series**	**Enterprise AXE Series**
8 Simultaneous VoIP Calls	8 FXS/8 FXO	8 FXS/0 FXO	8 FXS/2 FXO
12 Simultaneous VoIP Calls	12 FXS/8 FXO	12 FXS/0 FXO	12 FXS/2 FXO
16 Simultaneous VoIP Calls	16 FXS/16 FXO	16 FXS/0 FXO	16 FXS/2 FXO
24 Simultaneous VoIP Calls	24 FXS/24 FXO	24 FXS/0 FXO	24 FXS/2 FXO
48 Simultaneous VoIP Calls	N/A	N/A	N/A

Tenor AF Series	**MultiPath AFM Series**	**Station AFG Series**	**Enterprise AFE Series**
6 VoIP Calls	N/A	N/A	6 FXS/2 FXO
8 VoIP Calls	N/A	8 FXS/0 FXO	N/A

2 Special Notes

2.1 Emergency 911/E911 Services Limitations

While AT&T IP Flexible Reach services support E911/911 calling capabilities in certain circumstances, there are significant limitations on how these capabilities are delivered. Please review the AT&T IP Flexible Reach Service Guide in detail to understand these limitations and restrictions.

The AT&T IP Flexible Reach Calling Plan A does not support 911 calling. See Section 7 of this document for more details on configuring the Tenor to work with AT&T Calling Plan A.

With AT&T Calling Plans B & C the customer must choose how to route 911 calls based on the dial-plan intelligence capabilities resident in the Customer Premises Equipment (CPE), including Analog Key System, Analog PBX, Analog Phones, and Quintum Tenor. The following scenarios are provided as examples but do not represent a comprehensive list of alternatives. Each customer installation will be unique based on the communications facilities available, the dial-plan intelligence in the CPE, and the service reliability and availability needs of the customer.

2.1.1 911 Service Routing with Analog Key System and Tenor

The customer can choose to route 911 Service calls to the AT&T IP Flexible Reach Network or may alternately deploy (purchase) additional PSTN Lines to provide the 911 calling service. The dial plan intelligence to route 911 calls to AT&T IP Flexible Reach or the PSTN Lines can be provided by the Analog Key System or the Quintum Tenor. Some of the possible scenarios are illustrated below.

Figure 1 illustrates the scenario where the customer Analog Key System and Tenor are configured to route 911 calls to AT&T IP Flexible Reach. Note that this is scenario utilizes the standard Tenor configuration defined in Section 5.2. In this scenario, the network can process as many 911 calls as the customer's Concurrent Call capacity.

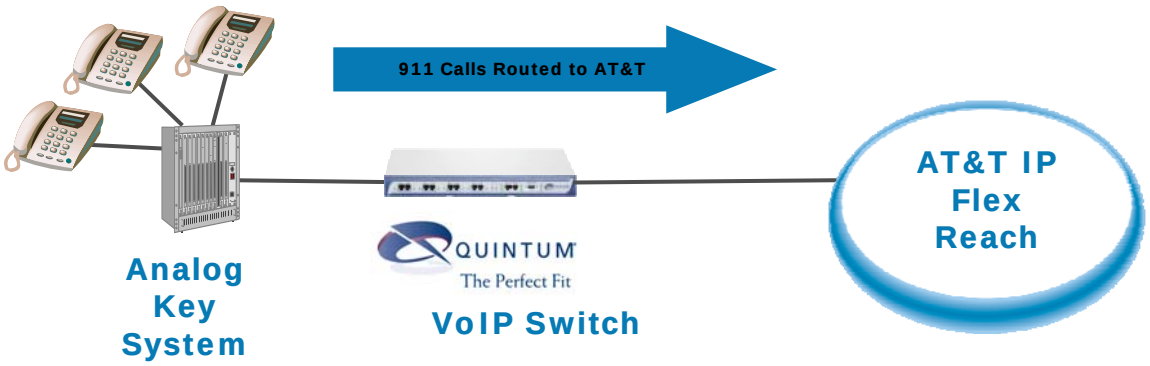


Figure 1 - 911 Calls Routed to AT&T

Figure 2 illustrates the scenario where the customer Tenor is configured to route 911 calls to the PSTN. Note that this scenario utilizes the Tenor ByPass Routing Configuration defined in Section 7.1.1. In this scenario, the system will only be able to handle as many simultaneous 911 calls as there are POTS lines connected to the Tenor.

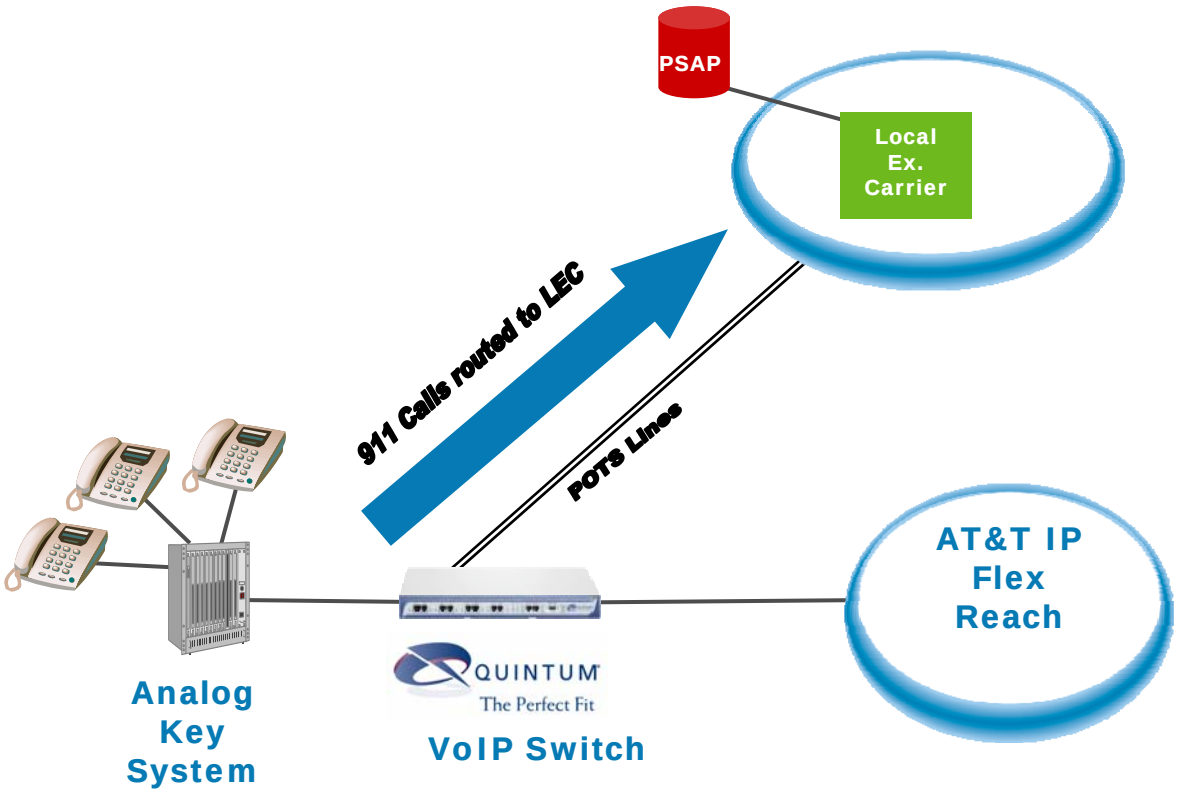


Figure 2 – 911 calls routed to LEC by Quintum Tenor

Figure 3 illustrates the scenario where the customer Analog Key System is configured to route 911 calls to the PSTN. In this scenario, the system will only be able to handle as

many simultaneous 911 calls as there are POTS lines connected to the Analog Key System.

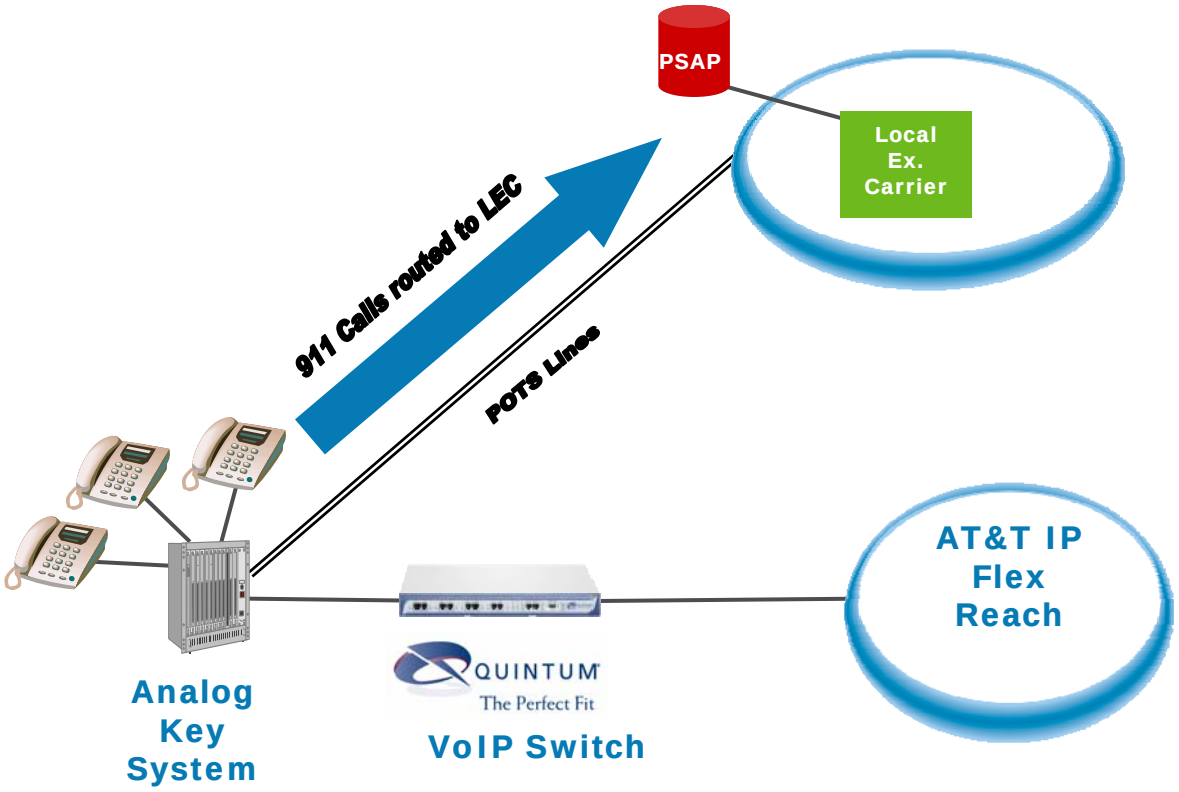


Figure 3 - 911 calls routed to LEC by Analog Key System

2.1.2 911 Service Routing with Analog PBX with Tenor

Figure 4 illustrates the scenario where the customer Analog Key System and Tenor are configured to route 911 calls to AT&T IP Flexible Reach. Note that this is scenario utilizes the standard Tenor configuration defined in Section 5.2. In this scenario, the network can process as many 911 calls as the customer's Concurrent Call capacity.

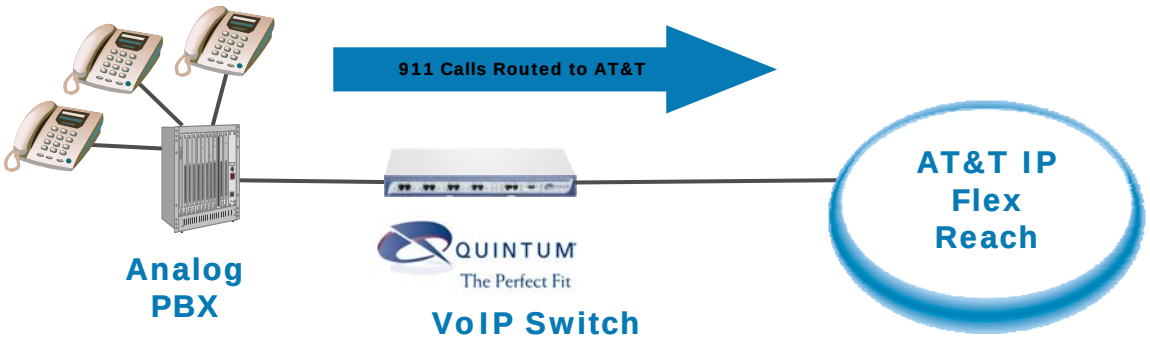


Figure 4 – 911 Calls Routed to AT&T

Figure 5 illustrates the scenario where the customer Tenor is configured to route 911 calls to the PSTN. Note that this scenario utilizes the Tenor ByPass Routing Configuration defined in Section 7.1.1. In this scenario, the system will only be able to handle as many simultaneous 911 calls as there are POTS lines connected to the Tenor.

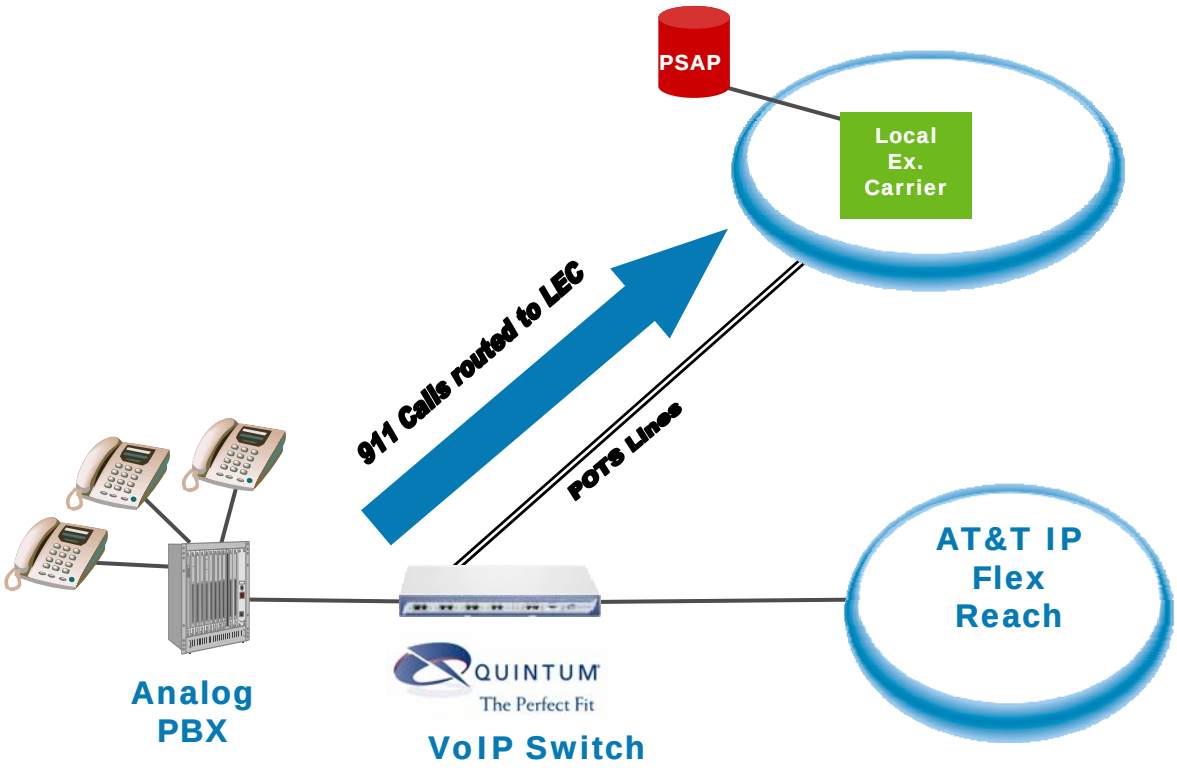


Figure 5 -911 Calls Routed to LEC by Quintum Tenor

Figure 6 illustrates the scenario where the customer Analog PBX is configured to route 911 calls to the PSTN. In this scenario, the system will only be able to handle as many simultaneous 911 calls as there are POTS lines connected to the Analog PBX.

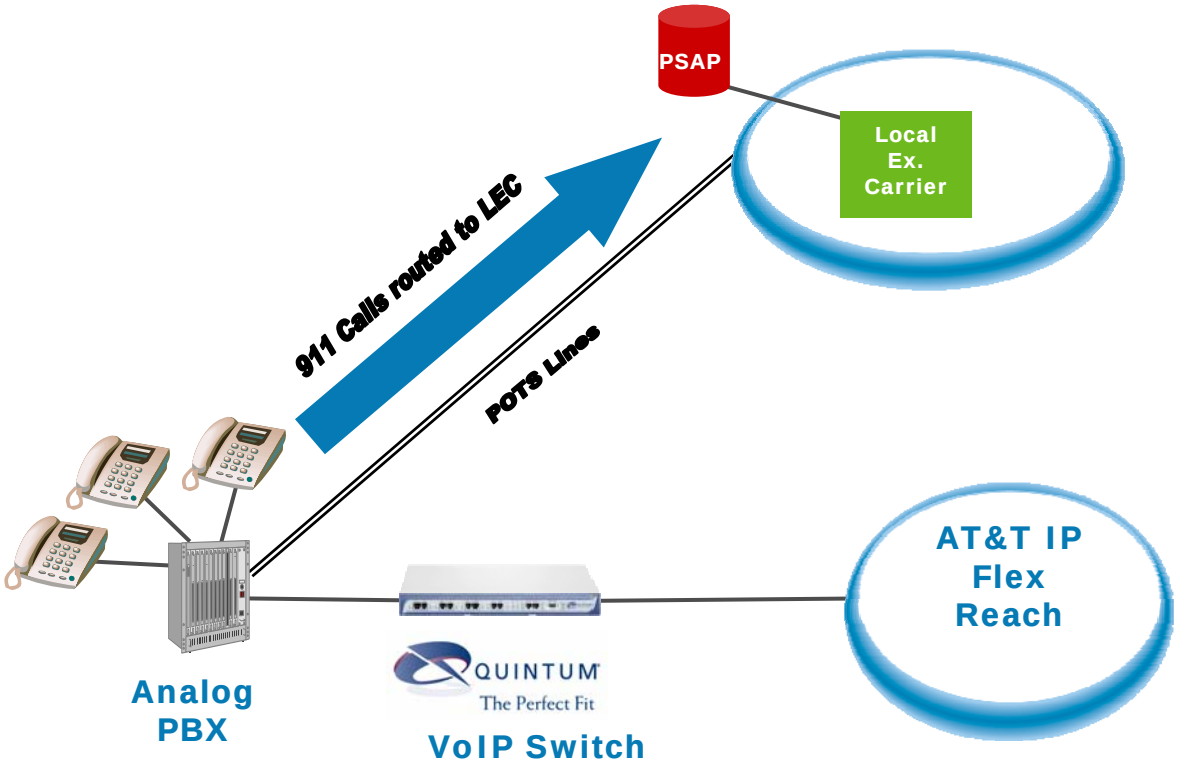


Figure 6 -911 calls routed to LEC by Analog Key System

2.1.3 911 Service Routing with Analog Phones with Tenor

Figure 7 illustrates the scenario where the Tenor is configured to route 911 calls to AT&T IP Flexible Reach. Note that this is scenario utilizes the standard Tenor configuration defined in Section 5.2. In this scenario, the network can process as many 911 calls as the customer's Concurrent Call capacity.

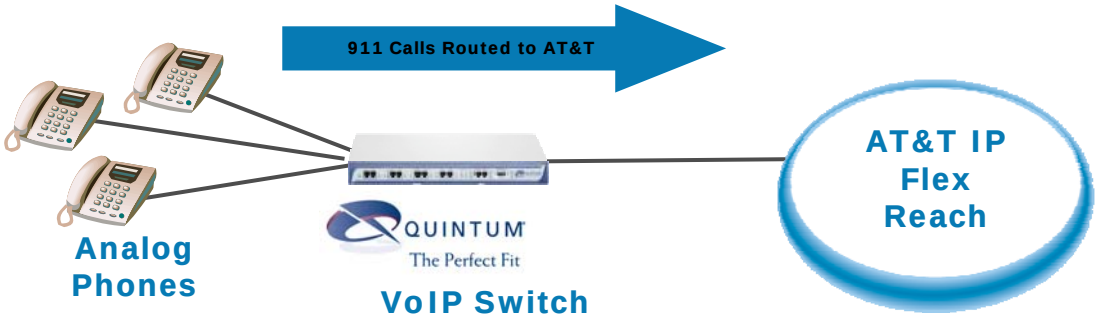


Figure 7 - 911 Calls Routed to AT&T

Figure 8 illustrates the scenario where the customer Tenor is configured to route 911 calls to the PSTN. Note that this scenario utilizes the Tenor ByPass Routing Configuration defined in Section 7.1.1. In this scenario, the system will only be able to handle as many simultaneous 911 calls as there are POTS lines connected to the Tenor.

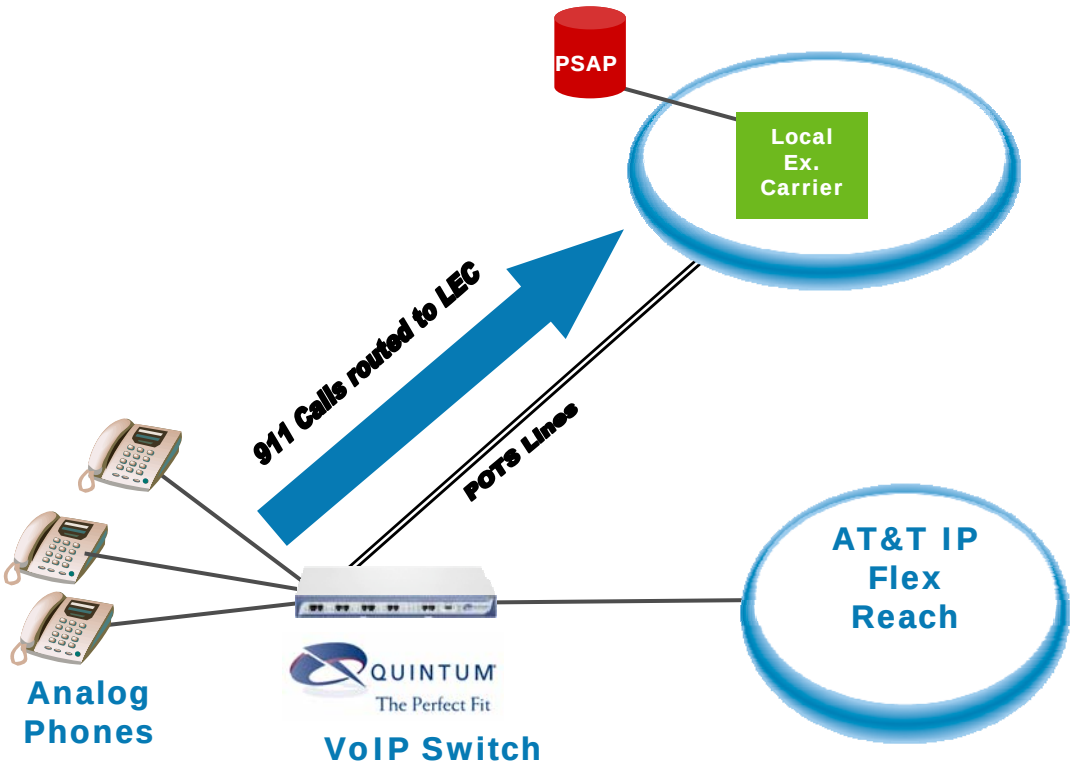


Figure 8 - 911 Calls Routed to LEC by Quintum Tenor

2.2 Tenor Transfer Feature Must be Turned Off

The Tenor IP transfer feature must be turned off. Quintum and AT&T do not currently have a compatible mechanism for handling IP transfers.

2.3 Session Description Protocol in Session Progress and Ringing Messages

Session description protocol in the session progress and ringing messages must be turned off in the Tenor.

3 Capabilities Overview

The Quintum Technologies Tenor supports the following capabilities in conjunction with the AT&T Flexible Reach service.

3.1 Calling Scenarios Supported

Inbound Calls to the Quintum Tenor

- Offnet gateway inbound to TENOR
  - Local, Long Distance, International, FAX
- TENOR to TENOR
  - Local, Long Distance, International, FAX
- TDM PBX to TENOR
  - Local, Long Distance, International, FAX

Outbound Calls from the Tenor

- TENOR to PSTN Offnet gateway
  - Local, Long Distance, International, FAX
- TENOR to TENOR
  - Local, Long Distance, International, FAX
- TENOR to TDM PBX
  - Local, Long Distance, International, FAX

Support for AT&T IP Flexible Reach Calling Plan A Dial Plan

The AT&T IP Flexible Reach Calling Plan A will only provide calling service to On-NET end-points and Off-Net calling to Long Distance and International locations. The AT&T Calling Plan A will not terminate calls to N11 (ex. 211, 311, 411, 511, 611, 711, 811, 911), 8YY-XXX-XXXX, 500-NPA-NXX-XXXX , 700-NPA-NXX-XXXX, 900-NPA-NXX-XXXX, NPA-555-XXXX, and Operator (0, 0+, 00, 01) numbers. See the AT&T Business VoIP Service Guide for more details.

With Calling Plan A, the customer is responsible for providing PSTN lines to support 911 and the other calling services (N11, 8YY, etc.) not provided by AT&T. The customer must provision/configure his premise equipment (Analog PBX, Analog Phones, Key System, Tenor, etc.) to properly route 911 and other non AT&T supported calling services to the PSTN lines.

Section 7 of this document provides details on how the Tenor can be configured to support the AT&T IP Flexible Reach Calling Plan A.

3.2 Routing Scenarios Supported

- Failover from Primary to Secondary AT&T IP Border Element
- Failover to PSTN when IP network unavailable
- Routing 911 calls for these scenarios:
    - IP FR with Key system and Tenor
    - IP FR with Analog phones
    - IP FR with analog PBXs

3.3 Codecs Supported

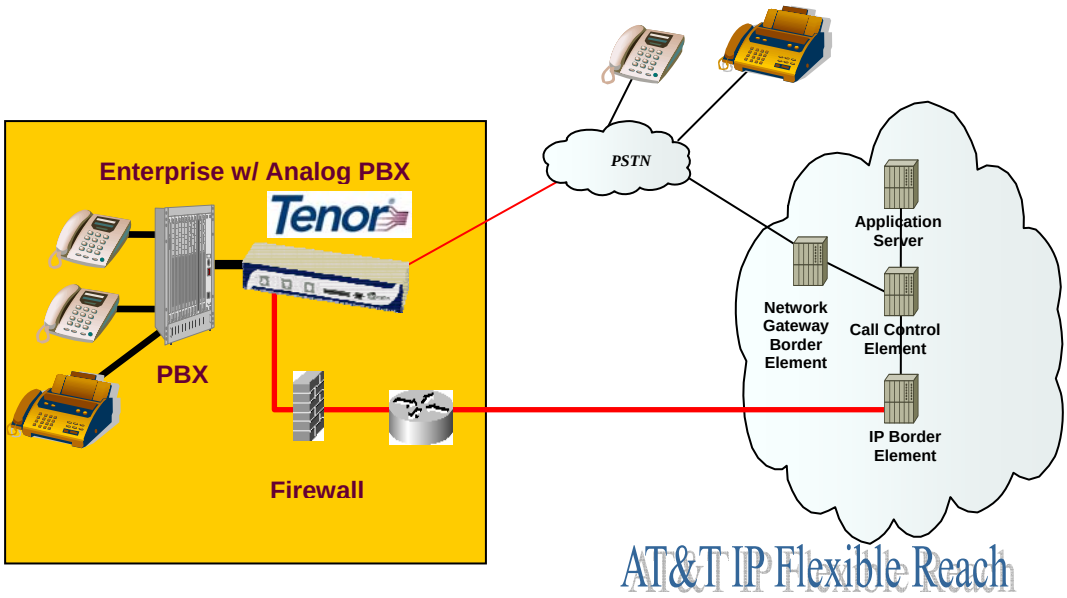
- G.729AB 8.0 Kbps
- G.711 A-law 64Kbps
- G.711 Mu-law 64Kbps

3.4 Features Supported

Virtual Telephone Number Support
Calling Name Delivery
Call Hold and Resume
DTMF Relay (midcall digits)
In Band DTMF
FAX over IP
Survivability Via PSTN on select models (AFM, AFE, AXM and AXE)

4 Configuration Component Overview

This section provides a service overview of the Vendor integration with AT&T IP Flexible Reach.



The customer premises equipment shall consist of the following components.

- Customer PBX or Key System with standard 2-wire Analog interface connections for FXO ports.
  Note: The Tenor AX provides a standard Centronics 50 pin male interface (50 pin / 25 pair male Amphenol connector).

  OR

  Customer Analog Phones and/or FAX machines with RJ11 interfaces.
  Note: The Tenor AX provides standard RJ11 interfaces.

- AT&T Managed Router

- Customer optional Firewall

- Quintum Technologies Tenor Analog Gateway (AX or AF).

Tenor Software release P104.12.02 was used when conducting interoperability testing with the AT&T IP Flexible Reach Service.

4.1 Tenor AX Overview

The *Tenor AX* is a high-density VoiP (Voice over Internet Protocol) SIP/H.323 switch that compresses and packetizes voice, fax, and modem data and transmits it over the IP network. The *Tenor AX* gives larger businesses with analog voice infrastructure a means to use Voice over IP (VoIP).

The Tenor's MultiPath architecture enables it to intelligently route calls between the FXS, FXO, and the VoIP network. The *Tenor AX* also routes calls over IP to reduce costs, and then transparently "hop off" to the PSTN, to reach off-net locations. Calls can be routed in any direction between any of the ports.

The unit's plug and play embedded system architecture brings VoIP technology to your network without changing your existing telephony infrastructure. The Customer's network stays as is, and the call type is transparent to the user.

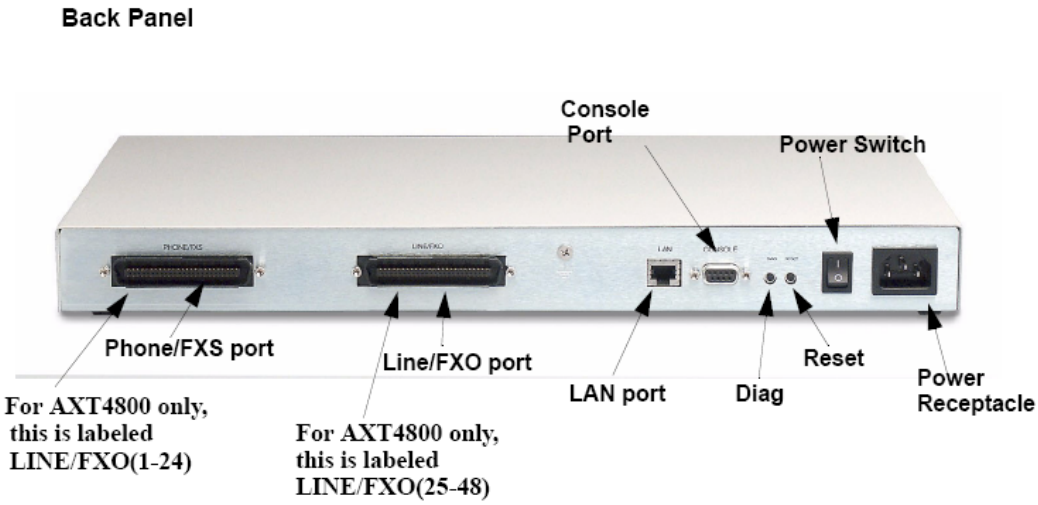


Figure 9 - Tenor AX Back Panel

- **Phone/FXS port -** Provides a 50 Pin Telco connector which supports up to 24 Phone/FXS connections for connecting to the analog PBX, Keyphone or phones.

- **Line/FXO port -** Provides a 50 Pin Telco connector which supports up to 24 Line/FXO connections for connection to the Central Office (connection to the PSTN).

- **LAN port -** 10/100 Base-T Ethernet port. This port provides an RJ-45 jack for individual connection to a 10/100 Ethernet LAN switch or hub via RJ-45 cable; it is individually configured with a unique IP and MAC address.

The Tenor AX will support 8, 16, 24 or 48 Simultaneous VoIP Calls.

AX GENERAL SPECIFICATIONS
Dimensions: 1U High Chassis
W 17 3/8" x H 1 3/4" x D 10 3/4"
W 44.5cm x H 4.5cm x D 27.6cm

- Maximum weight: 10 lbs. (4.55kg)
- AC Power: 100-240 Volts AC, 50/60 Hz, 60 watts

- Operating temperature: 40° - 104° F (5° - 40° C)
- Operating humidity: 20% - 80% non-condensing
- Telco: FCC Part 68, TS-016, TBR4, TS-038, CS03
- EMC: FCC Part 15 EN55022, EN55024, EN61000-2-3, EN61000-3-3, AS/NZS3260
- Safety: UL60950, EN60950, AS/NZS60950

TENOR AX CONFIGURATIONS

Tenor AX Series	MultiPath AXM Series	Station AXG Series	Enterprise AXE Series
8 Simultaneous VoIP Calls	8 FXS/8 FXO	8 FXS/0 FXO	8 FXS/2 FXO
12 Simultaneous VoIP Calls	12 FXS/8 FXO	12 FXS/0 FXO	12 FXS/2 FXO
16 Simultaneous VoIP Calls	16 FXS/16 FXO	16 FXS/0 FXO	16 FXS/2 FXO
24 Simultaneous VoIP Calls	24 FXS/24 FXO	24 FXS/0 FXO	24 FXS/2 FXO
48 Simultaneous VoIP Calls	N/A	N/A	N/A

For more details on the Tenor AX, consult with document [1].

4.2 Tenor AF Overview

The *Tenor AF* is a VoIP (Voice over Internet Protocol) H.323/SIP switch that digitizes voice, fax, and modem data and transmits it over the IP network. The *Tenor AF* gives small to medium sized businesses with analog voice infrastructure a means to use Voice over IP (VoIP).

The Tenor's MultiPath architecture enables it to intelligently route calls between the FXS, FXO, and the VoIP network to achieve the best combination of cost and quality. The *Tenor AF* also routes calls over IP to reduce costs, and then transparently "hop off" to the PSTN, to reach off-net locations. Calls can be routed in any direction between any of the ports.

The Tenor can be installed without upgrades to the existing voice or data network. You can install the unit in a home or office environment, without affecting the network infrastructure you already have in place.

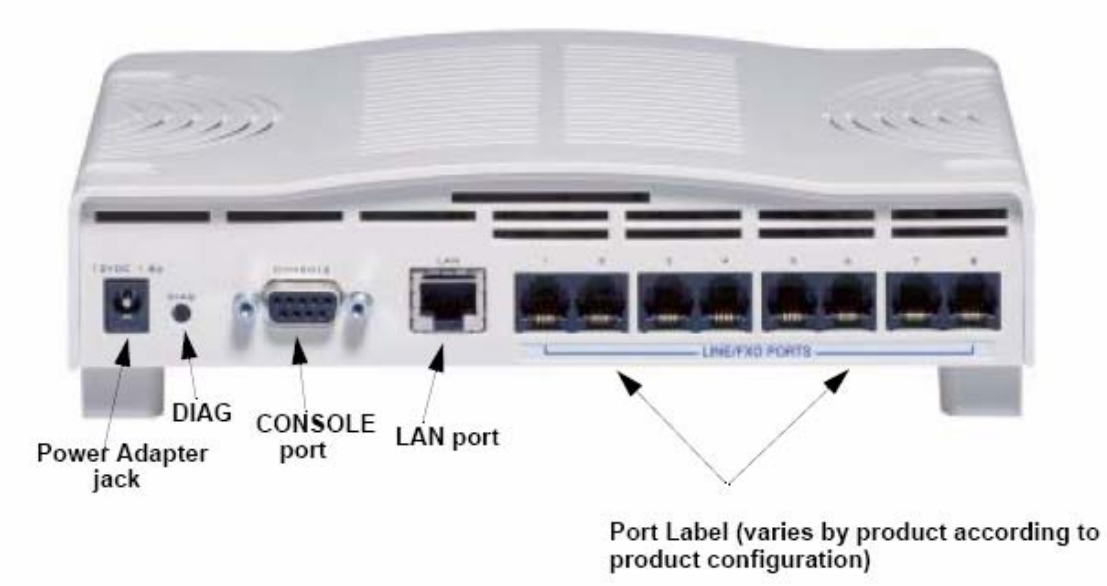


Figure 10 - Tenor AF Back Panel

- **Power Adapter jack -** Connection port to external power supply.
- **DIAG -** Enables you to perform software diagnostic procedures.
- **CONSOLE port -** This RS-232 connector is used for connection to a PC's serial port via a DB-9 serial cable at 38400 bps 8 N 1, no flow control.
- **LAN port -** 10/100 Base-T Ethernet port. This port provides an RJ-45 jack for an individual connection to a 10/100 Ethernet LAN switch or hub via RJ-45 cable; the interface is individually configured with a unique IP and MAC address.
- **Port Label (Phone/FXS or Line/FXO ports) - For Phone/FXS,** provides an RJ-11 jack for connection to an analog PBX, Keyphone or analog phone. **For Line/FXO**, enables connection to another piece of equipment that houses your telephone lines running to the PSTN, such as the patch panel.

AF GENERAL SPECIFICATIONS

Dimensions: 1U High Chassis
W 8 1/4" x H 2" x D 7"
W 21cm x H 5.1cm x D 18.73cm

- Maximum weight: 1.3 lbs. (0.6kg)
- AC Power: 100-240 Volts AC, 50/60 Hz, 22 watts
- Operating temperature: 40° - 104° F (5° - 40° C)
- Operating humidity: 20% - 80% non-condensing
- Telco: FCC Part 68, AS/ACIF S003, CS03, JATE, AS/ACIFS002:2001
- EMC: FCC Part 15 Class B, EN55022, EN55024,

- EN61000-2-3, EN61000-3-3, AS/NZS3260
- Safety: UL60950, EN60950, AS/NZS60950

The Tenor AF will support 6 or 8 Simultaneous VoIP Calls based on the configuration purchased.

TENOR AF CONFIGURATIONS

Tenor AF Series	MultiPath AFM Series	Station AFG Series	Enterprise AFE Series
6 VoIP Calls	N/A	N/A	6 FXS/2 FXO
8 VoIP Calls	N/A	8 FXS/0 FXO	N/A

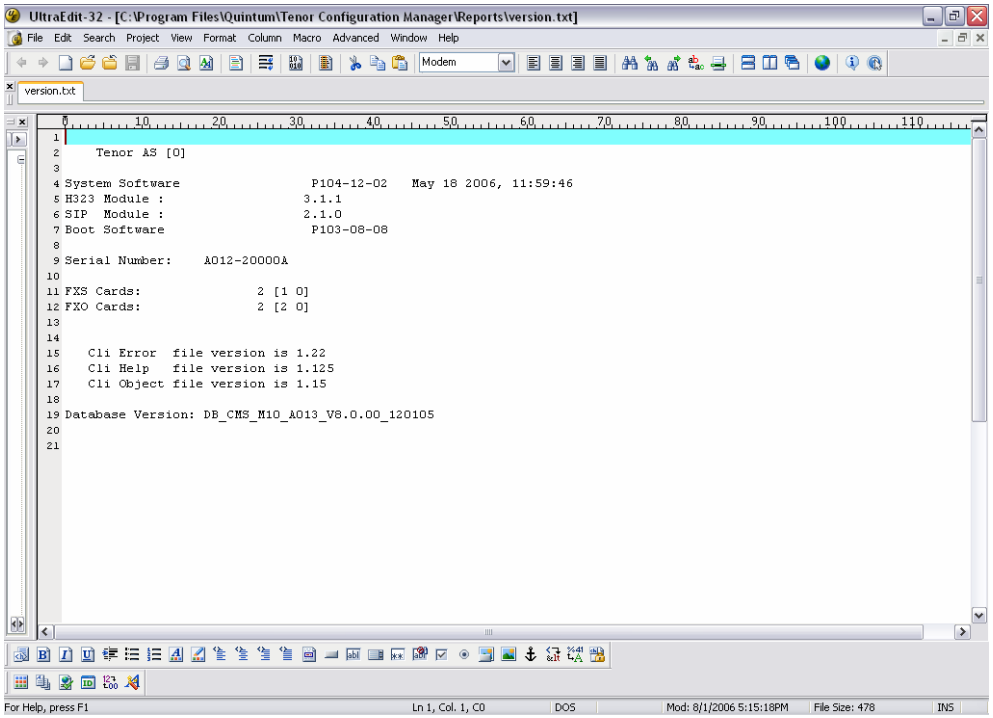
For more details on the Tenor AF, consult with document [2].

5 Configuration Guide

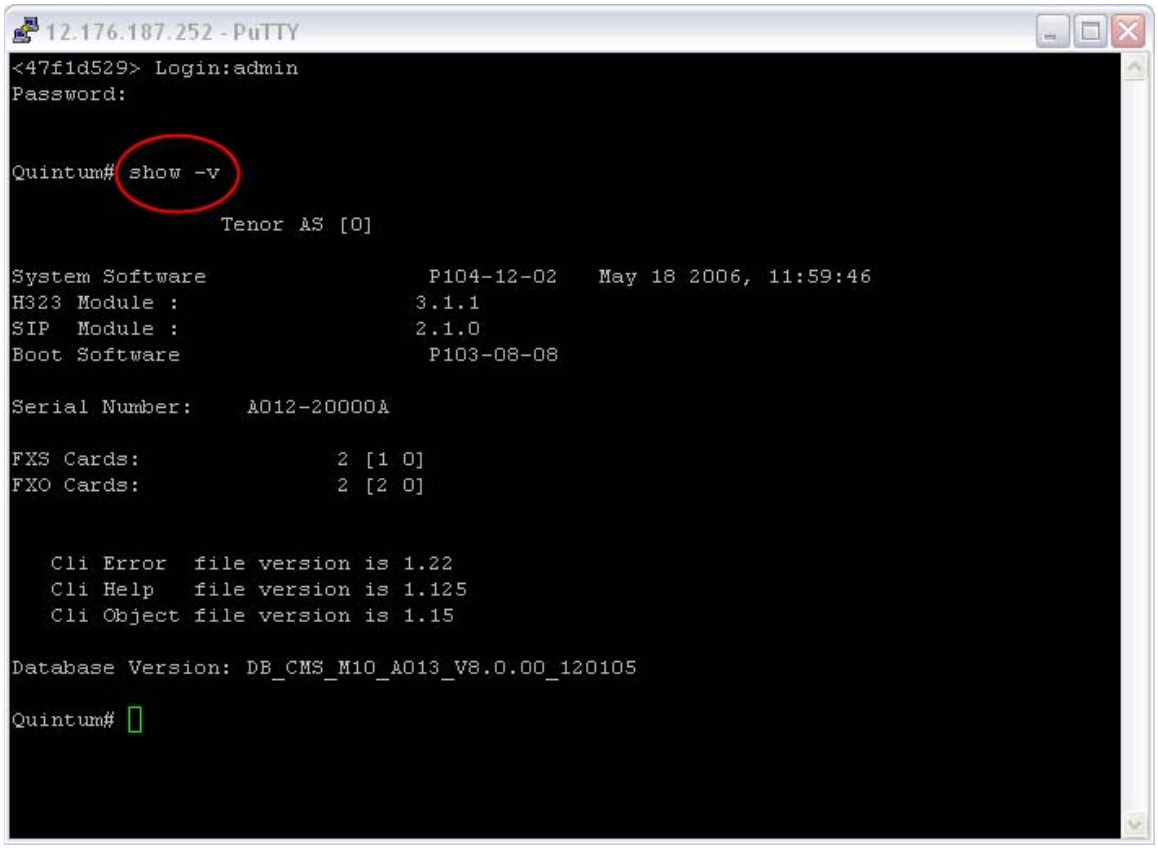
5.1 Tenor Software Version

The version of the Tenor Software can be obtained via the Tenor Configuration Manager GUI or the Command Line Interface (CLI).

From the Configuration Manager View Menu, click on “Tenor Version”.
A text file will open in a new window displaying the Software version as shown below.



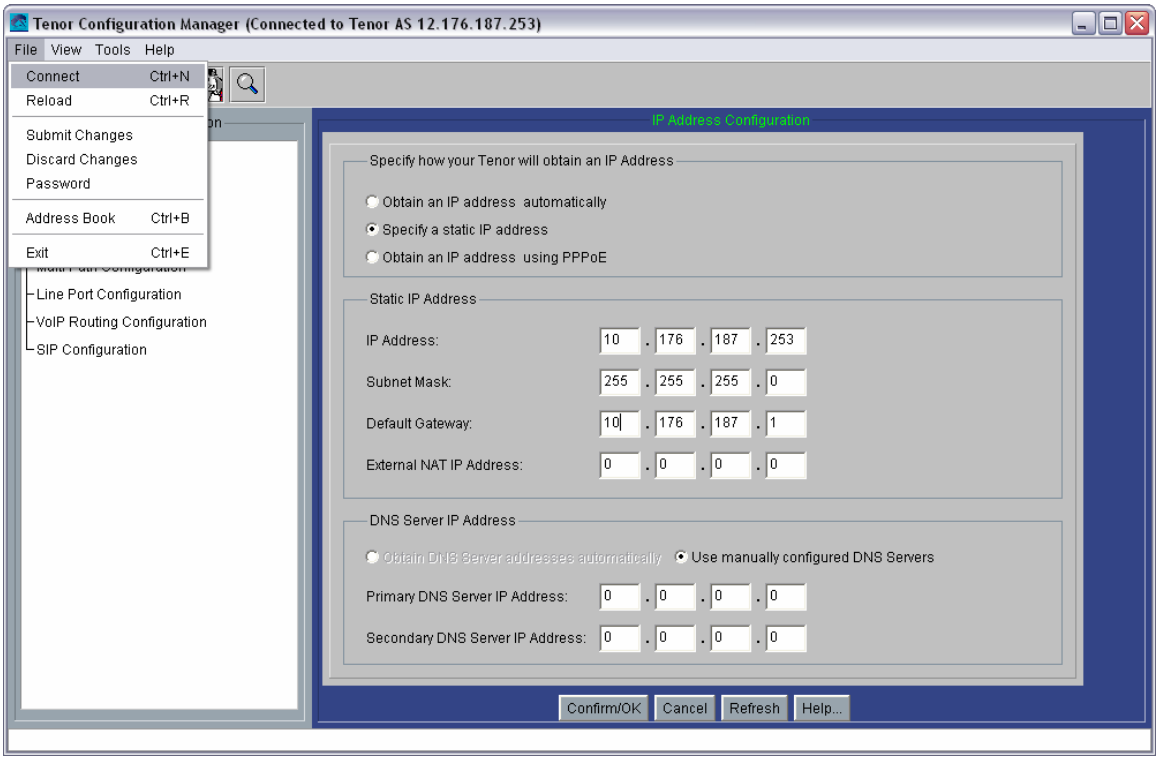
As shown below, the CLI Command to display the Tenor Software version information is “show –v”.

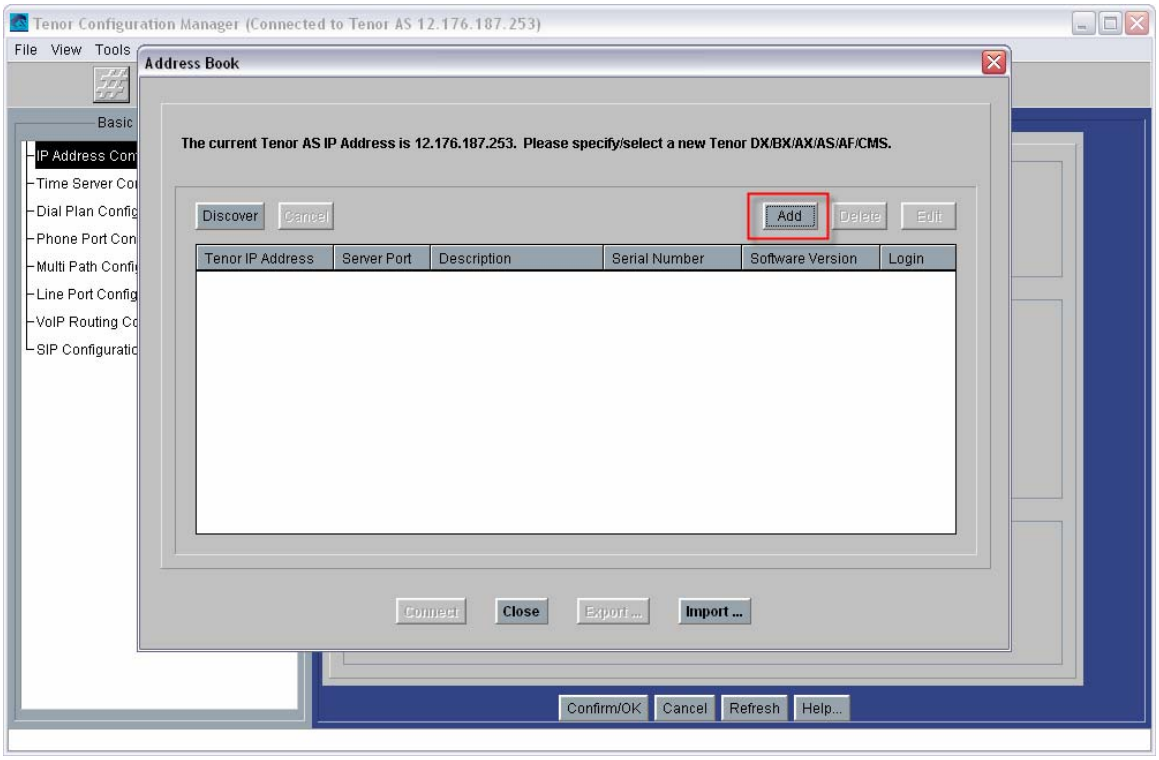


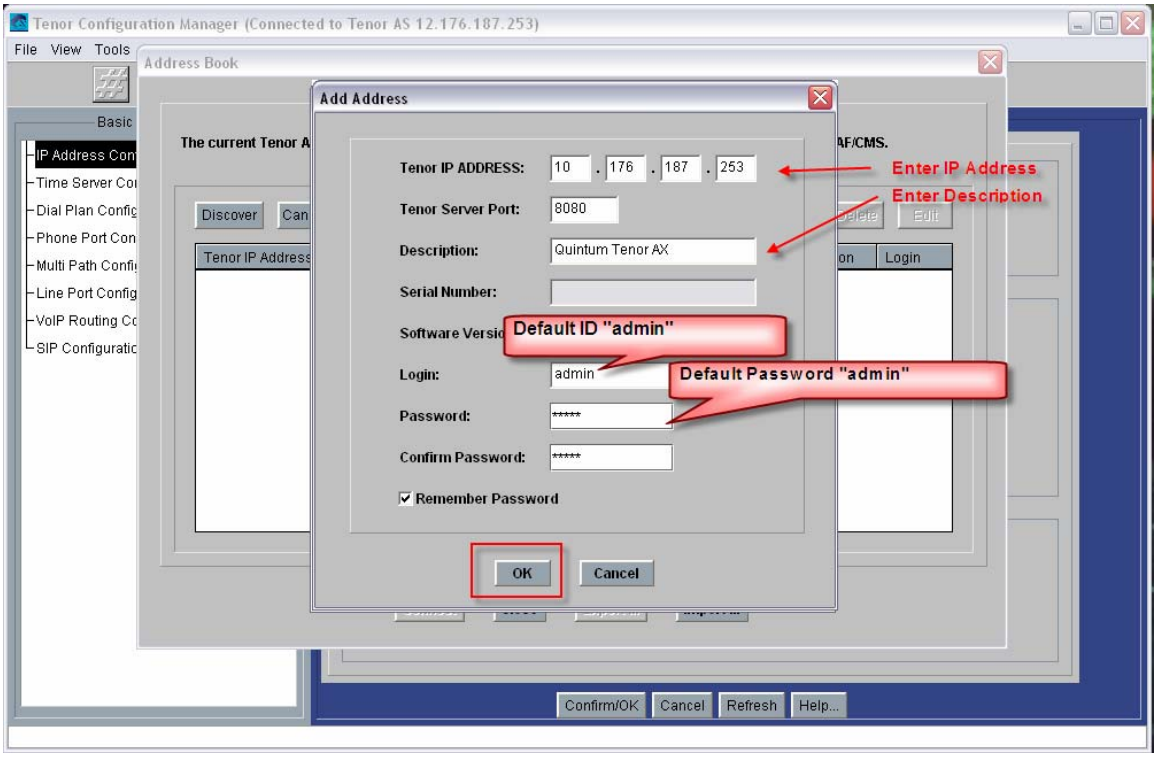
For technical support on the Quintum Tenor AF and Tenor AX, contact Quintum at 877-435-7553, and also refer to www.quintum.com

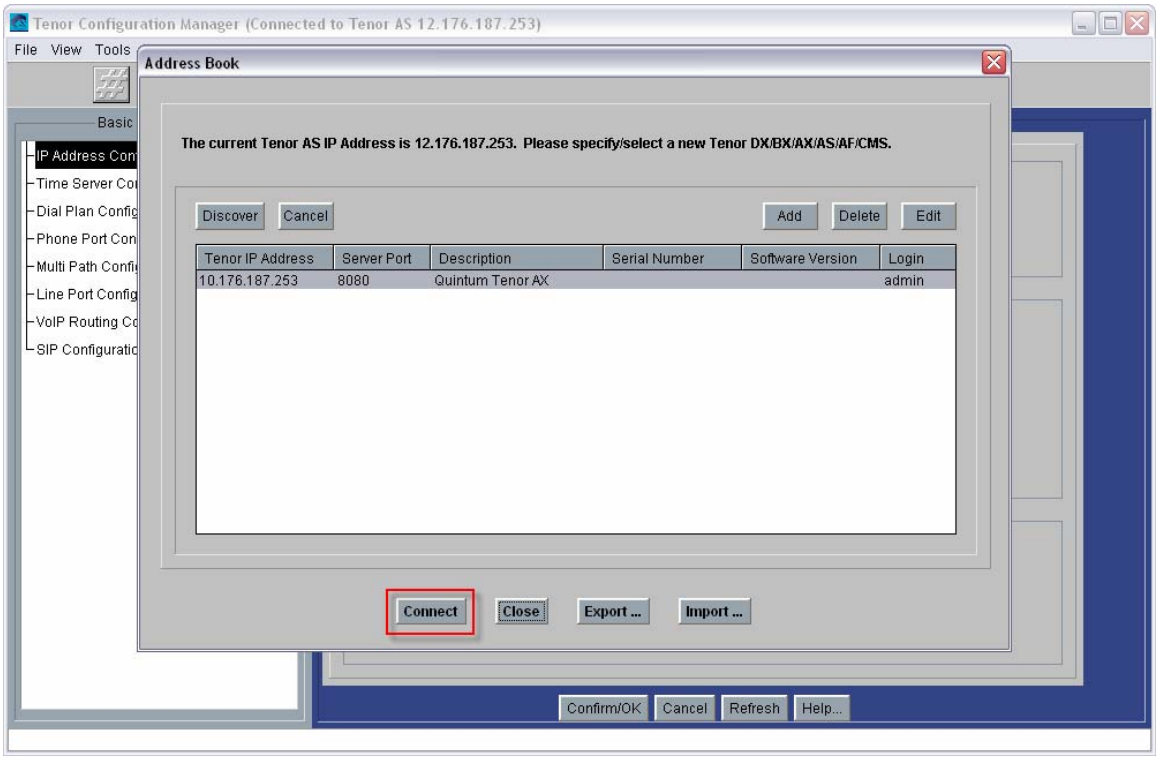
5.2 Standard Configuration

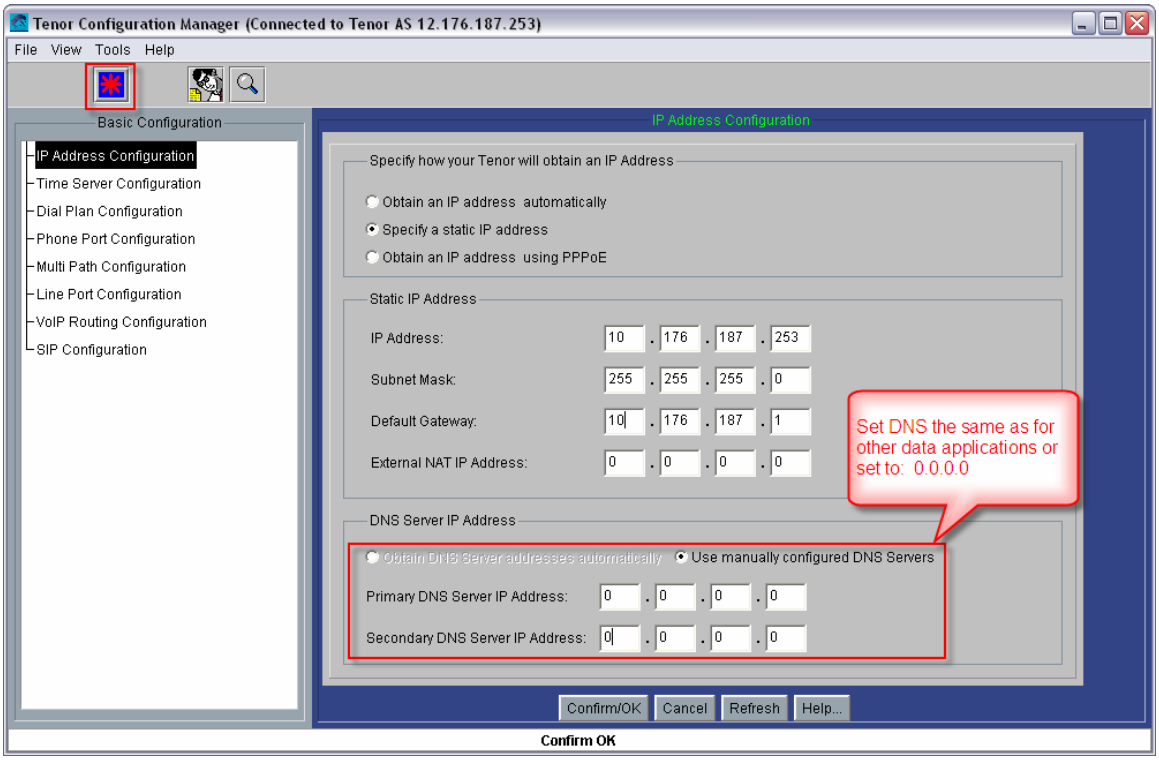
The following steps describe the configuration for the Tenor AF Multipath Gateway Switch verified to work with the AT&T IP Flexible Reach service. Configuration for the Tenor AX is the same as the Tenor AF described below. For detailed information on installing and running Tenor Configuration Manager, consult documents [1], [2] and [3].

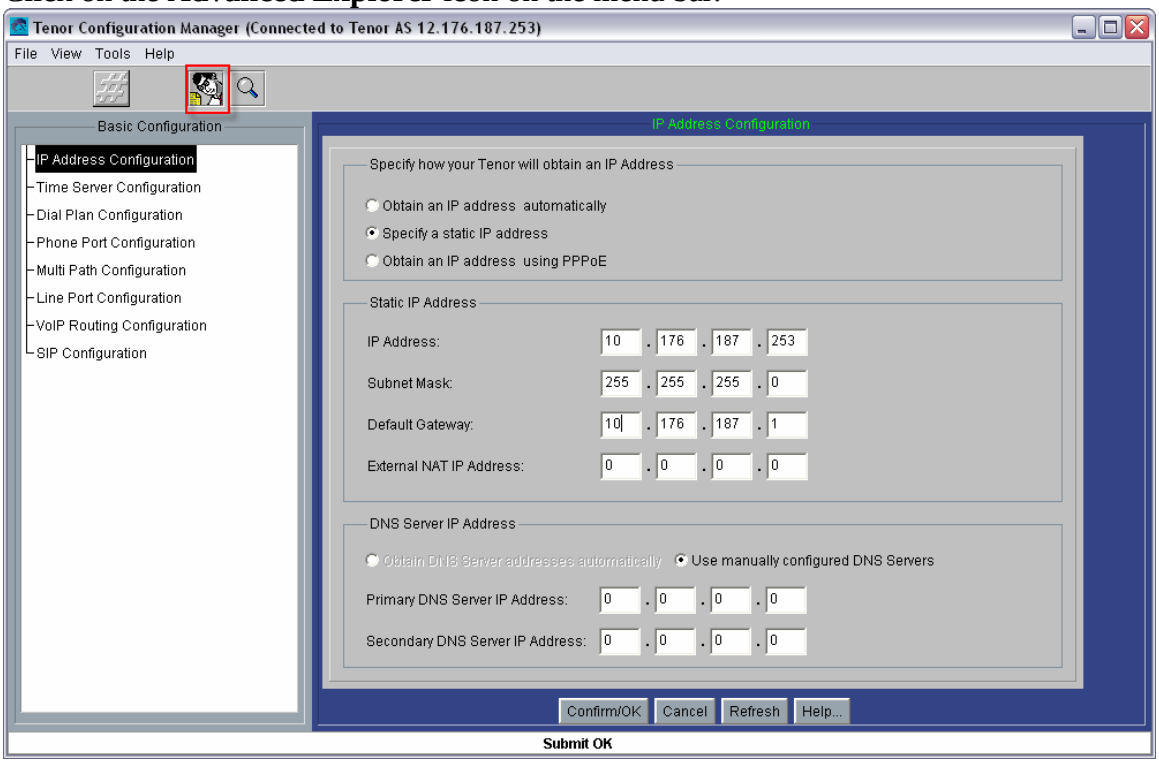
Step	Description
1.	Run the Tenor Configuration Manager. From the File Menu click on **Connect**. 

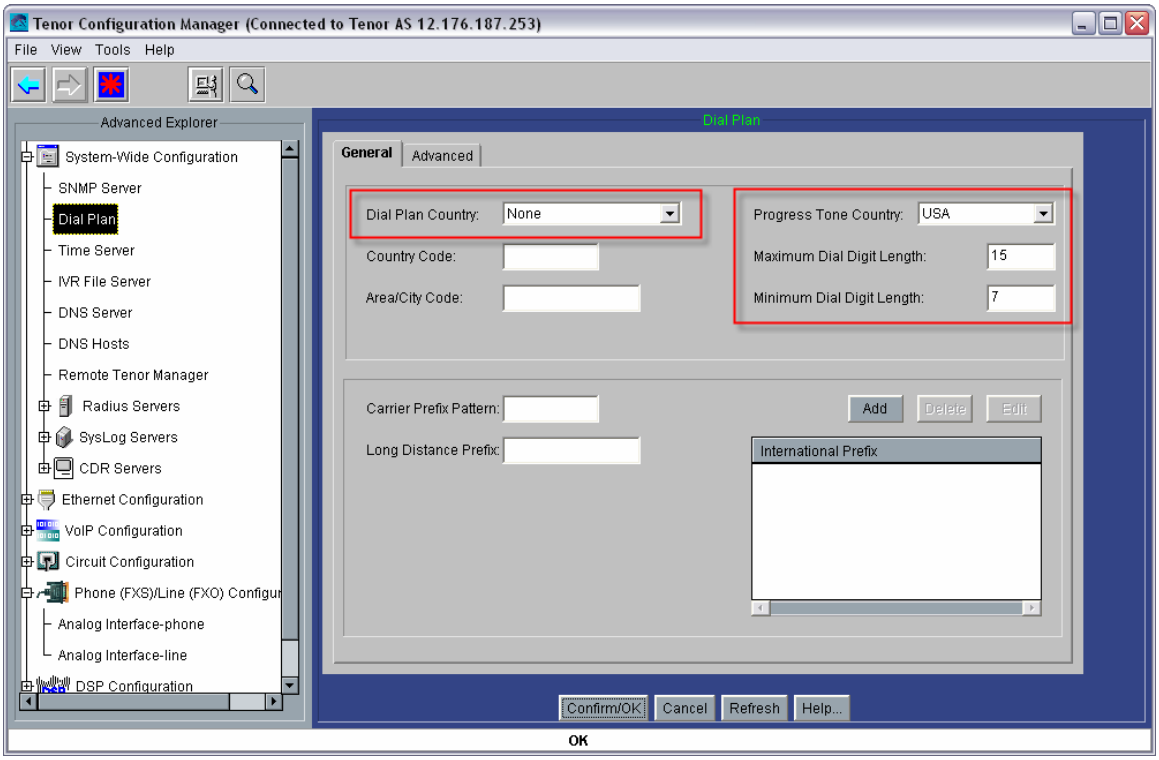
Step	Description
2.	Click on **Add.** 

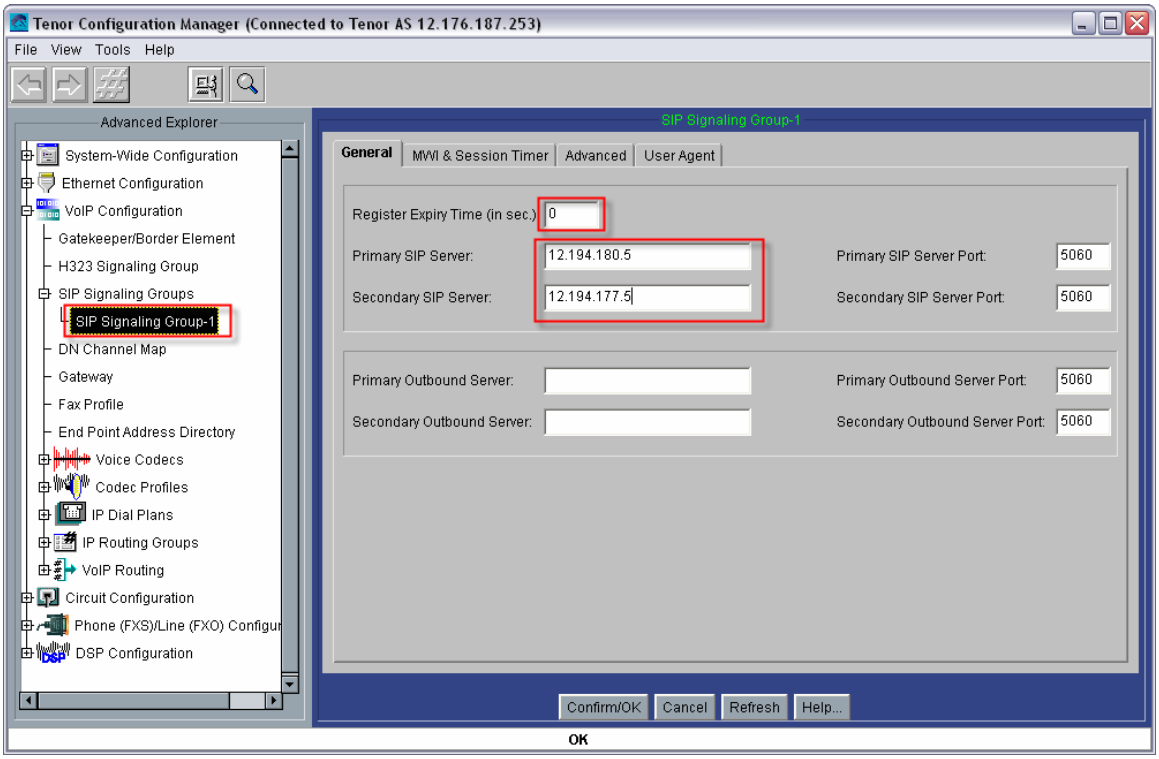
Step	Description
3.	Enter the Tenor **IP Address**, a **Description**, and the **Login ID** and **Password.** Click on **OK.** 

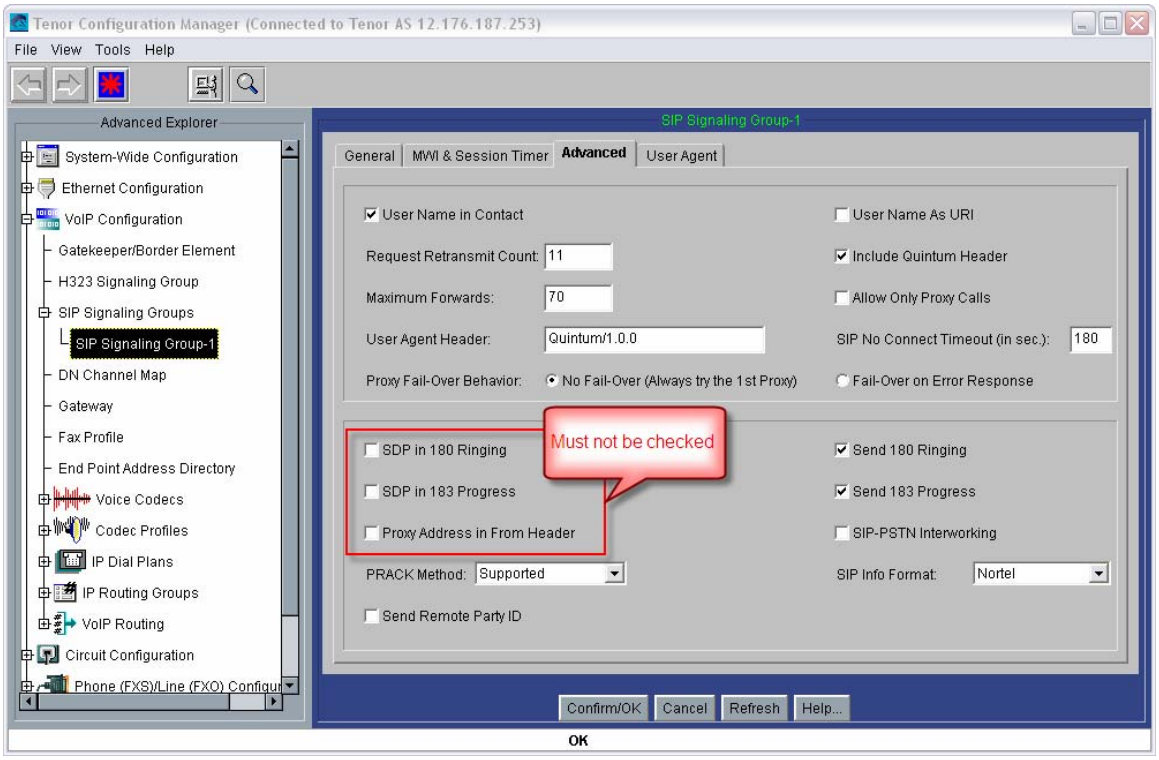
Step	Description
4.	Connect to the Tenor from the Tenor Configuration Manager. Highlight the Tenor switch and click on **Connect**. 

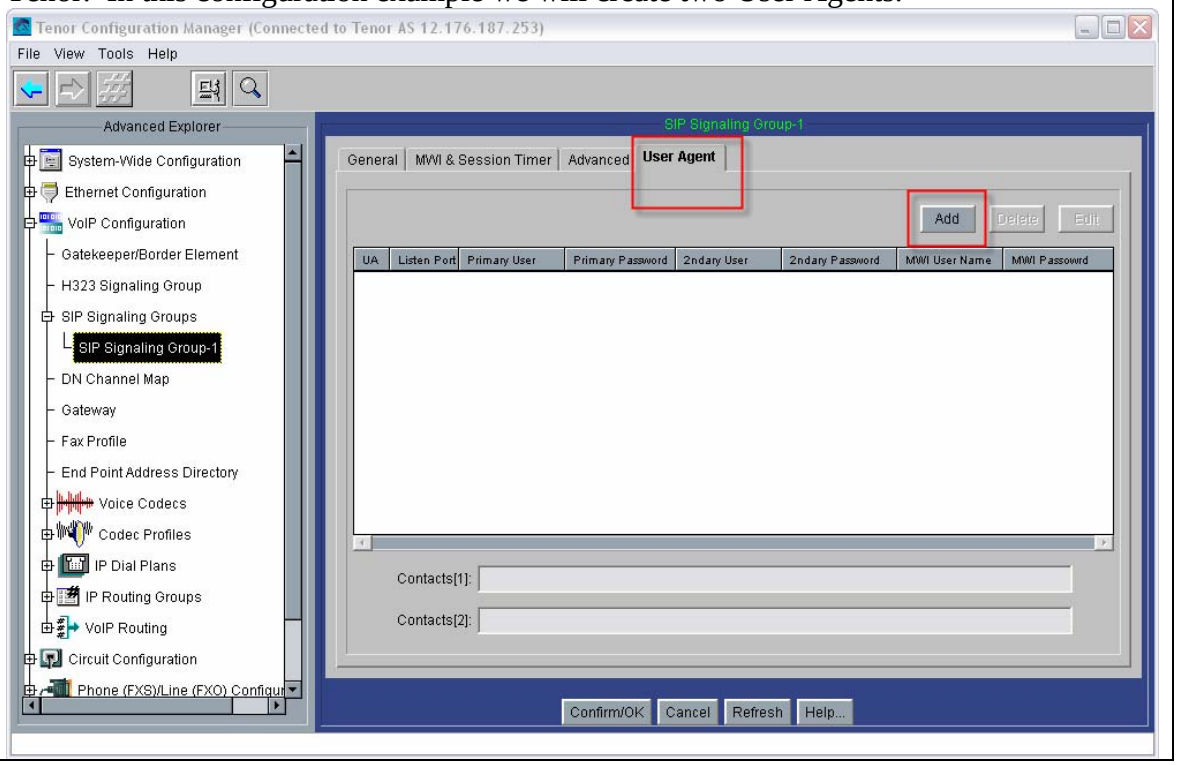
Step	Description
5.	Enter values for the Primary and Secondary DNS Server IP Address. If not using DNS enter: 0.0.0.0 Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change. 

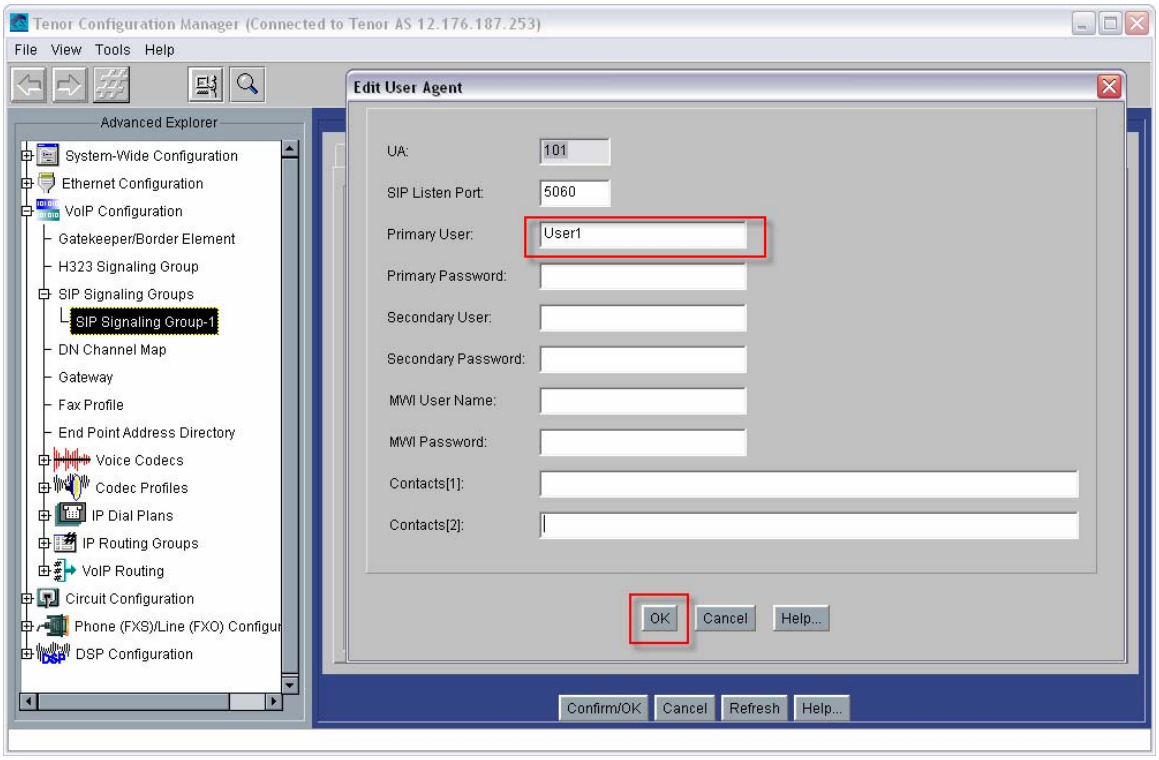
Step	Description
6.	Click on the **Advanced Explorer** icon on the menu bar. 

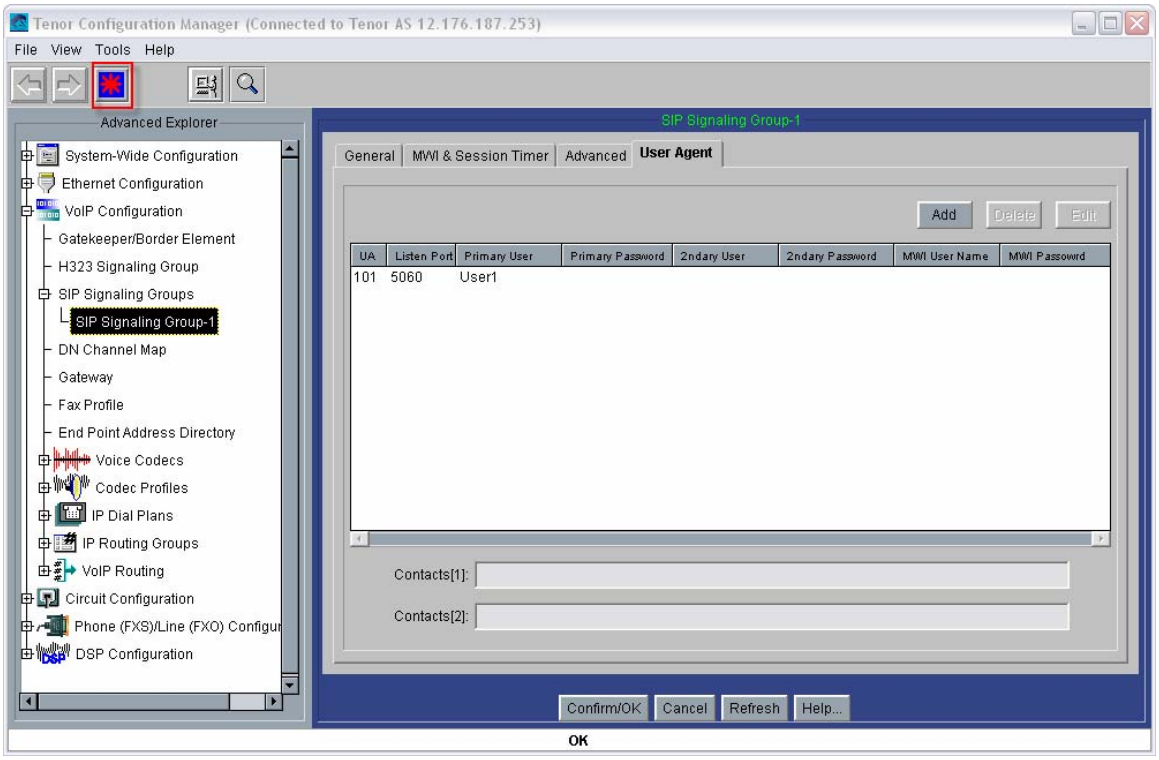
Step	Description
7.	From the **Advanced Explorer** panel on the left, highlight the **Dial Plan** field. Select the desired **Dial Plan Country** from the drop down menu. The sample configuration uses ***None***. Select the desired **Progress Tone Country** setting from the drop down menu. The sample configuration uses ***USA***. Enter values for the Minimum and Maximum dial digit string length. Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change. 

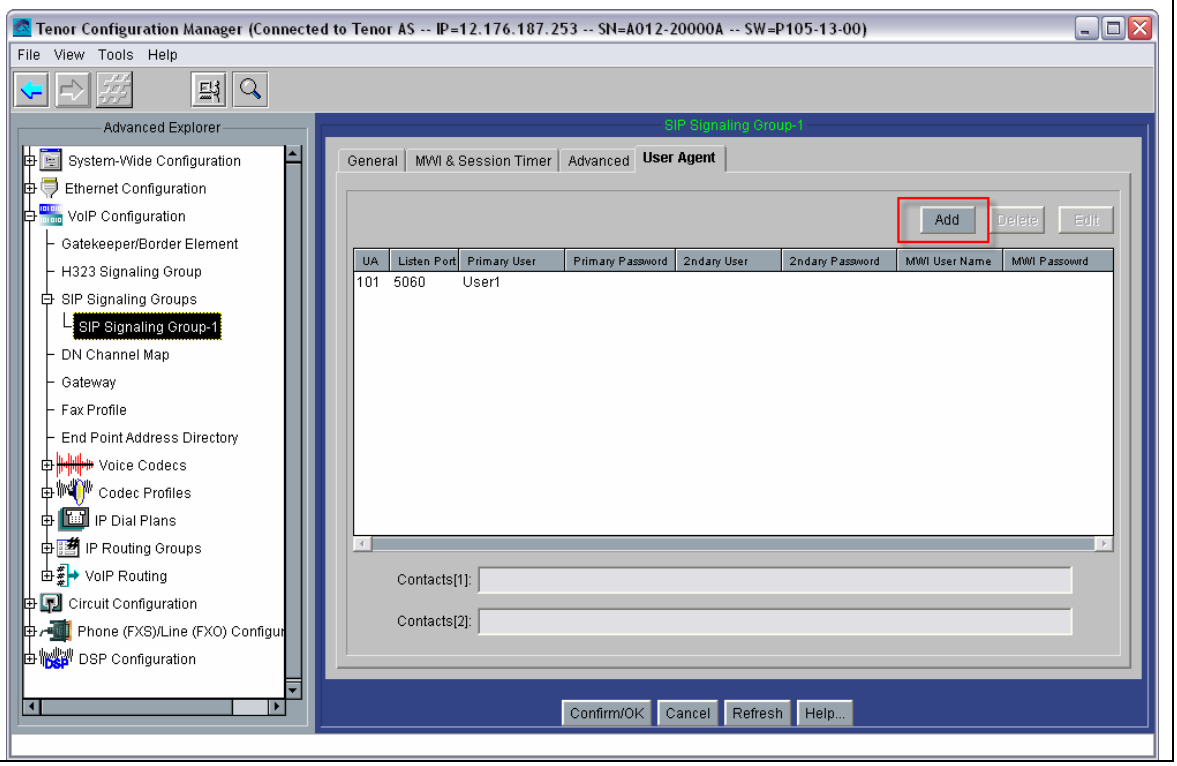
Step	Description
8.	From the **Advanced Explorer** panel on the left, click on the **+** sign next to **VoIP Configuration → SIP Signal Groups** to expand the field. Highlight the **SIP Signaling Group-1** field. Under the **General** tab, enter the **Primary SIP Server IP Address** and the **Secondary SIP Server IP Address** (IP Addresses of AT&T Primary and Secondary IP Border Elements). To disable Registration, enter the **Register Expiry Time** of **0.** 

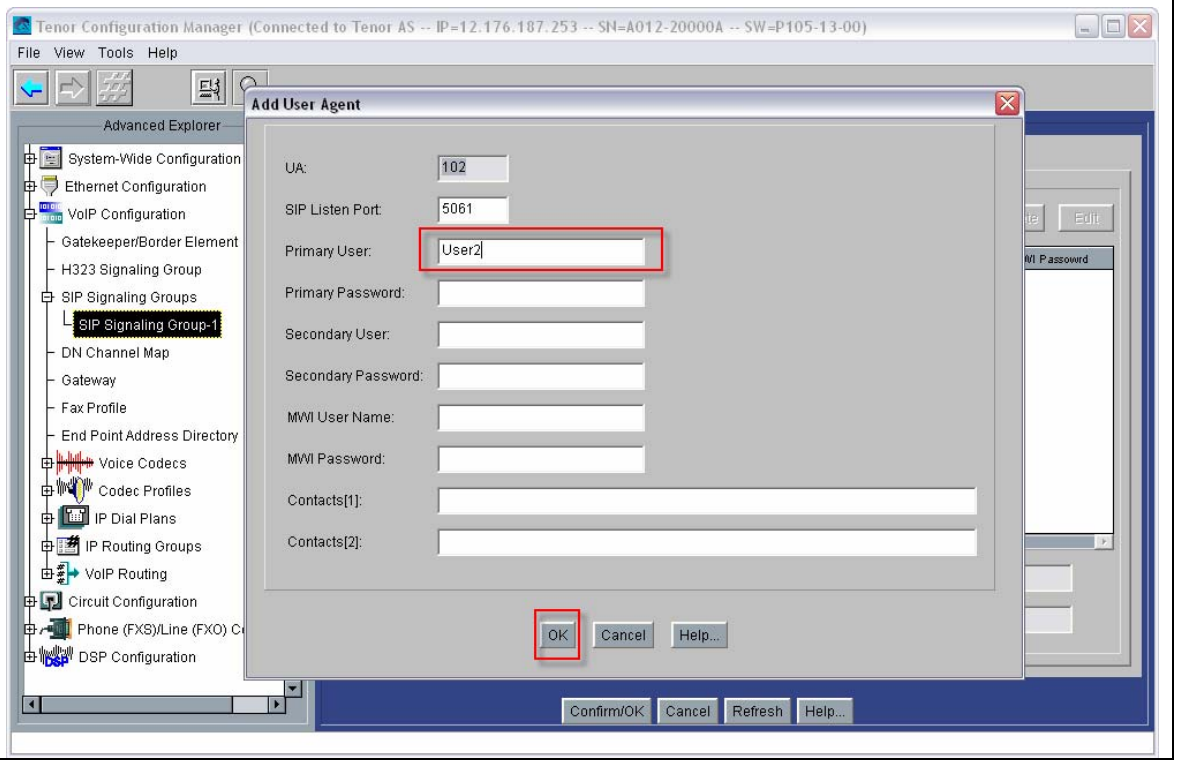
Step	Description
9.	Click on the **Advanced** tab. Un-check the boxes for: • **"SDP in 180 Ringing"** • **"SDP in 183 Progress"** • **"Proxy Address in From Header"** 

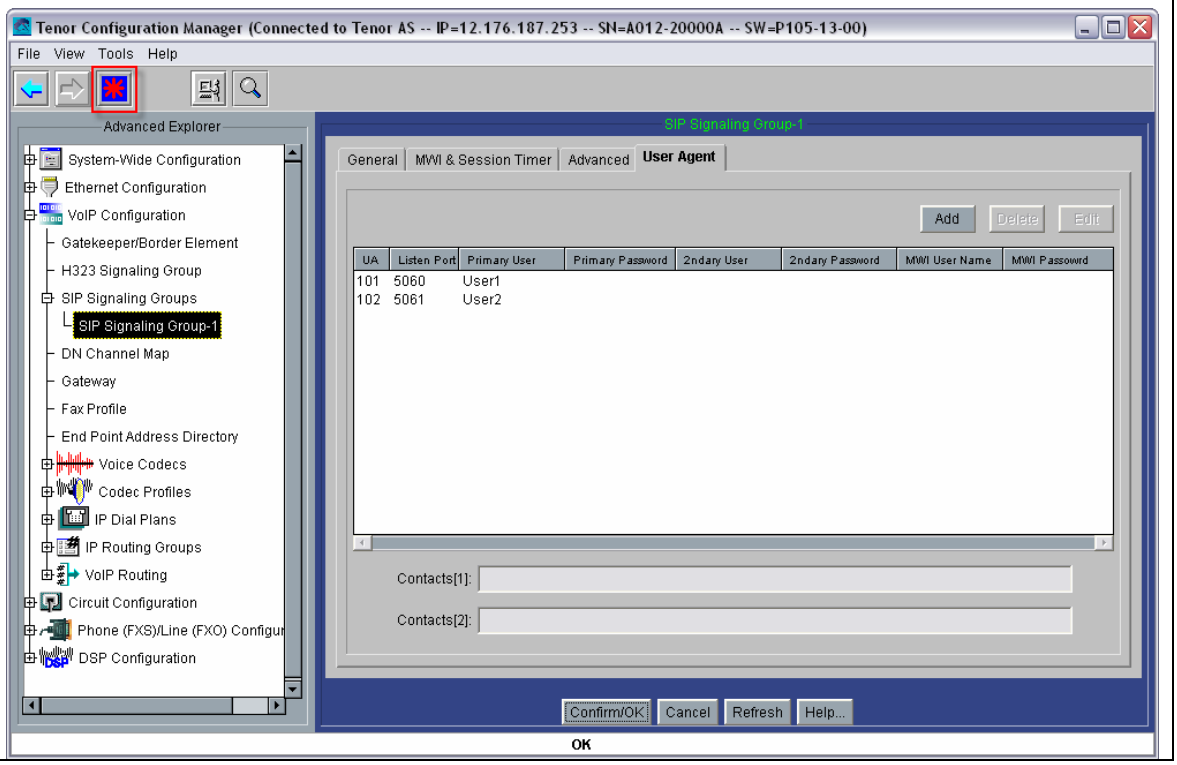
Step	Description
10.	Click on the **User Agent** tab. Click the **Add** button to add display the Add User Agent pop-up window. We will create One User Agent for each physical Analog Line that will be attached to the Tenor. In this configuration example we will create two User Agents. 

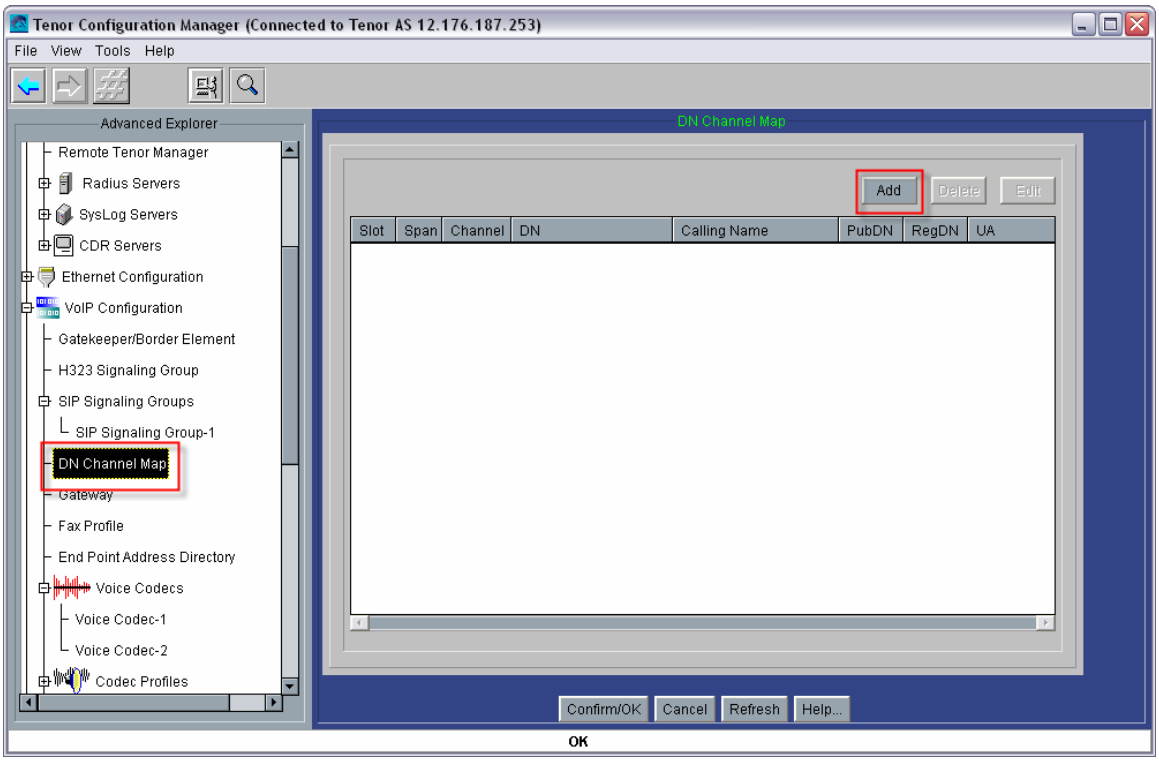
Step	Description
11.	In the **Add User Agent** pop-up window, enter the following information: **PrimaryUser** - The username for Registration and Authentication purposes. If Registration were enabled, the "username" will appear in the URI populated in the To and From headers of the REGISTER message. **Primary User:** ***User1*** < --- Any alpha-numeric string may be entered because SIP Registration and Authentication are not applicable to the AT&T IP Flexible Reach service. Click **OK** to continue. 

Step	Description
12.	At the **SIP Signal Group-1** panel click **Confirm/OK** to complete and the  sunburst icon to implement the change in the Tenor. 

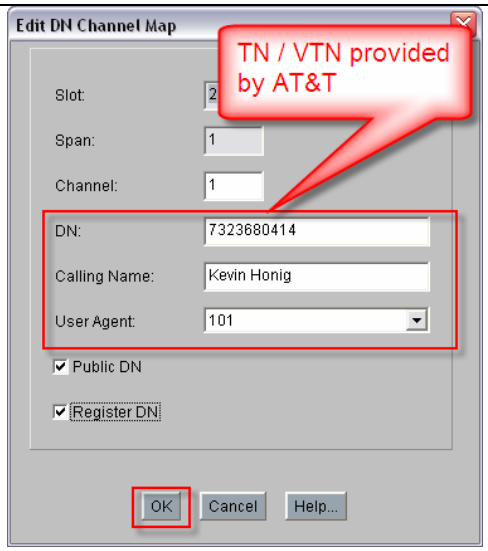
Step	Description
13.	Click the **Add** button to create the 2nd User Agent. The Add User Agent pop-up window will appear. 

Step	Description
14.	In the **Add User Agent** pop-up window, enter the following information: **Primary User:** ***User2*** < --- Any alpha-numeric string may be entered because SIP Registration and Authentication are not applicable to the AT&T IP Flexible Reach service. Click **OK** to continue. 

Step	Description
15.	At the **SIP Signal Group-1** panel click **Confirm/OK** to complete and the  sunburst icon to implement the change in the Tenor . 

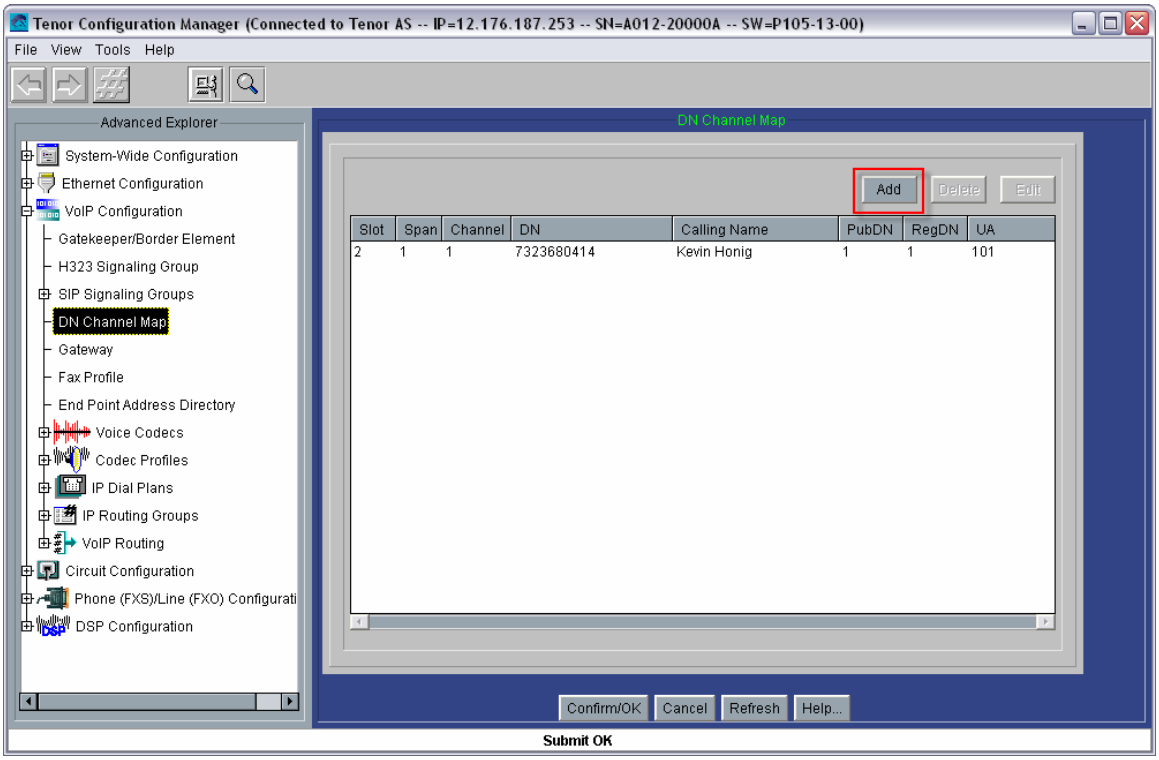
Step	Description
16.	From the **Advanced Explorer** panel on the left, highlight the **DN Channel Map** field. Click **Add** on the **DN Channel Map** panel on the right. 

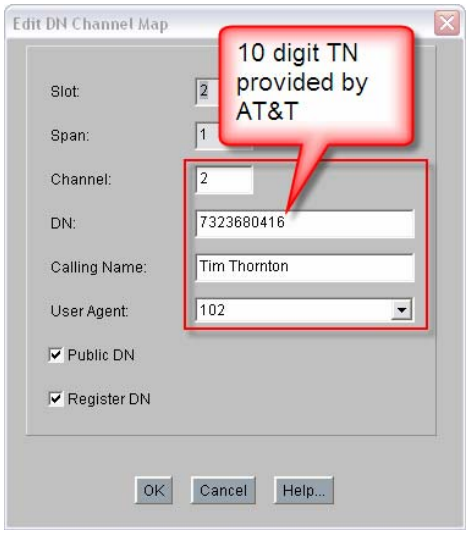
17.	Each of the physical Channels/Ports on the Tenor must be associated with a 10-digit TN (Telephone Number) or a 10-digit VTN (Virtual Telephone Number) provided by AT&T. When the Tenor initiates a call to the AT&T IP Flex Reach service the 10-digit TN / VTN is signaled to AT&T in the outgoing SIP INVITE message. When AT&T routes a call to the Tenor, the signaled DN will be a 10-digit VTN or a subset of the 10-digit TN (4 to 7 digits). The AT&T network **can not** presently signal the full 10-digit TN to the Tenor. But the AT&T network can signal the full 10-digit VTN to the Tenor. The consequence of this asymmetric digit string length for sending vs. receiving calls from the AT&T IP Flex Reach service requires that the Tenor be provisioned with multiple DN's for a given physical Channel/Port. To support outbound calling (from Tenor) the full 10-digit TN/VTN must be configured for the given channel/port. To support inbound calling to the Tenor, the channel/port must also be configured with the appropriate subset of the full TN (4 to 7 digits) or VTN (4 to 10 digits). For the sample configuration documented here, Channel 1 is assigned a 10-digit VTN (732-368-0414). When the Tenor Channel 1 places a call to the AT&T IP Flex reach service it will include the 10-digit VTN (732-368-0414) in the SIP signaling message. When the AT&T IP Flex Reach service routes a call to the Tenor Channel/Port 1 the SIP signaling message from AT&T will include the 10-digit VTN (732-368-0414). Channel 2 is assigned a 10-digit TN (732-368-0416). When the Tenor Channel 2 places a call to the AT&T IP Flex reach service it will include the 10-digit TN (732-368-0416) in the SIP signaling message. When the AT&T IP Flex Reach service routes a call to the Tenor Channel/Port 2 the SIP signaling message from AT&T will include a 4-digit subset of the TN (0416). At the **Add DN Channel Map** pop-up window, enter the following information. **Channel:** **1** < --- Physical port used on Tenor **DN:** **7323680414** < --- Phone Number (number provided by AT&T) **Calling Name: Kevin Honig** < --- Display Name **User Agent:** ***101*** < --- User Agent defined in Step 11. **Public DN** **checked** < --- default **Register DN** **checked** < --- default

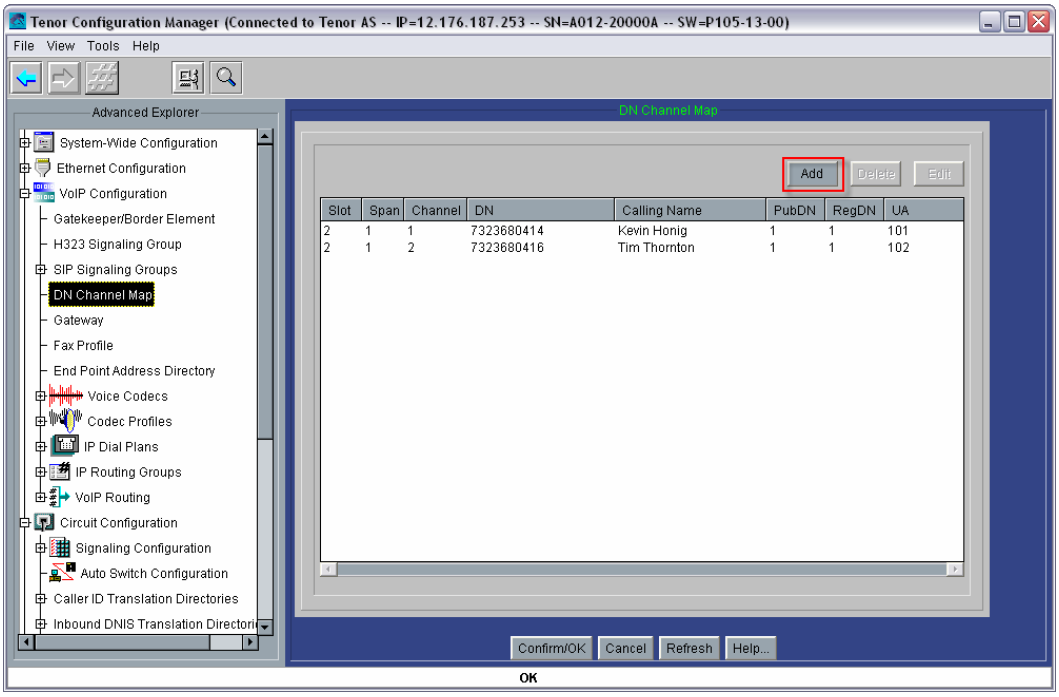


Click **OK** to continue. At the DN Channel Map panel click **Confirm/OK** and the  sunburst icon implements the change.

Note:
Slot *and* ***Span*** *are not relevant to the Analog Tenor.*
Channel: *Denotes the physical port that the analog device will be connected.*
DN: *TN number provided by AT&T. Populated in outgoing INVITE message (to AT&T) as the user part of the URI in the From and Contact headers. On inbound calls to Tenor, used to determine routing of calls to physical line. Should appear as user part of Request URI of incoming INVITE.*
Calling Name: *Will appear as the Display Name in the From header in outgoing INVITE messages.*
Public DN: *Indicates whether or not this is a Public DN*
Register DN: *Only relevant to H.323*

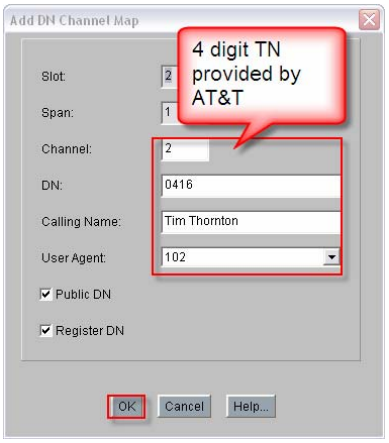
18.	Enter the 10 digit TN associated with Cannel/Port 2. Click the **Add** button. 

19.	At the **Add DN Channel Map** pop-up window, enter the following information. **Channel: 2** < --- Physical port used on Tenor **DN: 7323680416** < --- Phone Number (number provided by AT&T) **Calling Name: Tim Thornton**< --- Display Name **User Agent: *102*** < --- User Agent defined in Step 14. **Public DN checked** < --- default **Register DN checked** < --- default  Click **OK** to continue. At the DN Channel Map panel click **Confirm/OK** and the  sunburst icon to implement the change.

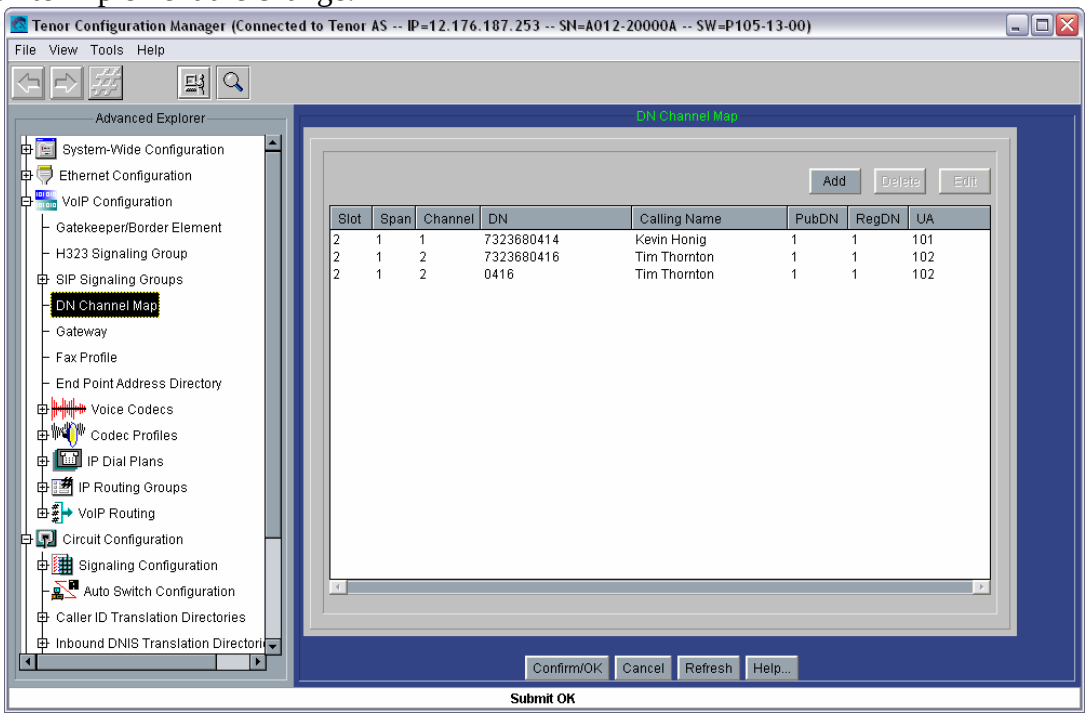
20.	It is necessary to define another DN Channel Map entry to provide the 4-digit routing (0416) of calls to the Tenor Channel/Port 2. Click the **Add** button. 

21. At the **Add DN Channel Map** pop-up window, enter the following information.

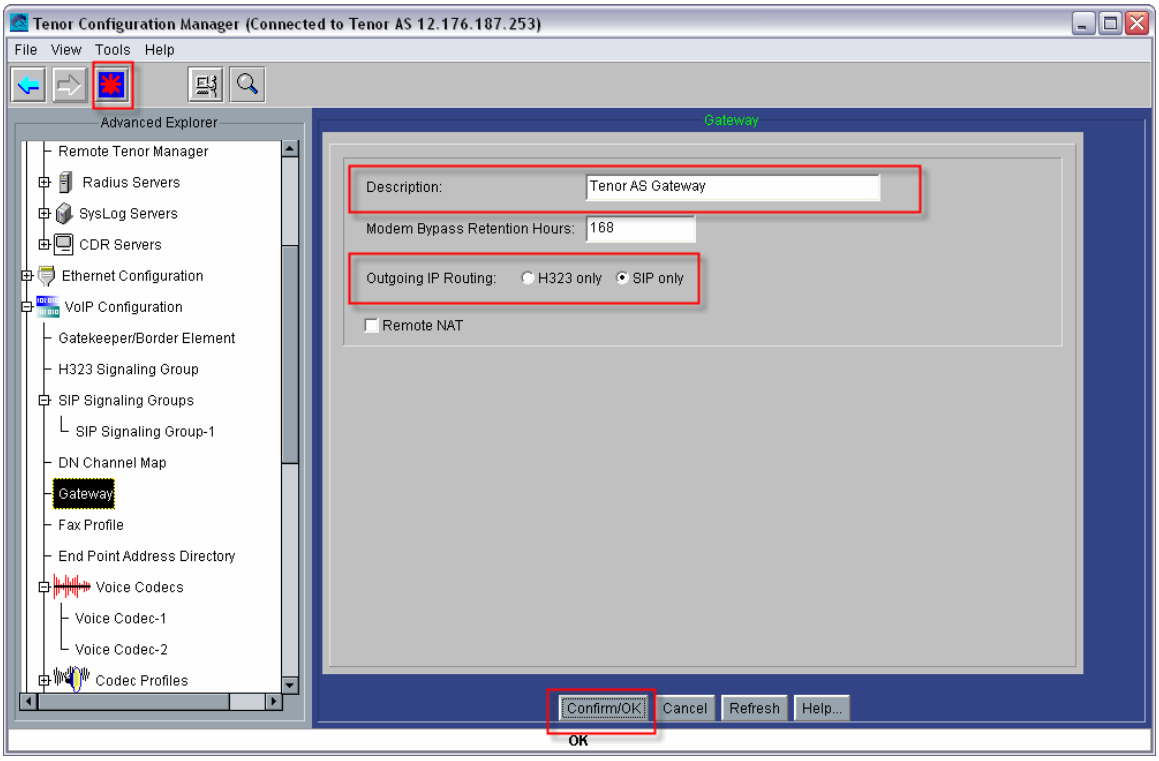
Channel: **2** < --- Physical port used on Tenor
DN: **0416** < --- Phone Number (number provided by AT&T)
Calling Name: Tim Thornton< --- Display Name
User Agent: ***102*** < --- User Agent defined in Step 14.
Public DN **checked** < --- default
Register DN **checked** < --- default

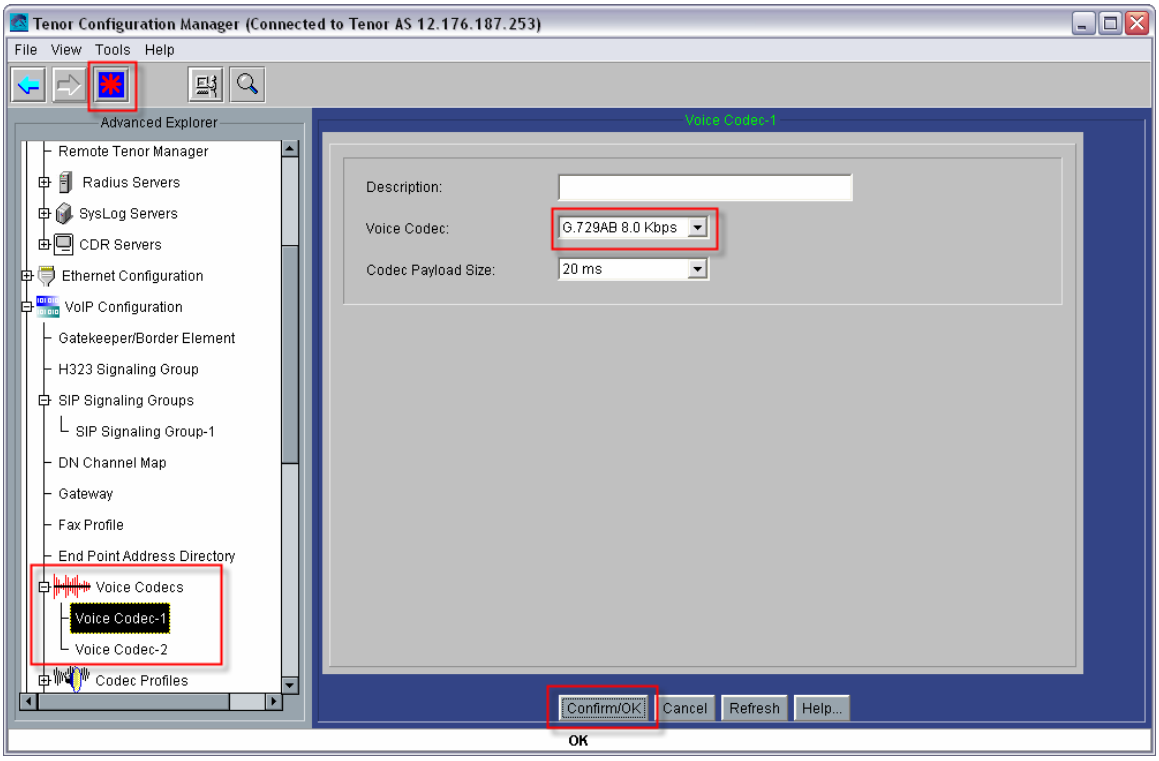


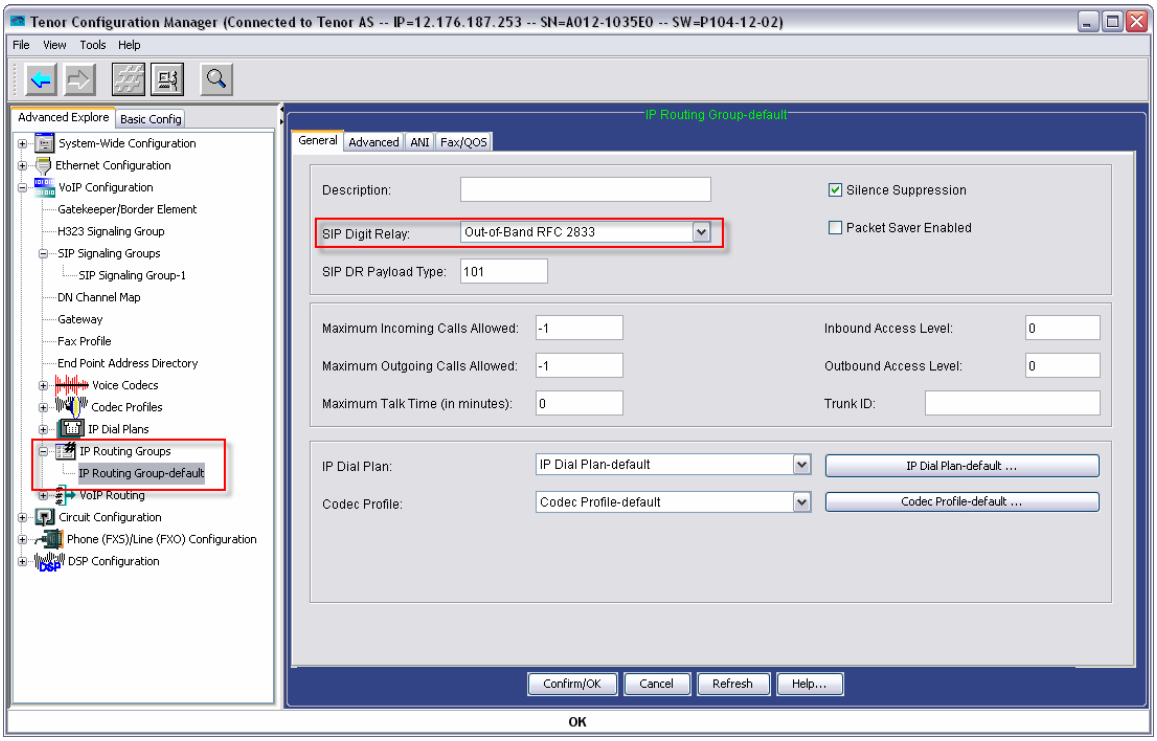
Click **OK** to continue. At the DN Channel Map panel click **Confirm/OK** and the  sunburst icon to implement the change.

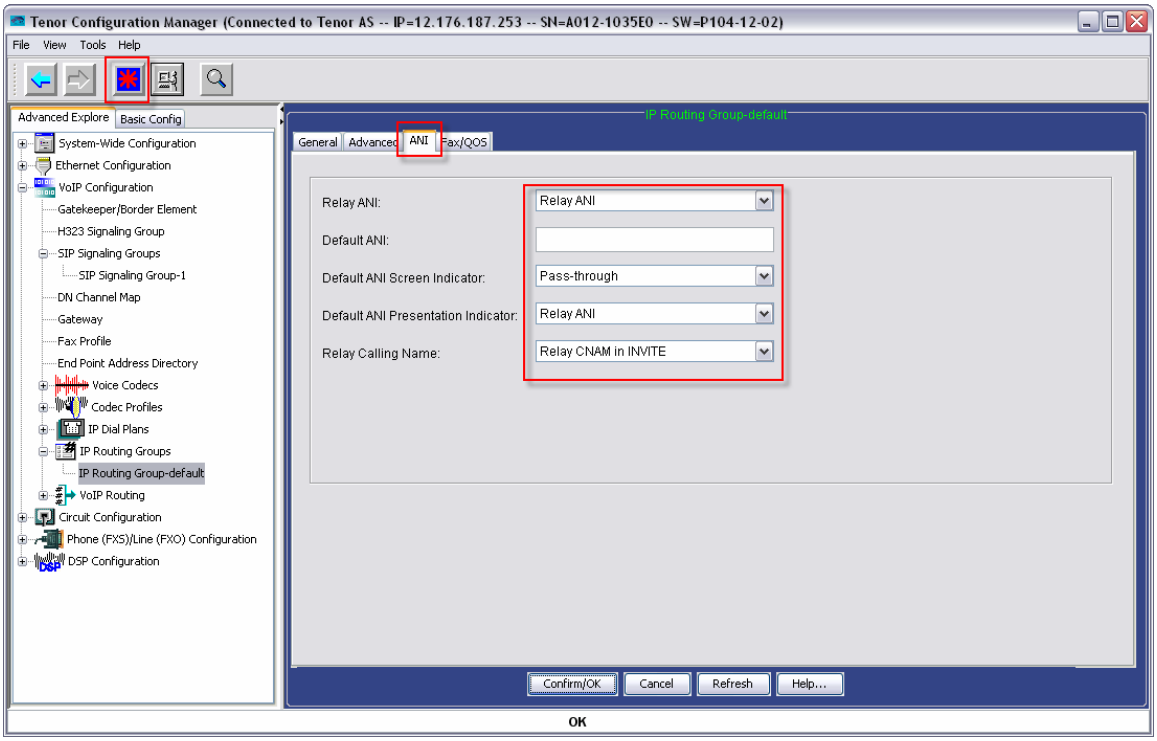


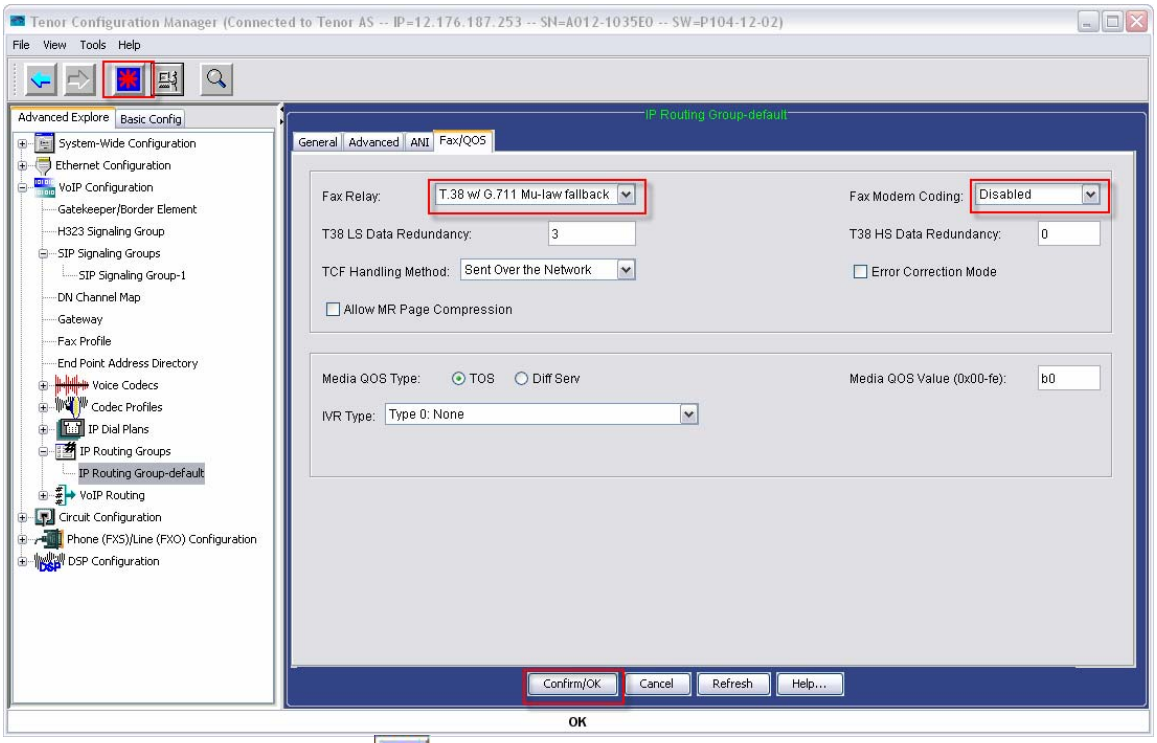
Note that 0416 and 7323680416 are the two DN's associated with channel 2 and User Agent (UA) 102. The DN 7324600416 is associated with channel 1 and User Agent 1.

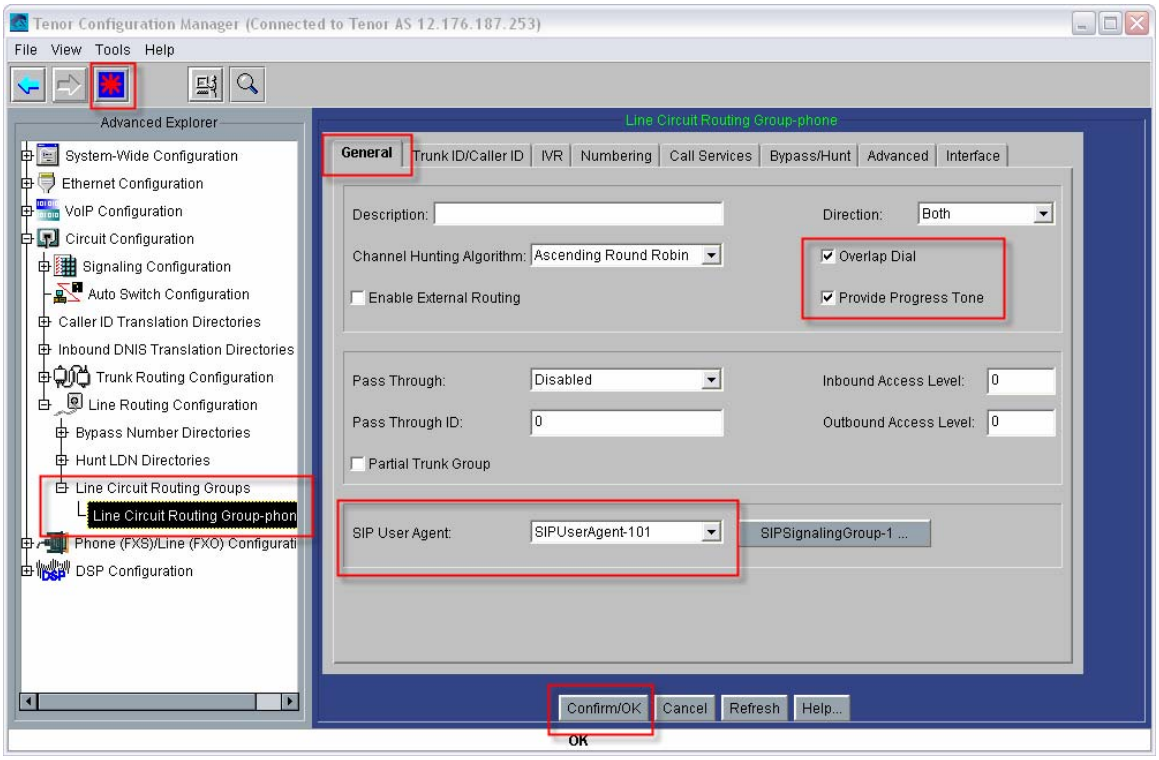
22.	From the **Advanced Explorer** panel on the left, highlight the **Gateway**. Enter a **Description** and ***check*** the **SIP only** radio button for the **Outgoing IP Routing** field under the Gateway screen panel on the right. Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change. 

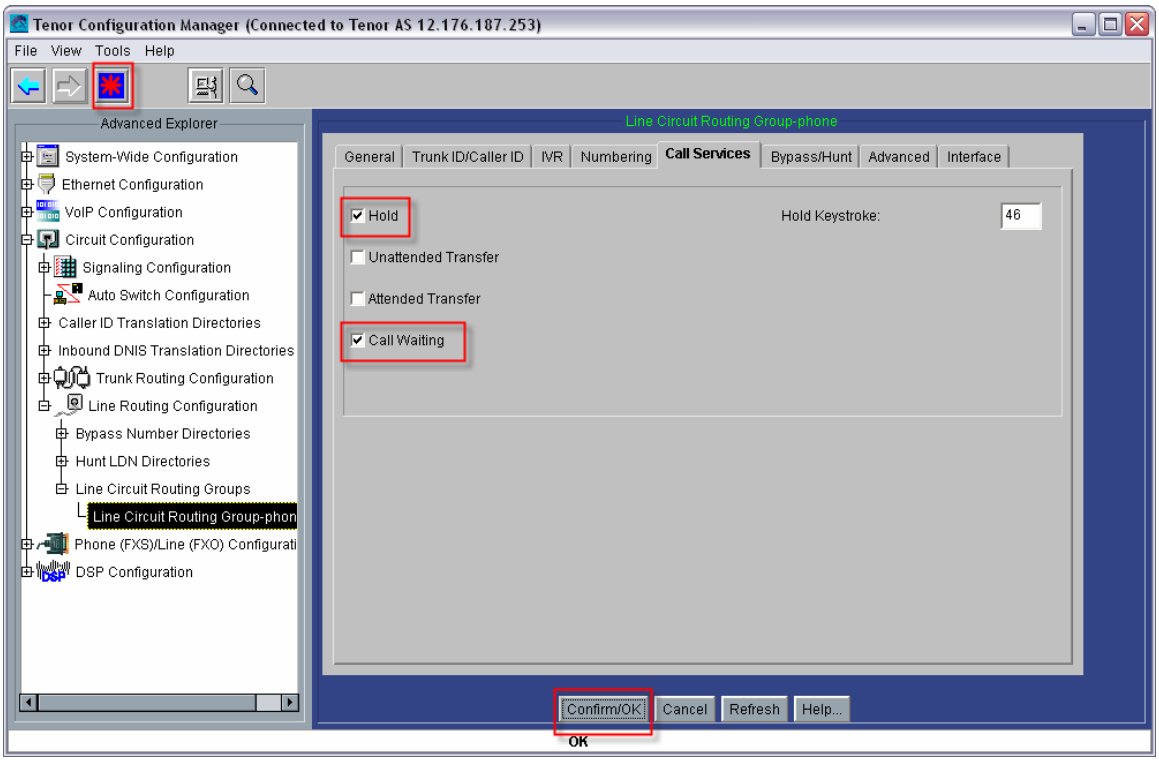
23.	From the **Advanced Explorer** panel on the left, click on the **+** sign to expand the **Voice Codecs** field. Highlight the **Voice Codec-1** field. Select the desire **Voice Codec** field from the drop down menu. The sample configuration uses the ***G.729*** codec. Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change. 

24.	From the **Advanced Explorer** panel on the left, highlight the **IP Routing Group-default** field under **IP Routing Groups**. Under the **General** tab in the **IP Routing Group-default** panel on the right, select ***Out-of-Band RFC 2833*** for **SIP Digit Relay** from the drop down menu. 

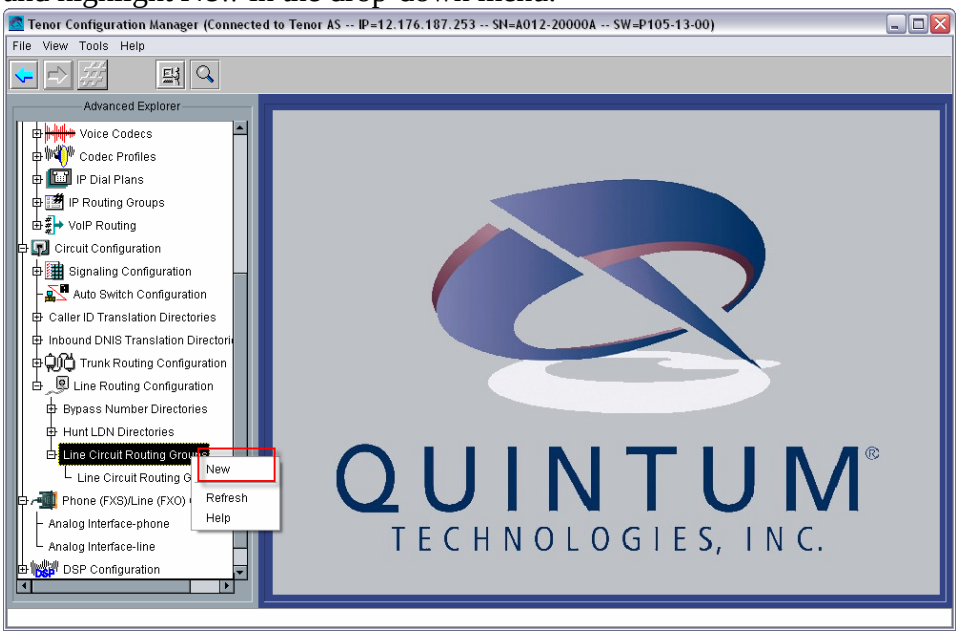
25.	Click on the **ANI** tab under the **IP Routing Group-default** panel on the right. Select ***Relay ANI*** for **Relay ANI** from the drop down menu. Select ***Pass-through*** for **Default ANI Screen Indicator** from the drop down menu. Select ***Relay ANI*** for **Default ANI Presentation Indicator** from the drop down menu. Select ***Relay CNAM in INVITE*** for **Relay Calling Name** from the drop down menu. Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change. 

26.	Click on the **FAX/QOS** tab under the **IP Routing Group-default** panel on the right. Select ***T.38 w/G.711 Mu-law fallback*** for **Fax Relay** from the drop down menu. Select ***Disabled*** for **Fax Modem Coding** from the drop down menu.  Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change.

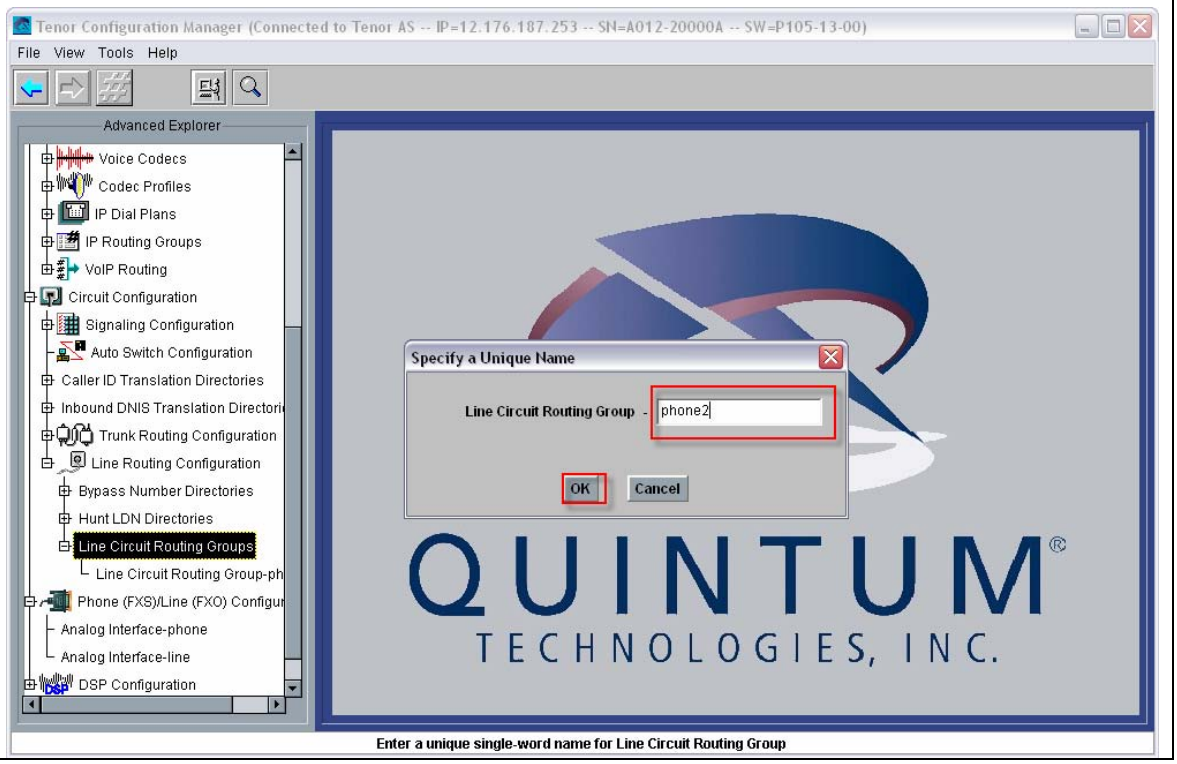
27.	From the **Advanced Explorer** panel on the left, expand **Circuit Configuration → Line Routing Configuration → Line Circuit Routing Groups**, and highlight the **Line Circuit Routing Group-phone** field. Click on the **General** tab under the **Line Circuit Routing Group-phone** panel on the right. From the **SIP User Agent** drop down menu, select ***SIPUserAgent-101*** and ***check*** the boxes for **Overlap Dial** and **Provide Progress Tone**. Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change. Click the **Call Services** tab. 

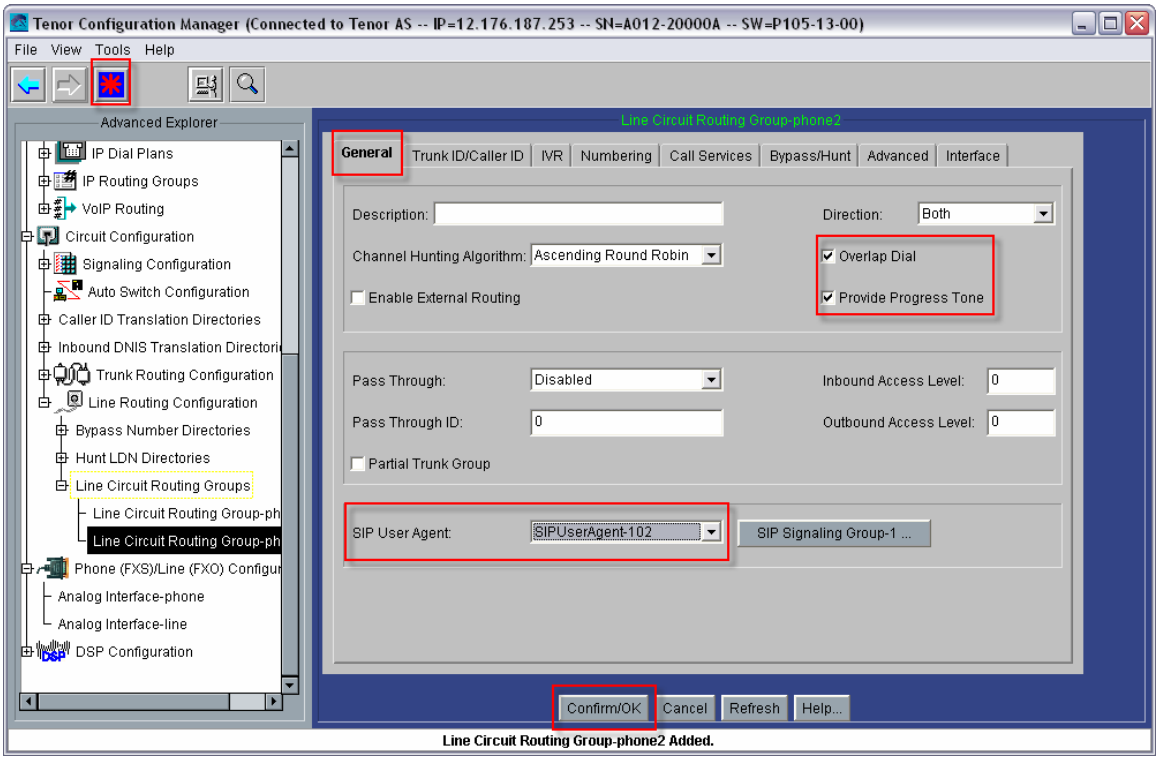
28.	From the **Call Services** tab under the **Line Circuit Routing Group-phone** panel on the right. Check to enable **Hold**, and **Call Waiting**. Disable **Unattended Transfer** and **Attended** Transfer if checked. Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change. 

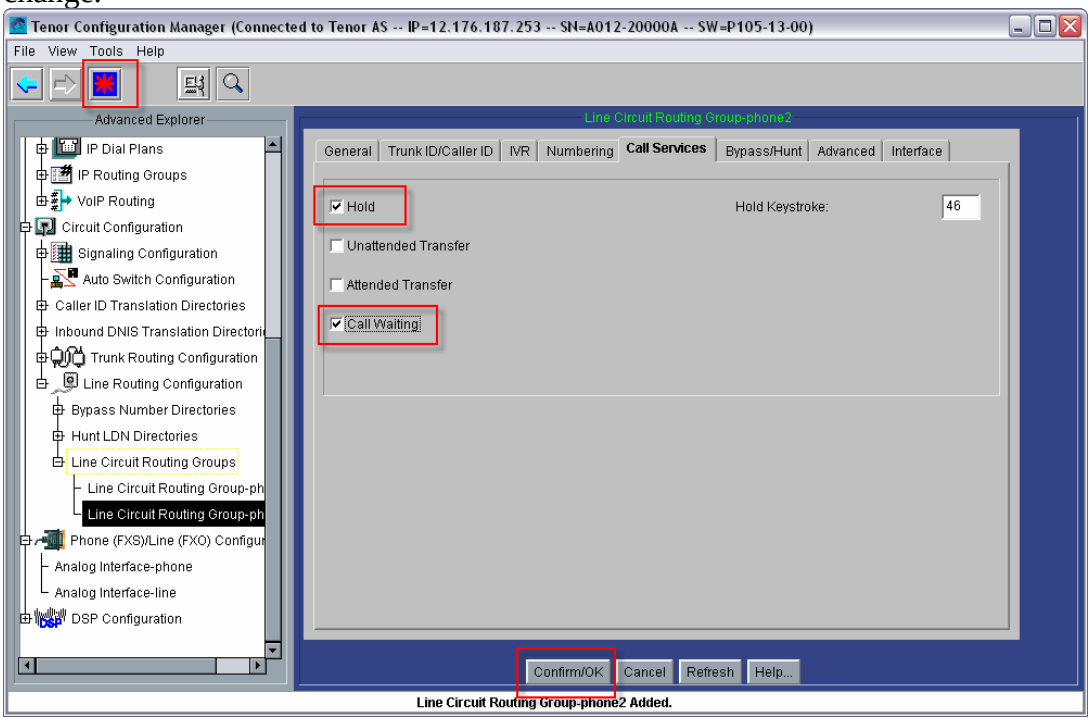
29. Create a second Line Circuit Routing Group to define the settings for Channel 2. From the **Advanced Explorer** panel on the right, click on **Line Circuit Routing Groups** and highlight **New** in the drop-down menu.

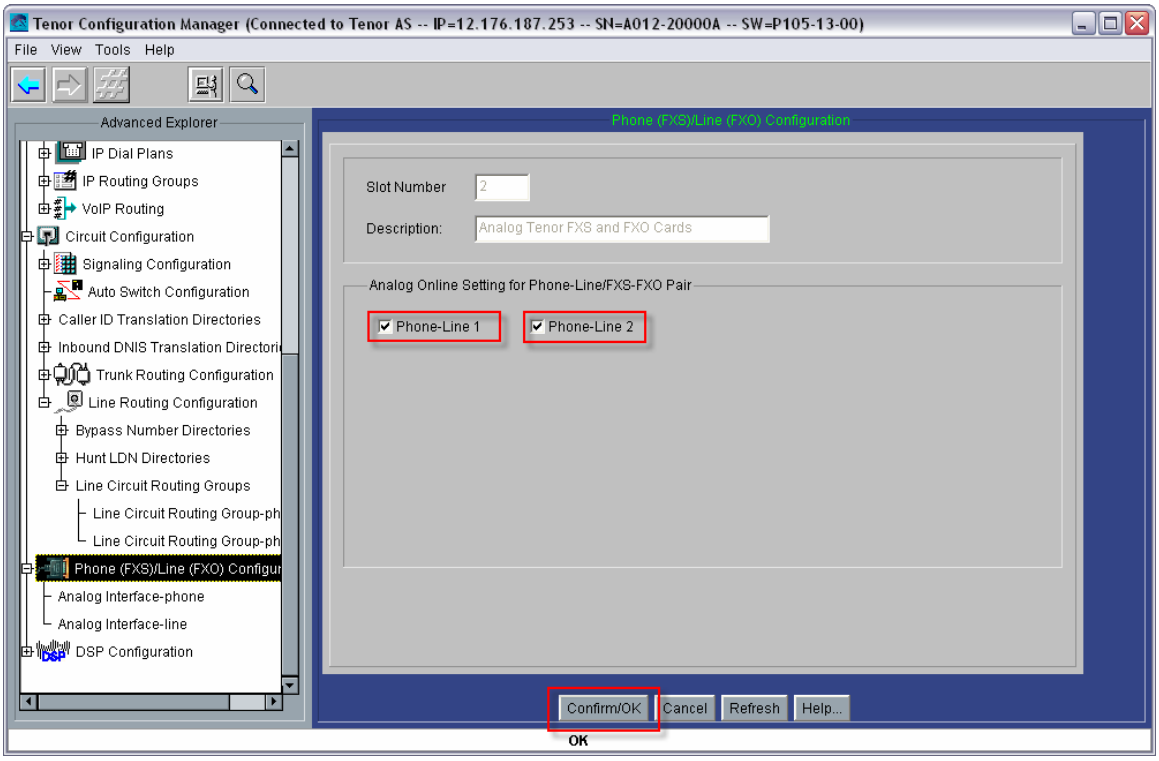


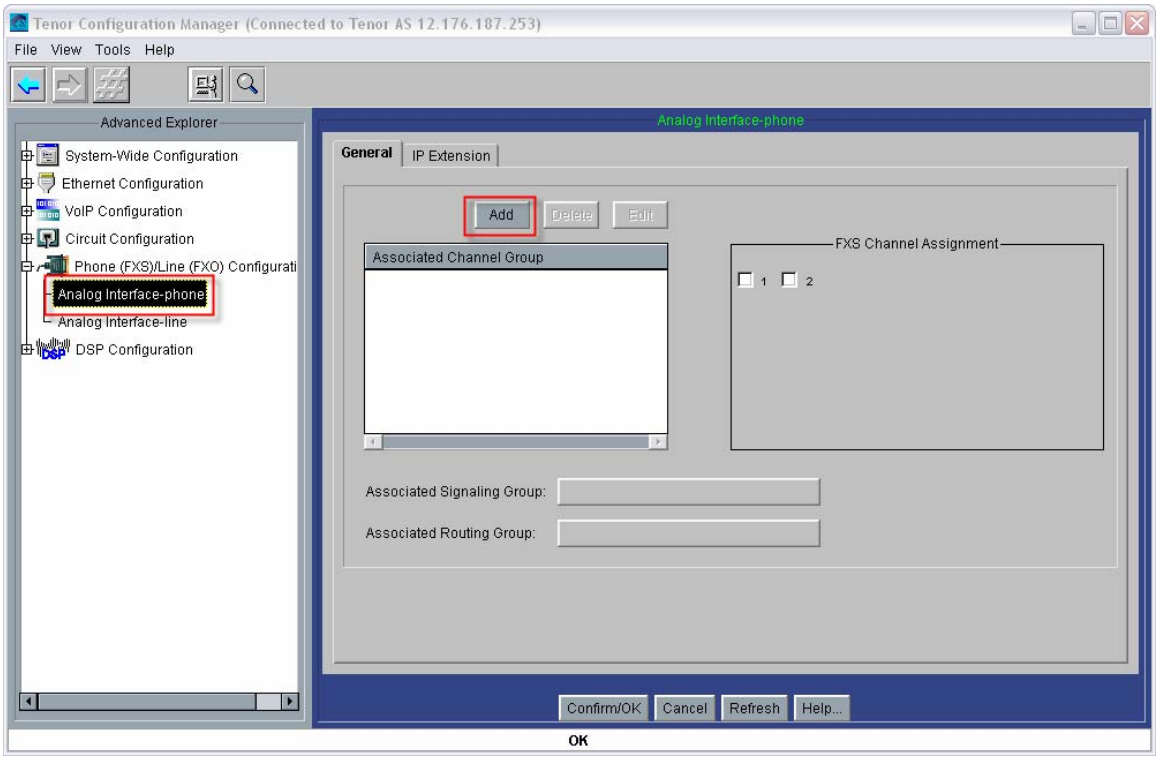
Enter a name for the new **Line Circuit Routing Group (ex. "phone2")** and click **OK.**



30.	From the **Advanced Explorer** panel on the left, expand **Circuit Configuration → Line Routing Configuration → Line Circuit Routing Groups**, and highlight the **Line Circuit Routing Group-phone2** field. Click on the **General** tab under the **Line Circuit Routing Group-phone** panel on the right. From the **SIP User Agent** drop down menu, select ***SIPUserAgent-101*** and ***check*** the boxes for **Overlap Dial** and **Provide Progress Tone**. Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change. Click the **Call Services** tab. 

31.	From the **Call Services** tab under the **Line Circuit Routing Group-phone** panel on the right. Check to enable **Hold**, and **Call Waiting**. Disable **Unattended Transfer** and **Attended** Transfer if checked. Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change. 

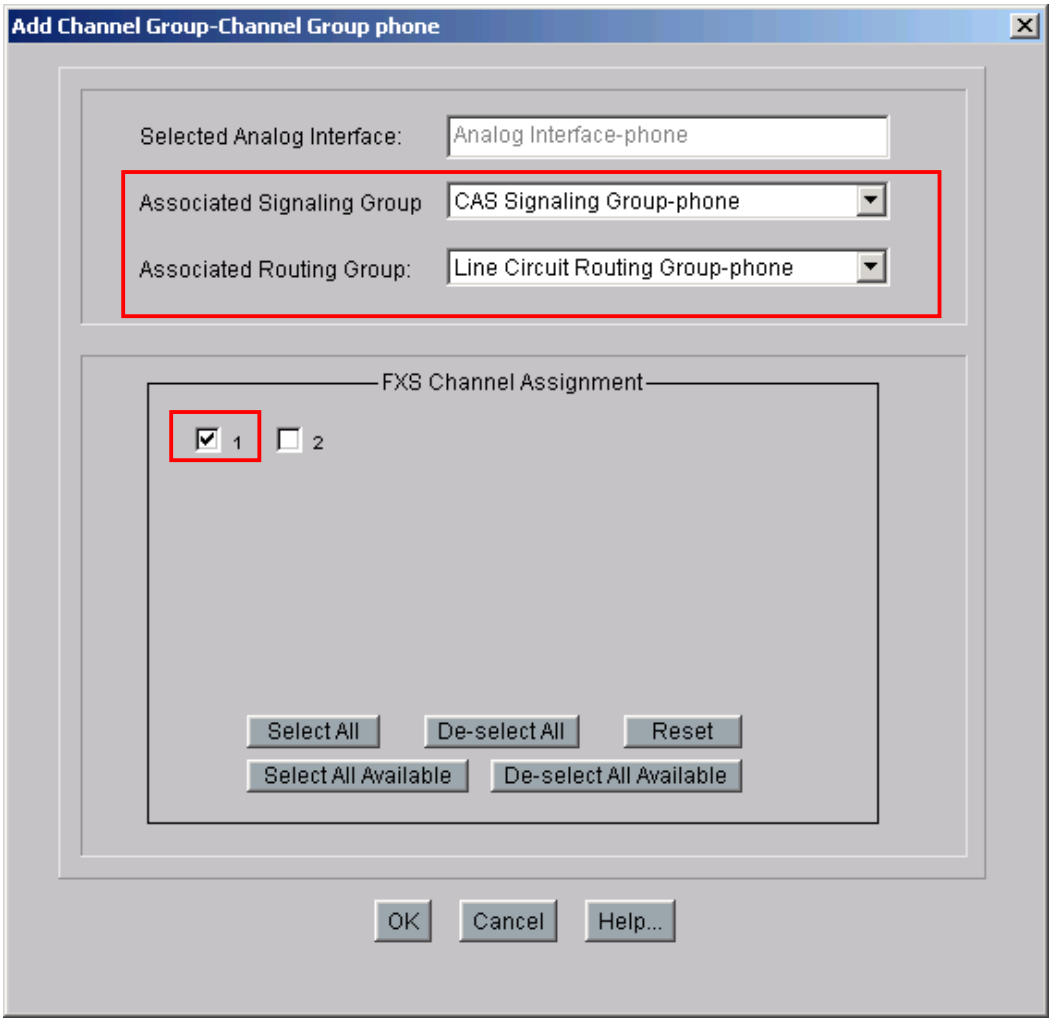
32.	Under the **Advanced Explorer** panel on the left, highlight the **Phone (FXS)/Line (FXO) Configuration**. Check the box to enable **Phone-Line 1 and Phone-Line 2**. Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change. 

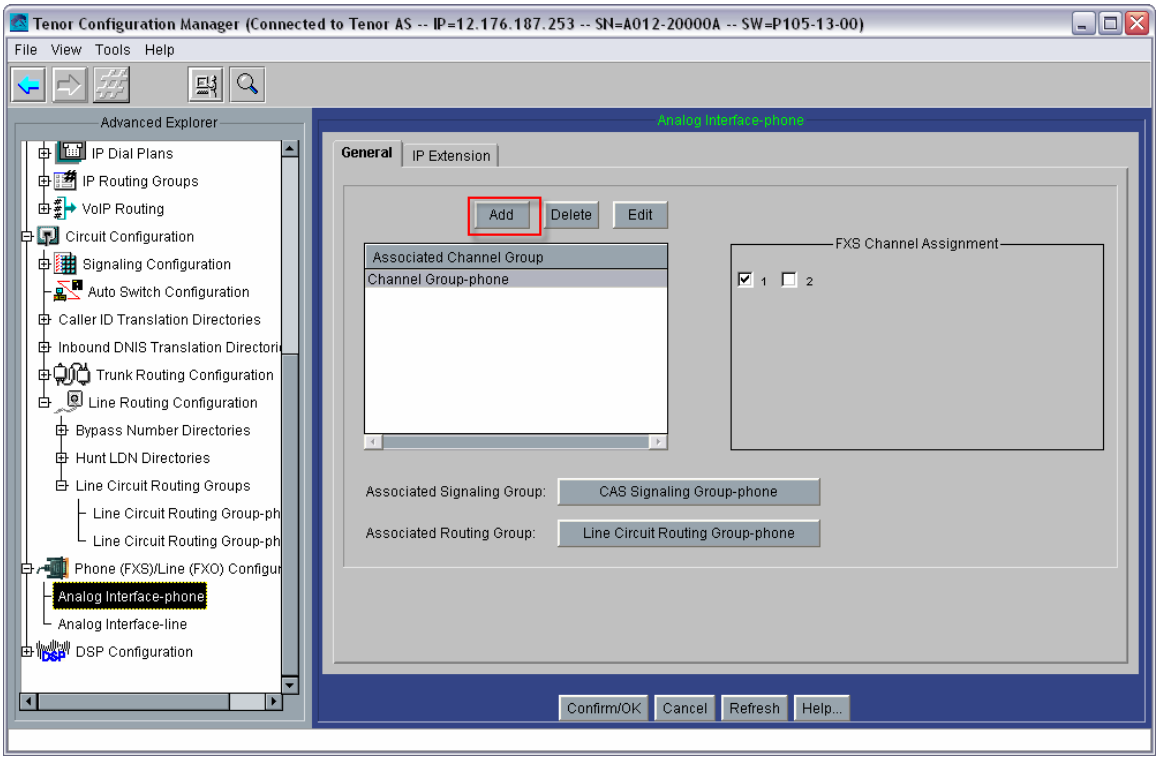
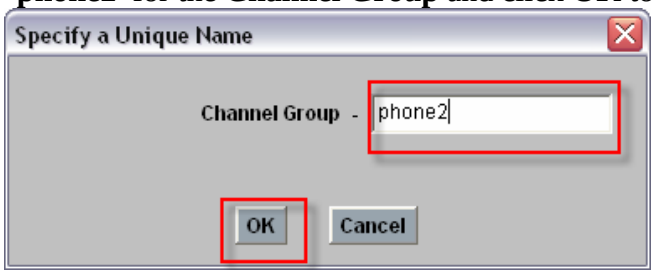
33.	Under the **Advanced Explorer** panel on the left, expand **Phone (FXS)/Line (FXO) Configuration**, and highlight the **Analog interface-phone** field. Highlight **Channel Group-phone** then click **Add**. 
34.	Enter a description **"phone"**for the **Channel Group** and click **OK** to continue. 

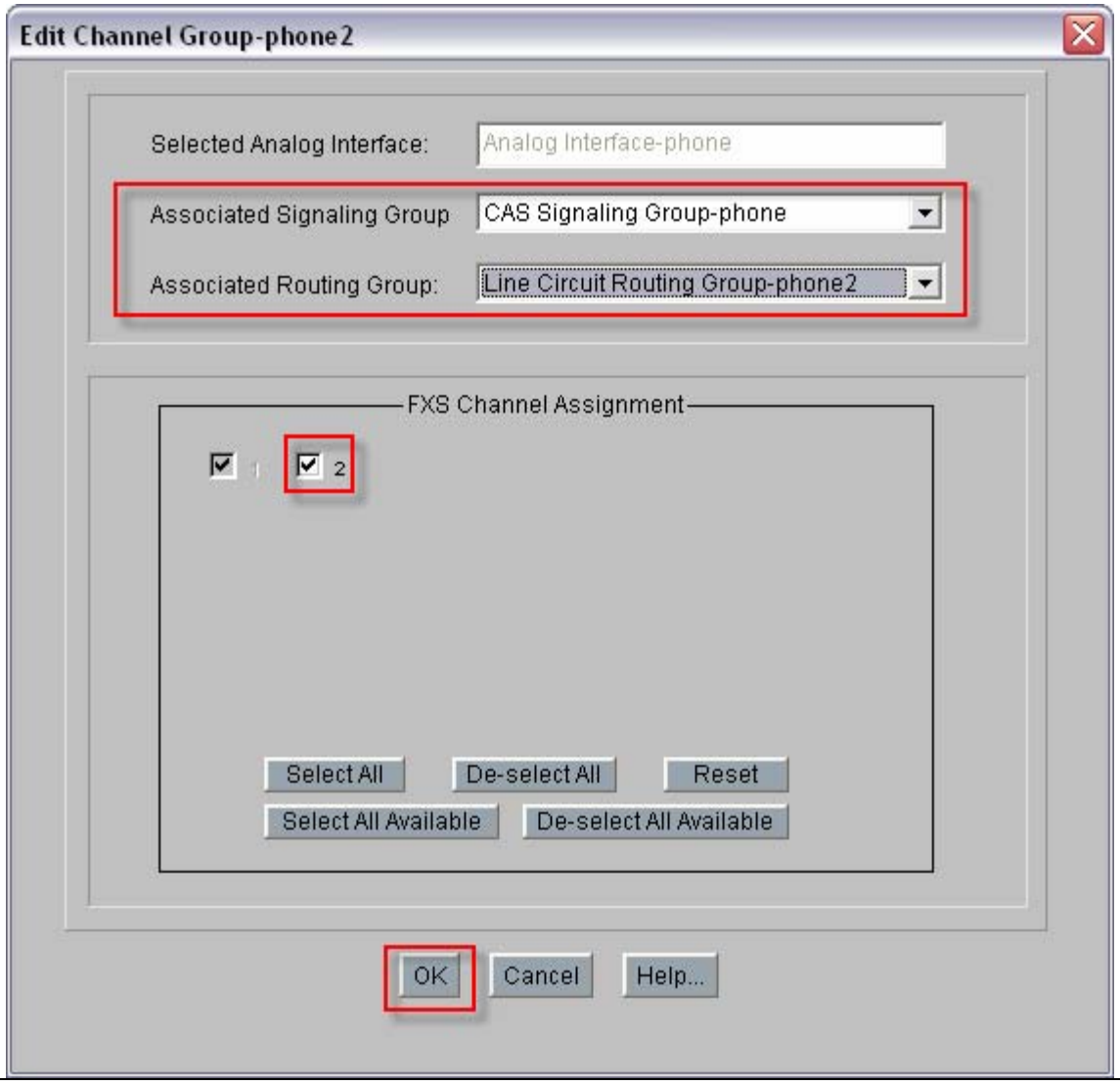
35. In the **Add Channel Group-Channel Group phone** pop-up window, select the following information.

Associated Signaling Group:	***CAS Signaling Group-phone***
Associated Routing Group:	***Line Circuit Routing Group-phone***
FXS Channel Assignment	***Check*** radio button for **1**

Click **OK** to complete.



36.	Under the **Advanced Explorer** panel on the left, expand **Phone (FXS)/Line (FXO) Configuration**, and highlight the **Analog interface-phone** field. Highlight **Channel Group-phone** then click **Add**. 
37.	Enter a description **“phone2”**for the **Channel Group** and click **OK** to continue. 

38.	In the **Add Channel Group-Channel Group phone2** pop-up window, select the following information. **Associated Signaling Group:** ***CAS Signaling Group-phone*** **Associated Routing Group:** ***Line Circuit Routing Group-phone2*** **FXS Channel Assignment** ***Check*** radio button for **2** Click **OK** to complete. 

6 Survivability

Select Quintum Tenor models (AFM, AFE, AXM and AXE) offer protection from catastrophic IP network outages and power failures. The Tenor MultiPath Call Routing architecture provides the ability to complete calls via PSTN lines when the IP network is not available. What about the 911 calls? How are those calls routed when IP is not available ? The Tenor Automatic Protection Switching (APS) capability provides PSTN connectivity in the event of a power failure.

This section describes the Tenor Survivability features and provides a guide for enabling and testing these capabilities.

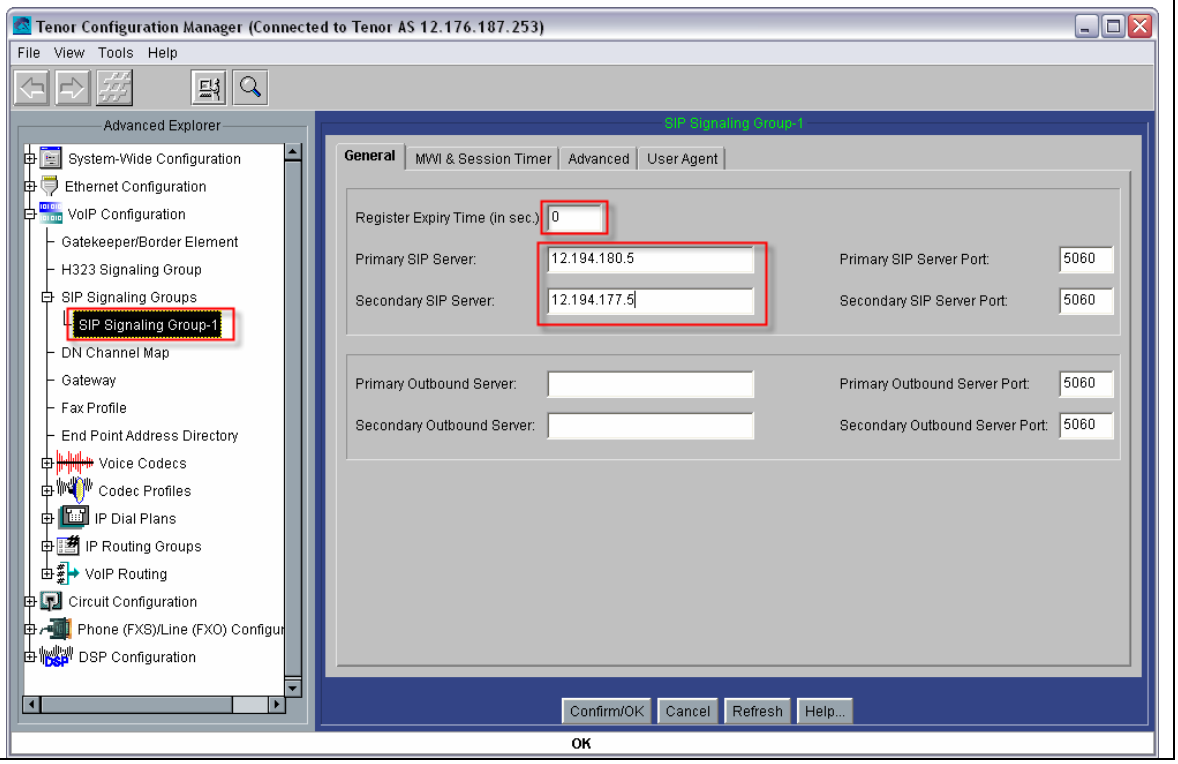
6.1 MultiPath Survivability

The Tenor MultiPath Call Routing architecture available in select Tenor models, provide the ability to complete calls via PSTN lines when the IP network is not available. With AT&T IP Flexible Reach the Tenor will attempt to complete a call via the Primary and Secondary AT&T IP Border Element before trying the PSTN line.
What about the 911 calls? How are those calls routed when IP is not available ?

6.1.1 Configuring MultiPath Survivability

To enable MultiPath Survivability, requires that at least one FXO line be attached to the Tenor. The following steps describe how to configure MultiPath Survivability on the Tenor.

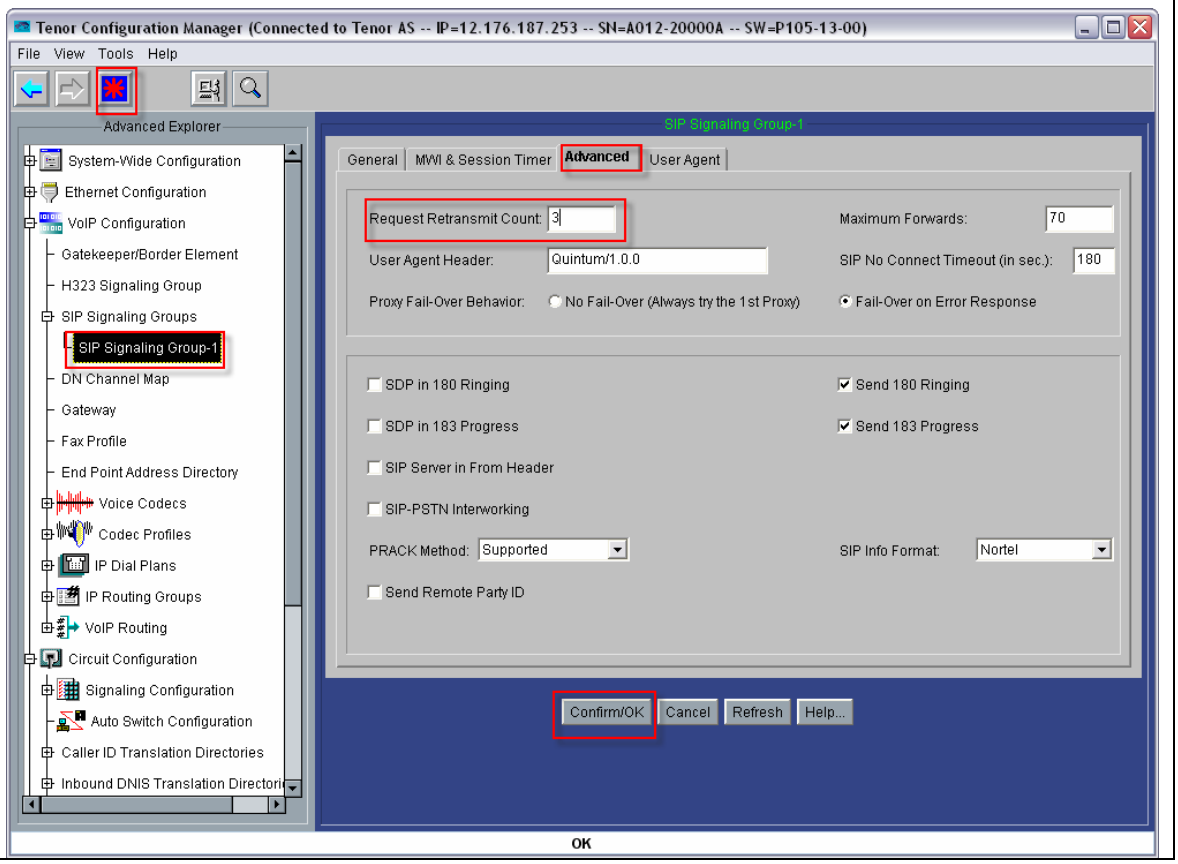
Step	Description
1.	As in Section 5, Step 8 above, insure that the **Primary SIP Server** and **Secondary SIP Server** are set to the IP addresses of the AT&T Primary and Secondary IP Border Elements.



2. To avoid a long post dial delay when completing a call over the PSTN line, set the Request Retransmit Count to "3".

From the **Advanced Explorer** panel on the left, click on the **+** sign next to **VoIP Configuration → SIP Signal Groups** to expand the field. Highlight the **SIP Signaling Group-1** field. Under the **Advanced** tab, set the **Request Retransmit Count** to "**3**".

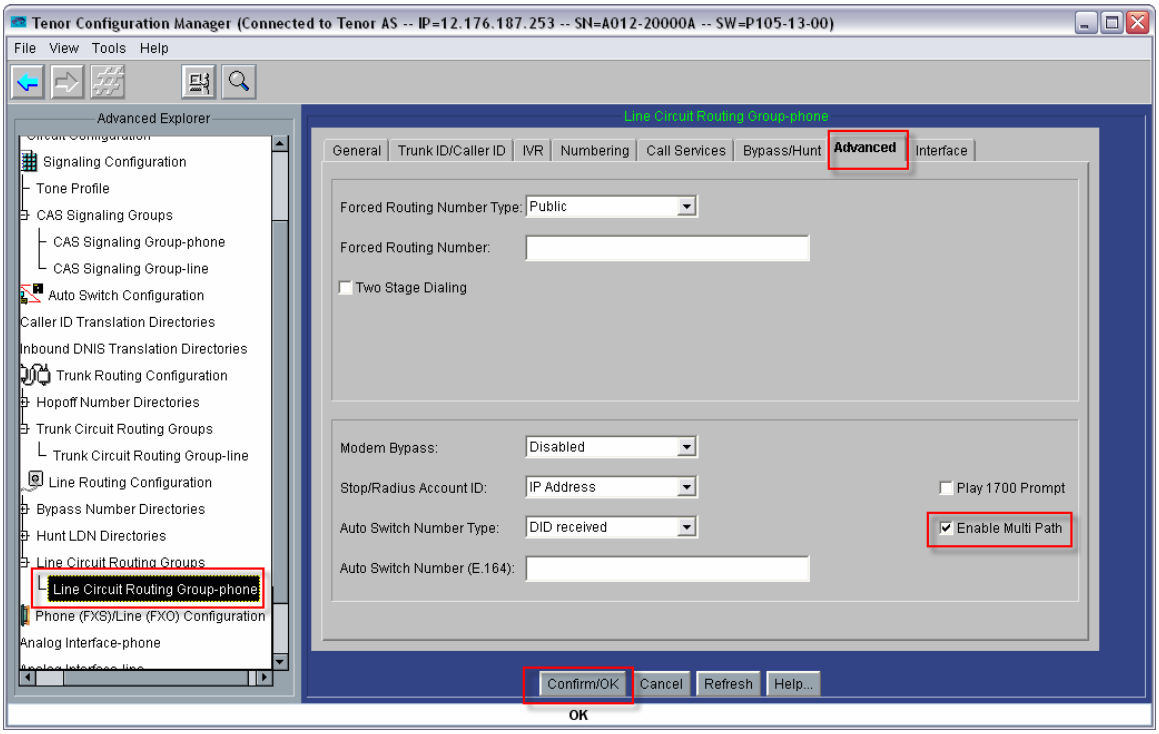
Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change.



3. Enable MultiPath for the FXS phone line.
From the **Advanced Explorer** panel on the left, expand **Circuit Configuration → Line Routing Configuration → Line Circuit Routing Groups**, and highlight the **Line Circuit Routing Group-phone** field.

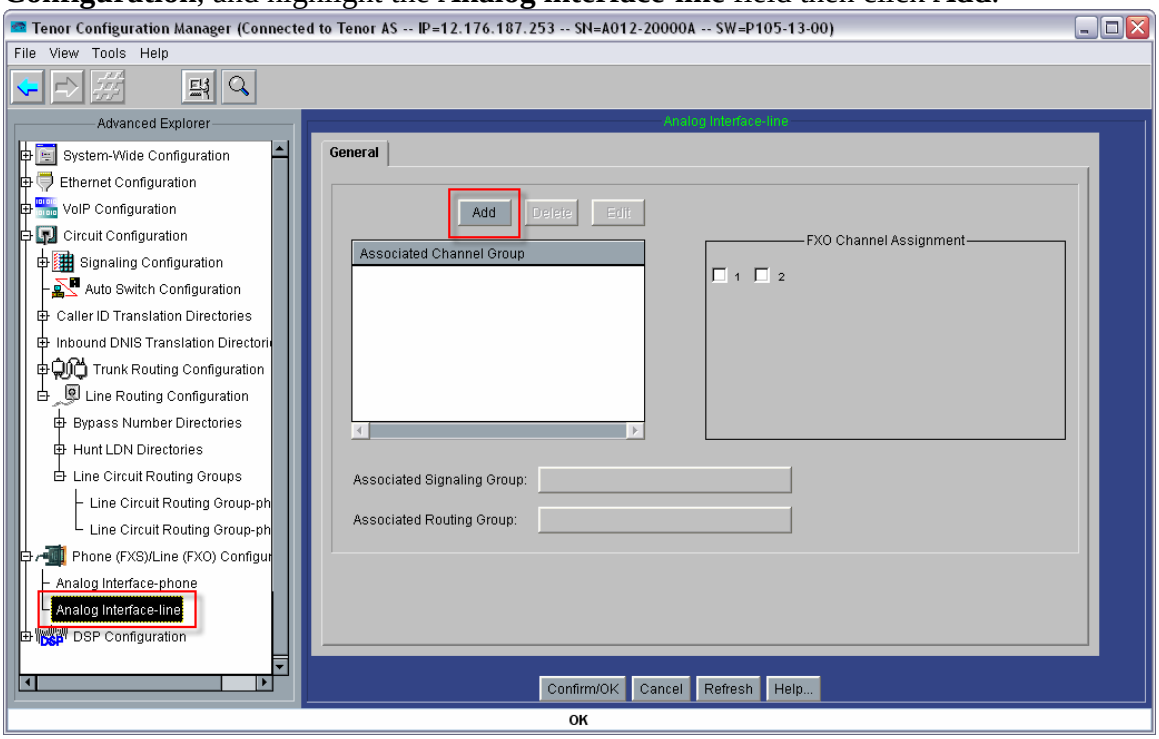
Click on the **Advanced** tab and ***check*** the boxes for **Enable Multi Path**.

Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change.

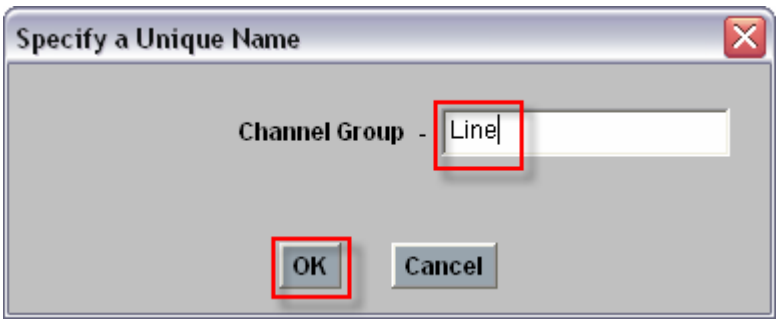


4. Enable one or more FXO lines on the Tenor. The configuration example in Section 5 above only utilized FXS ports. Here are the replacement Steps to configure one FXS port and one FXO port for MultiPath Survivability.

5. Under the **Advanced Explorer** panel on the left, expand **Phone (FXS)/Line (FXO) Configuration**, and highlight the **Analog interface-line** field then click **Add**.



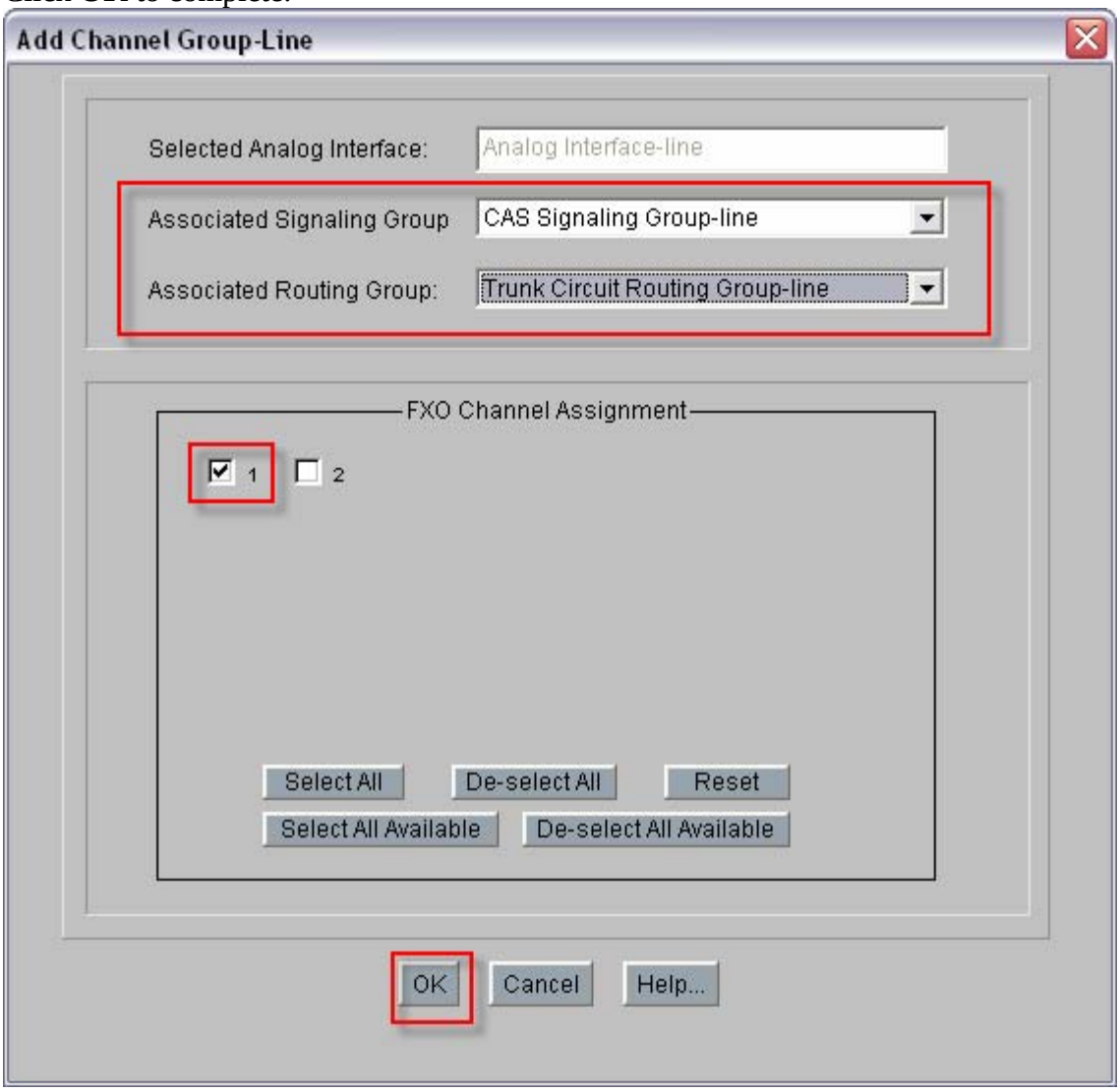
6. Enter a description **"Line"** for the **Channel Group** and click **OK** to continue.



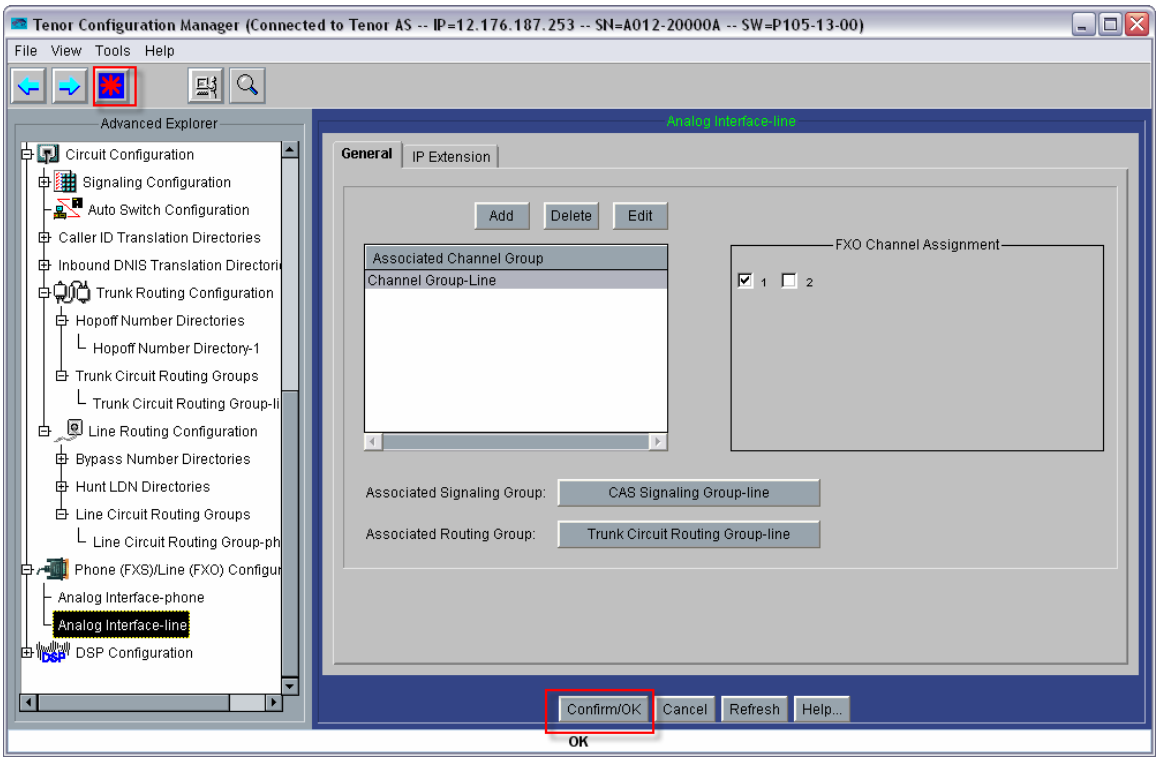
7. In the **Add Channel Group-Channel Group Line** pop-up window, select the following information**.**

Associated Signaling Group: ***CAS Signaling Group-line***
Associated Routing Group: ***Trunk Circuit Routing Group-line "***
FXO Channel Assignment ***Check*** radio button for **1**

Click **OK** to complete.



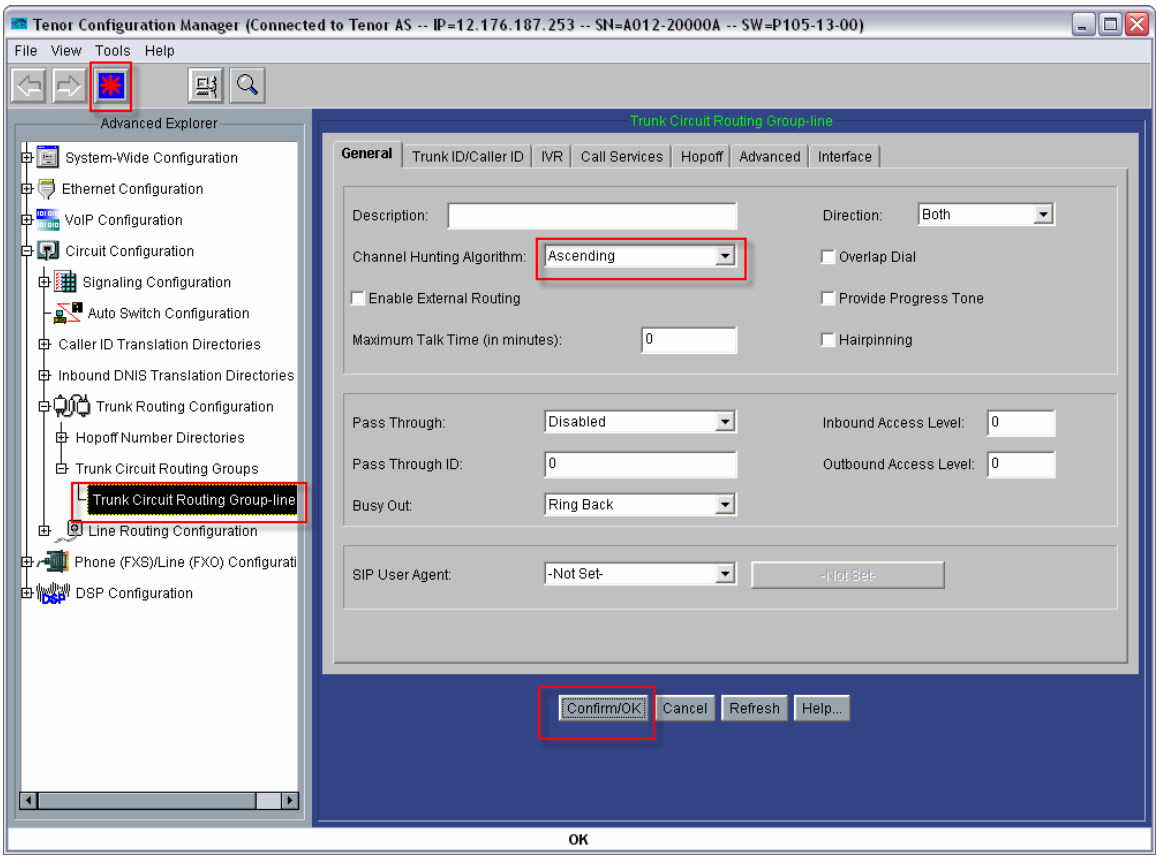
8. Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change.



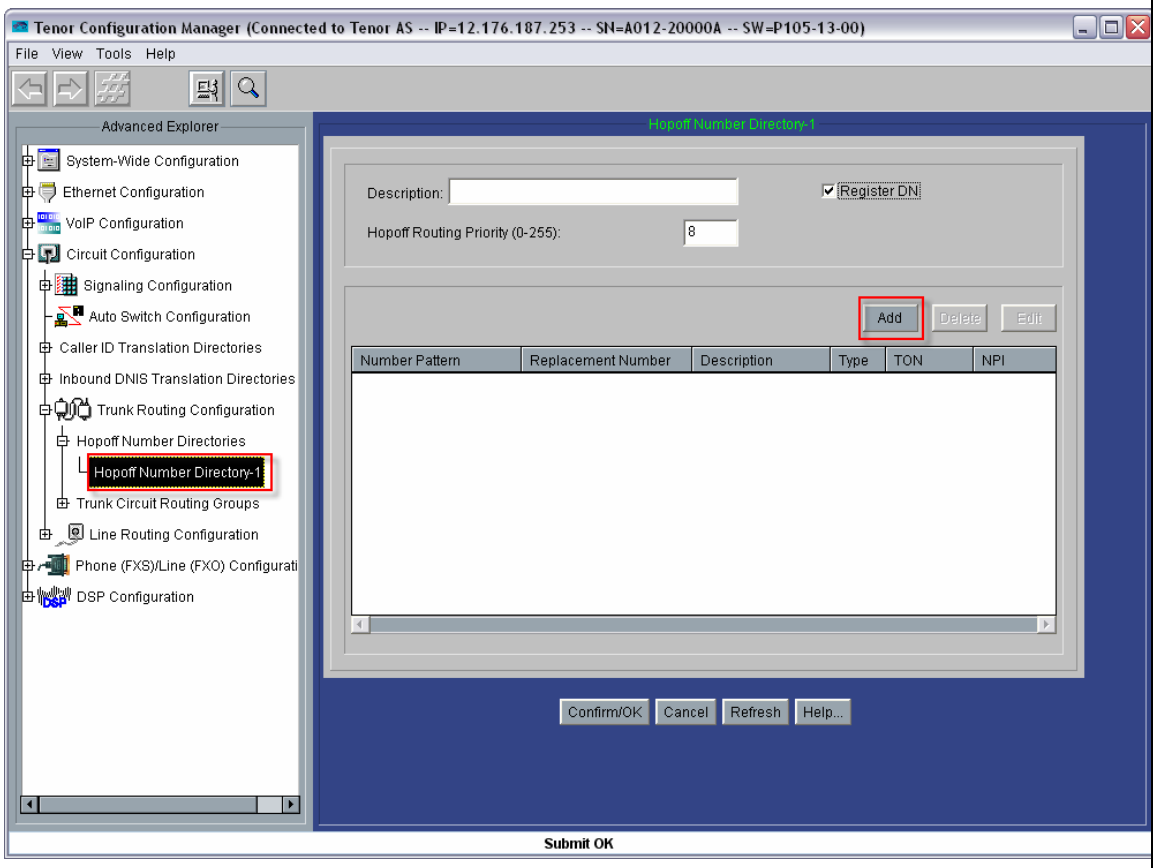
9. Define the correct mapping between FXS and FXO ports by using the same Channel Hunt Algorithm for each set of ports.
From the **Advanced Explorer** panel on the left, expand **Circuit Configuration → Trunk Routing Configuration → Trunk Circuit Routing Groups**, and highlight the **Trunk Circuit Routing Group-line** field.

Click on the **General** tab and ***select*** **Ascending** for the **Channel Hunting Algorithm**.

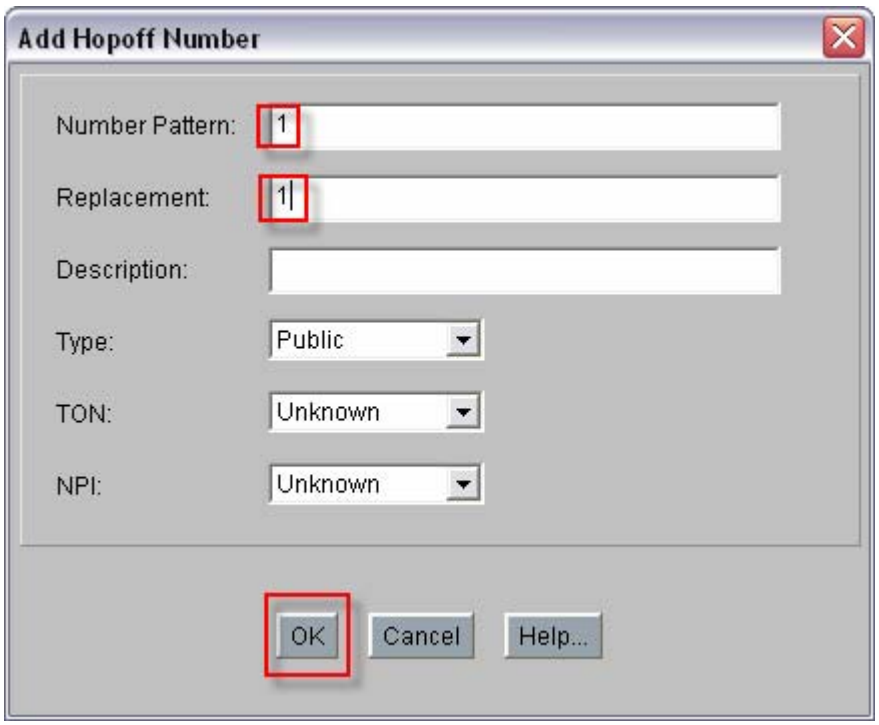
Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change.



10. Define the phone numbers (or number pattern) that will be permitted to Hopoff to the PSTN network when the IP network is not available.
Under the Advanced Explorer panel on the left, expand **Circuit Configuration → Trunk Routing Configuration → Hopoff Number Directories**, and highlight the **Hopoff Number Directory-1** then click **Add**.

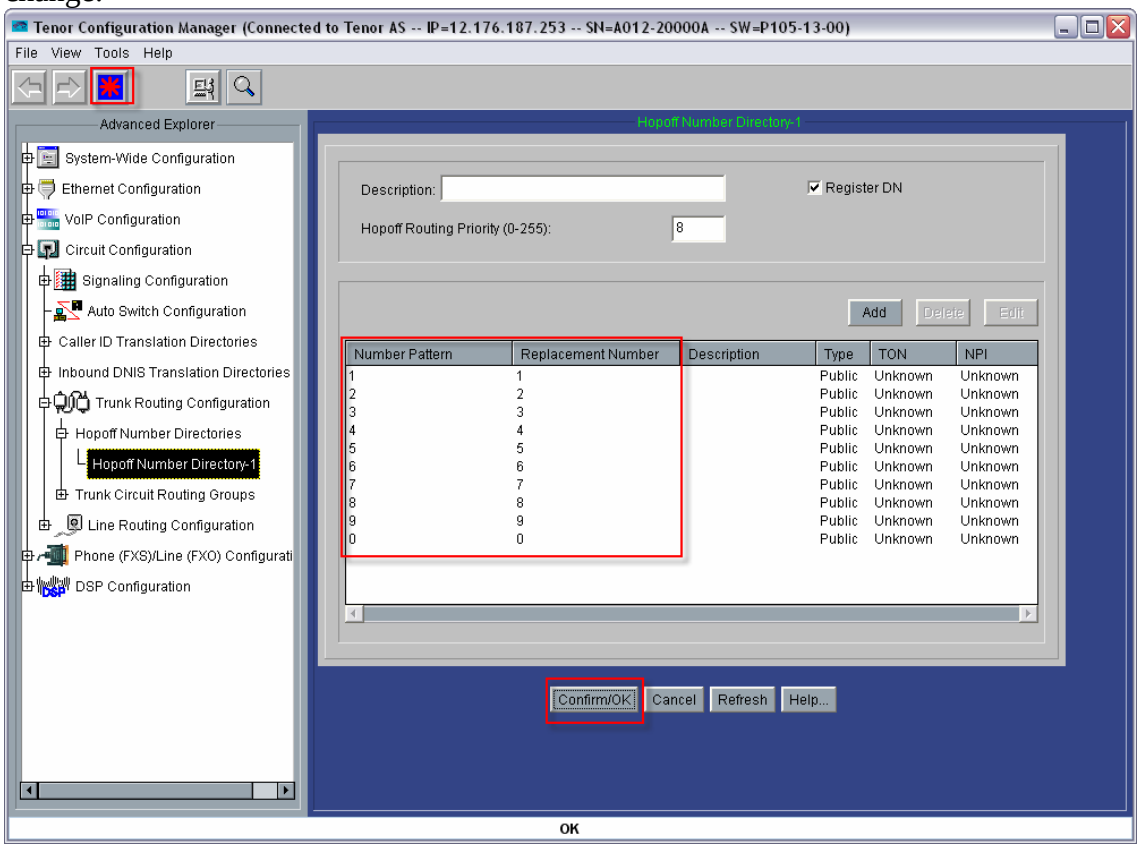


11. Provision numbers in the Hopoff table that will be allowed to terminate via PSTN. Including the digits 0 through 9 in the Hopoff table will insure that all phone calls will Hopoff to the PSTN network when the IP network is not available.
In the **Add Hopoff Number** pop-up window, insert **"1"** in the Number Pattern entry field and the Replacement window field.
Click **OK** to complete.



12. Repeat **steps 10 and 11 above** to add entries in the **Hopoff Table** for **2, 3, 4, 5, 6, 7, 8, 9, 0**.

Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change.



6.1.2 Testing MultiPath Survivability

Note: The following test procedure will disconnect all in progress VoIP calls. Please contact Quintum Technologies customer support if a non-disruptive test procedure is required.

The easiest way to test the operation of MultiPath Survivability is to temporally disconnect the cable from the LAN port. See the diagram below.

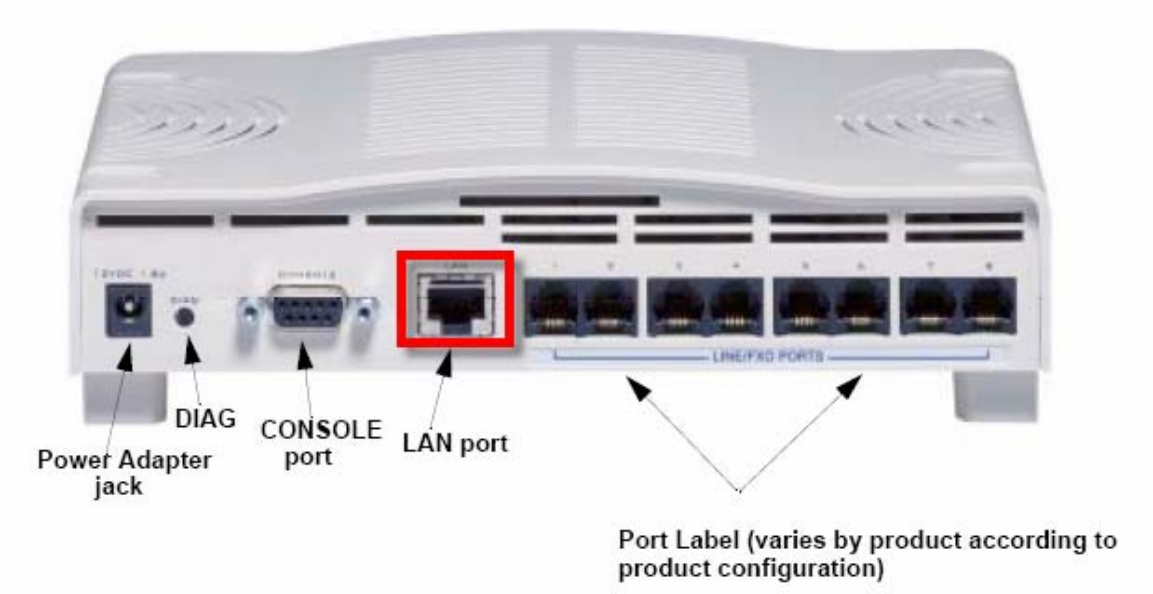


Figure 11 – Testing MultiPath by Disconnecting LAN port

Place a call while the Tenor is in this state and confirm that the call is completed. A visual indication that the call is being completed via the Tenor MultiPath capability will be two Analog Port LED's illuminating green, showing activity. See the figure below.

Figure 12 - Testing MultiPath – Active LED's

6.2 Automatic Protection Switching (APS) Survivability

The Tenor Automatic Protection Switching (APS) capability available in select Tenor models (AFM, AFE, AXM and AXE), provides the ability to complete calls via PSTN lines when power is disrupted or disconnected from the Tenor.

Within the Tenor mechanical relays are used to provide the APS capability. When power is removed from the Tenor's relays, the relays close, connecting the FXS ports directly to the FXO ports. This operational state is referred to as "Bypass Mode" and is depicted in the following diagram.

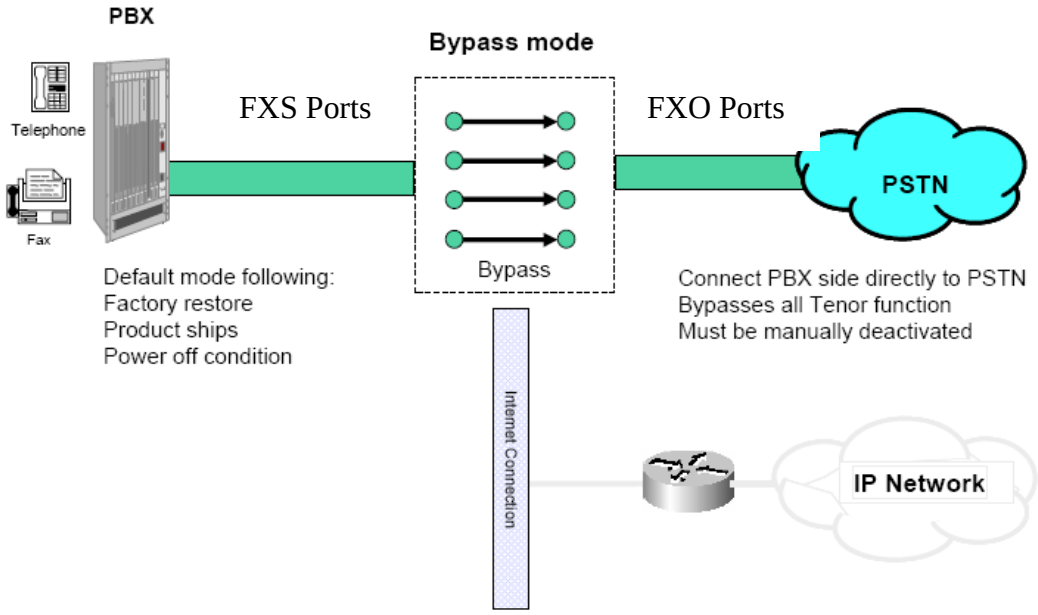


Figure 13 – Tenor with APS capability in Bypass mode

6.2.1 Configuring APS Survivability

Tenors that are manufactured with the APS capability require no additional configuration to enable the Power Off switch to Bypass Mode.

6.2.2 Testing APS Survivability

Note: The following test procedures will disconnect all in progress VoIP calls. This survivability feature can not be tested without disrupting active calls.

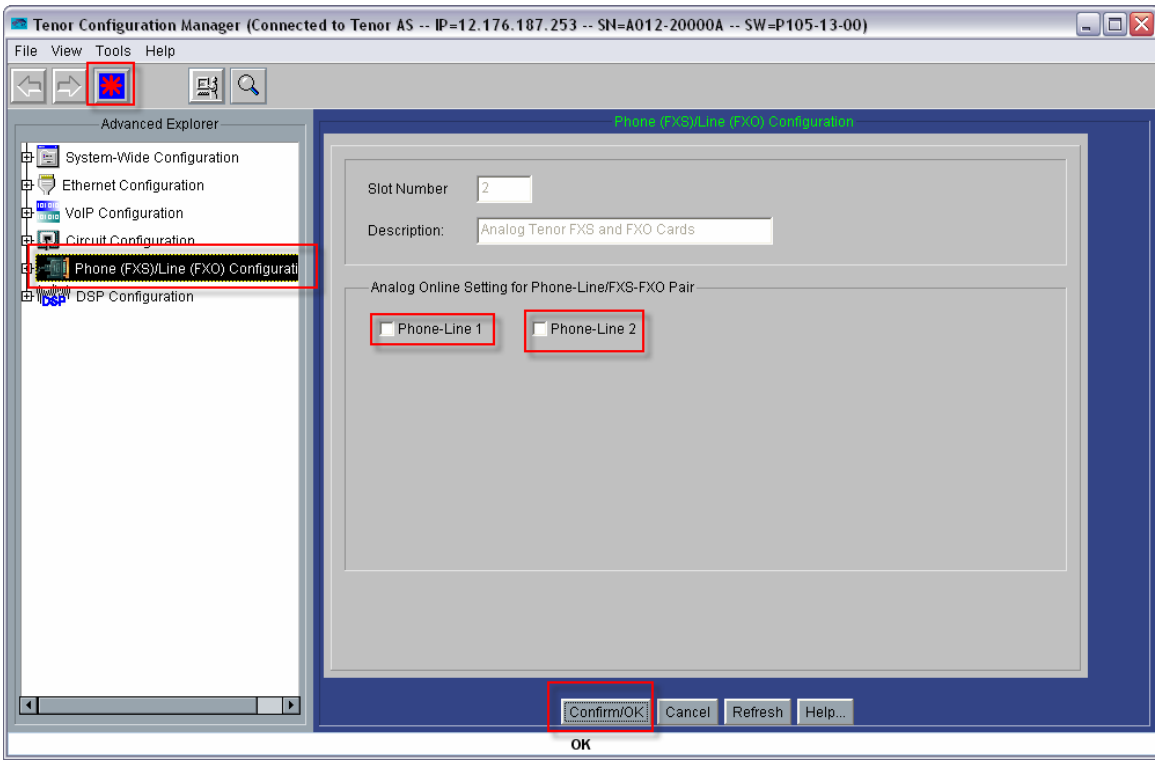
The easiest way to test the operation of APS Survivability is to temporally disconnect the power from the Tenor. See the diagrams in this guide for the location of the power cord/adapter.

It is also possible to put the Tenor into Bypass Mode by setting all the Tenor ports "offline" via the Configuration Manager.

Under the **Advanced Explorer** panel on the left, highlight **Phone (FXS)/Line (FXO) Configuration**, and **Un-Check** the **Analog Online Setting for Phone-Line/FXS-FXO Pair**.

Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change.

An auditable click will be heard as the relays close and the Front Panel Port Activity LED's will blink on and off.



7 Support for IP Flexible Reach Calling Plan A Dial Plan

The AT&T IP Flexible Reach Calling Plan A will only provide calling service to On-NET end-points and Off-Net calling to Long Distance and International locations. The AT&T Calling Plan A will not terminate calls to N11 (ex. 211, 311, 411, 511, 611, 711, 811, 911), 8YY-XXX-XXXX, 500-NPA-NXX-XXXX , 700-NPA-NXX-XXXX, 900-NPA-NXX-XXXX, NPA-555-XXXX, and Operator (0, 0+, 00, 01) numbers. See the AT&T Business VoIP Service Guide for more details.

The customer is responsible for providing PSTN lines to support 911 and the other calling services (N11, 8YY, etc.) not provided by AT&T. The customer must provision/configure his premise equipment (Analog PBX, Analog Phones, Key System, Tenor, etc.) to properly route 911 and other non AT&T supported calling services to the PSTN lines.

Typically the customer's PBX will provide the necessary dial-plan intelligence to route 911 and other non AT&T supported calling services to the PSTN circuits. Similarly, the customer's key system will provide the necessary user selective options that would allow the 911 calls to be routed to a PSTN line. This architecture is show below in figure 14.

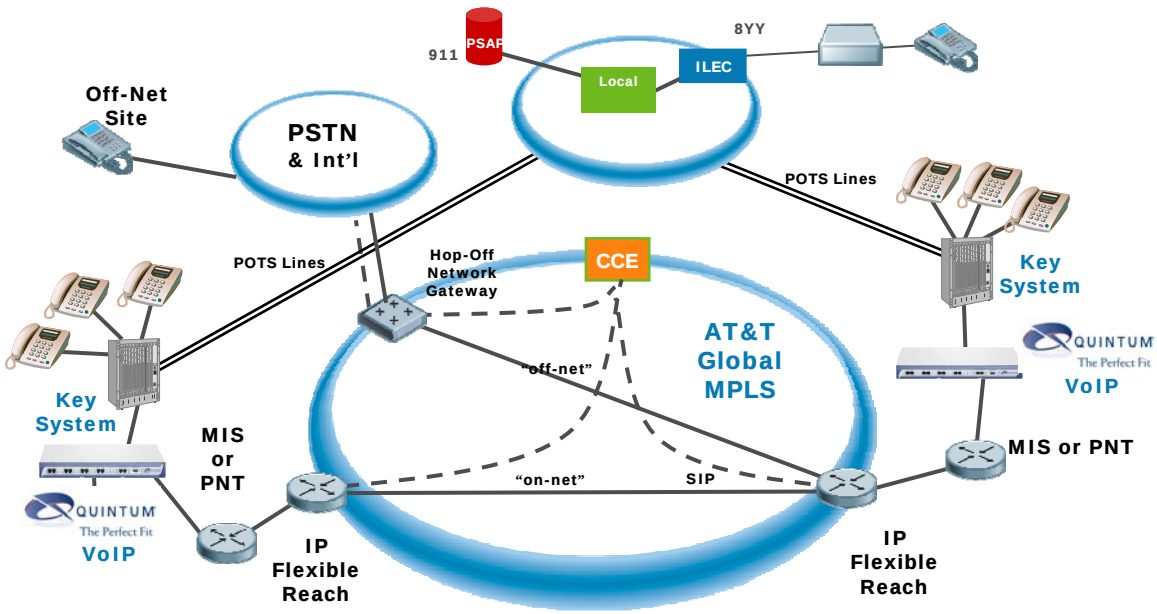


Figure 14 - Key System routes calls to POTS lines

If the customer Key System does not have the capability or port capacity to route 911 calls and other non AT&T supported calling services to the PSTN circuits, the Quintum Tenors can be configured to provide this functionality. Figure 15 below shows this architecture.

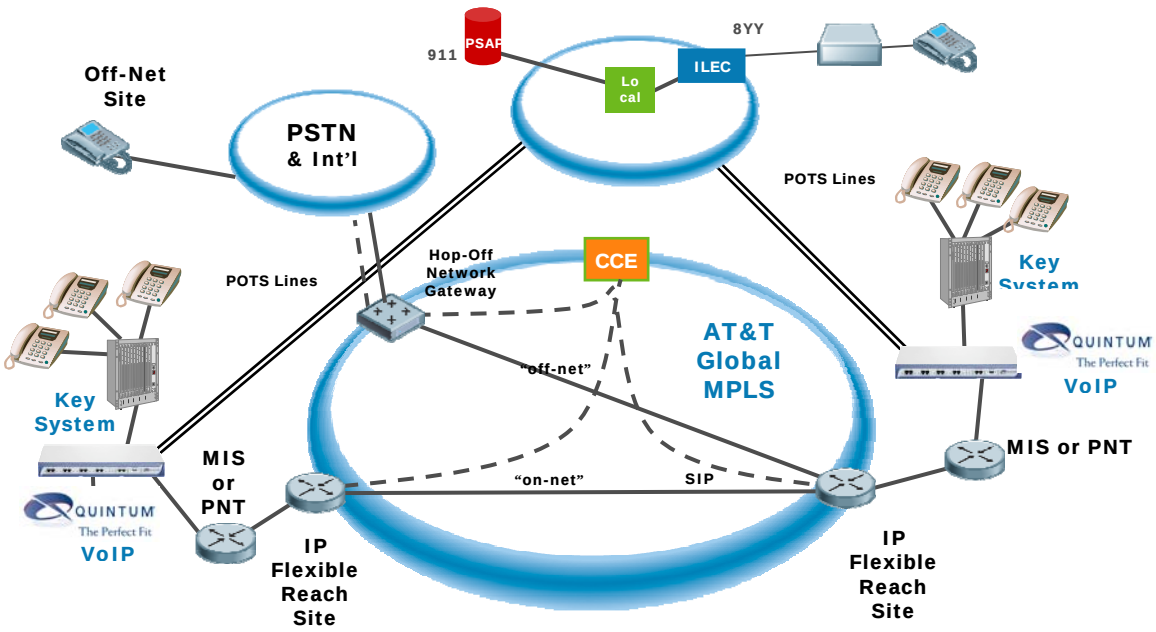


Figure 15 - Tenor routes calls to POTS lines

If the customer has Analog Phones the Quintum Tenor can be configured to route 911 calls and other non AT&T supported calling services to PSTN circuits. Figure 16 below shows this architecture.

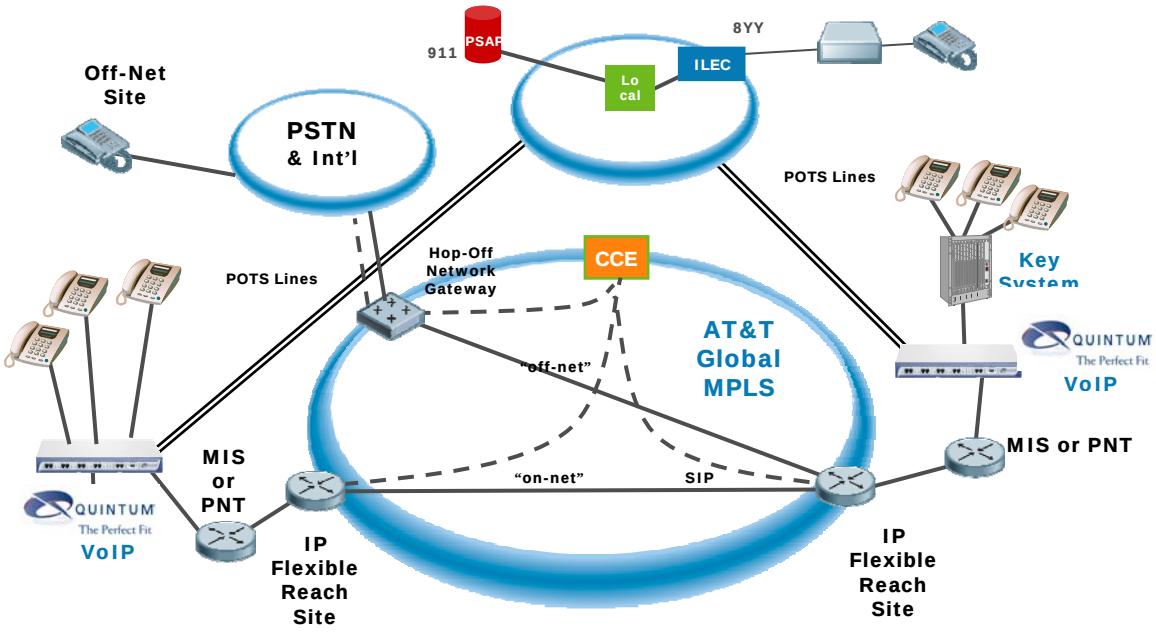


Figure 16 - Tenor routes calls to POTS lines

7.1 Tenor ByPass Routing Capability

The Tenor ByPass Routing Capability available in select Tenor models (AFM, AFE, AXM and AXE), provide the ability to complete calls via PSTN / POTS circuits instead of via the VoIP network.

A Bypass Directory Number (BND) is a telephone number that is automatically sent to the trunk-side (PSTN); it will not be routed by VoIP. Some examples of typical bypass numbers include toll-free calls, emergency calls (911), or high security calls. A bypass number should be specified in the format dialed from the line-side (Key System / PBX). It is permissible to use a "*" as a wildcard digit (e.g., 1800*).

7.1.1 Configuring the ByPass Routing Capability

To enable ByPass routing requires that at least one FXO line be attached to the Tenor. The following steps describe how to configure ByPass Routing on the Tenor to support the AT&T IP Flexible Reach Calling Plan A.

In order for the Bypass Number Directory (BND) to function, the following conditions should be met:

- The Trunk Circuit Routing Group (TCRG) and Line Circuit Routing Group (LCRG) should have a matching PassThroughID.
- The dialed number should match a number in the BND.

The literal characters "." (period) and "*" (asterisk) are supported. The "." matches a single digit (e.g., ".11" will match all N11 calls: 011, 111, 211, 311, 411 511, 611,711, 811, 911 calls). The "*" (asterisk) is interpreted as a wild card for 0 or more digits at the end of a string (e.g., 800* will match all 800 calls).

Step	Description
1.	Enable one or more FXO lines on the Tenor. Steps 5 through 8 in section 6 give the details on how to enable an FXO line.

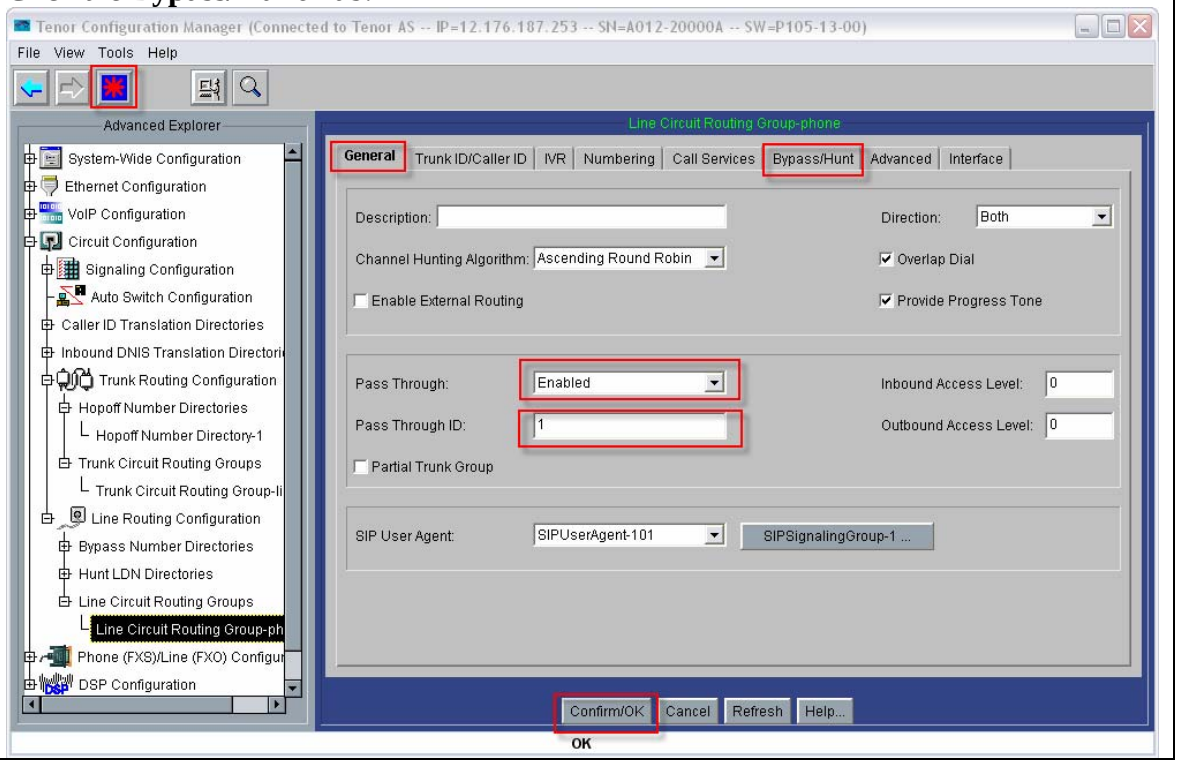
2. Enable PassThrough and configure matching PassThroughIDs in the LCRG and TCRG.

From the **Advanced Explorer** panel on the left, expand **Circuit Configuration → Line Routing Configuration → Line Circuit Routing Groups**, and highlight the **Line Circuit Routing Group-phone** field.

Click on the **General** tab under the **Line Circuit Routing Group-phone** panel on the right. From the **Pass Through** drop down menu, select ***Enabled*** and enter a ***"1"*** in the **Pass Through ID** field**.**

Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change.

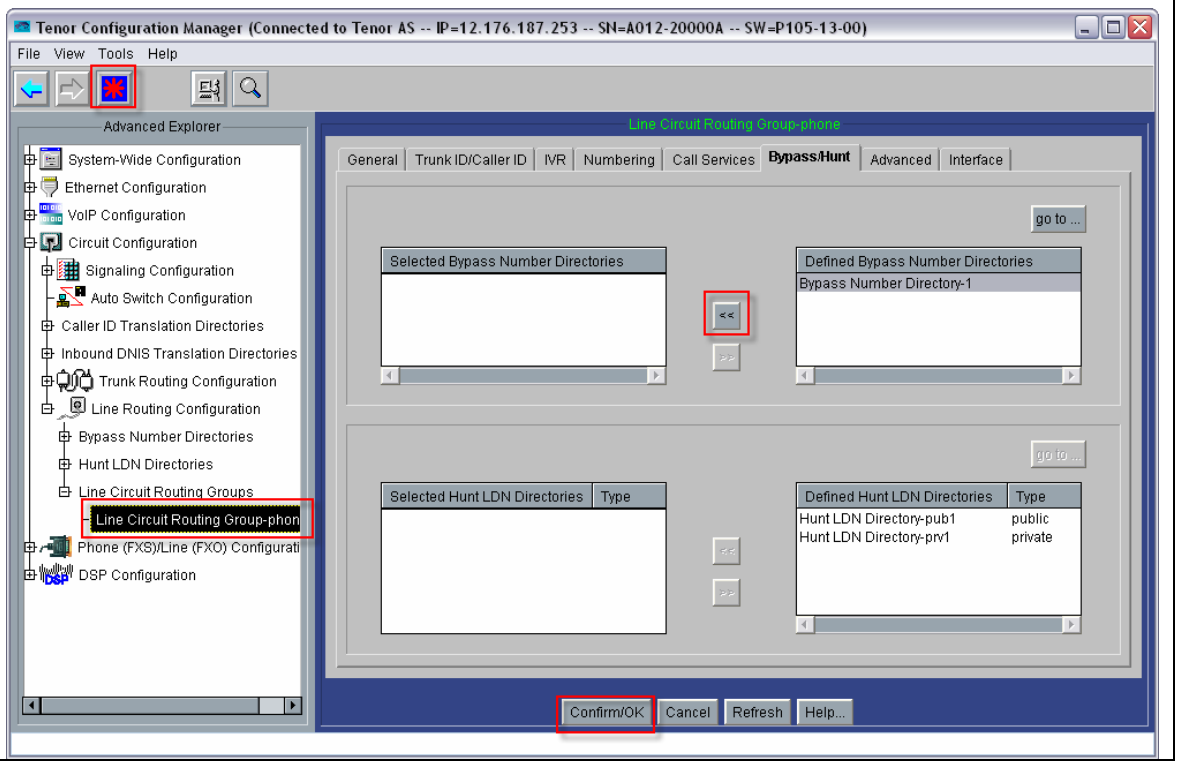
Click the **Bypass/Hunt** Tab.



3. Associate a **Bypass Number Directory** with the **Line Circuit Routing Group-phone**.

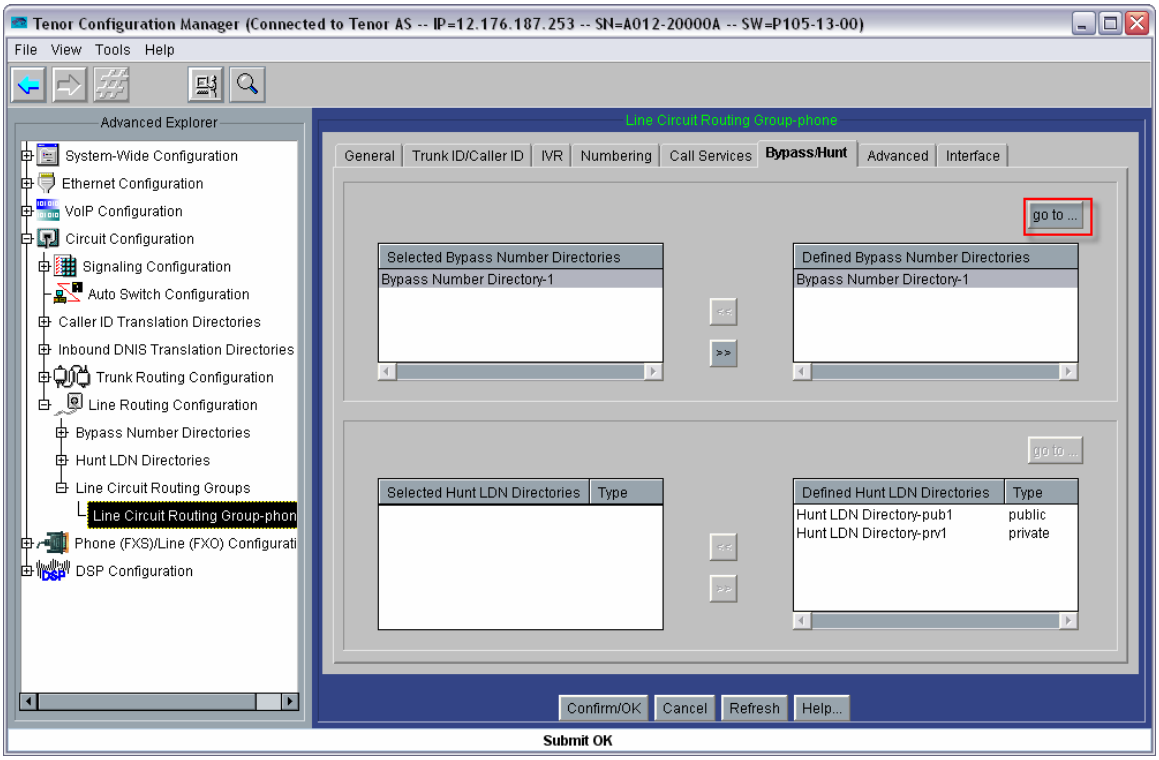
From the **Bypass/Hunt** tab under the **Line Circuit Routing Group-phone** panel on the right. Highlight the **Bypass Number Directory-1** in the Defined List and click the arrow button << to move the **Bypass Number Directory-1** Selected List.

Click **Confirm/OK** then the sunburst icon on the menu bar to implements the change.

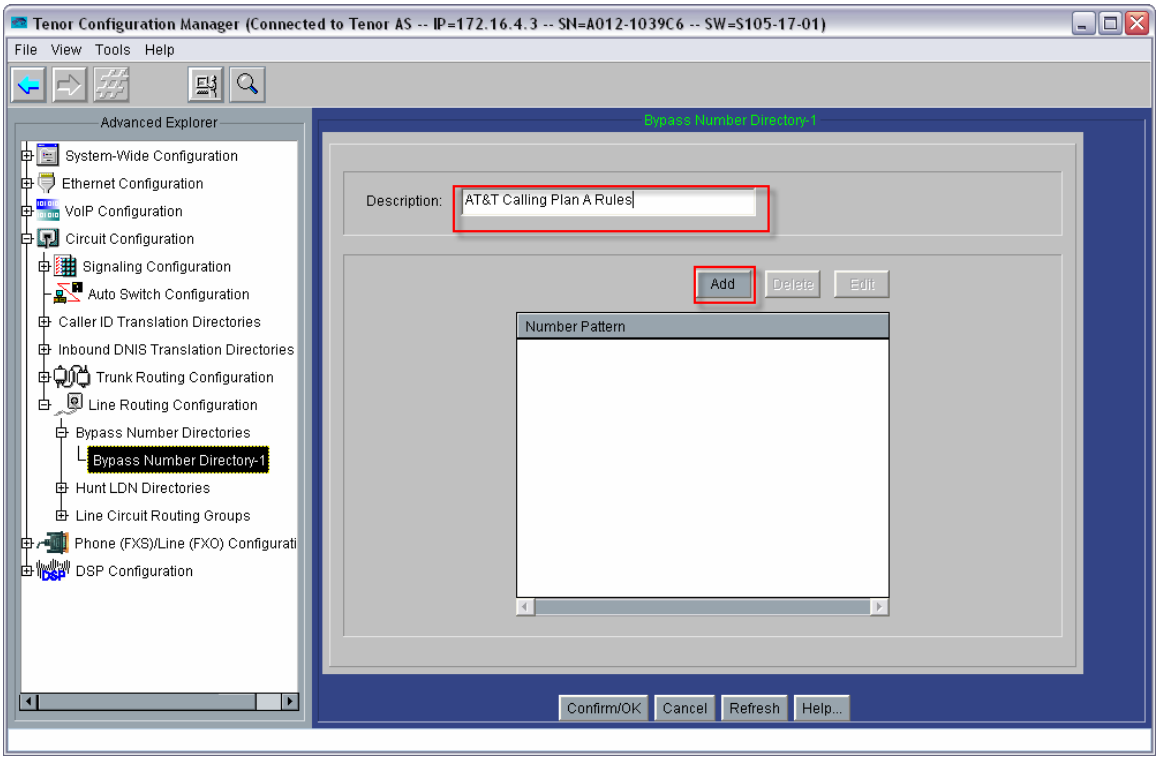


4. Click on the ***go to*** button 

to enter Numbers in the Bypass Directory-1.



5.	Enter a **Description** for the **Bypass Number Directory-1** (AT&T Calling Plan A Rules) and Click the **Add** button.



6. In the **Add Bypass Number** pop-up window, enter the number or number patterns that can not be completed via that AT&T IP Flex Reach VoIP Network.

Click **OK** to continue.

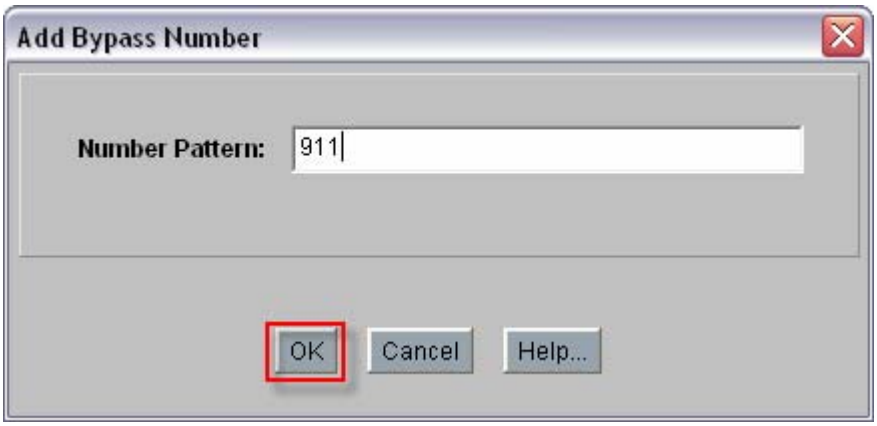
A Bypass Directory Number is a telephone number that is automatically sent to the Trunk Side (PSTN); it will not be routed via VoIP. Some examples of Bypass Numbers include local calls, emergency calls (911), or high security calls. A Bypass Number should be specified in the format dialed from the line-side (Key System / PBX).

The literal characters "." and "*" are supported. The "." (period) matches a single digit. The "*" (asterisk) is interpreted as a wild card for 0 or more digits at the end of a string (e.g., 1800*).

The Tenor will support 8 Bypass Directories with 32 numbers per directory.

You may associate all 8 **BypassNumberDirectories** on the Tenor with a LCRG.

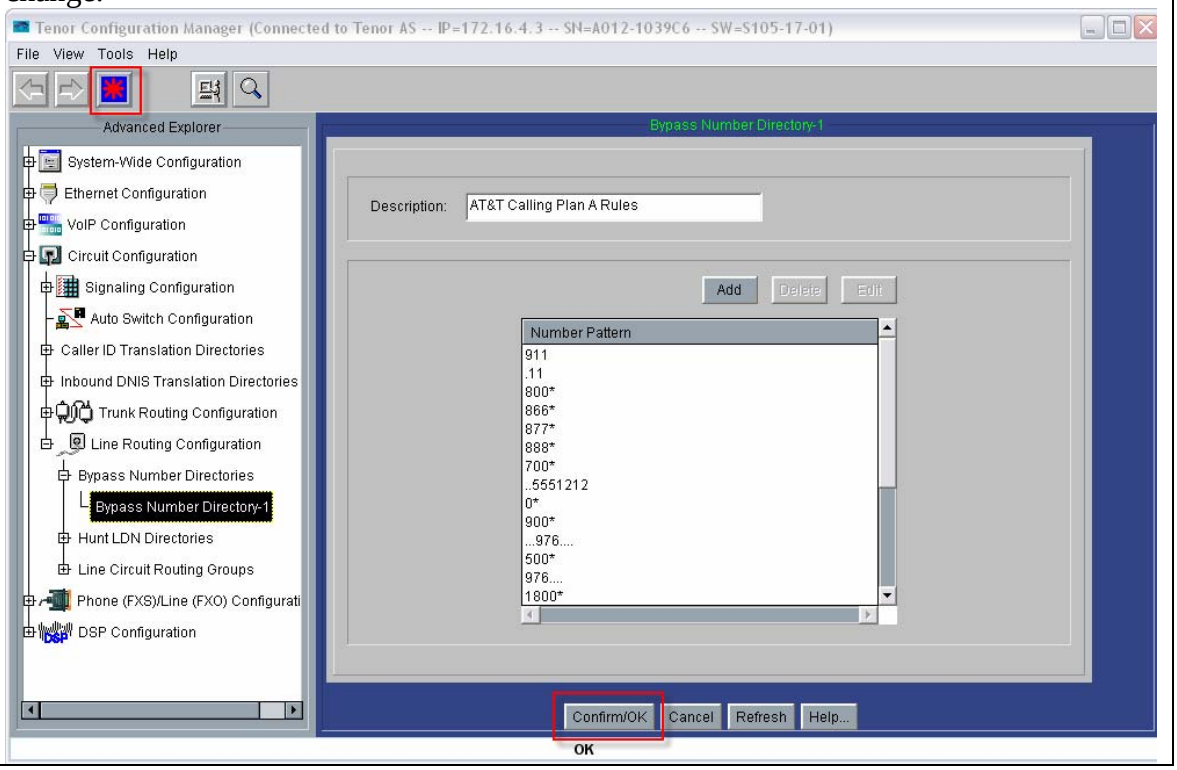
For a **BypassNumberDirectory** to be "in effect," you must associate it with a Line Circuit Routing Group.



7. The customer must evaluate his businesses calling needs and local line connectivity to populate the BypassNumberDirectory with the correct numbers. Below are example entries:

Entry	**Description**
911 911#	Bypass Emergency calls *Note: The "#" (pound sign) is defined as the End of Dial Digit by default.*
.11 .11#	Bypass all N11 calls *Note: The "#" (pound sign) is defined as the End of Dial Digit by default.*
800* 1800* 866* 1866* 877* 1877* 888* 1888*	Bypass Toll Free calls
700* 1700*	Bypass 700 calls (special network)
500* 1500*	Bypass 500 calls (one number service)
…5551212 1…5551212	Bypass directory assistance calls
0*	Bypass Operator calls
900* 1900* 976…. 1976…. …976… 1…976…..	Bypass pay for service calls

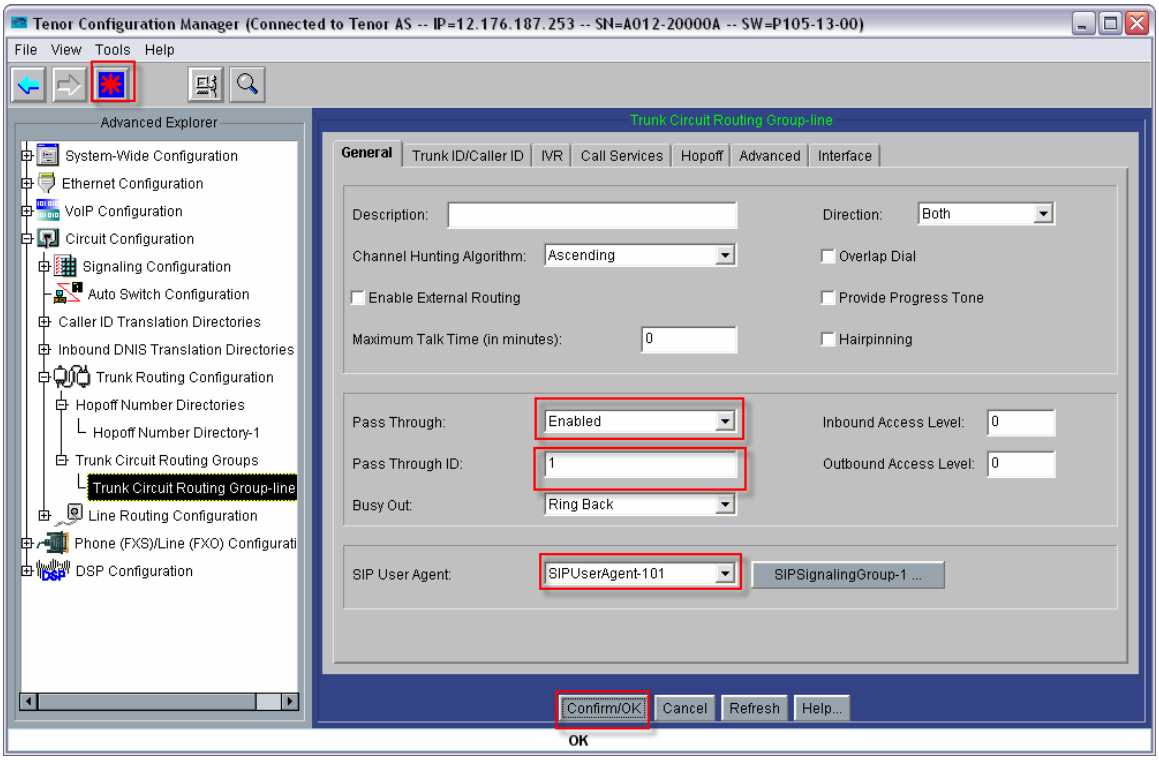
8. Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change.



9. Enable PassThrough and configure matching PassThroughIDs in the TCRG and LCRG.

From the **Advanced Explorer** panel on the left, expand +**Circuit Configuration → Trunk Routing Configuration → Trunk Circuit Routing Groups**, and highlight the **Trunk Circuit Routing Group-line** field.
Click on the **General** tab under the **Trunk Circuit Routing Group-line field"** on the right. From the **Pass Through** drop down menu, select ***Enabled*** and enter a ***"1"*** in the **Pass Through ID** field**.** From the **SIP User Agent** drop down menu, select ***SIPUserAgent-101.***

Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change.



8 Troubleshooting

For technical support on the Quintum Tenor AF and Tenor AX, contact Quintum at 877-435-7553, and also refer to www.quintum.com

9 Acronyms List

Acronym	Definition
DTMF	Dual Tone Multi Frequency – (midcall digits)
FXO	Foreign eXchange Office - Interface that receives telephone service, typically from a Central Office of the Public Switched Telephone Network (the plug on the phone).
FXS	Foreign eXchange Subscriber - Interface that delivers telephone service from the local phone company's Central Office (the plug on the wall).
LCRG	Line Circuit Routing Group
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
TCRG	Trunk Circuit Routing Group

10 Additional References

[1] Tenor AX VoIP Multipath/Gateway Switch Product Guide, P/N 480-0062-00-12
http://www.quintum.com/support/products/2G/tenor_2G/sysdoc/Tenoraxuserguide.pdf

[2] Tenor AF VoIP Multipath/Gateway Switch Product Guide, P/N 480-0084-00-11
http://www.quintum.com/support/products/2G/tenor_2G/sysdoc/TenorAFUserGuide.pdf

[3] Tenor Configuration Manager/Tenor Monitor Product Guide. P/N 480-0028-00-05
http://www.quintum.com/support/mgmt/TenorConfigManagerUsersGuide.pdf

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Appendix 1 Dump of Tenor Database Configuration

The following embedded document contains a complete printout of the Tenor database configuration used for testing with the AT&T IP Flexible Reach Service.

C:\Documents and Settings\kevin-honig\