

Avaya one-X Quick Edition and Quintum Tenor SSG

Configuration Guide for use with AT&T's IP Flexible Reach Service

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1. OVERVIEW

This document describes the configuration procedures required for an Avaya one-X Quick Edition and Quintum Tenor Survivable SIP Gateway (SSG) to make full use of the capabilities of AT&T Flex Reach Trunk solution.

The Quick Edition telephone uses the Session Initiation Protocol (SIP) to communicate with Tenor Gateway for call control. It also translates voice to audio packets for transmission across a packet network.

This guide describes the specific configuration items that are important for use with AT&T Flex Reach solution. It does not describe the purpose and use of all configuration items on the Quick Edition and Tenor Gateway. For those details, see the *Avaya one-X Quick Edition Release 3.0.0 System Administrator Guide 1* supplied by Avaya and the *Tenor AF Hardware Guide 2* supplied by Quintum Technologies Inc.

2. NETWORK CONFIGURATION

Figure 1 Typical network layout

Figure 1 shows a typical network configuration on the customer premise. The Quick Edition telephones and the Tenor gateway can be connected to the customer LAN using an Ethernet switch and can reside behind the AT&T managed router. Power is ideally supplied to Quick Edition telephones by plugging the telephones into 802.3af PoE-enabled Ethernet wall jacks using the supplied CAT5 modular line cords.

The Tenor gateway acts as a registrar and proxy to the Quick Edition phones and provides connectivity to AT&T Flex Reach trunk. AT&T Flex Reach service provides NAT traversal capability and connectivity to PSTN, other **Avaya QE/Quintum** IP-PBXs, and TDM PBXs.

A DHCP server is required on the customer LAN to assign the IP addresses to the Quick Edition phones. The DHCP server can either be an existing device on the customer site, or be enabled on the AT&T managed router.

If the DHCP server is not available when the system boots up, the Quick Edition phones will automatically configure their network interfaces with the IP addresses in the range of 169.254.x.x. This feature allows QE phones to communicate with each other on the local network. When the DHCP server is available, recycling the power on the QE phones will enable them to acquire the standard private IP addresses from the DHCP server.

3. ONE-X QUICK EDITION 4610/4621 CONFIGURATION


The Quick Edition is a distributed system, with the intelligence and the site configuration held and stored on each Quick Edition IP telephone that comprises that site. When a configuration change is made on a telephone in this site, the change is automatically sent to all the other Quick Edition IP telephones in this site using multicast communications.

The Quick Edition is configured through its embedded Web Server. Because of its distributed nature, the configuration can be done to any Quick Edition device IP address.

The capabilities of the Quick Edition have been verified for use with Quintum Tenor Gateway and AT&T Flex Reach service based on the settings described in the following table. For more information on the meaning, purpose, and applicability of the individual configuration items, see the *Avaya one-X Quick Edition Release 3.0.0 System Administrator Guide* [1] supplied by Avaya.

3.1. Verifying QE telephone version

Perform the following steps to verify that the QE 4610 and 4621 IP phones are loaded with compatible software.

1. Click the options button () on the telephone to bring up the options menu.
2. Select the following menu items: Set Details->Release Ver
3. Verify the following items:

Release Ver	3.0.6
Firmware Version	7.0.52
Boot	2.2.14

3.2. Accessing System Level Configuration

Step	Command	Purpose
Step 1	Find out IP address of a QE set Click on Options button. Select Set Details → Ext IP and MAC address would be listed.	The IP address of a QE telephone is needed to access the system level configuration.
Step 2	Access the web portal of the QE set Enter http://<ip address of the QE set> Select System Options Enter your Admin Password Click Login.	This is where we will be able to configure the system level configuration.

3.3. Codec Preference Configuration

Setting codec sequence preference is a system level configuration. Perform the following steps to set G729 to be the first codec after you have logged into the system option web portal of a QE telephone.

Step	Command	Purpose
Step 1	Select Networking on System Options Menu	Gets you to the menu item to configure codec preference
Step 2	Select audio Bandwidth Click Change Details Set Audio Bandwidth to Low	Sets G729A codec preference over G711u/A codec

Figure 2 Setting Codec Preference

3.4. Service Provider Configuration

This section describes the service provider configuration that is required to connect to AT&T IP Flex Reach Service through the Quintum Tenor gateway. The Tenor gateway is acting as a SIP registrar and proxy server for QE phones. This example is based on the network topology shown in Figure 1. The IP Address used in this example may be different depending on your site configuration. Please see figure 3 below for reference.

In order to forward incoming SIP message through the WAN port to the Tenor gateway, it needs to configure managed router to create a mapping between WAN port 5060 to Tenor gateway's internal IP address and port.

Step	Command	Notes
Step 1	Add a Service Provider Domain. Set System Options/Service Provider/Add Configuration/Domain Name = "172.16.4.3"	Set the Quick Edition service provider domain name. The domain name must match Tenor gateway's IP address.
Step 2	Set Proxy Host. Set System Options/Service Provider/Add Configuration/Proxy Host = "172.16.4.3"	The proxy host must match Tenor gateway's proxy IP address.
Step 3	Set Proxy Port. Set System Options/Service Provider/Add Configuration/Proxy Port = "5060"	The proxy port must match Tenor gateway's proxy port.
Step 4	Set Registrar Host. Set System Options/Service Provider/Add Configuration/Registrar Host = "172.16.4.3"	The registrar host must match Tenor gateway's registrar IP address.
Step 5	Set Registrar Port. Set System Options/Service Provider/Add Configuration/Registrar Port = "5060"	Set the registrar port to match the Tenor gateway's registrar port.
Step 6	Leave Outbound Proxy Host and Port empty	
Step 7	Set Realm. Set System Options/Service Provider/Add Configuration/Realm = "172.16.4.3"	Set the realm to match the domain configured in Step 2.
Step 8	Set Register Expiry Time. Set System Options/Service Provider/Add Configuration/Register Expiry Time = 60	60 seconds registration expiry time is adequate in most cases where phones and the gateway reside in the same network.



Figure 3 Service Provider Configuration

3.5. Service Provider Identity

AT&T provides two types of telephone numbers (TNs) --- virtual telephone numbers (VTNs) and non virtual telephone numbers. VTNs are usually comprised of 10 digits while non-VTNs are usually comprised of 7 or less digits. For example, if a QE telephone is associated with a VTN, the number received from AT&T would be 10 digits (i.e. 732-216-2700). If a QE telephone is associated with a non virtual TN, the number received from AT&T would be 7 digits or less (i.e. 216-2701 for a 908-216-2701 TN). However, in both cases, the 10 digit number must be sent as calling party number when originating calls to AT&T. The implication of having both VTN and non-VTN identities is that the QE telephone should be able to receive calls from both numbers but will only use the VTN number to make outgoing calls..

3.5.1. Configuring VTN

This section describes how to provision VTN from the QE System Options Web Portal. Please see figure 4 below for reference.

Step	Command	Notes
Step 1	<p>Add Service Provider Identity.</p> <p>Set System Options/Service Provider/Domain 172.16.4.3/Identities/Add Identity/Identity</p>	This identity should match a VTN provisioned for this AT&T trunk.

	= "7322162700"	
Step 2	Set Authentication Password. Set System Options/Service Provider/Domain 172.16.4.3/Identities/Add Identity/Password = "7322162700"	Any password will do --- since registration and incoming request are not challenged.
Step 3	Verify Authentication Password Set System Options/Service Provider/Domain 172.16.4.3/Identities/Add Identity/Verify Password = "7322162700"	This password must match the password entered in the "Password" field in Step 2.
Step 4	Select Incoming Extension. Set System Options/Service Provider/Domain 172.16.4.3/Identities/Add Identity/Incoming Extension = "201"	Select the internal extension that will receive incoming calls targeted for this VTN. Select the "Global" option from drop down list to enable the auto attendant for this VTN.
Step 5	Select Outgoing Extension. Set System Options/Service Provider/Domain 172.16.4.3/Identities/Add Identity/Outgoing Extension = "201"	Select the internal extension that is allowed to make calls using this VTN.
Step 6	Leave AA Enabled and AA Script as it is	They are only used for Auto Attendant functionality.

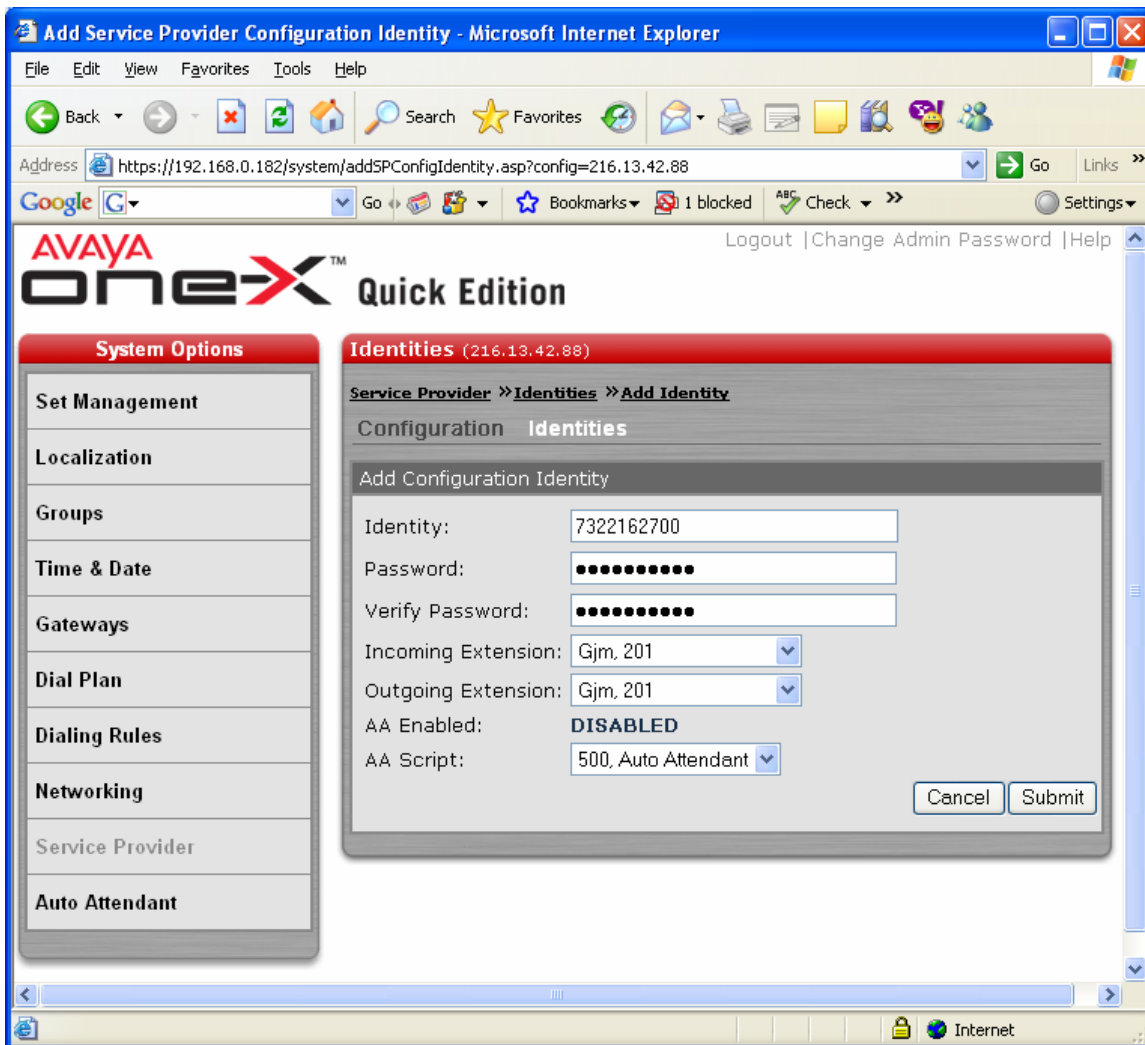


Figure 4 VTN configuration from QE Web portal

3.5.2. Configuring non-VTN

The first step in configuring a non-VTN number in a QE telephone is to create a “non-existing” extension which will ultimately be used for outgoing extension for the non-VTN identity. The main purpose of creating such a non existing extension is to disallow any QE telephone from making outgoing calls using the non-VTN number. This “non-existing” extension can be used for all non-VTN identities in one QE network. In section 3.5.1 we have configured the VTN number (9082162700) for QE extension 201. Follow the steps below to provision non-VTN number (2162700) for the same QE extension 201. Please see figure 5, 6 and 7 for reference.

Step	Command	Notes
Step 1	<p>Add a “non existing” extension manually</p> <p>Set System Options/Set Management/Add Extension/Extension = “303”</p>	<p>Make sure the extension doesn't exist and is in the valid “Extension Range” configured in System Options/DialPlan/Extension Range.</p>
Step 2	<p>Set the name of the “non existing” extension</p> <p>Set System Options/Set Management/Add Extension/Name = “303”</p>	

Step 3	<p>Set the MAC address of the "non existing" extension</p> <p>Set System Options/Set Management/Add Extension/MAC Address = "00:00:00:00:00:00"</p>	Setting MAC address to an invalid value like 00:00:00:00:00:00 ensures that there won't be any conflict with existing MAC devices.
Step 4	<p>Create the "non existing" extension</p> <p>Click the "Validate" button followed by the "Add Extension" button to create the extension.</p>	This non existing extension can be used for the outgoing extensions of all the non-VTNs in the PBX.
Step 5	<p>Add Service Provider Identity.</p> <p>Set System Options/Service Provider/Domain 216.13.42.88/Identities/Add Identity/Identity = "2162700"</p>	This identity should match a non-VTN provisioned for this AT&T trunk.
Step 6	<p>Set Authentication Password.</p> <p>Set System Options/Service Provider/Domain 216.13.42.88/Identities/Add Identity/Password = "2162700"</p>	Any password will do --- since registration and incoming request are not challenged.
Step 7	<p>Verify Authentication Password</p> <p>Set System Options/Service Provider/Domain 216.13.42.88/Identities/Add Identity/Verify Password = "2162700"</p>	This password must match the password entered in the "Password" field in Step 6.
Step 8	<p>Select Incoming Extension.</p> <p>Set System Options/Service Provider/Domain 216.13.42.88/Identities/Add Identity/Incoming Extension = "201"</p>	<p>Select the internal extension that will receive incoming calls targeted for this non-VTN.</p> <p>Select the "Global" option from drop down list to enable the auto attendant for this non-VTN.</p>
Step 9	<p>Select Outgoing Extension.</p> <p>Set System Options/Service Provider/Domain 216.13.42.88/Identities/Add Identity/Outgoing Extension = "303"</p>	Select the "non-existing" extension that is allowed to make calls using this non-VTN i.e. no extension is allowed to make calls using the non-VTN as its identity.
Step 10	<p>Leave AA Enabled and AA Script as it is</p>	They are only used for Auto Attendant functionality.

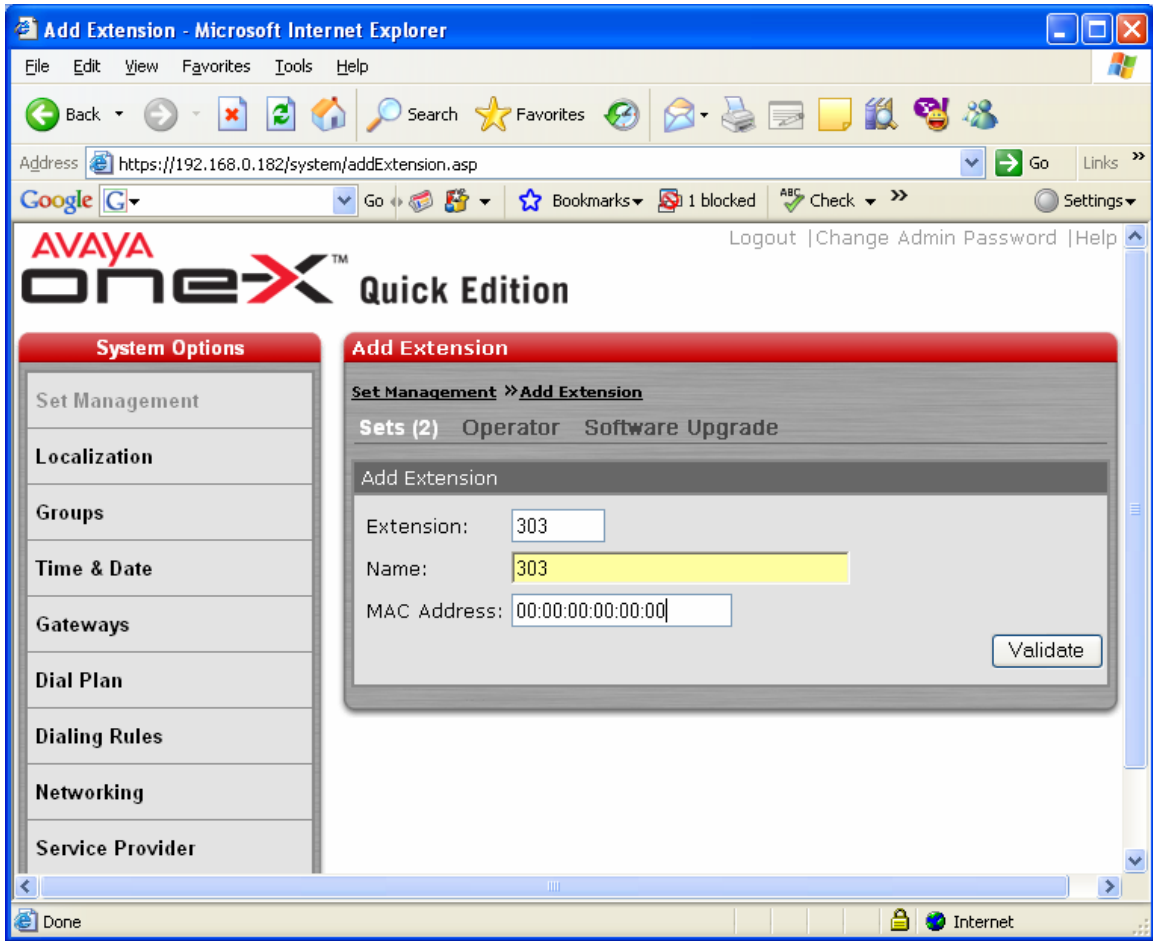


Figure 5 Provisioning a non-existing extension

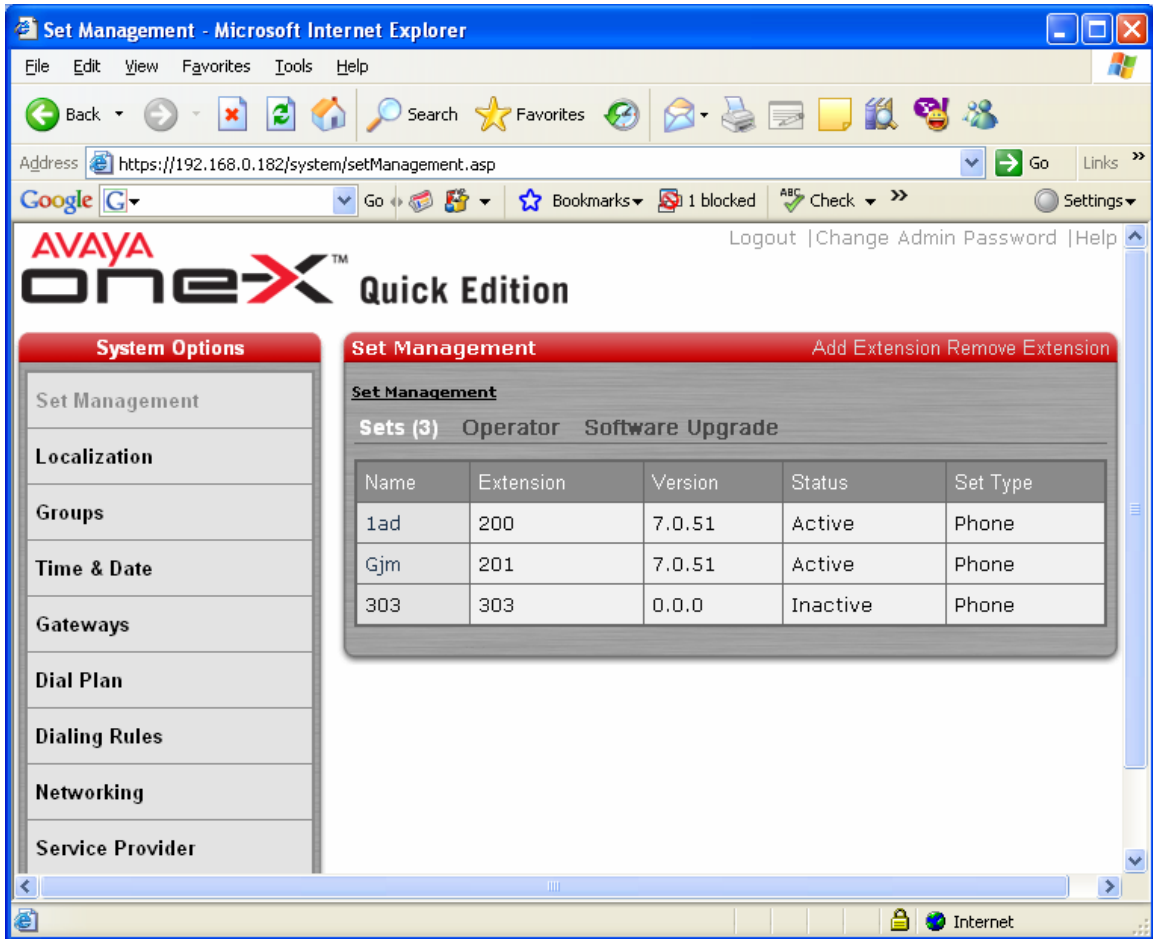


Figure 6 Non-existing extension is shown along with the existing extensions

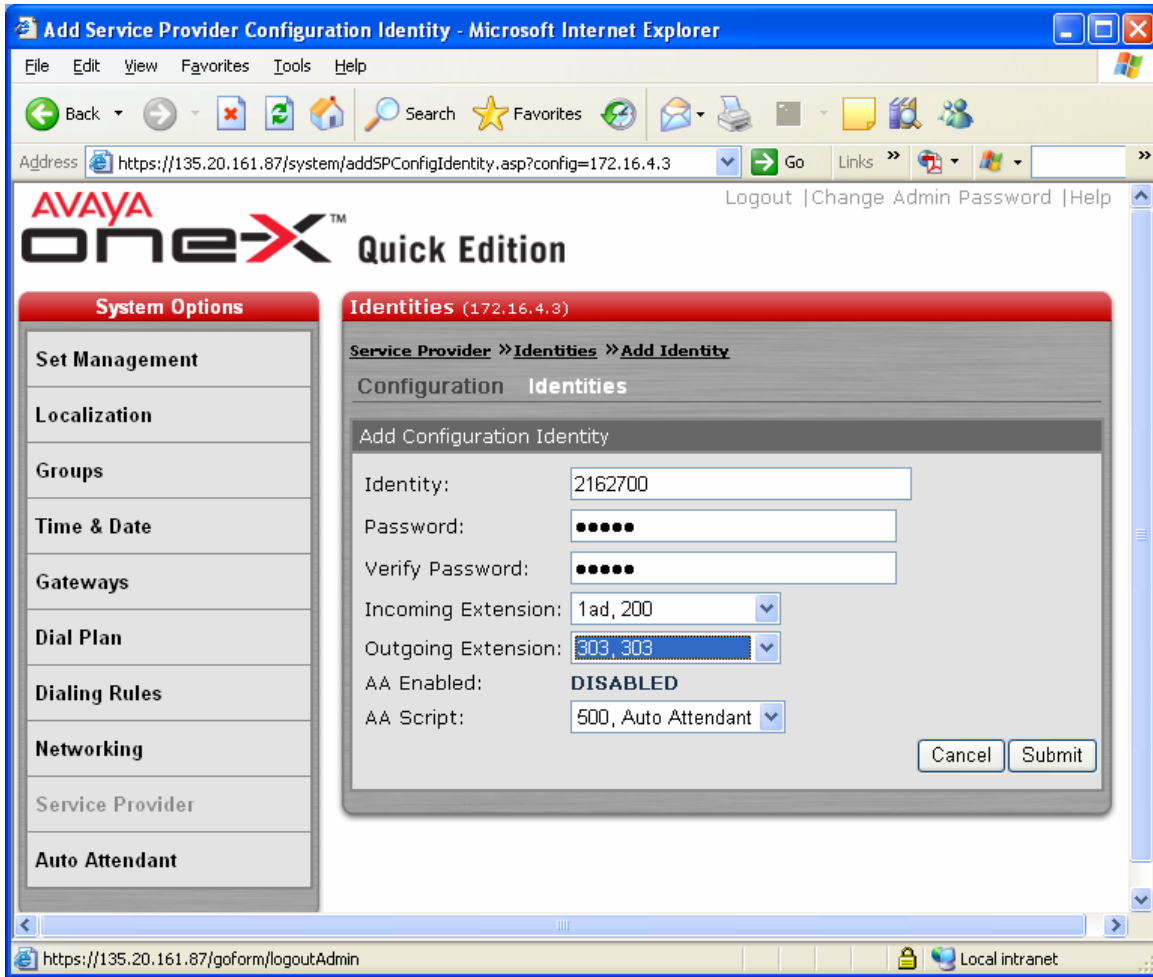


Figure 7 Provisioning non-VTN using the “non-existing” extension in Outgoing Extension

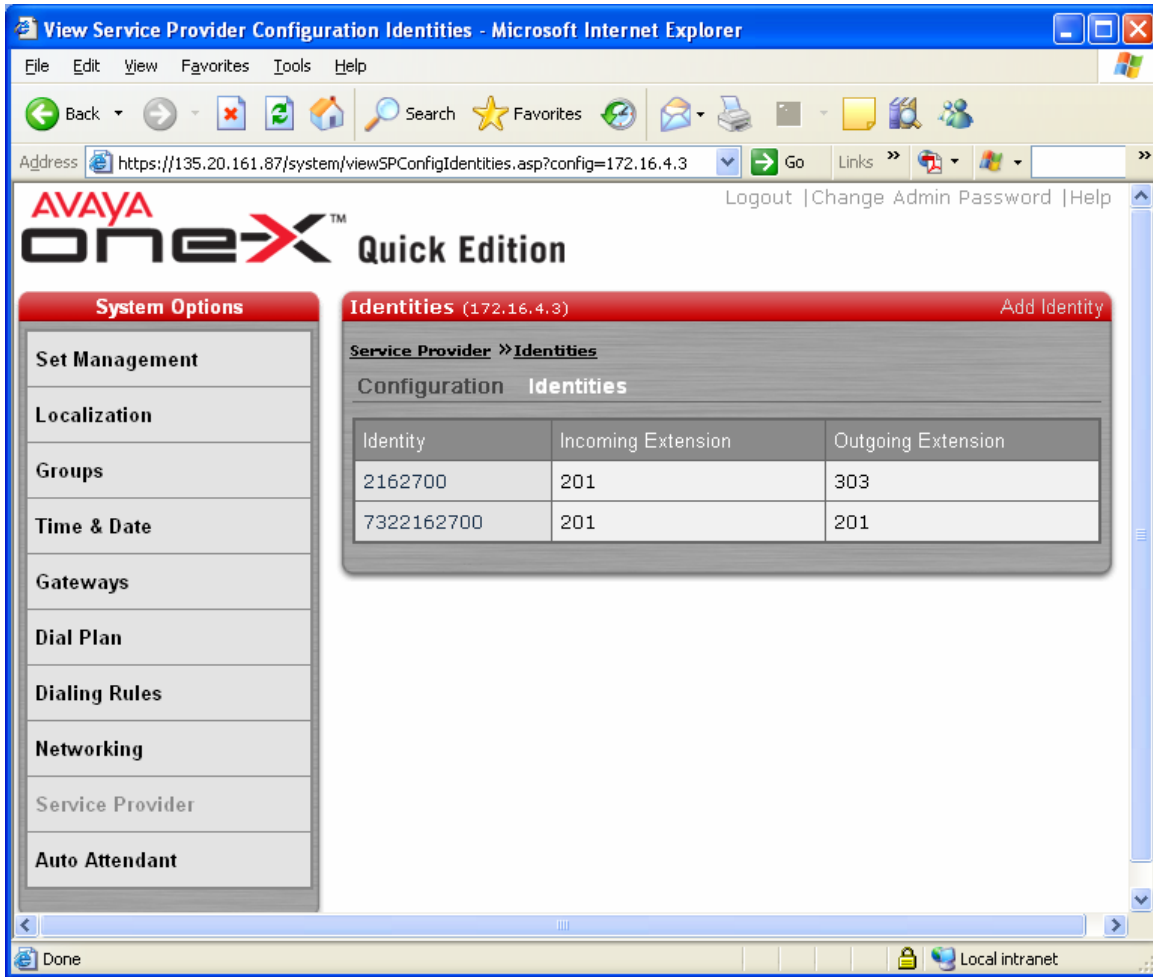


Figure 8 VTN and non-VTN identities provisioned for extension 201

4. MAKING CALLS USING QE

After all of the extensions have been properly configured from a QE web portal, you are ready to make calls. To make calls using the AT&T trunk, complete the following steps:

Step	Command	Notes
Step 1	<p>Find the SIP Code from the Dial Plan Settings</p> <p>Go to System Options/Dial Plan/View Dial Plan Settings from the QE web portal and look up the SIP code</p>	<p>By default the SIP code is 8 unless changed by the administrator.</p>
Step 2	<p>Dial a number</p> <p>Using the telephone keypad, first enter the SIP Code (e.g. 8), and then the country code and then the VTN/non-VTN for calls within North America. For international call, enter SIP code, then 011 and then the country code followed by VTN/non-VTN.</p> <p>e.g.</p> <p>In order to dial a NJ number (732-216-1203), you have to enter 817322161203</p> <p>In order to dial a UK number (+44-2476824003), you have to enter 8011442476824003</p>	<p>The prefix 1 (for North American calls) and 011 (for international calls) are being used by Quintum Tenor gateway to decide whether to route the call to the AT&T trunk or route it internally.</p> <p>Those prefixes have to be configured in the Quintum gateway also.</p>

4.1. Making N11 Calls (211, 311, 411, 511, 611, 711, 811,911)

The Avaya one-X Quick Edition IP Phones will complete N11 calls via the AT&T IP Flexible Reach Network. See the network diagram in figure 1.

1. To place an N11 call, perform one of the following actions:
 - handset operation: pick up the handset.
 - hands-free operation: press the Speaker () button or a Line/Feature () button.
 - headset operation: press the Headset () button.
2. Dial 8-N11

5. QUINTUM TENOR SURVIVABLE SIP GATEWAY (SSG)

This section describes the steps for configuring the Quintum Tenor AF Survivable SIP Gateway (SSG), to work with the Avaya one-X Quick Edition and AT&T's IP Flexible Reach Service. The Quintum Tenor AF Software release S105.17.01 was tested with the Avaya one-X Quick Edition version 3.0/3.1 and the AT&T IP Flexible Reach Service.

The Tenor AF Survivable SIP Gateway (SSG) is a signaling intermediary between the local IP endpoints (Avaya one-X phones) and the central SIP Server (AT&T IP Flexible Reach). SSG technology (internal to the Tenor) provides site survivability for branch offices in a single, integrated access solution. The SSG acts as an outbound proxy--all IP endpoints register to the SIP Server via SSG and all SIP signaling passes through it. It is an intelligent local agent that gathers and maintains routing information for all endpoints passing registrations and call signaling through it to the central SIP proxy.

In the event of a loss of connection with the central SIP Server, SSG takes control as the SIP session manager and uses its internal database of stored routing information to provide local routing support for all local IP endpoints (this routing information is automatically built from the registration and signaling information passed through the SSG acting as an outbound proxy).

5.1. Routing Scenarios Supported

- Failover from Primary to Secondary AT&T IP Border Element

5.2. Quintum Tenor AF Overview

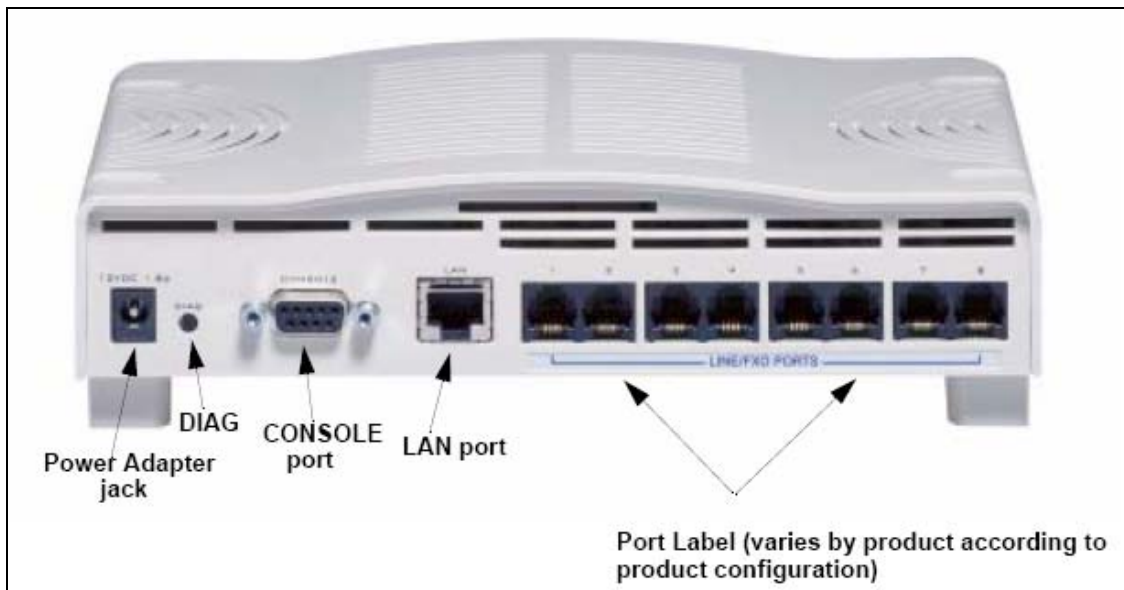


Figure 9: Tenor AF Back Panel

- **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 a 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 ¼" 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 SSG will support 50 simultaneous VoIP Calls.

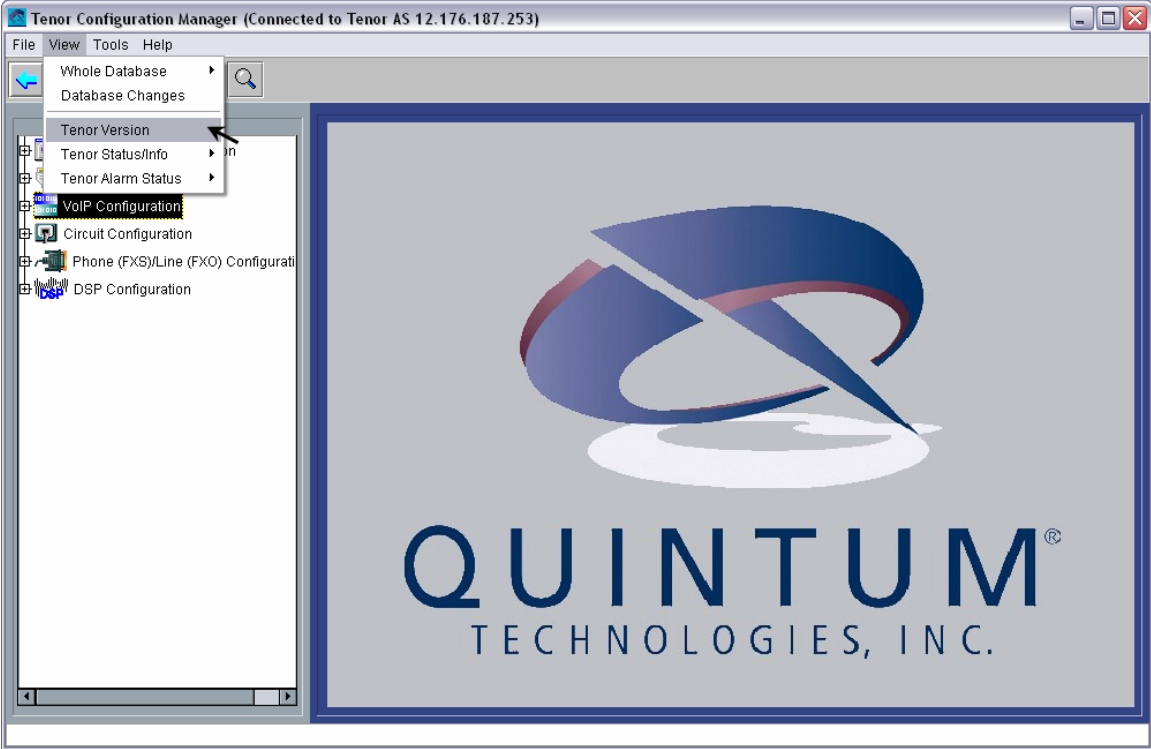
The Tenor AF SSG will also provide two analog FXS ports for connecting FAX machines.

For more details on the Tenor AF SSG, consult documents "*Tenor AF Product Guide*" and "*Tenor S Application Guide*".

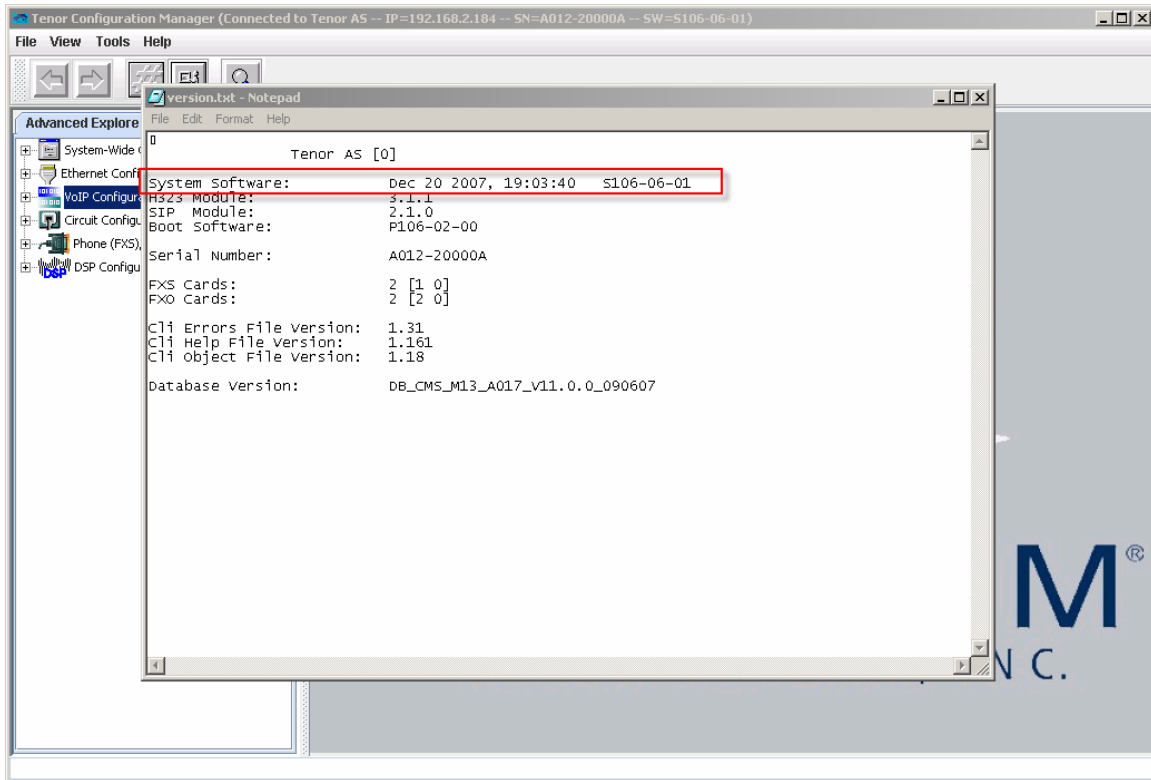
6. QINTUM TENOR CONFIGURATION GUIDE

6.1. 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 **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”.

```
192.168.2.184 - PuTTY
<8585a15d> Login:admin
Password:
Quintum# show -v
Tenor AS [0]
System Software:      Dec 20 2007, 19:03:40  S106-06-01
H323 Module:         3.1.1
SIP Module:          2.1.0
Boot Software:       P106-02-00
Serial Number:       A012-20000A
FXS Cards:           2 [1 0]
FXO Cards:           2 [2 0]
Cli Errors File Version: 1.31
Cli Help File Version: 1.161
Cli Object File Version: 1.18
Database Version:    DB_CMS_M13_A017_V11.0.0_090607
Quintum#
```

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

6.2. Standard Configuration

The following steps describe the configuration for the Tenor AF Survivable SIP Gateway to work with the AVAYA one-X Quick Edition and AT&T IP Flexible Reach service. For detailed information on installing and running the Tenor Configuration Manager, consult the *Tenor Configuration Manager Product Guide* available at the Quintum Technologies web site <http://www.quintum.com/support/mgmt/index.shtml>.

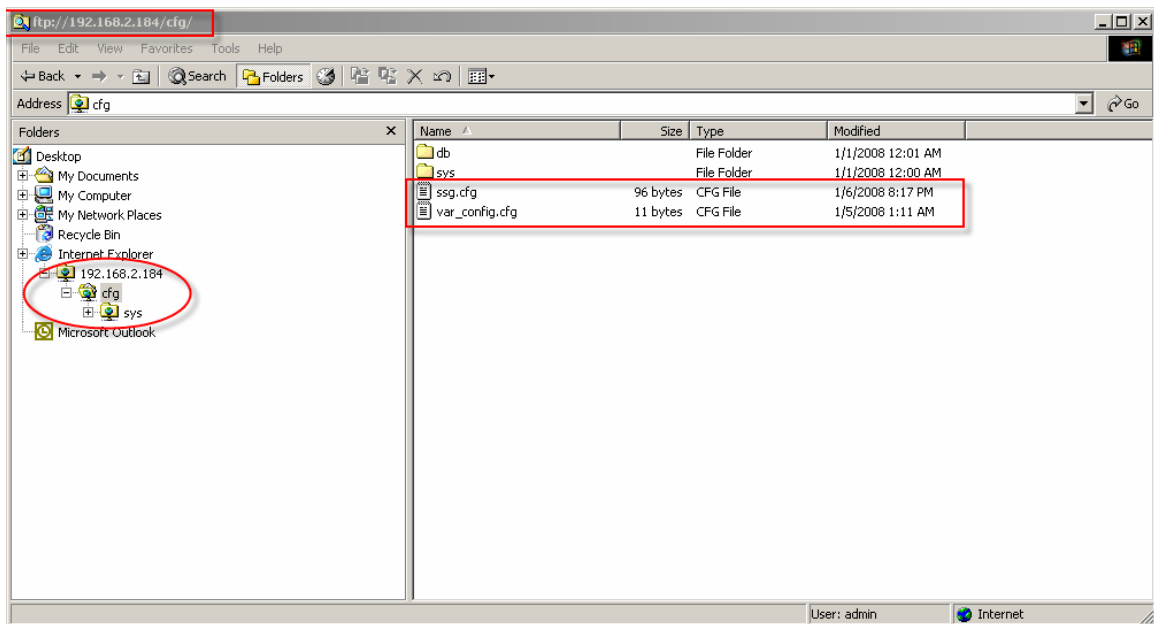
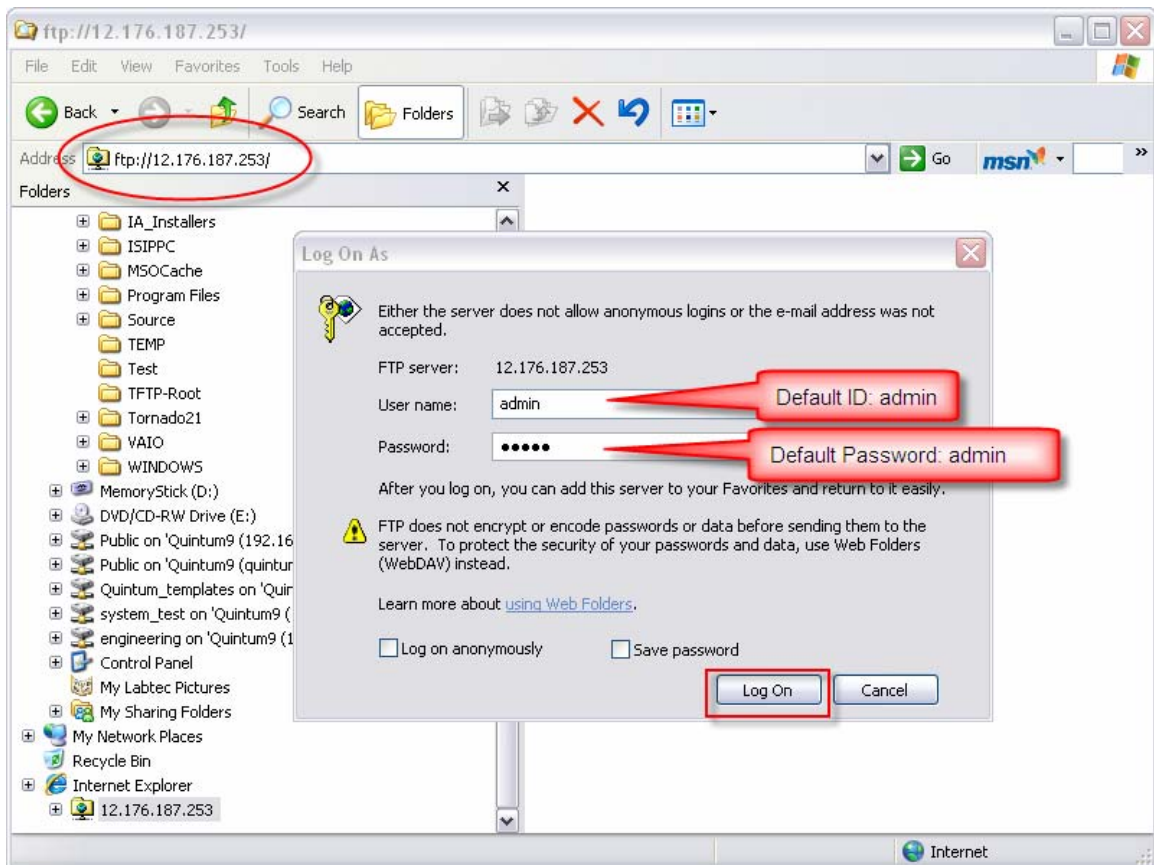
Create an ASCII text file with the file name “var_config.cfg”.
Copy the following line into the file.

```
SecureSSG0 0
```

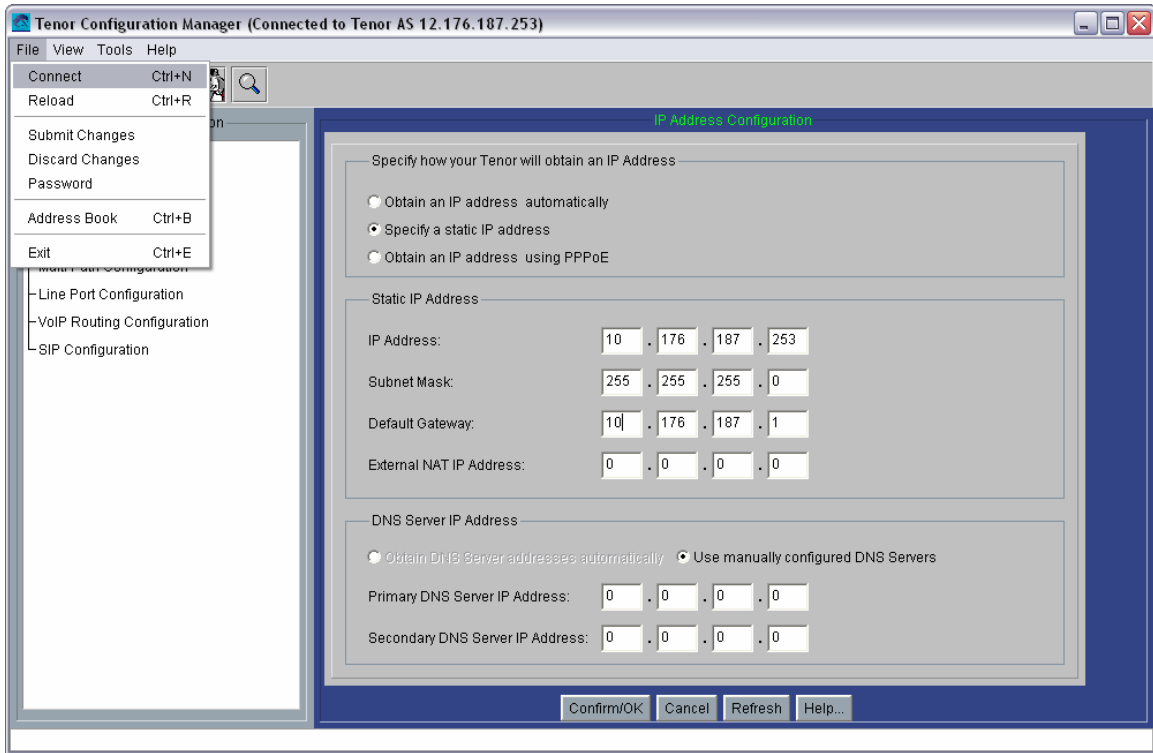
Create an ASCII text file with the file name “ssg.cfg”.
Copy the following four lines into the file replacing the “xxx.xxx.xxx.xxx” with the IP Address of the Primary AT&T IP Border Element and “yyy.yyy.yyy.yyy” with the IP Address of the Secondary AT&T IP Border Element.

```
* ^sip:\+?([0-1][0-9]*)@+ sip:$1@XXX.XXX.XXX.XXX
* ^sip:\+?9?([1-9]11)@+ sip:$1@XXX.XXX.XXX.XXX
* ^sip:\+?([0-1][0-9]*)@+ sip:$1@YYY.YYY.YYY.YYY
* ^sip:\+?9?([1-9]11)@+ sip:$1@YYY.YYY.YYY.YYY
```

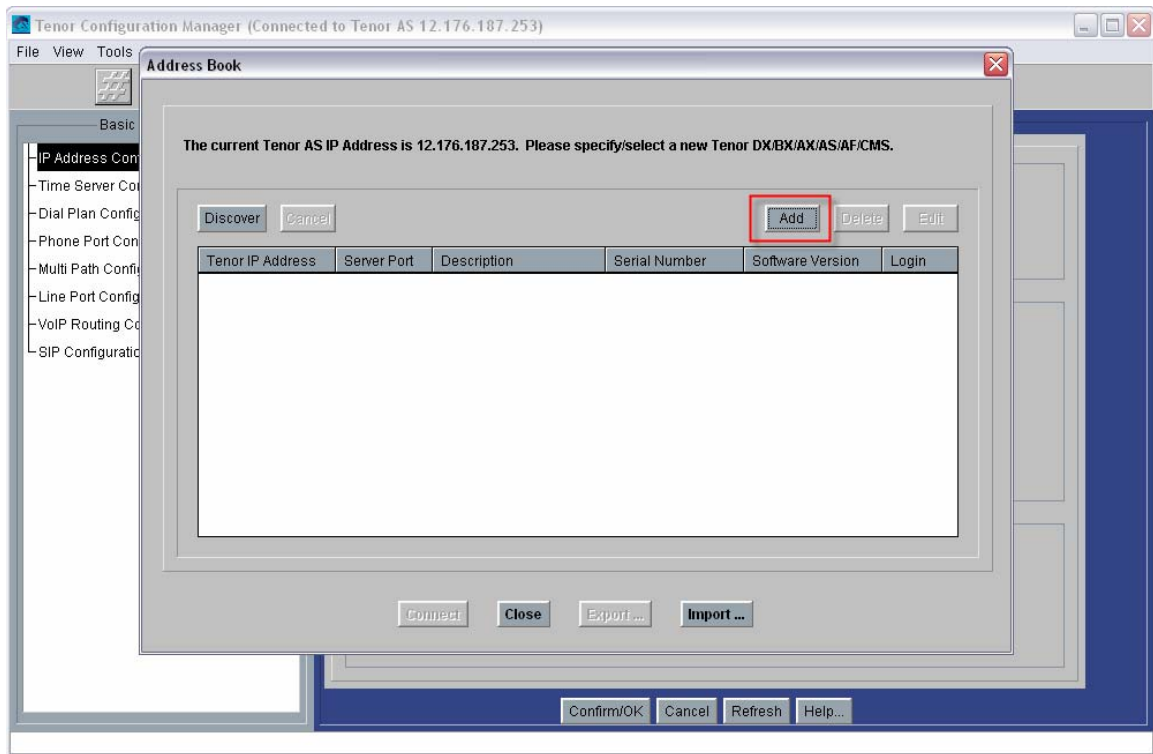
FTP the “var_config.cfg” file and the “ssg.cfg” file into the Tenor /cfg directory.
MS Windows Explorer can be used to perform the FTP with the Tenor.



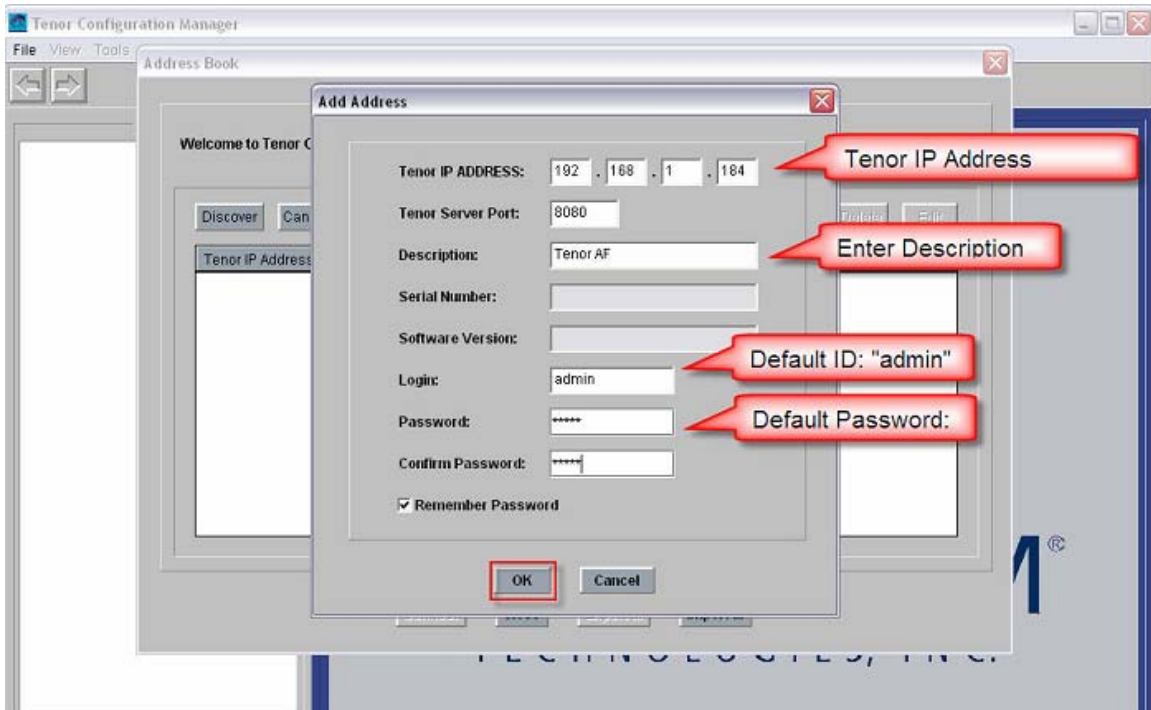
1. Run the Tenor Configuration Manager.
From the File Menu click **Connect**.



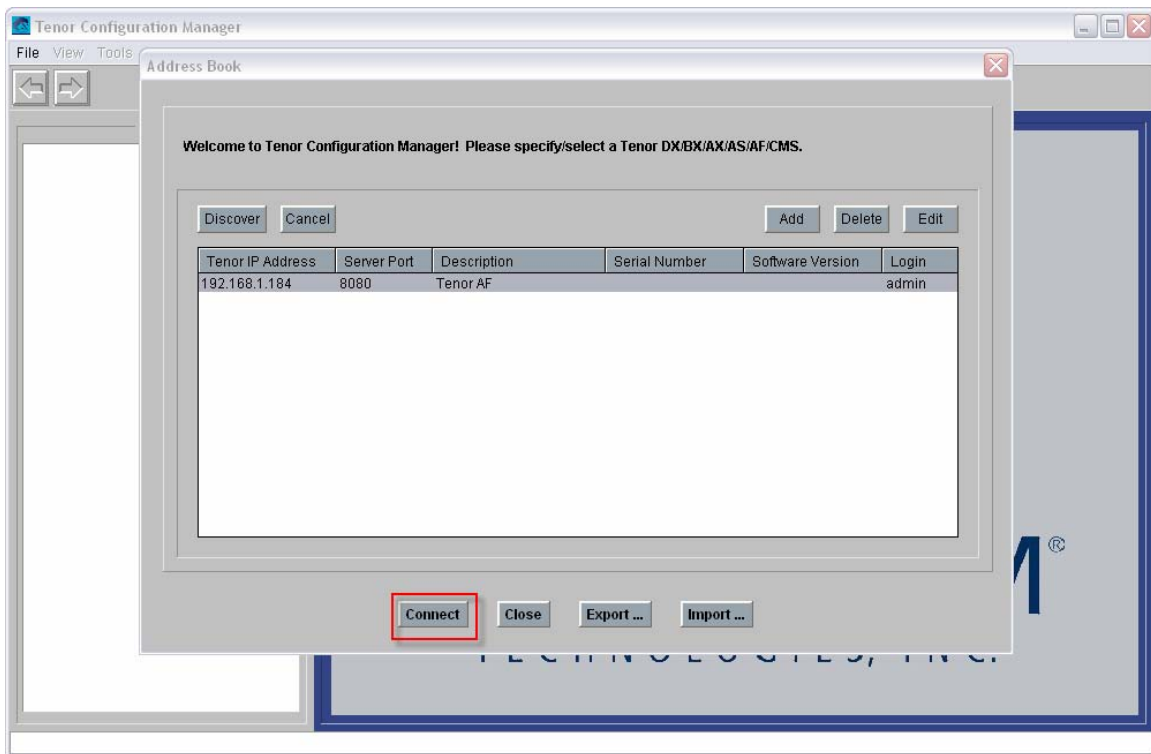
2. Click **Add**.



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



4. Connect to the Tenor AF from the Tenor Configuration Manager. Highlight the Tenor AF and click **Connect**.



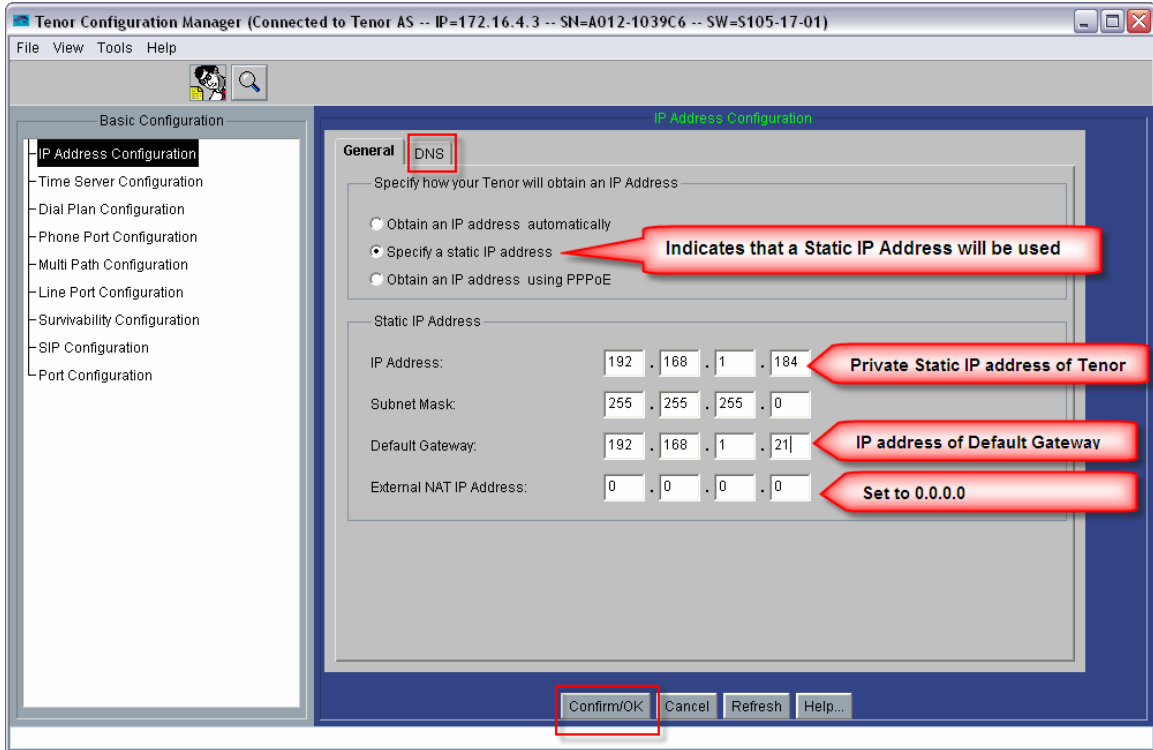
5. Confirm that the Radio Button for "Specify a static IP address" is selected. Populate the IP Address fields with the correct numbers.
IP Address: This is the Private Network IP Address assigned to the Tenor.

Subnet Mask: This is the Subnet Mask used on the customer private network.


Default Gateway: Default Gateway at Customer location.

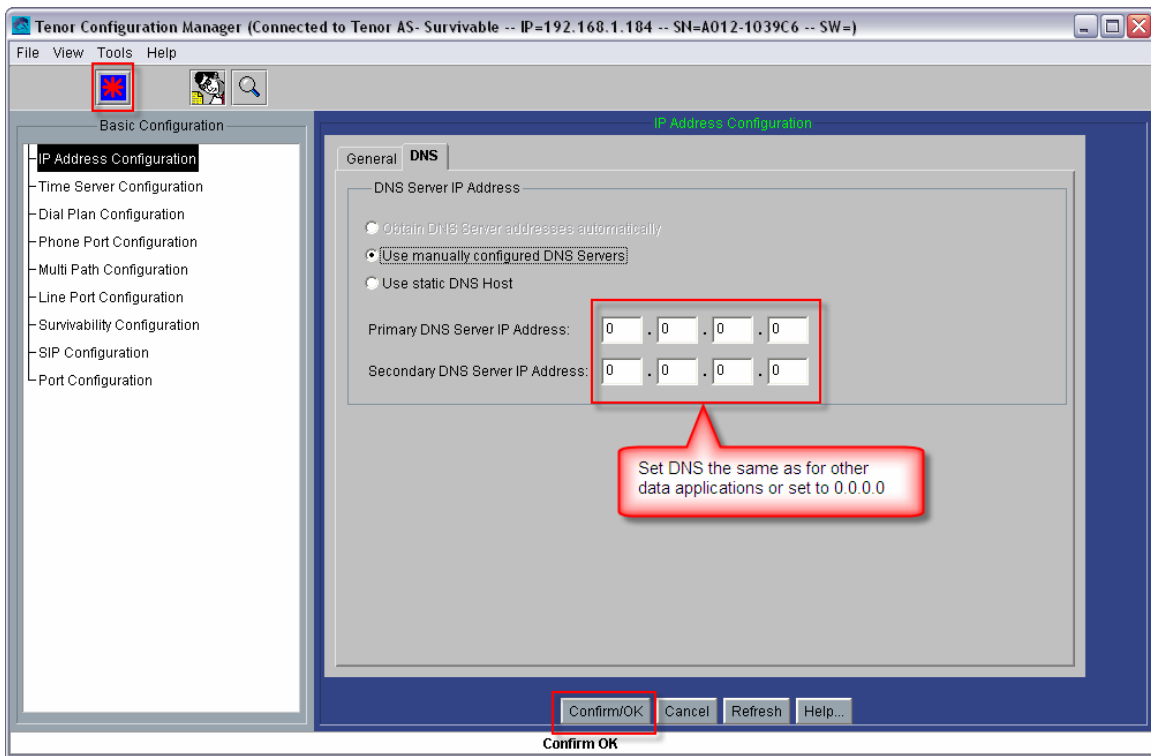
External NAT IP Address: Set to "0.0.0.0" to disable the Tenor NAT capability.

Click **Confirm/OK** then Click the DNS tab

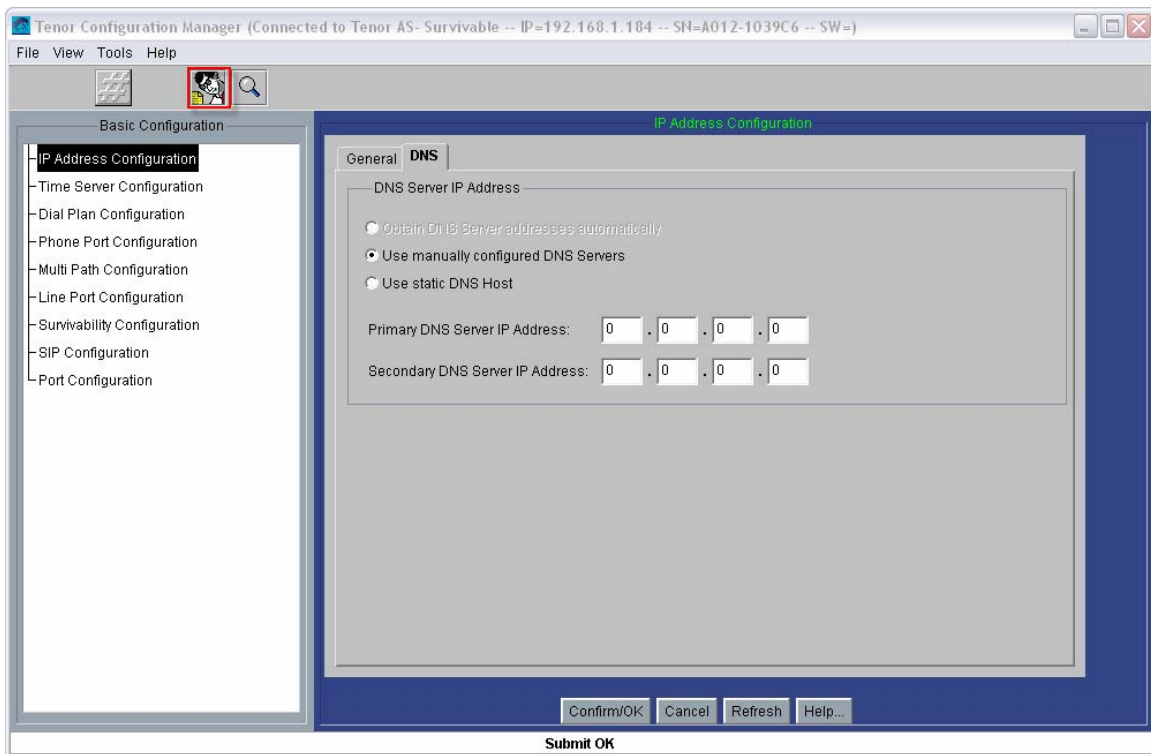


6. Enter values for the Primary and Secondary DNS Server IP Address. If not using DNS enter: 0.0.0.0

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



7. Click on the **Advanced Explorer** icon on the menu bar.

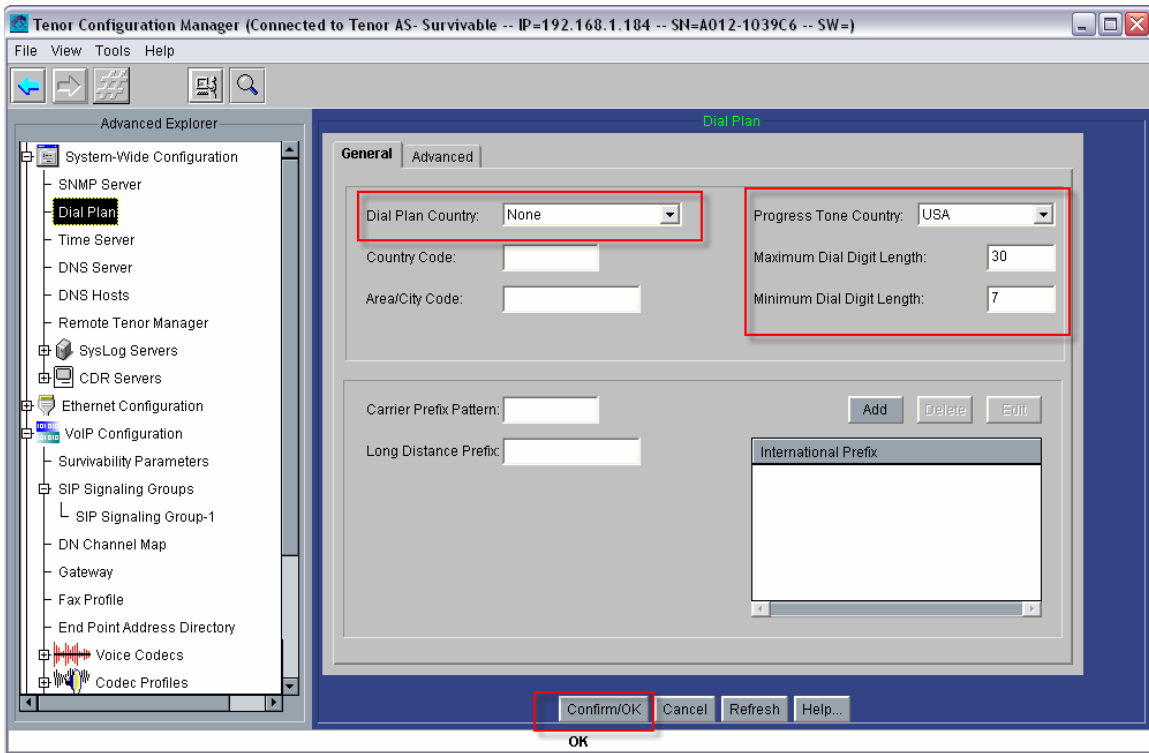


8. In the **Advanced Explorer** panel on the left, highlight the **Dial Plan** field. Select the desired **Dial Plan Country** in the drop down menu. The sample configuration uses **None**.


Select the desired **Progress Tone Country** setting from the drop down menu. The sample configuration uses **USA**.

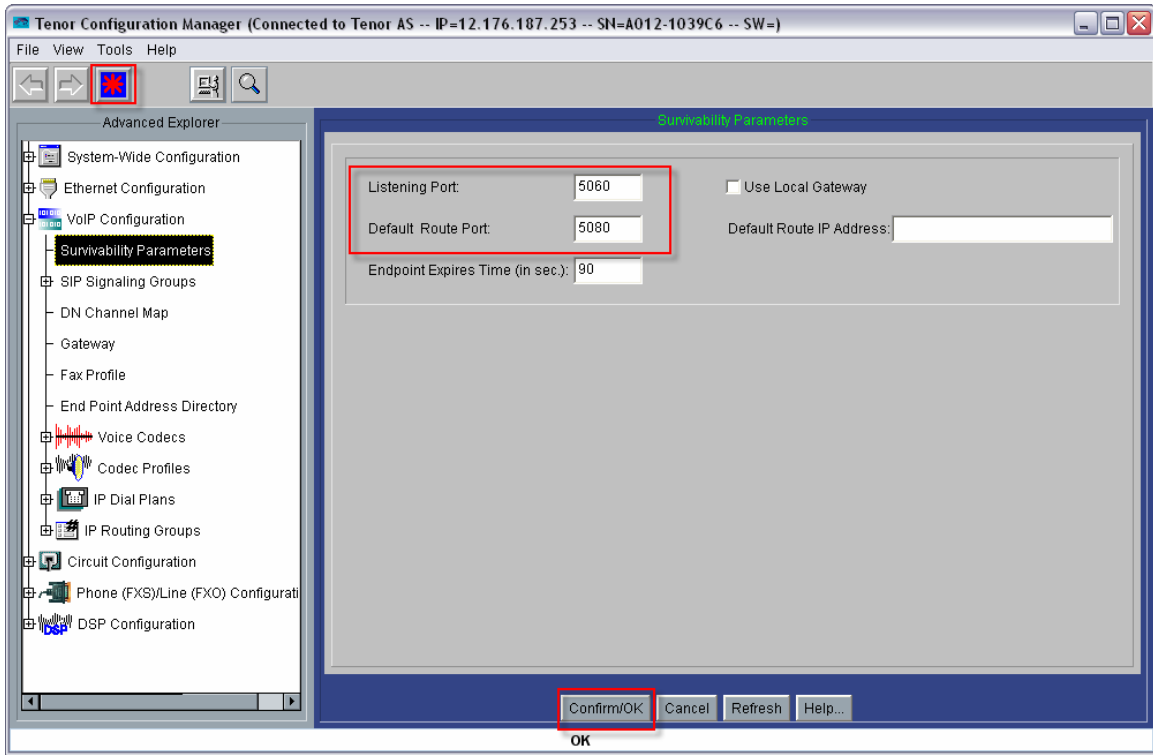
Enter values for the Minimum and Maximum dial digit string length.

To implement the change, click **Confirm/OK** then the  sunburst icon.




9. In the **Advanced Explorer** panel on the left, click the **+** sign next to **VoIP Configuration** to expand the field. Highlight **Survivability Parameters**. Set the Listening Port to "5060" and the Default Route Port to "5080".

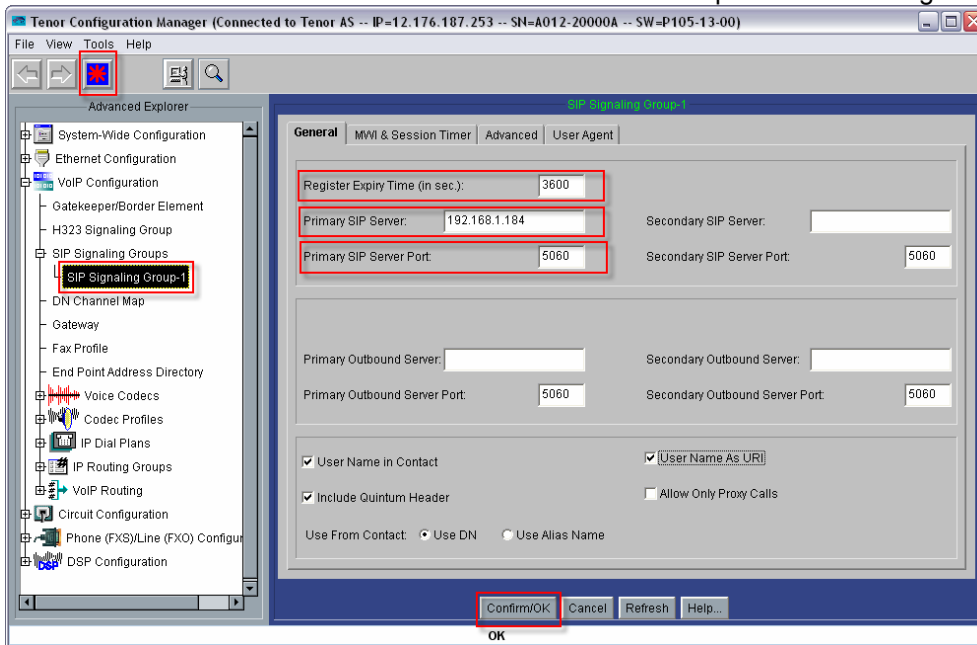
Click **Confirm/OK** then the  sunburst icon to implement the change.



6.2.1. Additional Steps for VoIP FAX Configuration

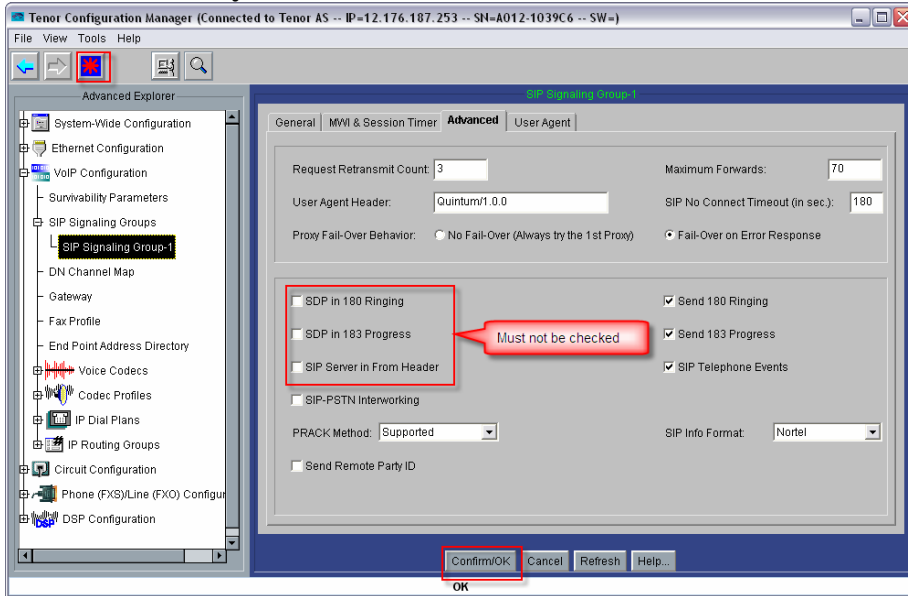
10. On the **Advanced Explorer** panel on the left, click the **+** sign next to **VoIP Configuration** → **SIP Signal Groups** to expand the field. Highlight **SIP Signaling Group-1**. Under the General tab, enter the **IP Address** and **Port** of the Tenor SSG. Set the Register Expiry Time to 3600.

Click **Confirm/OK** then the  sunburst icon to implement the change.

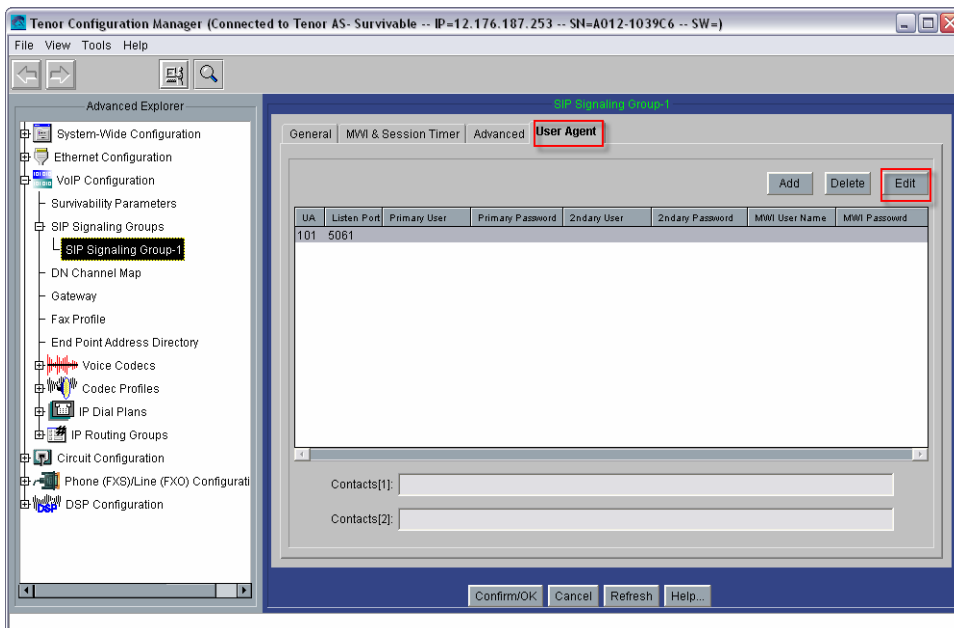


11. Click on the **Advanced** tab. Un-check the boxes for:

- “SDP in 180 Ringing”
- “SDP in 183 Progress”
- “Proxy Address in From Header”

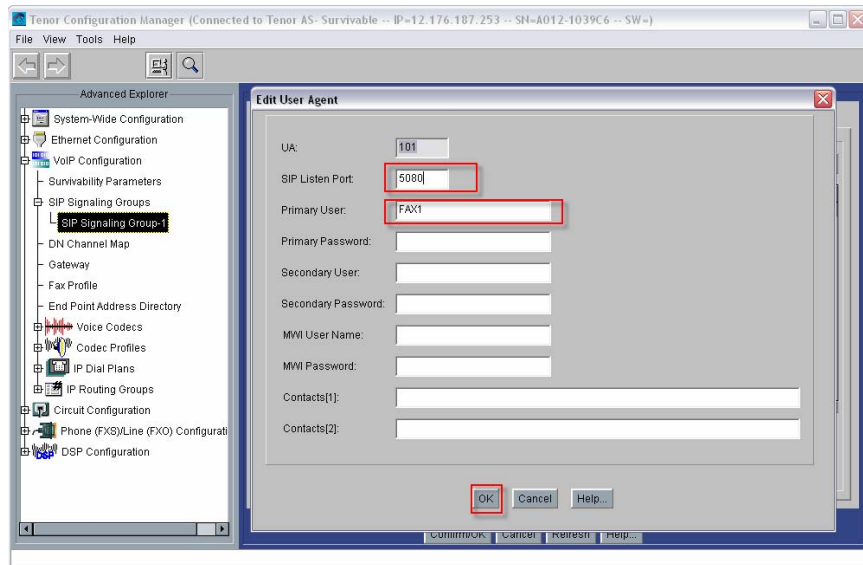



12. Click on the **User Agent** tab. Highlight the **User Agent 101** entry. Click the **Edit** button to display the Edit User Agent pop-up window. We will create One User Agent for each FAX Machine that will be attached to the Tenor. In this configuration example we will create one User Agent.

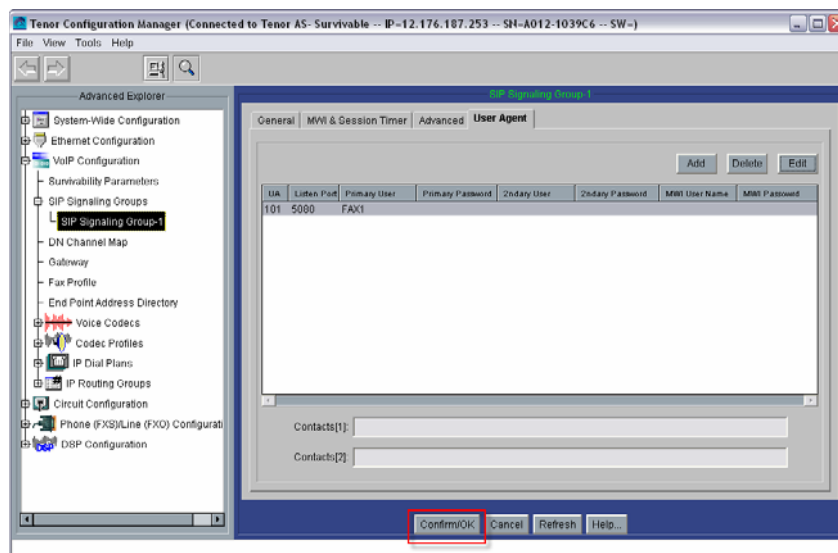


13. In the **Edit User Agent** pop-up window, enter the following information:
- PrimaryUser** - The username for Registration and Authentication purposes. If Registration were enabled, the “username” will appear in the URI populated in the To and From headers of the REGISTER message.
 - SIP Listen Port:** **5080** Set the SIP Listen Port to 5080. *Any value other than 5060 will be acceptable.*
 - Primary User:** **FAXI** < --- Any alpha-numeric string may be entered because SIP Registration and Authentication are not applicable to the AT&T IP Flexible Reach service.

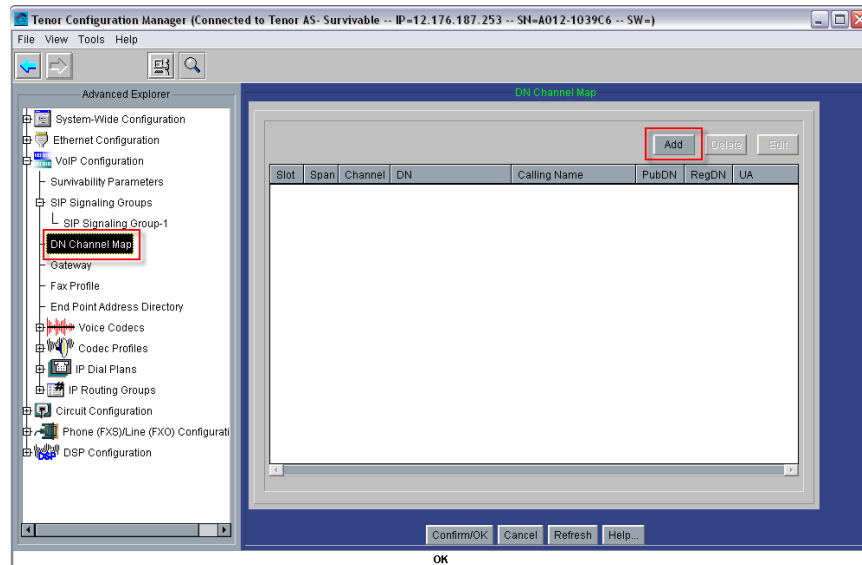
Click **OK** to continue.



14. At the **SIP Signaling Group-1** panel click **Confirm/OK** to complete and the  to implement the change in the Tenor SSG.



15. 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.



16. Each of the FAX machines physically connected to a Channel/Port 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 FAX machine / 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 FAX machine (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 TN (732-368-0416). When the FAX machine connected to Tenor Channel 1 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 FAX machine connected to the Tenor Channel/Port 1 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: **7323680416** < --- Phone Number (number provided by AT&T)

Calling Name: Quintum FAX < --- Display Name
User Agent: 101 < --- User Agent defined in Step 13.
Public DN checked < --- default
Register DN checked < --- default

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

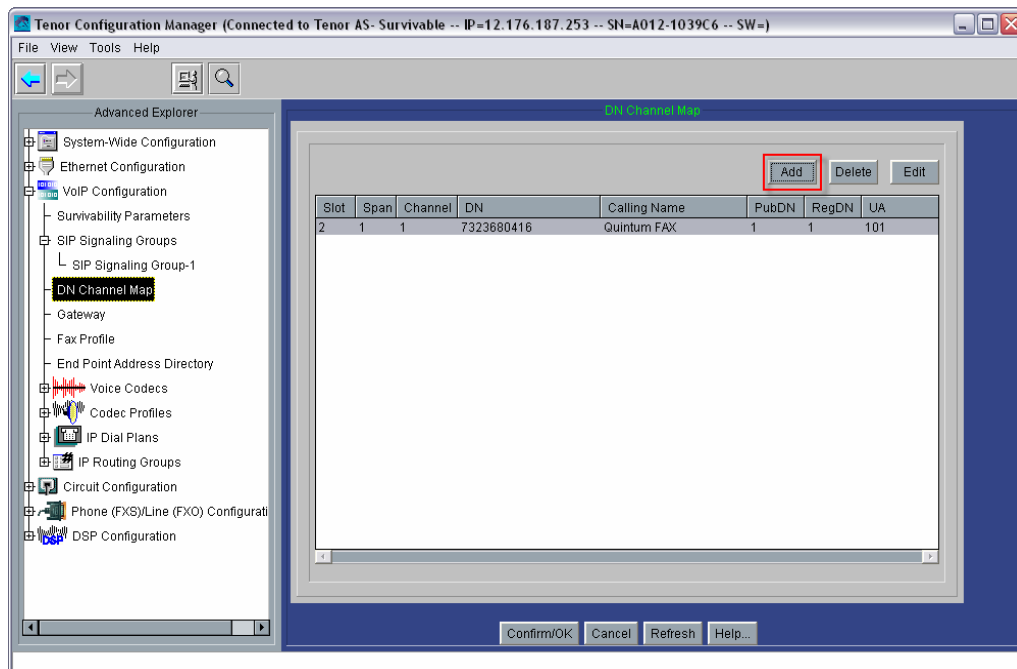
Calling Name: Will appear as the Display Name in the From header in outgoing INVITE messages.

Public DN: Indicates whether or not this is a Public DN

Register DN: Only relevant to H.323

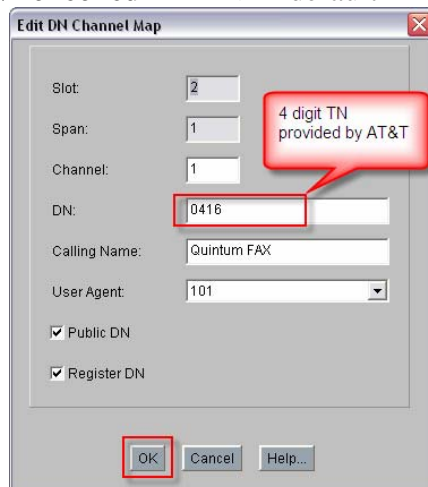
17. It is necessary to define another DN Channel Map entry to provide the 4-digit routing (0416) of calls to the FAX machine connected to Tenor Channel/Port 1.


Click the **Add** button.

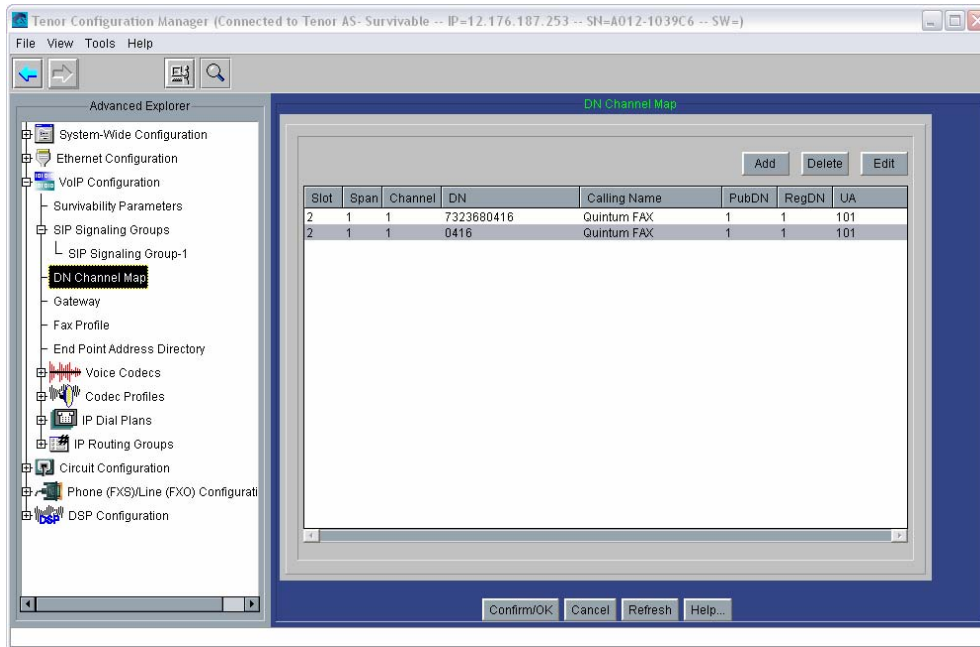


18. At the **Add DN Channel Map** pop-up window, enter the following information.

Channel: 1 < --- Physical port used on Tenor
DN: 0416 < --- Phone Number (number provided by AT&T)
Calling Name: Tim Thornton < --- Display Name
User Agent: 101 < --- User Agent defined in Step Error! Reference source not found.
Public DN checked < --- default
Register DN checked < --- default




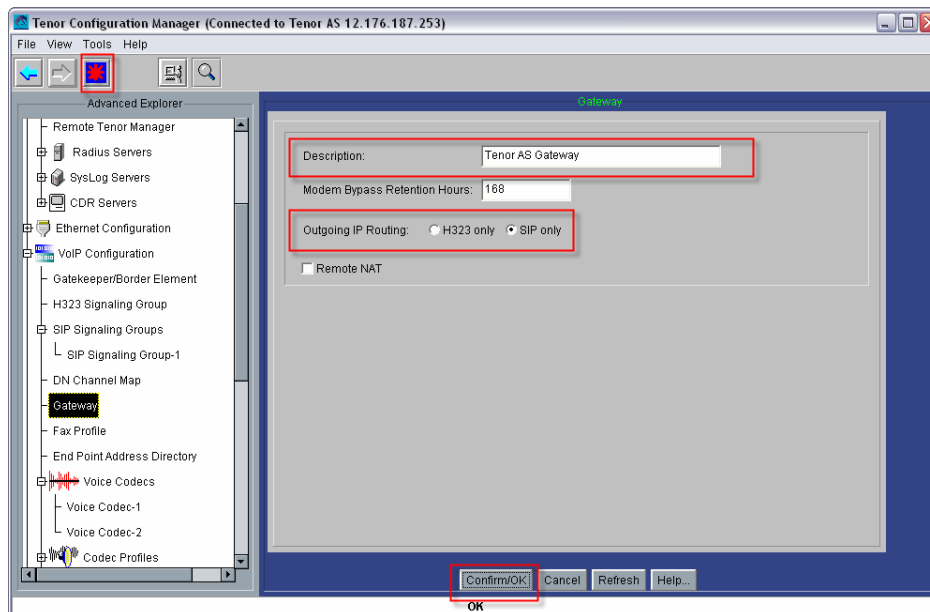
Click **OK** to continue. At the DN Channel Map panel click **Confirm/OK** and the  to implement the change.



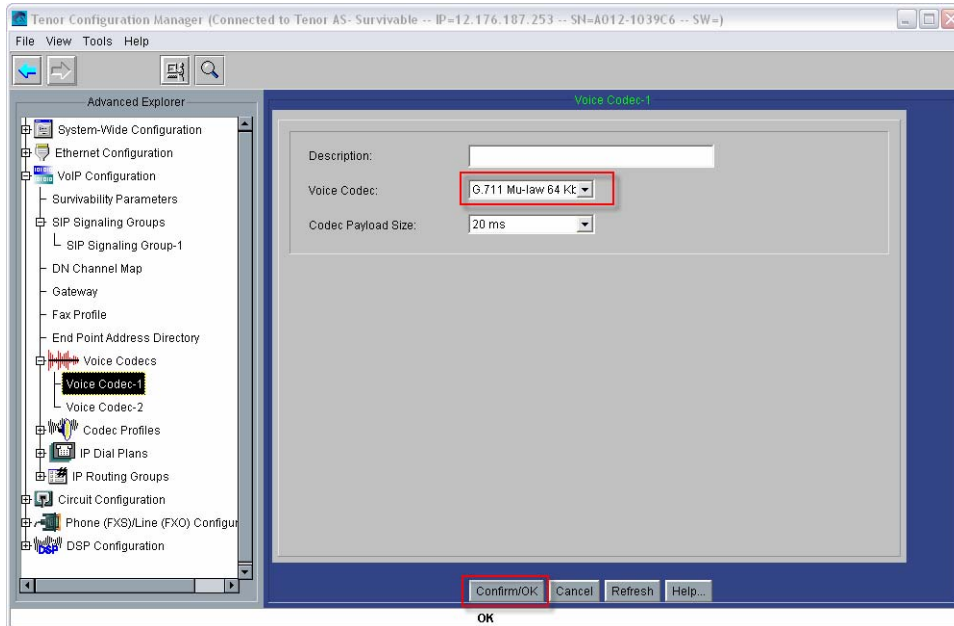
Note that 0416 and 7323680416 are the two DN's associated with channel 1 and User Agent 101 (UA 101).


- 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 implement the change.

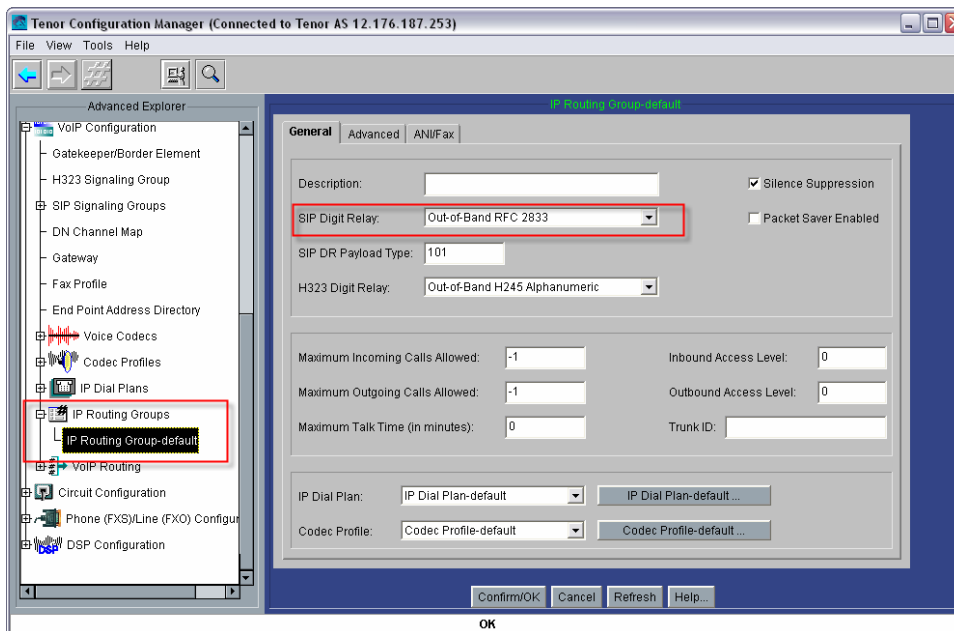


20. From the **Advanced Explorer** panel on the left, click on the + sign to expand the **Voice Codecs** field. Highlight the **Voice Codec-1** field. For FAX select the **G.711** codec.




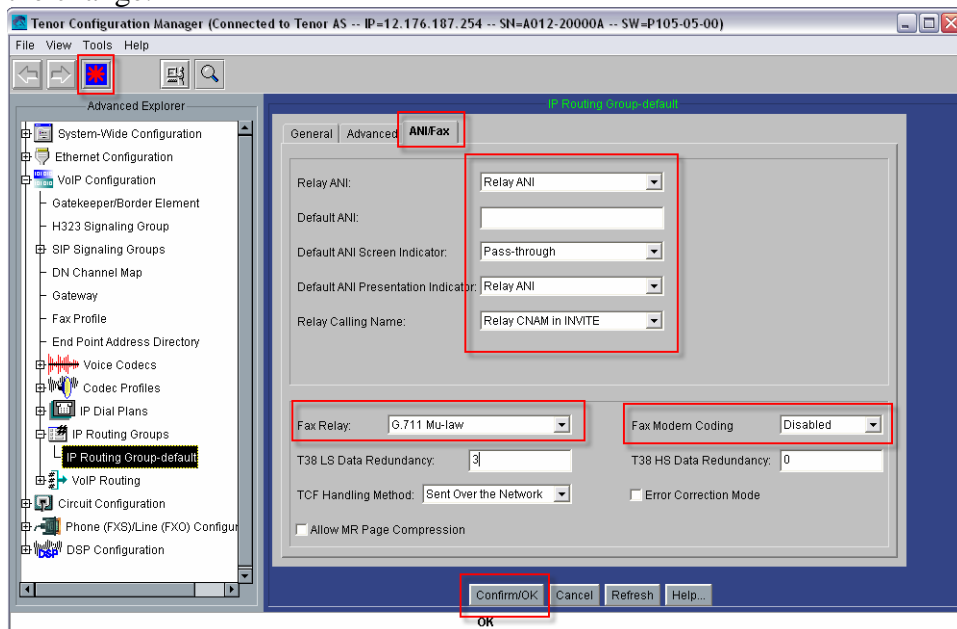
Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change.

21. 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.




22. Click on the **ANI/FAX** 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. Select **G.711 Mu-law** 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.

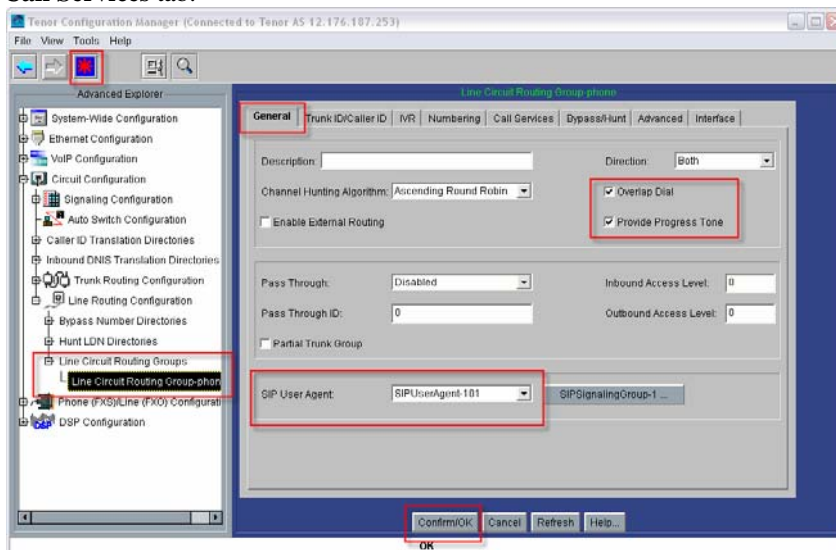


23. From the **Advanced Explorer** panel on the left, expand **Circuit Configuration** → **Line Routing Configuration** → **Line Circuit Routing Groups**, and highlight the **Line Circuit Routing Group-phone** field.


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**.

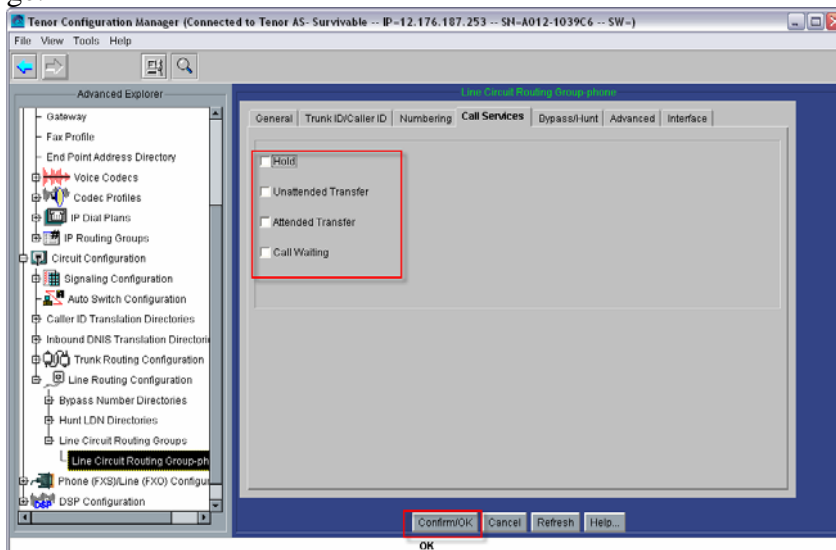
Click **Confirm/OK** then the  sunburst icon on the menu bar to implements the change.

Click the **Call Services** tab.




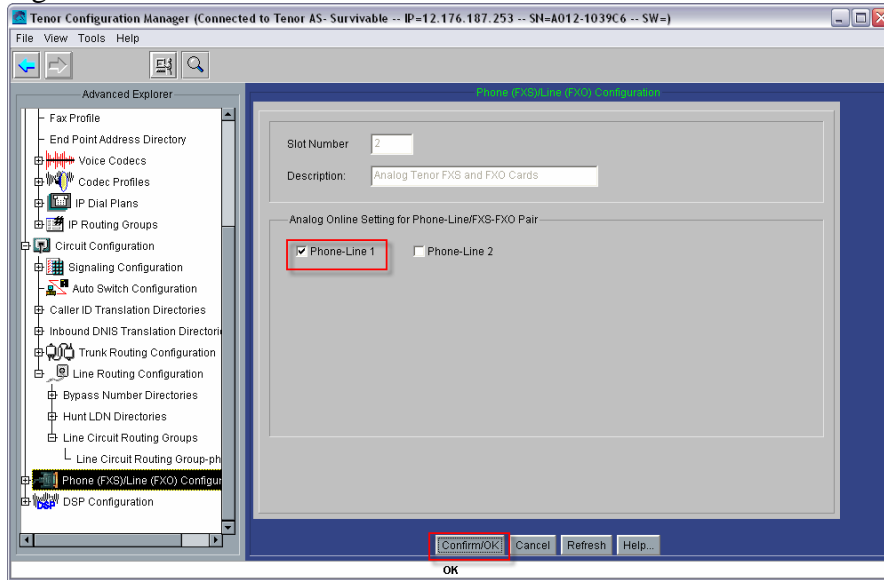
24. From the **Call Services** tab under the **Line Circuit Routing Group-phone** panel on the right. Disable all Call Service because this is a FAX line.

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

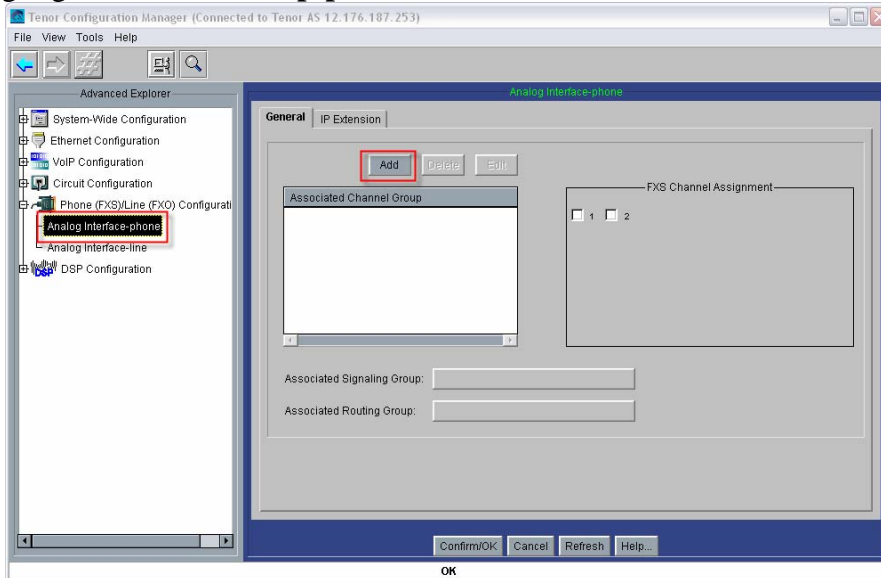


25. Under the **Advanced Explorer** panel on the left, highlight the **Phone (FXS)/Line (FXO) Configuration**. Check the box to enable **Phone-Line 1**.

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



26. 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**.



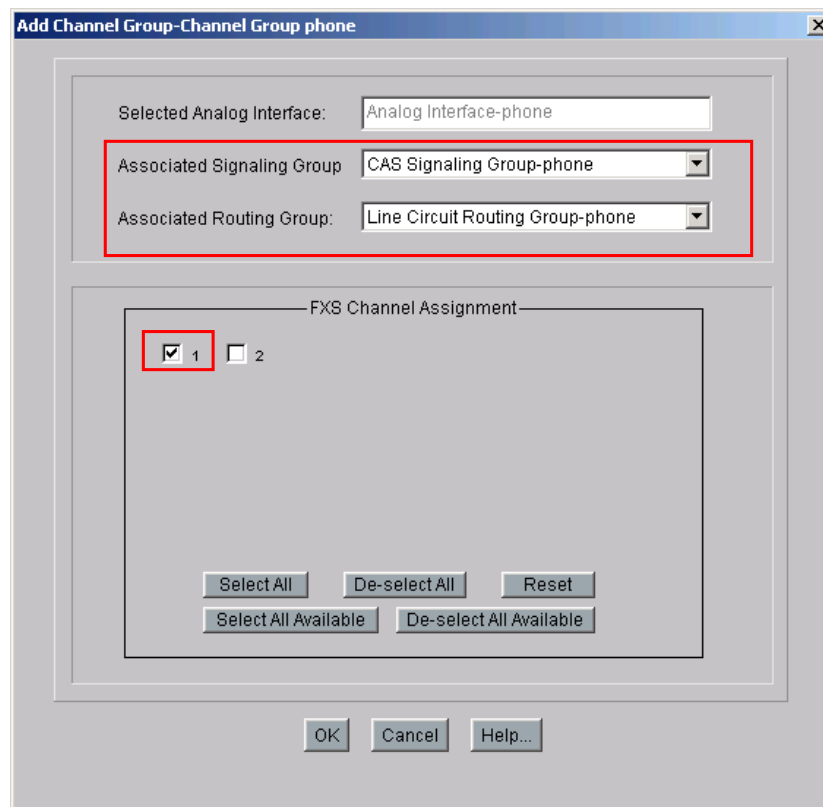
27. Enter a description “**phone**” for the **Channel Group** and click **OK** to continue.



28. 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.



7. REMOTE CONFIGURATION

The CPE Router that provides connectivity to the AT&T trunk must be configured properly to allow remote monitoring of the Tenor gateway as well as the one-X Quick Edition telephones. The gateway and one of the one-X QE telephones need to be configured with a static IP address so that static port mapping can be configured on the router.

The following steps will configure the static IP address on the one-X QE telephones. The steps need to be executed on the phone itself.

Step	Command	Notes
Step 1	Go to Network Options Click the Options button on the telephone. From the screen menu, go to Options/System Options/Network Options	Enter the administration password to log into System Options.
Step 2	Change IP Set IP Settings/Chg/Edit IP Address = "172.16.4.20"	Make sure the IP assigned is from the static IP address block and not from the DHCP IP address block.
Step 3	Change Netmask Set IP Settings/Chg/Edit Netmask= "255.255.255.0"	Make sure the netmask is the one configured in your router.
Step 4	Change Gateway Set IP Settings/Chg/Edit Gateway = "172.16.4.1"	Make sure the gateway is the IP address of the router.
Step 5	Save the settings Press IP Settings/Save	

In Section 6.2 Step 1, find the instructions to configure the static IP address of the Quintum Tenor gateway. Now create a mapping between the router's external 5060 port and the internal Tenor gateway so that incoming SIP signals can be routed from the AT&T trunk to the Tenor Gateway.

To access the web portal of one-X QE telephone, it is necessary to configure two more port mappings on the router. One port is mapped to the http port (80) of the QE telephone while the port is mapped to the https port (443). The following procedure will configure the port mapping on the router whether it is a customer managed router or an AT&T managed router.

Step	Command	Notes
Step 1	Login to the router's administrator portal	Depending on the router it could be a CLI interface or web based interface. For remote management of the router, you may have to configure the IP addresses from which the router can be managed.
Step 2	Set the port mapping for http port on the router Map port 80 on the public side to port 80 on the QE phone with the static IP.	Make sure port 80 is not assigned to an http server other than the QE telephones.
Step 3	Set the port mapping for https port on the router Map port 443 on the public side to port 443 on the QE phone with static IP address.	Make sure port 443 is not used by other internal servers except QE.
Step 4	Access the QE web portal from a remote site From a web-browser enter http://<wan ip address of the router>/ and that should redirect the request to the QE web portal	Only System Options on QE phone are configurable from a remote site.
Step 5	Login to System Options Select System Options and enter the system password	

The remote configuration is only allowed for system options on the QE telephone --- individual phones cannot be configured through this telephone. All of the configurations related to AT&T Flex Reach solution and Quintum Tenor gateway are available through System Options in QE web portal.

The Tenor Survivable SIP Gateway (SSG) can also be configured and monitored remotely, in two ways.

One way to achieve this functionality without modifying the router is by using the Quintum Remote Management Session Server (RMSS). The Remote Management Session Server resides either on the public IP network or within the Service Provider DMZ. The RMSS sets up a secure management channel between the Remote Management Session Server platform and the customer Tenor that may be located behind a NAT firewall. After the Tenor SSG is configured with the RMSS's IP address, the Tenor will open a UDP session through the CPE router's firewall (port 2300) to the RMSS. Then the remote user can connect to the Tenor SSG through the RMSS server to perform system configuration, performance monitoring, diagnostics, troubleshooting and remote upgrade functionality.

Another method for providing remote configuration and monitoring of the Tenor SSG is to configure port mapping on the CPE router. After an external port (e.g. 8080) on the router is mapped to the Tenor gateway's management port (usually 8080), a remote user can access the Tenor gateway through that router's external mapped address (e.g. 207.15.13.45:8080).

8. TROUBLESHOOTING

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

9. ADDITIONAL REFERENCES

1. Avaya. 2006. *Avaya one-X Quick Edition Release 3.0.0 System Administrator Guide*, P/N16-601412, Release 3.0.0, November 2006, Issue 1. Available from Avaya at support.avaya.com/quickedition.
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 S Application Guide*
http://www.quintum.com/support/products/2G/tenor_s/sysdoc/SSGAppNote.pdf
4. *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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