Application Note - IP Trunking

End-to-End Configuration for IP Trunking

This document gives you a detailed description of how to configure IP Trunking in a Tenor VoIP system.

The following topics are included in this Application Guide:

Quintum Technologies
71 James Way
Eatontown, NJ 07724
1.732.460.9000 / 1.732.544.9119 (fax)

Technical Support
Toll Free (U.S. only) 1.877.435.7553
International 1.732.460.9339
email:microsoft-service@quintum.com

IP Trunking - Application Description

An IP Trunk carries voice traffic across an IP network. This document provides the necessary information to configure end-to-end IP trunking between a *Tenor BX* (destination Tenor) and a *Tenor AF* (Remote Tenor). This sample application details how to configure a call originating from a remote extension (phone/fax connected to the *Tenor AF*) to go out over an IP trunk, bypass the PBX (*Tenor BX* in pass through mode), and out to the PSTN.

Destination Side (Tenor BX)

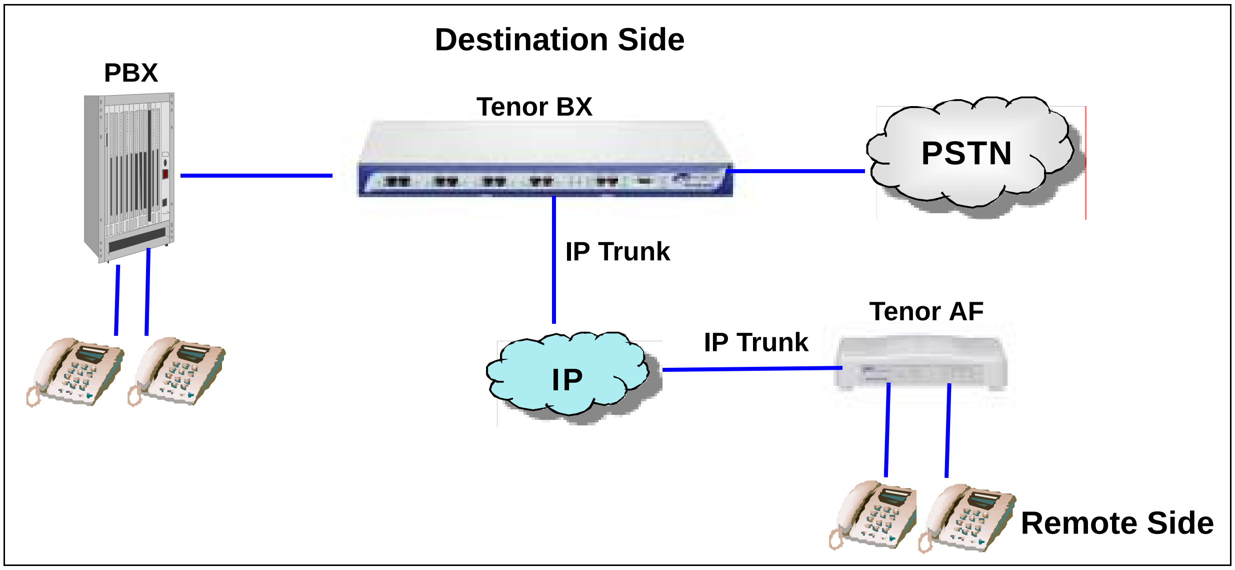
- Progress Tone Country: Czech/Slovakia
- BRI ETSI Signaling Type
- Pass Through Mode (Bypass PBX to go out to PSTN)
- Private Dial Plan

Remote Side (Tenor AF)

- Progress Tone Country: Czech/Slovakia
- Private Dial Plan
- CAS Signaling
- Analog trunk

See *End-to-End IP Trunking - Network Diagram* as an application example for IP trunking; the configuration will differ, depending upon the Tenor units and equipment in your network.

Figure 1-1 End-to-End IP Trunking - Network Diagram



Tenor AF. 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* supports FXO and FXS ports, and supports simultaneous VoIP calls (the number of voice calls is determined by the Tenor AF's configuration) Through the FXS port, you can connect a telephone, key system or PBX; through the FXO port, you can connect to the PSTN (through direct connection to the Central Office). A *Tenor AFM200* is used for the purposes of this example, which supports 2 FXO and 2 FXS ports.

Tenor BX. The *Tenor BX* is a high density VoIP (Voice over Internet Protocol) H.323/SIP switch that converts voice, fax, and modem data on digital circuit switched trunks, and transmits it over the IP network.

The *Tenor BX* supports 2, 4 or 8 BRI (Basic Rate Interfaces (2B+D channel) ports with S/T interface, supporting up to 16 voice channels. It enables connectivity between the customer equipment (i.e., PBX), PSTN and VoIP Network. A *Tenor BX204* is used for the purposes of this example, which supports 2 BRI ports (4 BRI channels).

Hardware Installation

The network application includes two Tenor units: *Tenor BX* and *Tenor AF.* This Application Note assumes both units are installed and running.

For information about installing the *Tenor BX*, see the *Tenor BX Product Guide*.

For information about installing the *Tenor AF,* see the *Tenor AF Product Guide.*

Configuration - Getting Started

In order for more than one Tenor unit to “communicate” over an IP trunk line, each Tenor unit needs to be configured. For the purposes of this document, all configuration for the Tenor is completed through the *Configuration Manager* software, available at *www.quintum.com*.

These instructions assume the *Tenor Configuration Manager* is downloaded, installed and running. For detailed information about how to install the software, see *Tenor Configuration Manager User Guide*. Basic configuration parameters for each of the required fields is included in this document; detailed information is available in the *Command Reference* guide

If this is the first time you are configuring the Tenor unit, when you log into the *Tenor Configuration Manager,* the Auto Discovery process displays a list of Tenors needing configuration. From this list, you select the desired Tenor. For detailed information about the AutoDiscovery process and launching the Configuration Wizard, see *http://www.quintum.com/support/mgmt/TenorConfigManagerUsersGuide.pdf.*

Setup Tenor 1 - Tenor BX204 (destination side)

Ensure the *Tenor Configuration Manager* is connected to the *Tenor BX*. For instructions on how to install/access the software or to connect to a Tenor, see *Tenor Configuration Manager User Guide*.

NOTE: These instructions assume the *Tenor BX204* is connected to the *Tenor Configuration Manager.*

Step 1: System Wide Configuration

System Clock

1. From the main menu, select *System-Wide Configuration> System Clock*.

2. Select a line source from the **Primary Clock Line ID** drop down box. See *Figure 1-2*. This is the line interface from which the Tenor derives primary network clocking. Ensure there is a single data clock source at each Tenor location. Otherwise, you may have poor voice quality and alarms will be generated.

3. Click **Confirm/Ok**.

Figure 1-2 Configure System Clock

Dial Plan

1. From the main menu, select *System-Wide Configuration> Dial Plan.*

2. Click on the *General* tab. See *Figure 1-3*.

3. From the **Dial Plan Country** drop down box, select **Generic CCITT.**

4. In the **Minimum Dial Digit Length** box, enter **3** (private dial plan).

5. Leave the **Dial Plan** information (**Country Code** and **Area Code**) blank. It is unnecessary to normalize numbers into E.164 for IP trunk calls.

6. From the **Progress Tone Country** drop down box, select **Czech/Slovakia.**

7. Click **Confirm/Ok**.

Figure 1-3 Configure Dial Plan

Step 2: VoIP Configuration

SIP Signaling Group

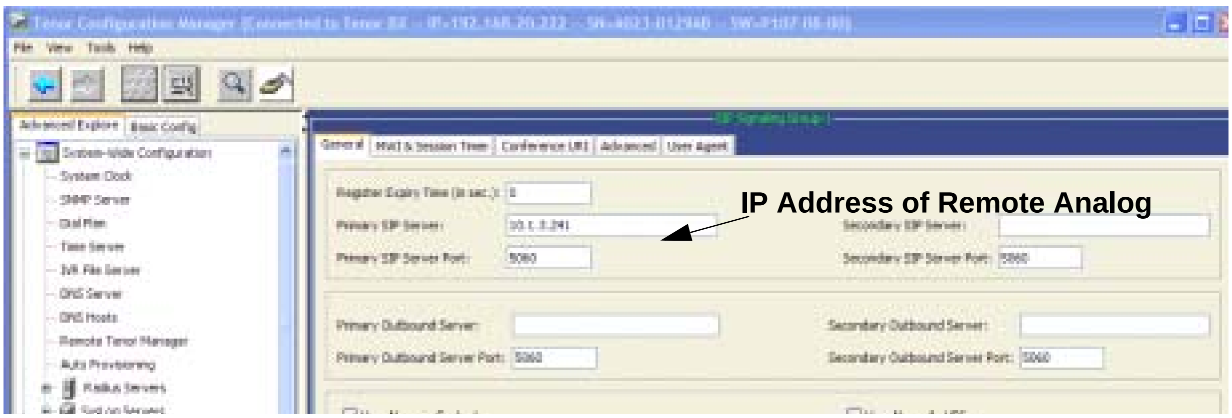
1. From the main menu, select *VoIP Configuration> SIP Signaling Group> SIP Signaling Group-1*.

2. Click on the *General* tab. See *Figure 1-4*.

3. In the **Primary SIP Server** field, enter an **IP address**. This IP address is for the remote *Tenor AFM200* (where calls destined to extensions 600 and 601 will be routed).

4. Click **Confirm/Ok**.

Figure 1-4 Configure SIP Signaling Group



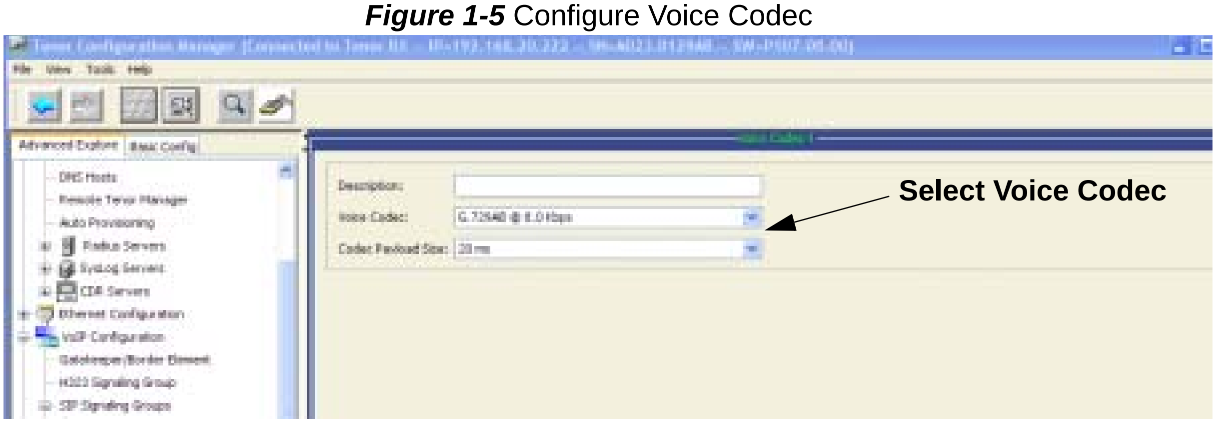
Voice Codec

1. From the main menu, select *VoIP Configuration> Voice Codecs> Voice Codec-1*.

2. From the **Voice Codec** drop down box, select the codec you will use as the voice compression algorithm (for IP trunk). See *Figure 1-5*.

3. Click **Confirm/Ok**.

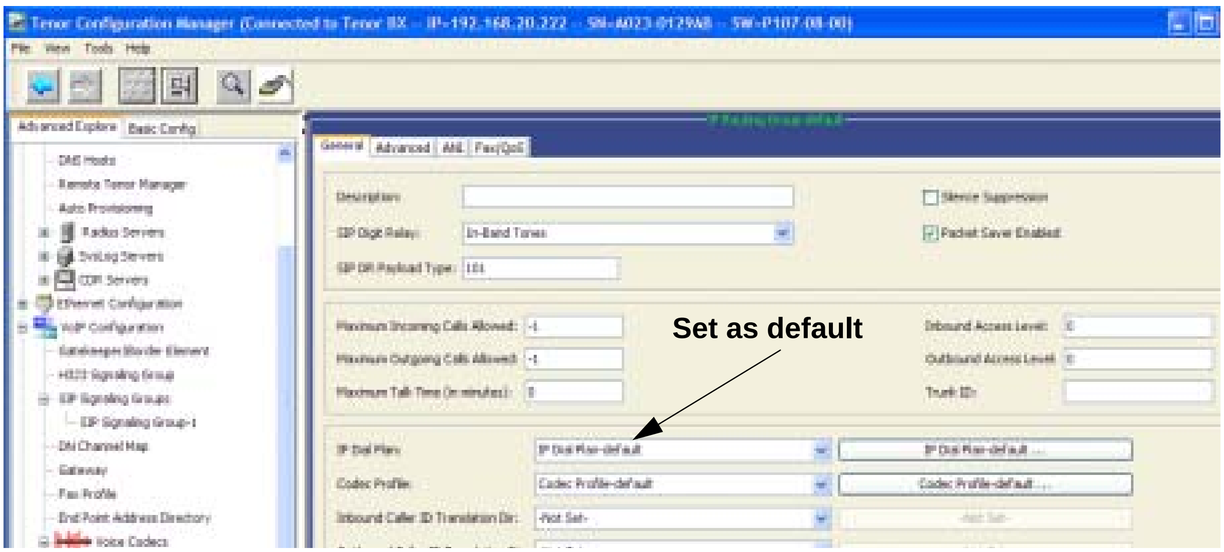
Figure 1-5 Configure Voice Codec



IP Routing Group

1. From the main menu, select *VoIP Configuration> IP Routing Group> IP Routing Group-default.*
2. Click on the *General* tab. See *Figure 1-6*.
3. From the **IP Dial Plan** drop down box, select **IP Dial Plan-default.**
4. Click **Confirm/Ok**.

Figure 1-6 IP Routing Group



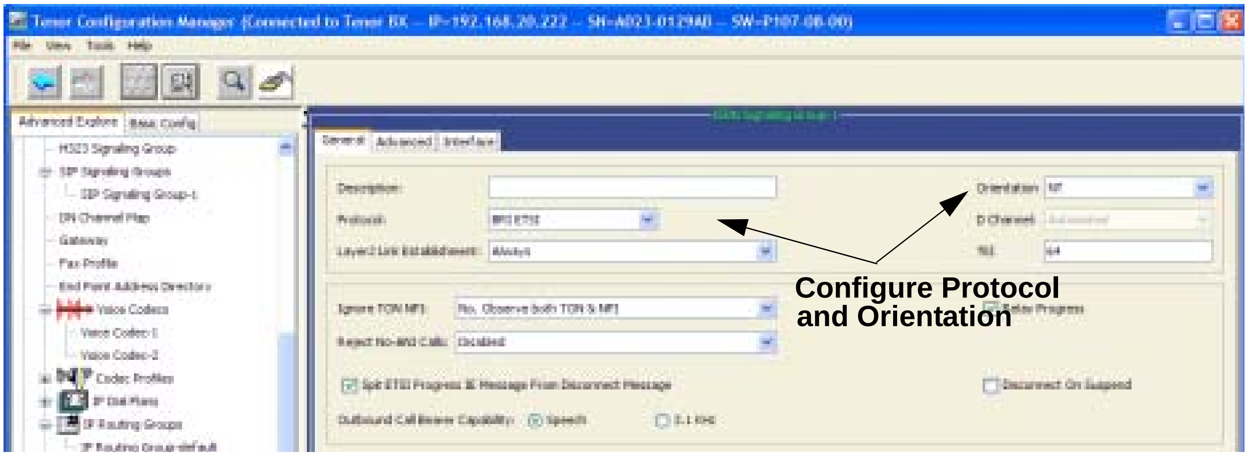
Step 3: Circuit Configuration

ISDN Signaling Group

You will need to create two ISDN Signaling Groups: **PBX** side (ISDN Signaling Group-1) and **PSTN** side (ISDN Signaling Group-2). One ISDN signaling group is created by default; you will need to create the second group. Use the default group (ISDN Signaling Group-1) as the PBX-side ISDN Signaling Group.

1. From the main menu, select *Circuit Configuration> ISDN Signaling Group> ISDN Signaling Group-1).* See *Figure 1-7*.
2. From the **Protocol** drop down box, select **BRI ETSI**.
3. From the **Orientation** drop down box, select **NT** to support the BRI port connected to the PBX side.
4. Click **Confirm/Ok**.

Figure 1-7 Configure ISDN Signaling Group - PBX Side



5. Add PSTN-side ISDN Signaling Group:

   From the main menu, right-click on *ISDN Signaling Groups (Circuit Configuration > Signaling Configuration> ISDN Signaling Groups)* and select **New**. Enter a name for the Signaling Group (i.e., ISDN Signaling Group-2) and click on **OK**. The Signaling group will be added.

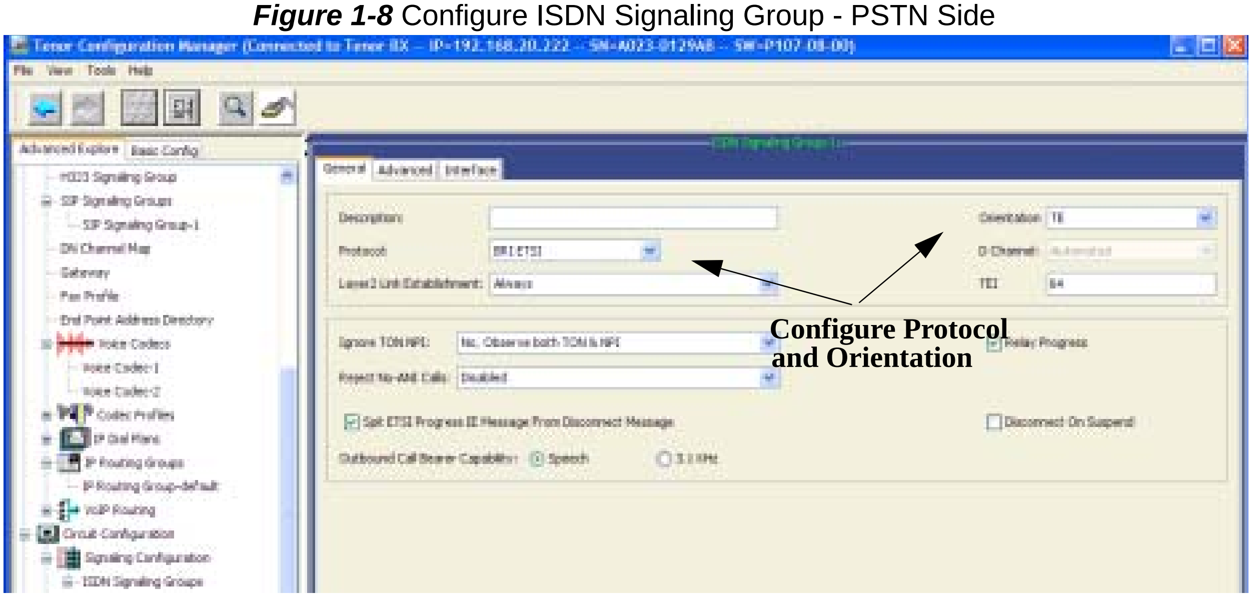
6. From the main menu, click on the newly added ISDN Signaling group (select *Circuit Configuration> ISDN Signaling Group> ISDN Signaling Group-2).* See *Figure 1-8*.

7. From the **Protocol** drop down box, select **BRI ETSI**.

8. From the **Orientation** drop down box, select **NT** to support the BRI port connected to the PSTN side.

9. Click **Confirm/Ok**.

Figure 1-8 Configure ISDN Signaling Group - PSTN Side



Bypass Number Directory

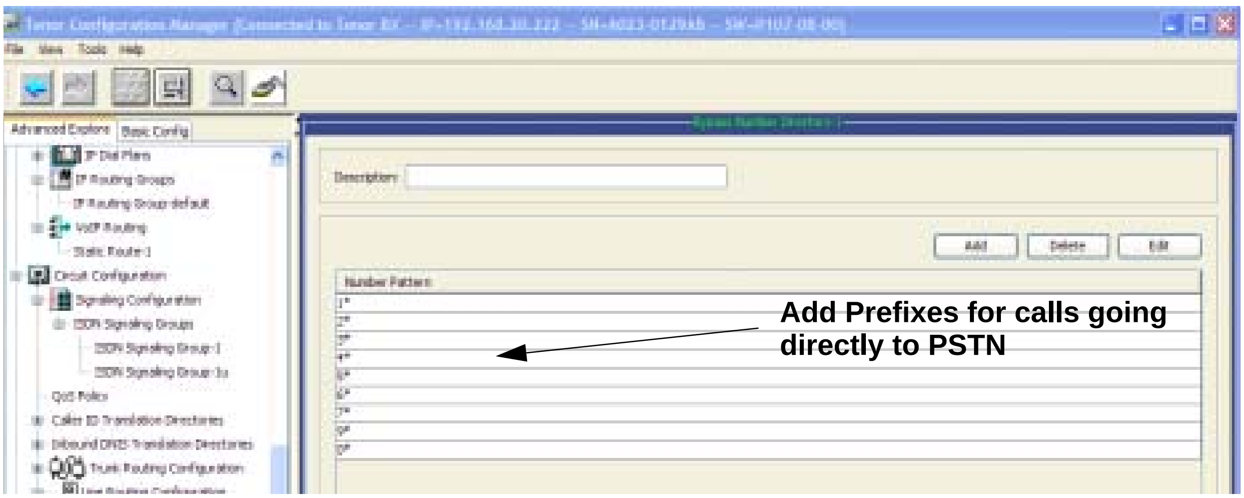
The Bypass Number Directory window contains telephone numbers for calls that are automatically sent to the PSTN and bypass VoIP.

1. From the main menu, select *Circuit Configuration> Line Routing Group> Bypass Number Directories> Bypass Number Directory-1*. See *Figure 1-9*.

2. Enter all prefixes (except 8). Calls with these prefixes will be routed from the PBX directly to the PSTN, without routing over VoIP. (With a prefix of 8, the IP trunk going to the PBX will be used.)

3. Click **Confirm/Ok**.

Figure 1-9 Bypass Number Directory



Hunt LDN Directory

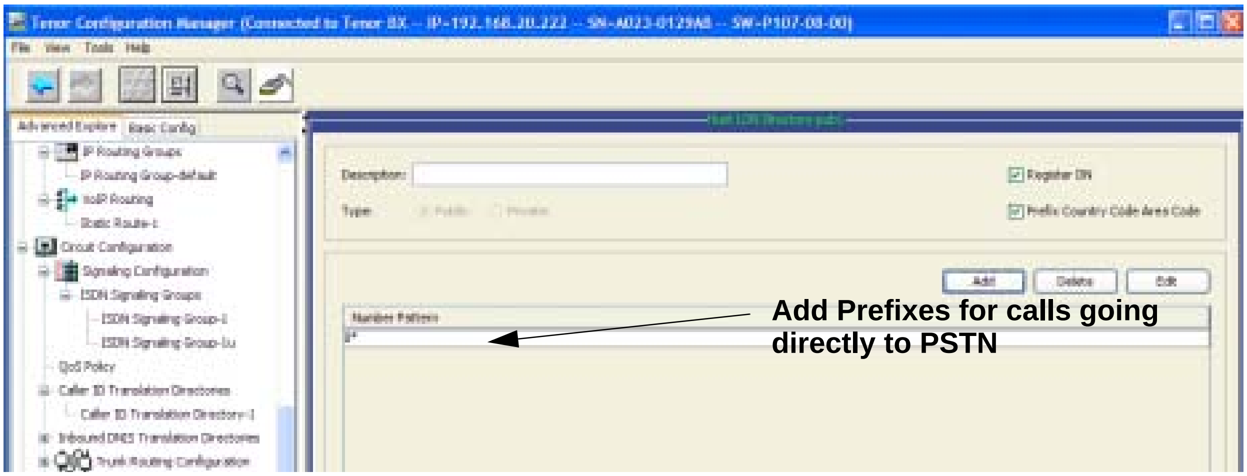
This directory designates the calls which will be routed from the Analog extension (at the *Tenor AF*) to the PBX (i.e., prefix of 8) For example, to reach extension 801 (at the PBX), you need to dial 8801 at the remote Analog extension.

1. From the main menu, select *Circuit Configuration> Line Routing Group> Hunt LDN Directories> Hunt LDN Directory-1*. See *Figure 1-10*.

2. Click on **Add** and enter the number pattern of **8**. Click on **Ok**.

3. Click **Confirm/Ok**.

Figure 1-10 Hunt LDN Directory



Hopoff Number Directory

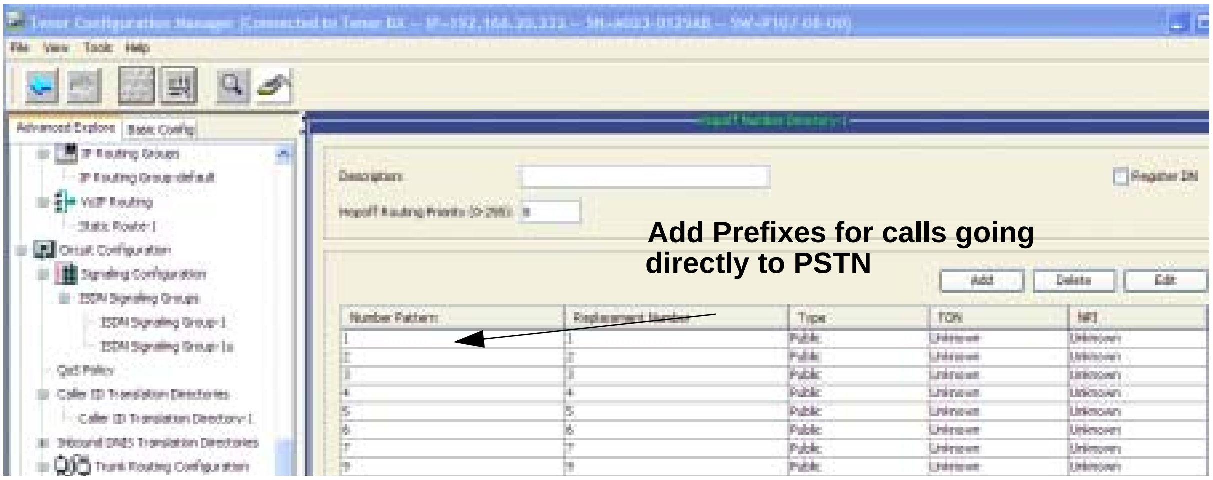
This directory designates the prefixes that are routed from the IP Trunk (from extension 600 and 601 at the *Tenor AF)* directly to the PSTN.

1. From the main menu, select *Circuit Configuration> Trunk Routing Group> Hopoff Number Directories> Hopoff Number Directory-1*. See *Figure 1-11*.

2. Enter all prefixes (except a prefix of 8). These prefixes will be routed from the Analog extension (at Tenor AF) to the PSTN. (With a prefix of 8, the IP trunk going to the PBX will be used).

3. Click **Confirm/Ok**.

Figure 1-11 Hopoff Number Directory



Trunk Circuit Routing Group

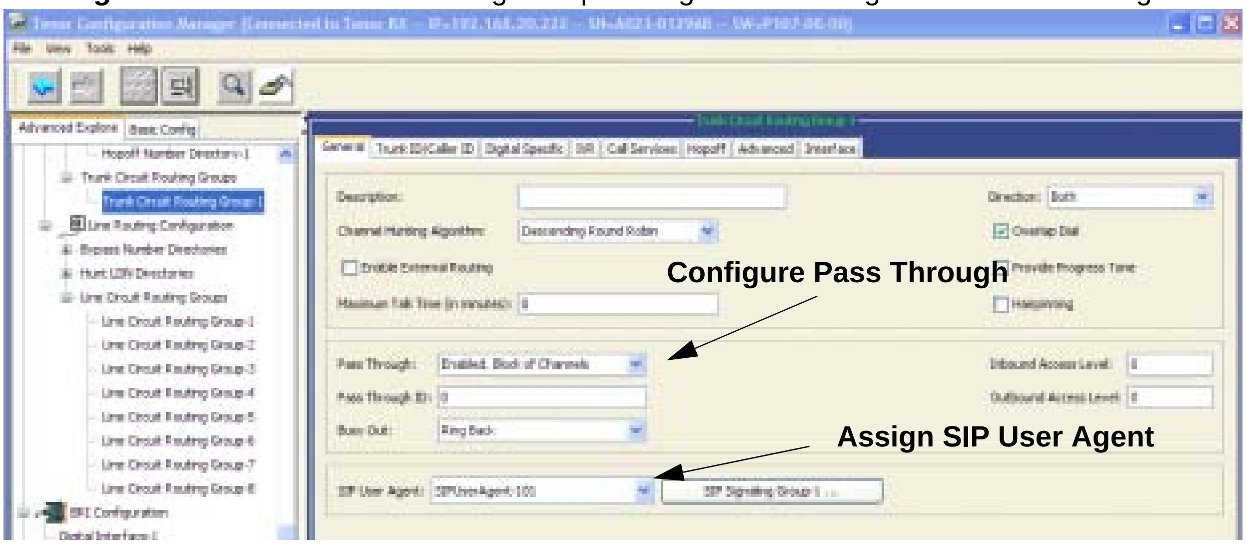
1. From the main menu, select *Circuit Configuration> Trunk Routing Configuration> Trunk Routing Groups> Trunk Routing Group-1*.

2. Click on the *General* Tab. See *Figure 1-12*.

3. From the **Pass Through** drop down box, select **Enabled, Block of Channels**. This allows PSTN calls to pass through to the PBX side. The **Pass through ID** must be the same value as the associated LCRG.

4. From the **SIP User Agent** drop down box, select/assign the **SIP User Agent** (you must first define a SIP Signaling Group containing SIP User Agents before attaching it to this routing group; see *SIP Signaling Group*).

Figure 1-12 Trunk Circuit Routing Group - Assign SIP User Agent and Pass Through

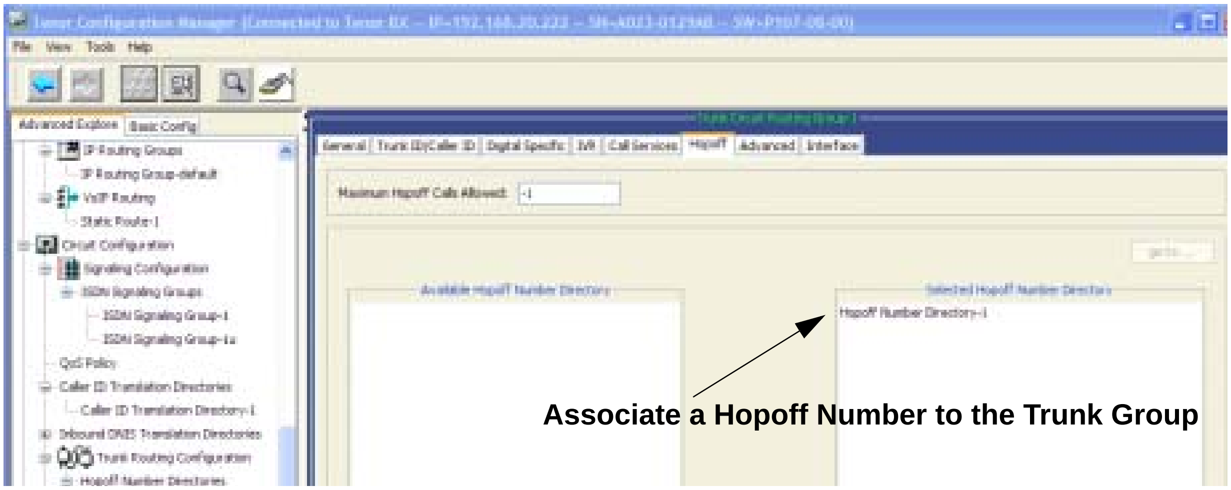


5. Click on the *Hopoff* tab. See *Figure 1-13*.

6. From the A*vailable Hopoff Number* directory window, select the desired **Hopoff Number Directory** (to associate the Hopoff Number directory to the trunk group) and click on **Add**. This is the Hopoff number you configured in the *Hopoff Number Directory*.

7. Click **Confirm/Ok**.

Figure 1-13 Trunk Circuit Routing Group - Assign Hopoff Number Directory



Line Circuit Routing Group

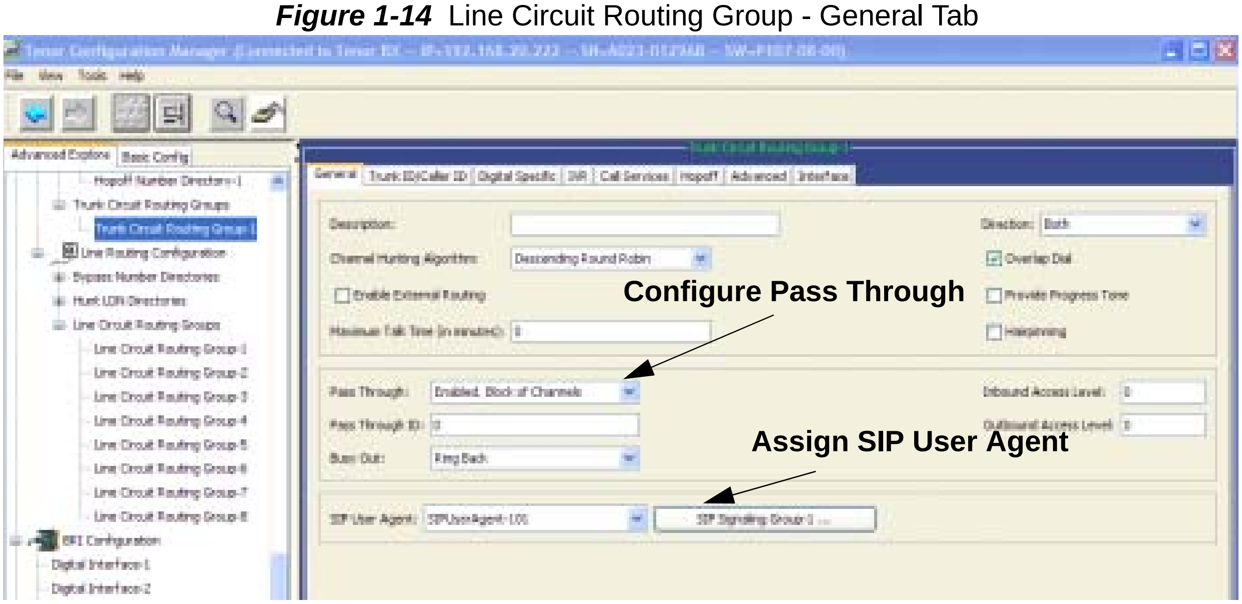
1. From the main menu, select *Circuit Configuration> Line Routing Configuration> Line Circuit Routing Groups> Line Circuit Routing Group-1*.

2. Click on the *General* tab. See *Figure 1-14*.

3. From the **Pass Through** drop down box, select **Enabled, Block of Channels**. This allows PSTN calls to pass through to the PBX side. The **Pass through ID** must be the same value as the associated LCRG.

4. From the **SIP User Agent** drop down box, select/assign the **SIP User Agent** (you must first define a SIP Signaling Group containing SIP User Agents before attaching it to this routing group; see *SIP Signaling Group*).

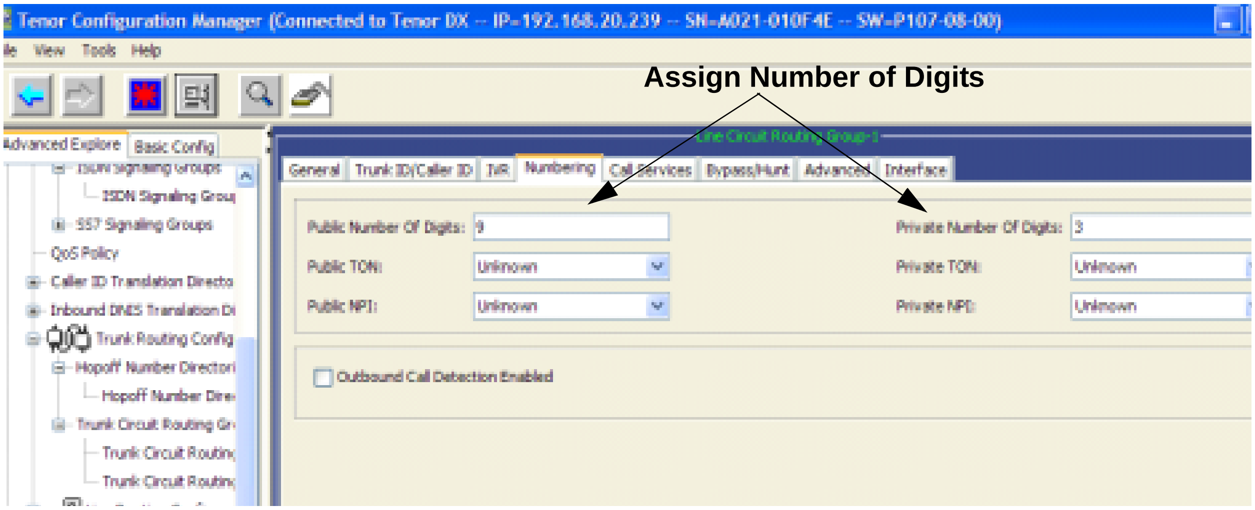
Figure 1-14 Line Circuit Routing Group - General Tab



5. Click on the *Numbering* Tab. See *Figure 1-15*.

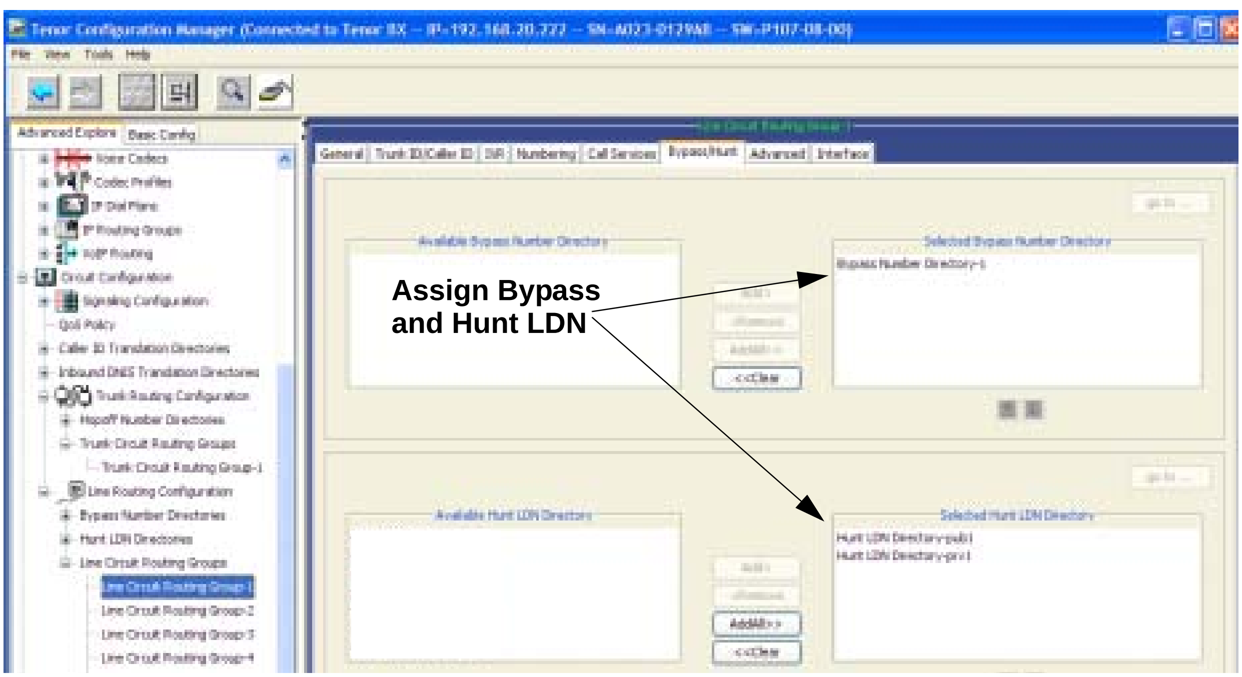
6. In the **Public Number of Digits** box, enter the number of public and private digits (i.e., public, 9 and private, 3).

Figure 1-15 Line Circuit Routing Group - Numbering Tab



7. Click on the *Bypass/Hunt* tab. See *Figure 1-16*.

Figure 1-16 Line Circuit Routing Group - Bypass/Hunt Tab



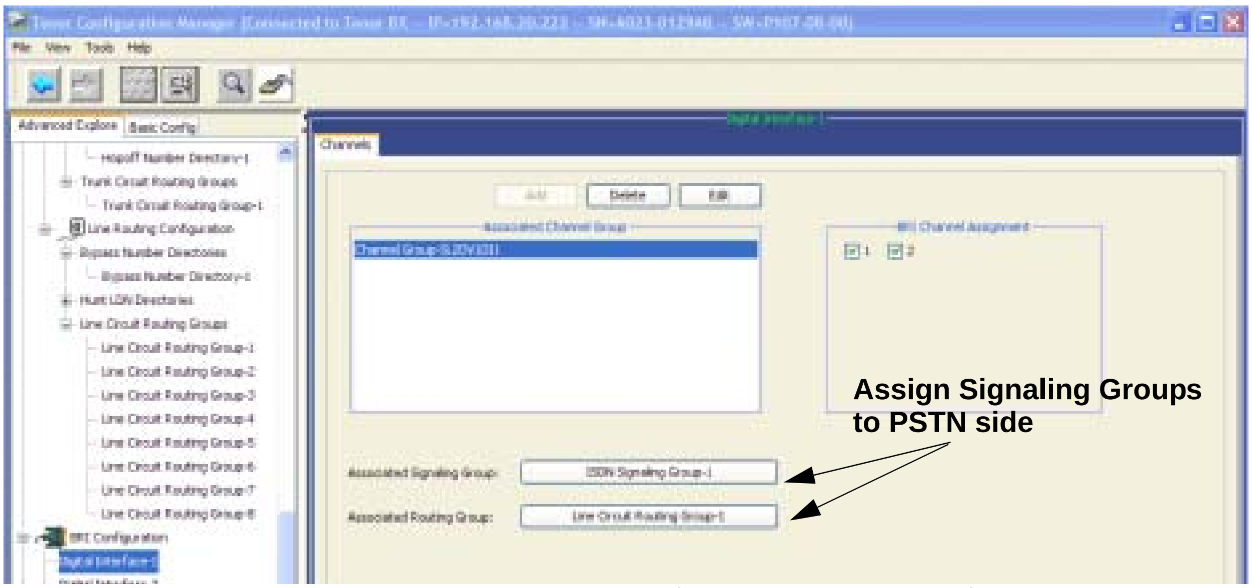
8. Ensure the Bypass Number Directory and Hunt LDN Directory are associated with the Line Circuit Routing Group. To ensure you choose the correct options, see *Bypass Number Directory* and *Hunt LDN Directory*.

9. Click **Confirm/Ok**.

Step 4: Digital Interface

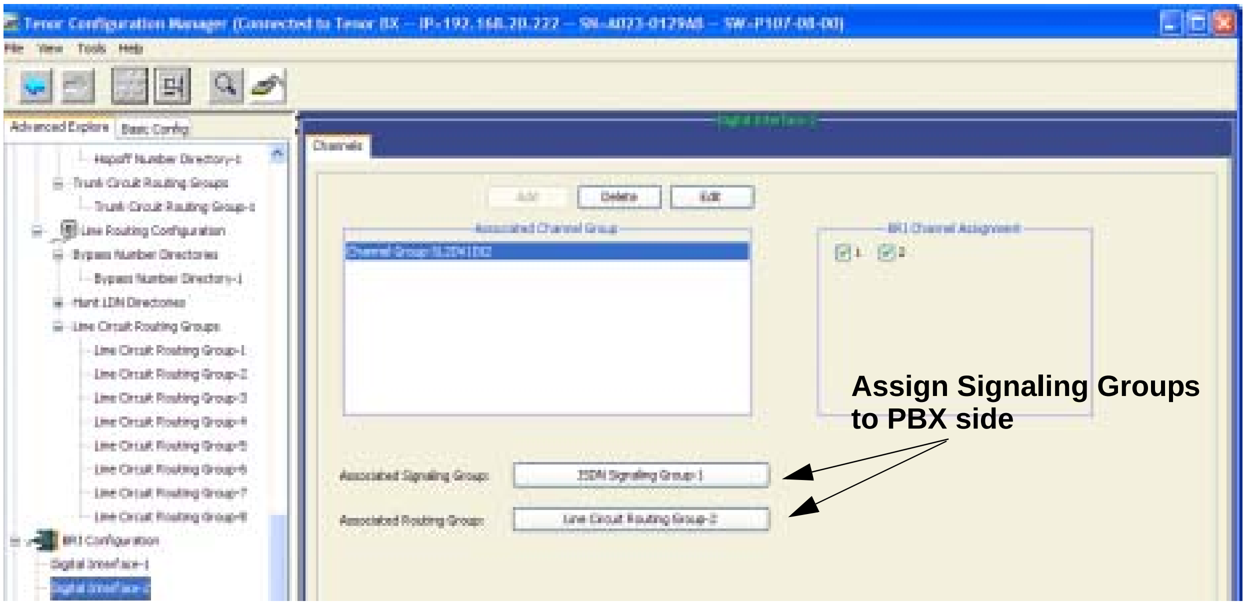
1. From the main menu, select *BRI Configuration> Digital Interface -1*.
2. Click on the *Channels* tab. See *Figure 1-17*.
3. Assign the LCRG and ISDN Signaling groups to the physical ports that lead to the BRI trunk line (PSTN).
4. Click on **Confirm/Ok**.

Figure 1-17 Assign Signaling Groups to BRI Trunk - PSTN



5. From the main menu, select *BRI Configuration> Digital Interface -2*.
6. Click on the *Channels* tab. See *Figure 1-18*.
7. Assign the LCRG and ISDN Signaling groups to the physical port that leads to wherever the PBX is connected.
8. Click on **Confirm/Ok**.

Figure 1-18 Assign Signaling Groups to BRI Trunk - PBX



Submit Tenor BX Configuration

Once all the configuration is made to the *Tenor BX*, you can submit the changes as follows:

From the main *Tenor Configuration Manager* window, click on *File > Submit Changes*. The changes are submitted to the Tenor BX.

Setup Tenor 2 - Tenor AFM200 (remote side, FXS)

Ensure the *Tenor Configuration Manager* is connected to the *Tenor AFM200*. For instructions on how to install and access the software or to connect to a Tenor, see *Tenor Configuration Manager User Guide*.

NOTE: These instructions assume the *Tenor AFM200* is connected to the *Tenor Configuration Manager.*

System Wide Configuration

Dial Plan

1. From the main menu, select *System-Wide Configuration> Dial Plan*. Click on the *General* tab. See *Figure 1-19*.

2. From the **Dial Plan Country** drop down box, select **Generic CCITT.**

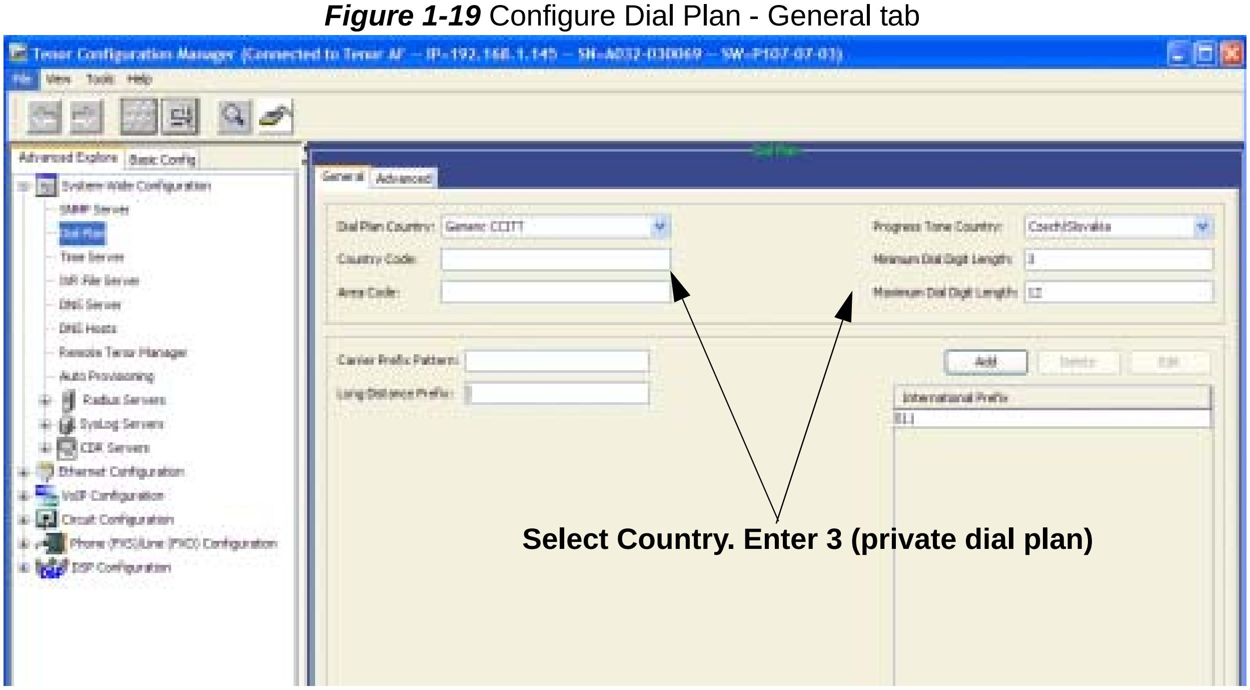
3. In the **Minimum Dial Digit Length** box, enter **3** (private dial plan).

4. Leave the **Dial Plan** information (**Country Code** and **Area Code**) blank. It is unnecessary to normalize numbers into E.164 for IP trunk calls.

5. From the **Progress Tone Country** drop down box, select **Czech/Slovakia.**

6. Click **Confirm/Ok**.

Figure 1-19 Configure Dial Plan - General tab



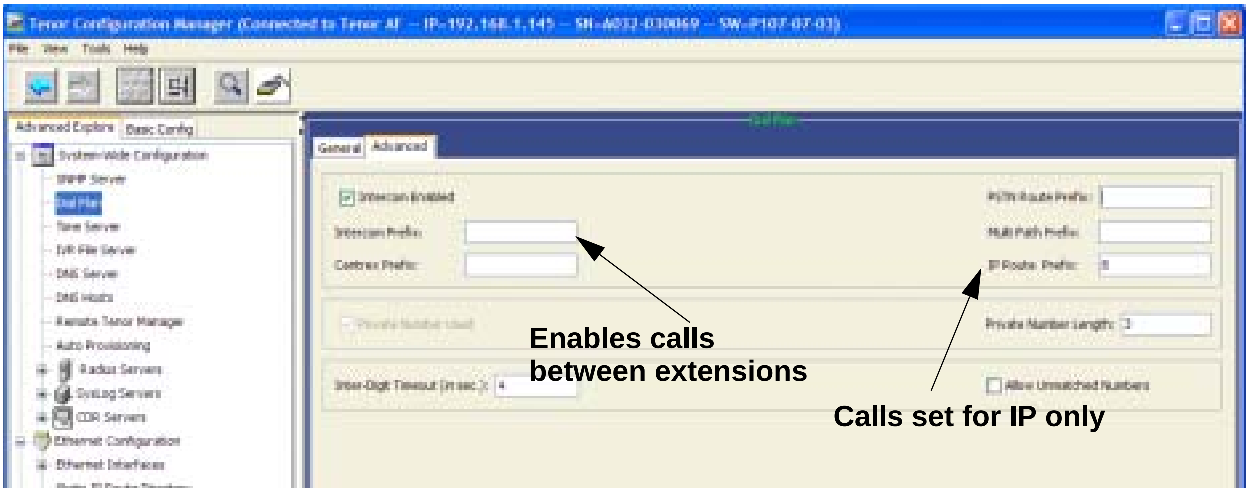
7. Click on the *Advanced* tab. See *Figure 1-20*.

8. Ensure the **Intercom Enabled** check box is marked (the **Private Number Used** box is marked automatically). This enables calls between the extensions 600 and 601.

9. In the **IP Route Prefix** box, enter **8**. This enables any call that has the first digit of 8 will be routed by IP only.

10. Click **Confirm/Ok**.

Figure 1-20 Configure Dial Plan - Advanced tab



VoIP Configuration

SIP Signaling Group

1. From the main menu, select *VoIP Configuration> SIP Signaling Group> SIP Signaling Group-1*.

2. Click on the *General* tab. See *Figure 1-21*.

3. Enter an **IP address** in the **Primary SIP Server** field. This IP address is for the *Tenor BX* (where calls from extension 600 and 601 will be routed).

4. Click **Confirm/Ok**.

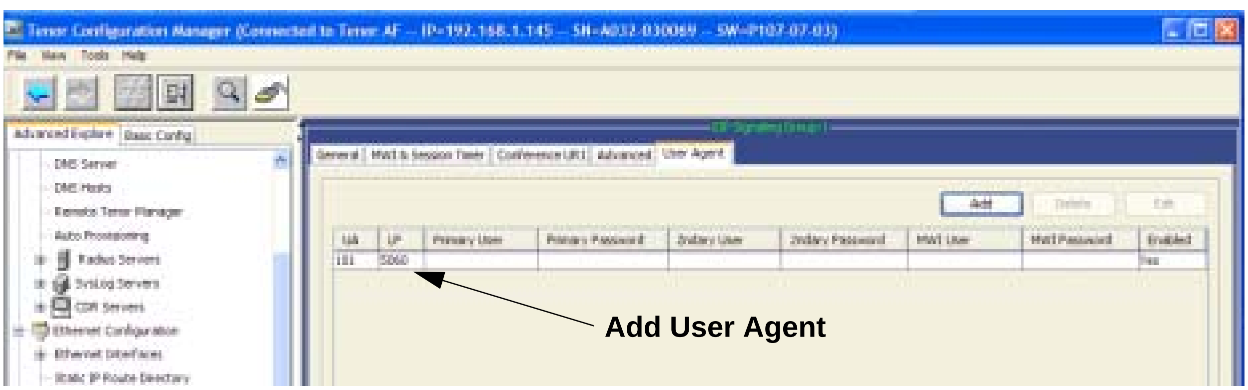
Figure 1-21 Configure SIP Signaling Group - General tab

5. Click on the *User Agent* tab. See *Figure 1-22*.

6. Ensure a User Agent has been created (To add a user Agent, click on **Add**. Enter the User Agent information. Detailed information about these fields is available in the *Command Reference* guide.)

7. Click on **Confirm/Ok**.

Figure 1-22 Configure SIP Signaling Group - User Agent tab



DN Channel Map

1. From the main menu, select *VoIP Configuration> DN Channel Map.*

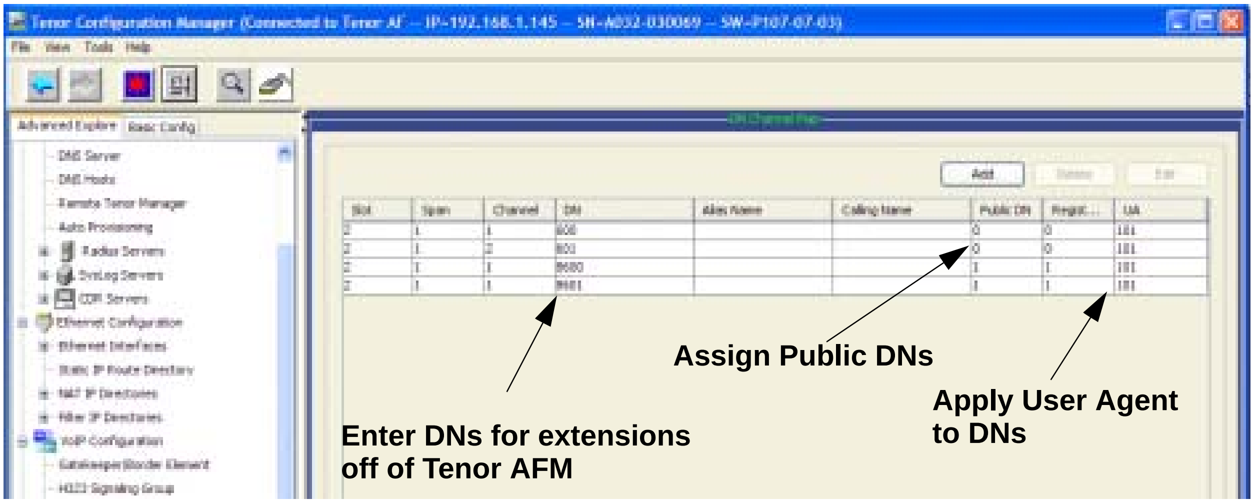
2. Click on **Add.** Through the *Add DN Channel Map* screen, add the following DN channel maps:

   - Extensions 600 and 601 (phone/fax connected to the *Tenor AFM*).
   - Public numbers (8600 and 8601) used in the public dial plan for calls from the IP trunk.

Ensure the **Public DN** box is unchecked (this is a private dial plan) and the **User Agent** (created in *SIP Signaling Group*) is selected. Detailed information about configuring these fields is available in the *Command Reference* guide.

3. Click **Confirm/Ok**.

Figure 1-23 Configure DN Channel Map



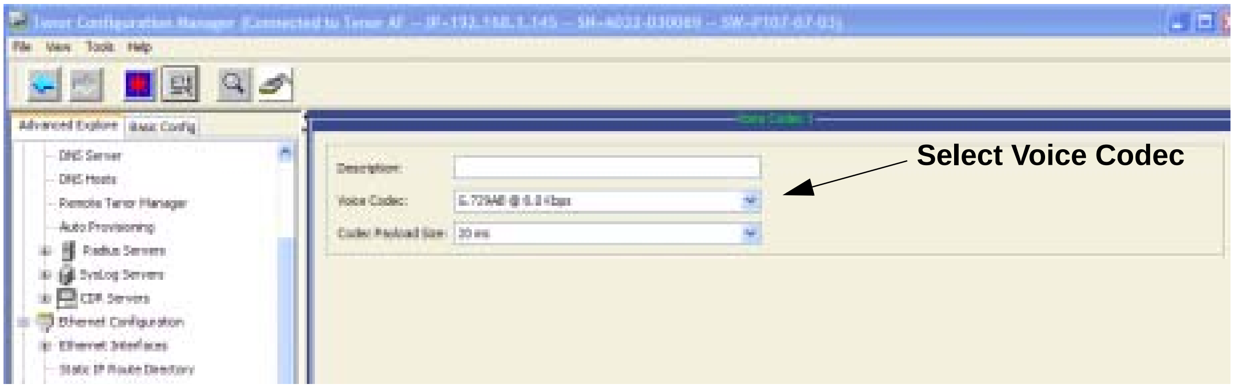
Voice Codec

1. From the main menu, select *VoIP Configuration> Voice Codecs> Voice Codec-1*. See *Figure 1-24*.

2. From the **Voice Codec** drop down box, select the codec you will use as the voice compression algorithm (for IP trunk). At least one of the codecs in the drop down list must match with one of the codecs defined in the *Tenor BX* configuration.

3. Click **Confirm/Ok**.

Figure 1-24 Configure Voice Codec



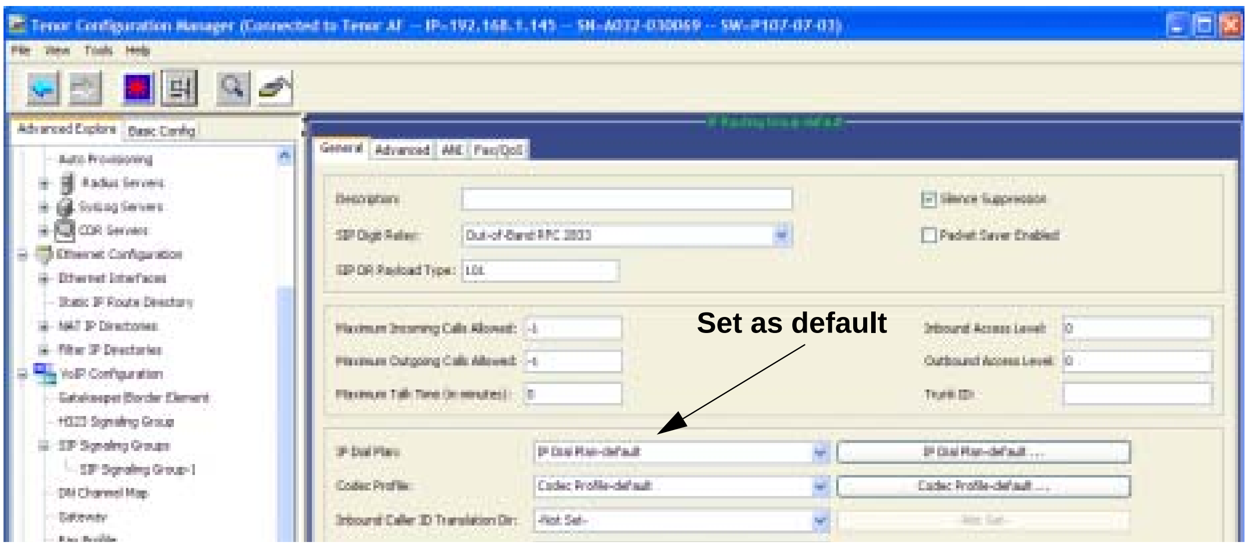
IP Routing Group

1. From the main menu, select *VoIP Configuration> IP Routing Group> IP Routing Group-1*. See *Figure 1-25*.

2. From the **IP Dial Plan** drop down box, ensure the IP Dial Plan is set to *IP Dial Plan-default.*

3. Click **Confirm/Ok**.

Figure 1-25 IP Routing Group



Circuit Configuration

CAS Signaling Group

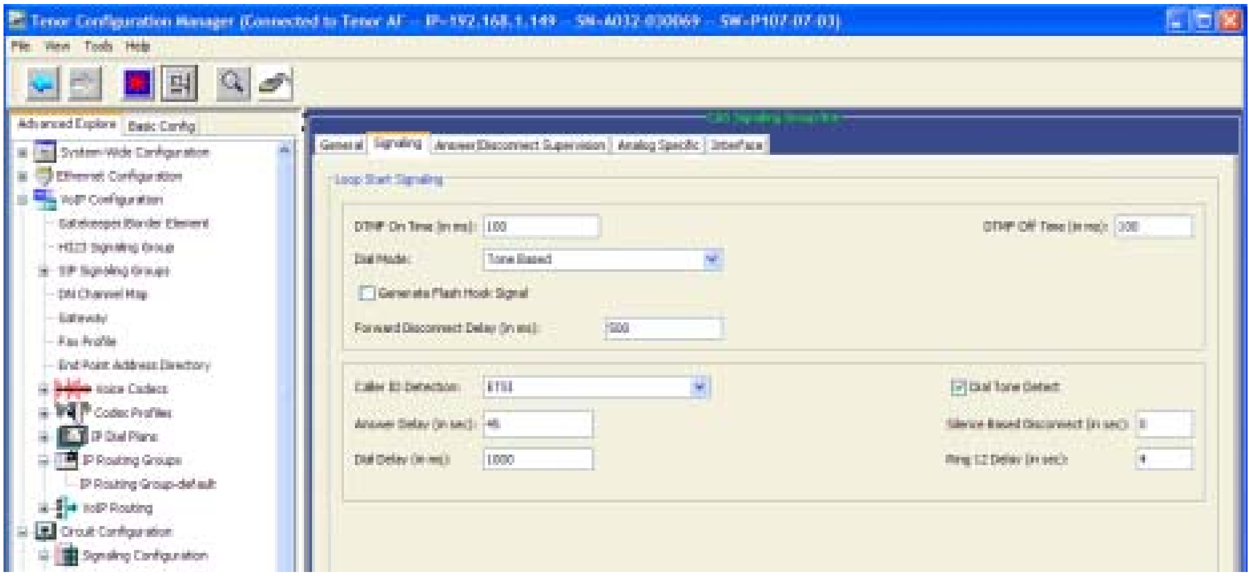
1. From the main menu, select *Circuit Configuration> CAS Signaling Group> CAS Signaling Group - Line.*

2. Click on the *Signaling* tab. See *Figure 1-26*.

3. To configure the screen, use the configuration options set in *Figure 1-26* as a guideline.

4. Click **Confirm/Ok**.

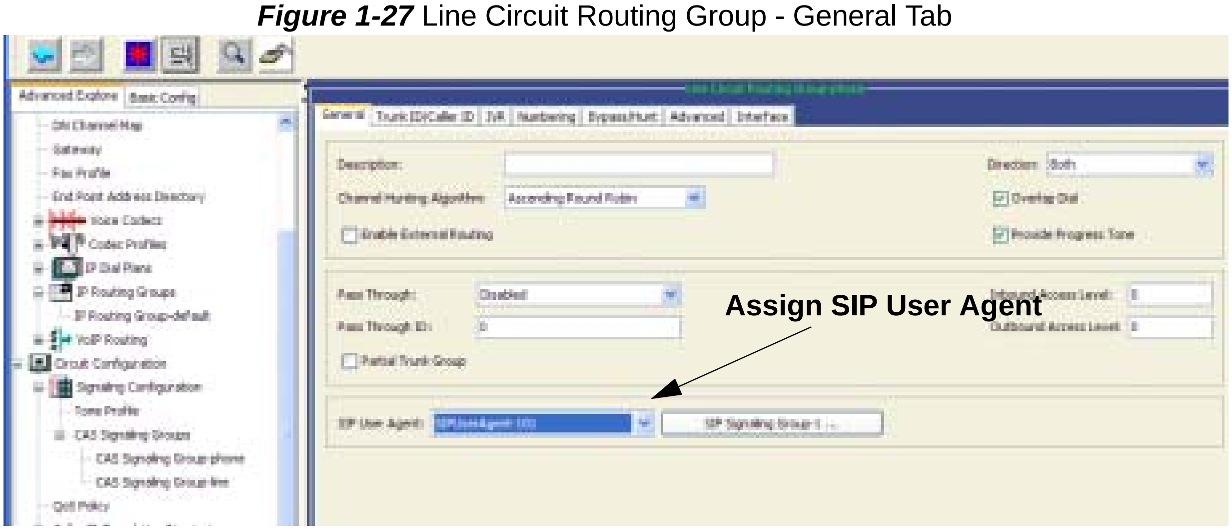
Figure 1-26 CAS Signaling Group



Line Circuit Routing Group

1. From the main menu, select *Circuit Configuration> Line Routing Configuration> Line Circuit Routing Groups> Line Circuit Routing Group-1*.
2. Click on the *General* tab. See *Figure 1-27*.
3. From the **SIP User Agent** drop down box, select/assign the **SIP User Agent** (a SIP Signaling Group containing SIP User Agents must be defined before attaching it to this routing group; see *SIP Signaling Group*).
4. Click **Confirm/Ok**.

Figure 1-27 Line Circuit Routing Group - General Tab

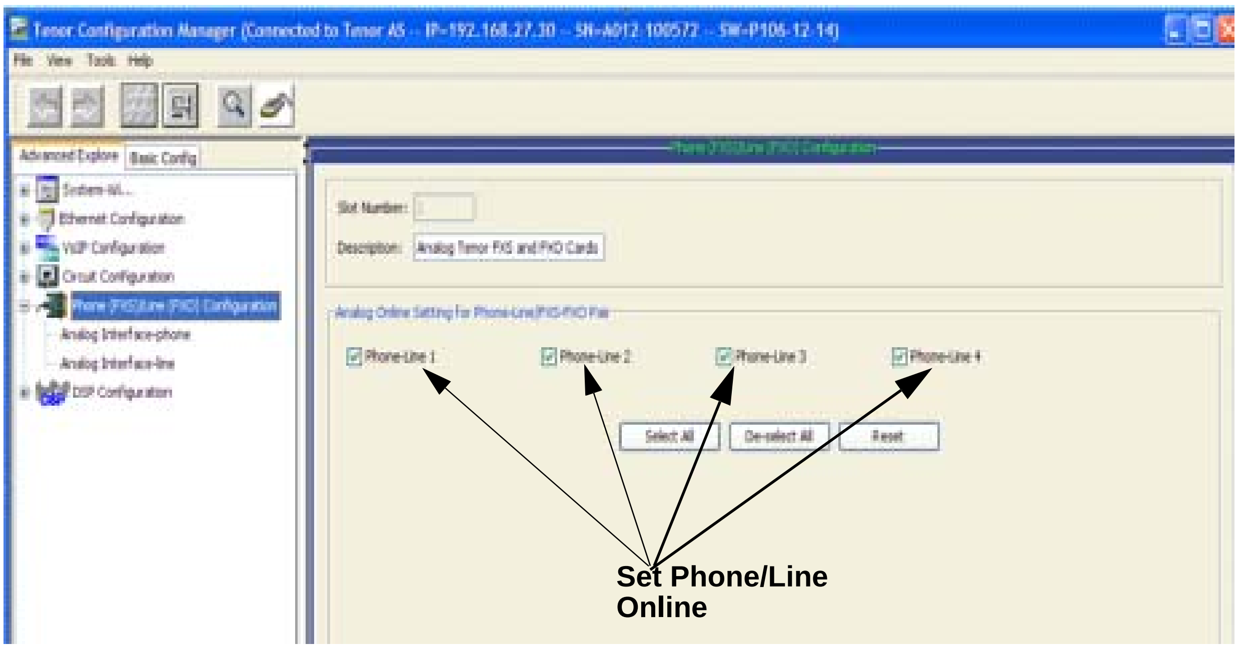


Phone (FXS) /Line (FXO) Configuration

Set Tenor AFM Online

1. From the main menu, select *Phone (FXS)/ Line (FXO)*. See *Figure 1-28*.
2. For the *Analog Online Setting for Phone-Line/FXS-FXO-Pair,* ensure all **Phone-Line** boxes are checked. This puts all Lines online.
3. Click **Confirm/Ok**.

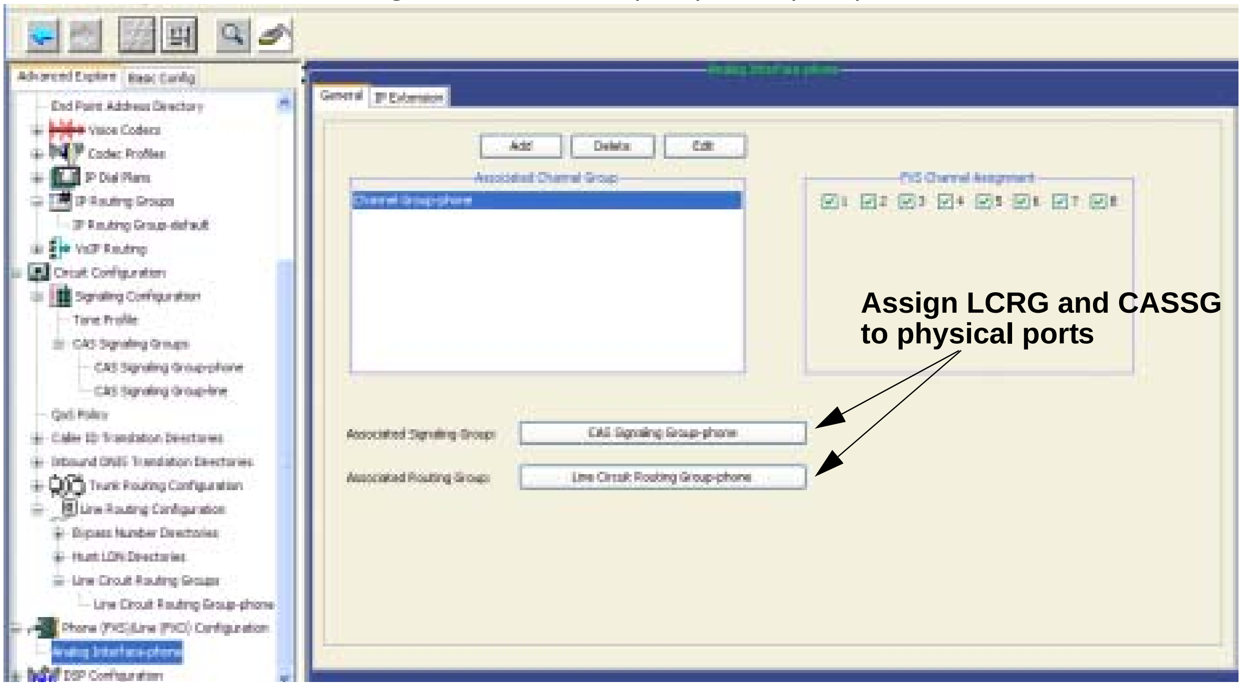
Figure 1-28 Set Phone (FXS)/ Line (FXO) Online



Analog Interface

1. From the main menu, select *Phone (FXS)/ Line (FXO) > Analog Interface-phone.*
2. Click on the *General* tab. See *Figure 1-29*.
3. Assign the LCRG (Line Circuit Routing Group) and CASSG (CAS Signaling Group) to the group of physical analog ports (extension 600 and 601).
4. Click **Confirm/Ok**.

Figure 1-29 Phone (FXS)/ Line (FXO)



Submit Tenor AF Configuration

Once all the configuration is made to the *Tenor AF*, you can submit the changes as follows:

From the main *Tenor Configuration Manager,* click on *File > Submit Changes*. The changes are submitted to the Tenor AF.