Mediant™ 800, 1000 and 3000

E-SBC Media Gateways

Connecting PAETEC SIP Trunking Service to Microsoft® Lync Server 2010

Configuration Guide

Version 6.2

October 2011

Document # LTRT-33420

Published by the AudioCodes Interoperability Group

Table of Contents

Table of Figures

Reader's Notes

Notice

This document describes the procedure for integrating the PAETEC SIP Trunking service with Microsoft® Lync Server using the AudioCodes Mediant 800 MSBG-E-SBC, Mediant 1000 MSBG E-SBC and Mediant 3000 E-SBC Media Gateway.

Information contained in this document is believed to be accurate and reliable at the time of printing. However, due to ongoing product improvements and revisions, AudioCodes cannot guarantee the accuracy of printed material after the Date Published nor can it accept responsibility for errors or omissions. Updates to this document and other documents can be viewed at http://www.audiocodes.com/downloads.

This document is subject to change without notice.

Date Published: October-27-2011

Trademarks

AudioCodes, AC, AudioCoded, Ardito, CTI2, CTI², CTI Squared, HD VoIP, HD VoIP Sounds Better, InTouch, IPmedia, Mediant, MediaPack, NetCoder, Netrake, Nuera, Open Solutions Network, OSN, Stretto, TrunkPack, VMAS, VoicePacketizer, VoIPerfect, VoIPerfectHD, What's Inside Matters, Your Gateway To VoIP and 3GX are trademarks or registered trademarks of AudioCodes Limited. All other products or trademarks are property of their respective owners.

WEEE EU Directive

Pursuant to the WEEE EU Directive, electronic and electrical waste must not be disposed of with unsorted waste. Please contact your local recycling authority for disposal of this product.

Customer Support

Customer technical support and service are provided by AudioCodes' Distributors, Partners, and Resellers from whom the product was purchased. For Customer support for products purchased directly from AudioCodes, contact support@audiocodes.com or from within North America toll free at 800-735-4588 or Internationally at +972-3-976-4343.

Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used.

Note: Throughout this guide, the term *E-SBC device* refers to AudioCodes' Mediant 800 MSBG E-SBC, Mediant 1000 MSBG E-SBC and the Mediant 3000 E-SBC Media Gateway.

Related Documentation

Manual Name
LTRT-26901 SIP CPE Release Notes v6.2
LTRT-52306 SIP CPE Product Reference Manual v6.2
LTRT-27001 Mediant 1000 MSBG User's Manual v6.2
LTRT-40809 Mediant 1000 MSBG Installation Manual v6.2
LTRT-89710 Mediant 3000 SIP User's Manual v6.2
LTRT-94708 Mediant 3000 SIP-MGCP-MEGACO Installation Manual v6.2

1 Introduction

This Configuration Guide describes a sample configuration scenario for a network that uses the AudioCodes Mediant 800 MSBG E-SBC, Mediant 1000 MSBG E–SBC or the Mediant 3000 E-SBC Media Gateway to facilitate a connection between Microsoft Lync 2010 and PATEC's SIP Trunking Service, for superior voice quality services.

The Mediant 800 MSBG E-SBC is a networking device that combines multiple service functions such as a Media Gateway, Session Border Controller (SBC), Data Router and Firewall, LAN switch, WAN access, Stand Alone Survivability (SAS) and an integrated general-purpose server.

The Mediant 1000 MSBG E-SBC is all-in-one multi-service access solution products for Service Providers offering managed services and distributed Enterprises seeking integrated services. This multi-service business gateway is designed to provide converged Voice & Data services for business customers at wire speed, while maintaining SLA parameters for superior voice quality.
The device is based on AudioCodes' VoIPerfect Media Gateway technology, combined with Enterprise class Session Border Controller, Data & Voice security elements, Data Routing, LAN Switching and WAN Access. These services allow smooth connectivity to cloud services, while providing protection to the end customer.

The Mediant 3000 E-SBC Media Gateway is a High Availability VoIP Gateway and Enterprise Class SBC for medium and large enterprises.

Reader's Notes

2 Testing Considerations

Note that for the PAETEC test environment:

- Fax was not tested
- G.711 U-law and G.729 were the codecs tested for this application
- Voice mail was not tested during the certification but it functions

Reader's Notes

3 Configuration Scenario Overview

The configuration scenario described in this guide includes:

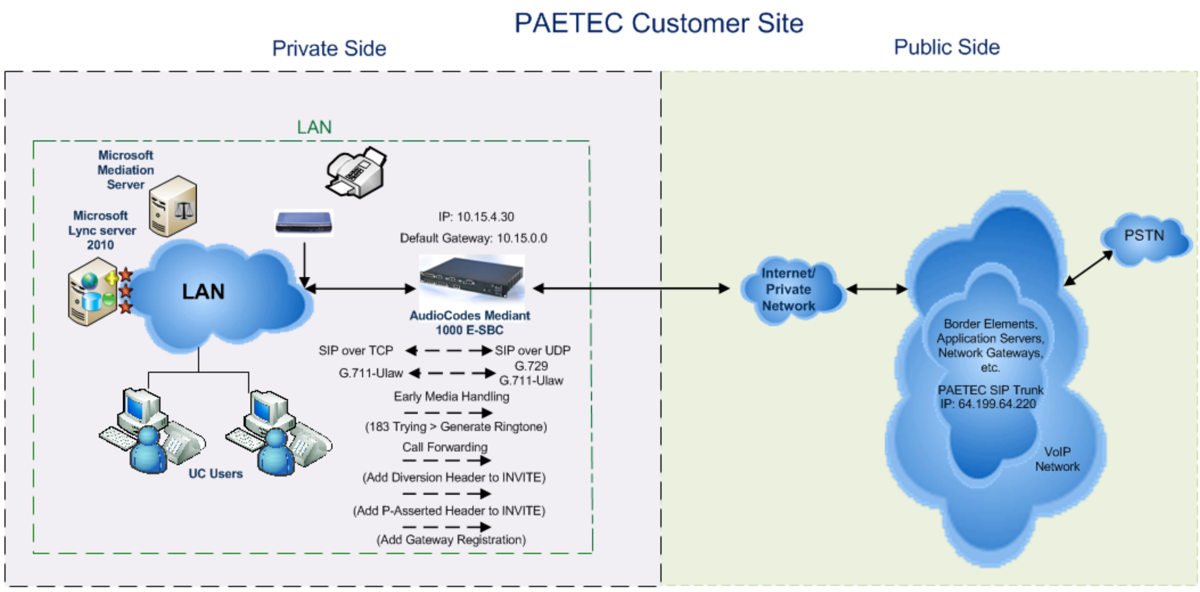
- An enterprise has deployed a Microsoft® Lync server 2010 in its private network for enhanced communication within the company.
- The enterprise decides to offer its employees VoIP and to connect the company to the PSTN using the PAETEC SIP Trunking Service.

Setup requirements are:

- While the Microsoft® Lync Server 2010 environment is located on the enterprise's Local Area Network (LAN), the PAETEC SIP Trunk is located on the WAN.
- Microsoft® Lync Server 2010 works with the TCP or TLS transport type, while the PAETEC SIP Trunk works on the SIP over UDP transport type.
- Microsoft® Lync Server 2010 supports G.711 - µ Law coder type while PAETEC SIP Trunk supports the G.729 and G.711- µ Law coder types.
- Support for early media handling
- Support for call forwarding
- Support of Gateway Registration

Figure **3-1** overviews the configuration scenario.

Figure 3-1**: Configuration Scenario Overview**



Reader's Notes

4 Configuring Microsoft Lync Server 2010

This section describes how to configure the Microsoft Lync Server 2010 to operate with the E-SBC device.

➤ **To do this:**

1. Configure the E-SBC device as an ‘IP/PSTN Gateway’
(see Section 4.1 on page 16)
2. Associate the ‘IP/PSTN Gateway’ with the Mediation Server
(see Section 4.2 on page 21)
3. Configure a ‘Route’ to utilize the SIP trunk connected to the E-SBC device
(see Section 4.3 on page 27)

Note: Dial Plans, Voice Policies, and PSTN usages are also necessary for enterprise voice deployment; however, they are beyond the scope of this document.

4.1 Configuring AudioCodes' E-SBC Device as 'IP/PSTN Gateway'

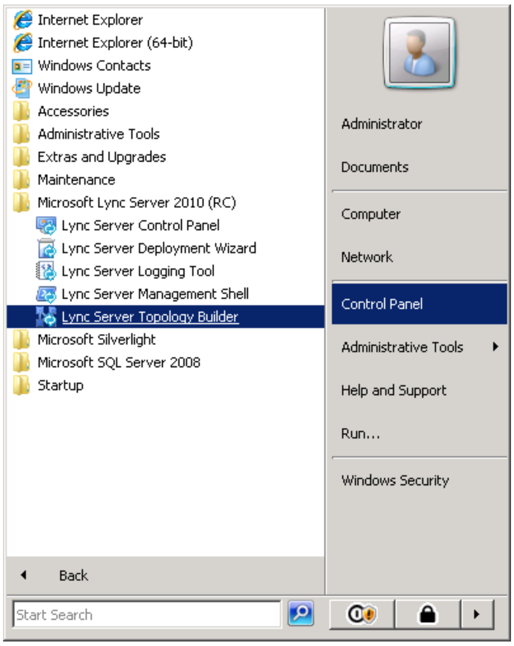
This section describes how to configure the AudioCodes E-SBC device as an IP/PSTN Gateway.

> **Note:** The Microsoft Lync Topology Builder interface dialogs refer to the E-SBC device as an 'IP/PSTN gateway' or 'PSTN gateway'.

➤ **To configure the E-SBC device as an IP/PSTN Gateway and associate it with the Mediation Server:**

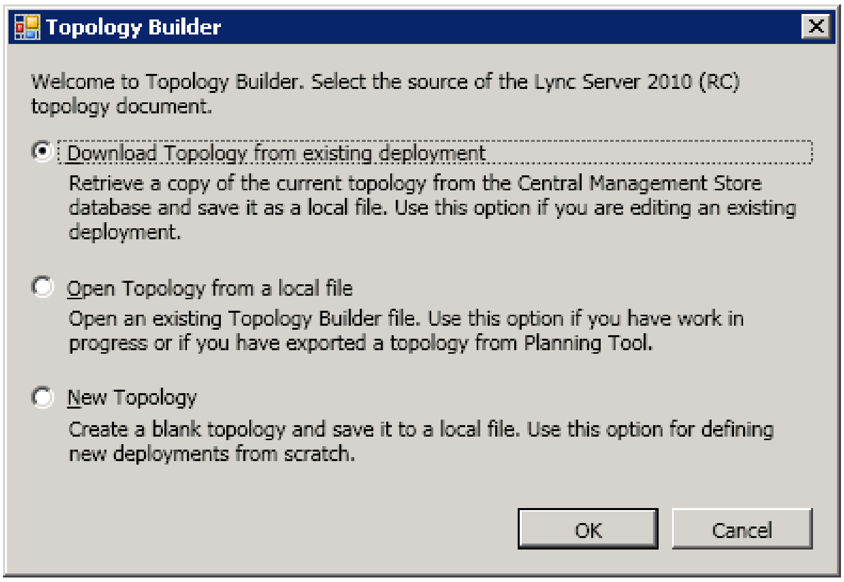
1. On the server where the Topology Builder is located, start the Microsoft Lync Server 2010 **Topology Builder**: Click **Start**, select **All Programs**, then select **Lync Server Topology Builder**.

Figure 4-1: Starting the Lync Server Topology Builder



The following screen is displayed:

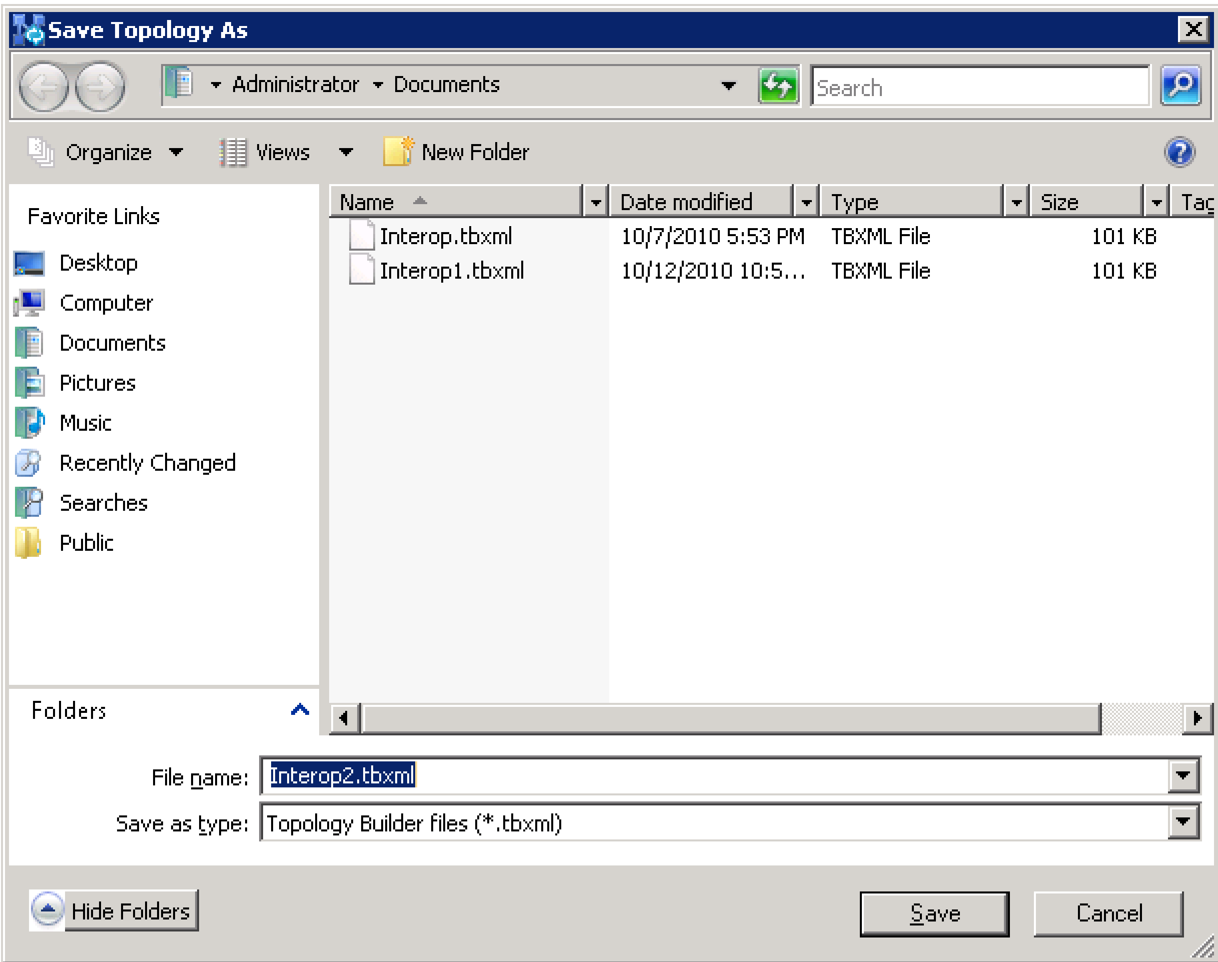
Figure 4-2: Topology Builder Options



2. Choose 'Download Topology from the existing deployment' and click **OK**.

   You are prompted to save the Topology which you have downloaded.

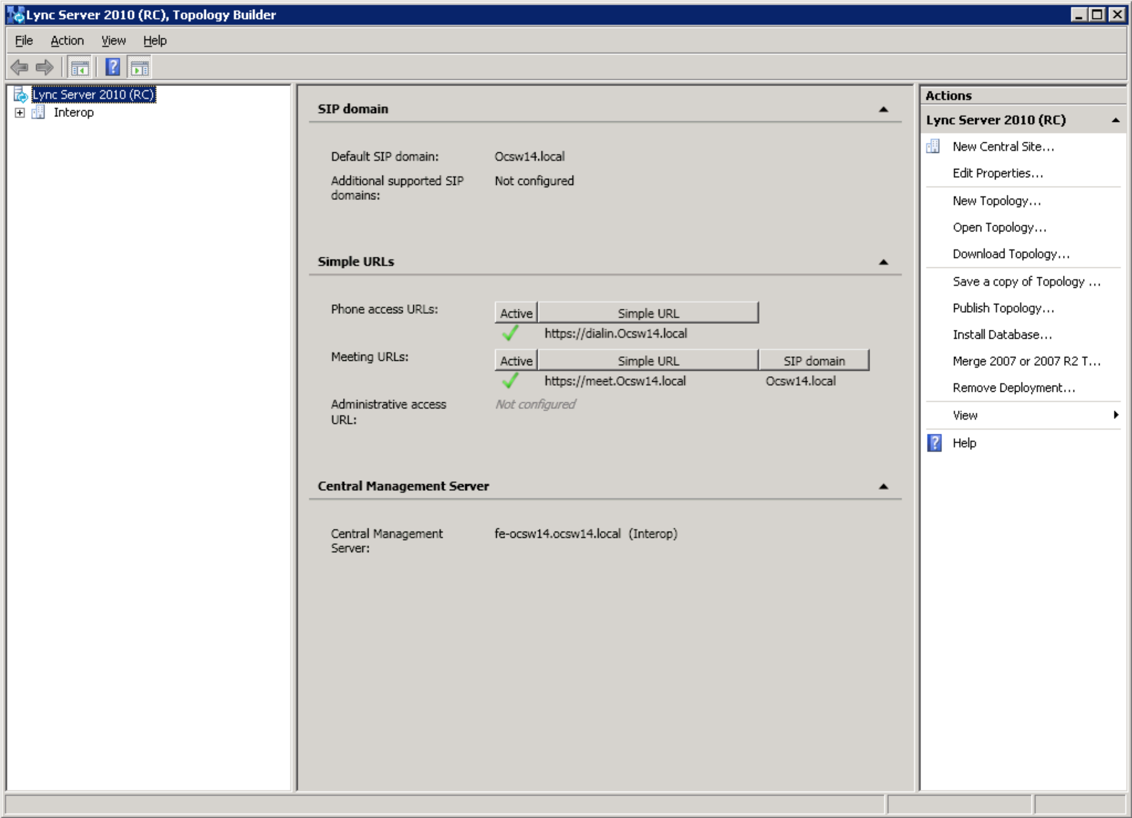
Figure 4-3: Save Topology



3. Enter a new **File Name** and **Save** - this action enables you to roll back from any changes you make during the installation.

   The Topology Builder screen with the downloaded topology is displayed.

Figure 4-4: Downloaded Topology



4. Expand the Site; right-click on the IP/PSTN Gateway and choose 'New IP/PSTN Gateway'.

Figure 4-5: New IP/PSTN Gateway

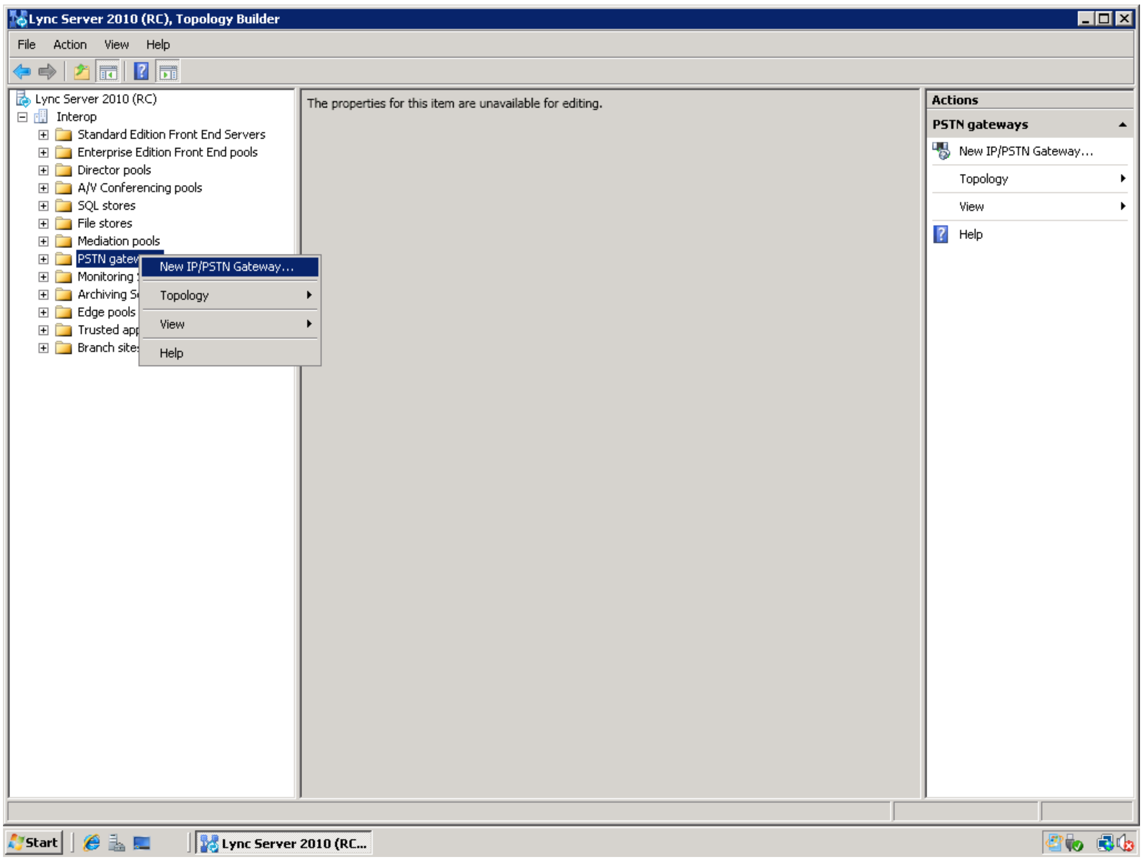
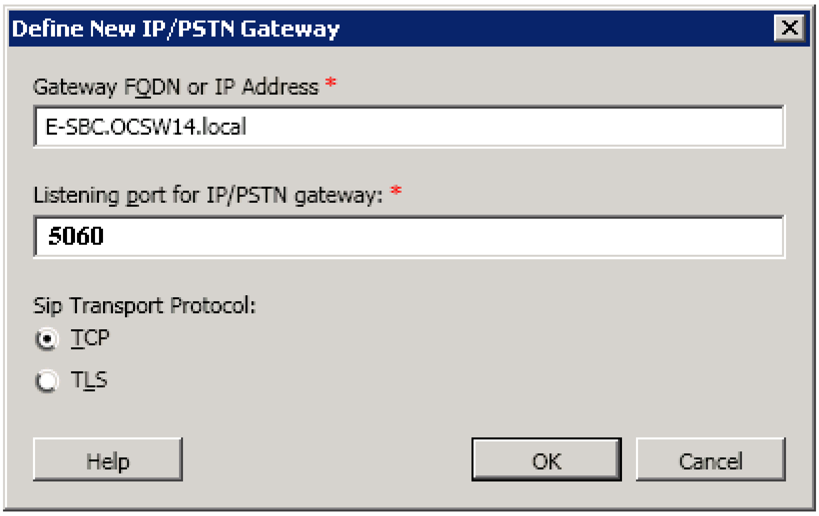


Figure 4-6: Define New IP/PSTN Gateway

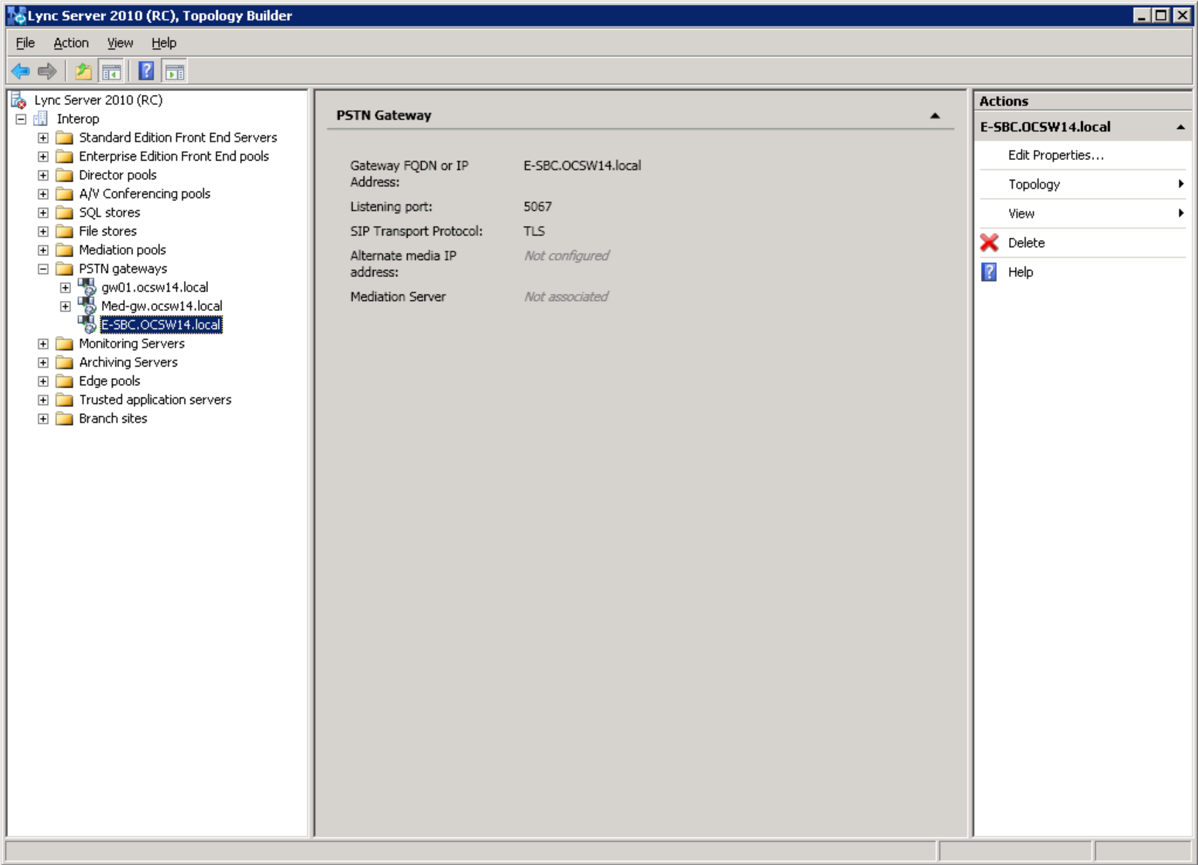


5. Enter the FQDN of the E-SBC device (i.e., 'E-SBC.OCSW14.local') and click **OK**.

   Note that the listening port for the Gateway is '5060' and the transport type is 'TCP'. In certification testing for the PAETEC SIP Trunk, listening port 5060 was used with transport protocol 'UDP'. To configure the E-SBC with the datafill in order to interface PAETEC, see Configuring the E-SBC Device.

   The E-SBC device is now added as an 'IP/PSTN Gateway'.

Figure 4-7: IP/PSTN Gateway



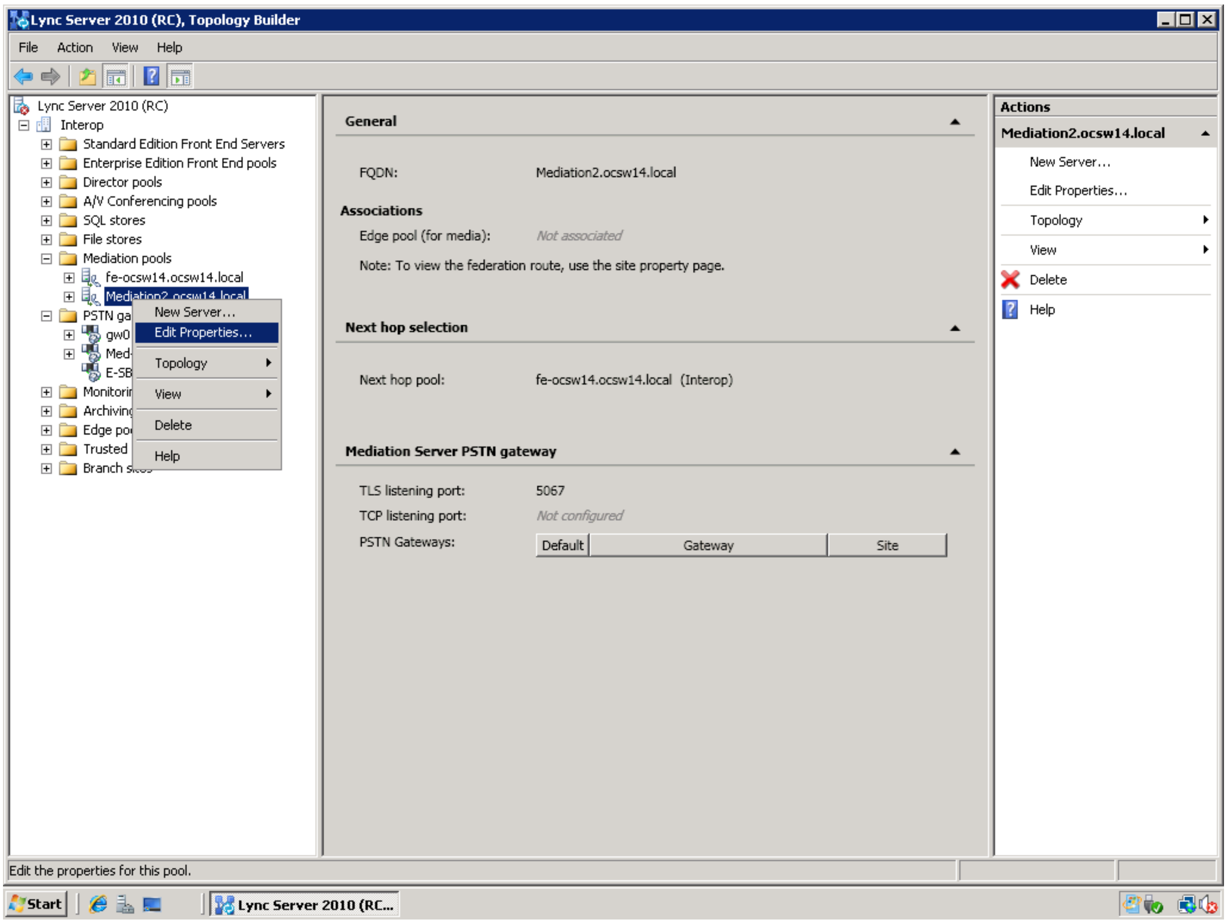
4.2 Associating the 'IP/PSTN Gateway' with the Mediation Server

This section describes how to associate the 'IP/PSTN Gateway' (E-SBC device) with the Mediation Server.

➢ **To associate the IP/PSTN Gateway with the Mediation Server:**

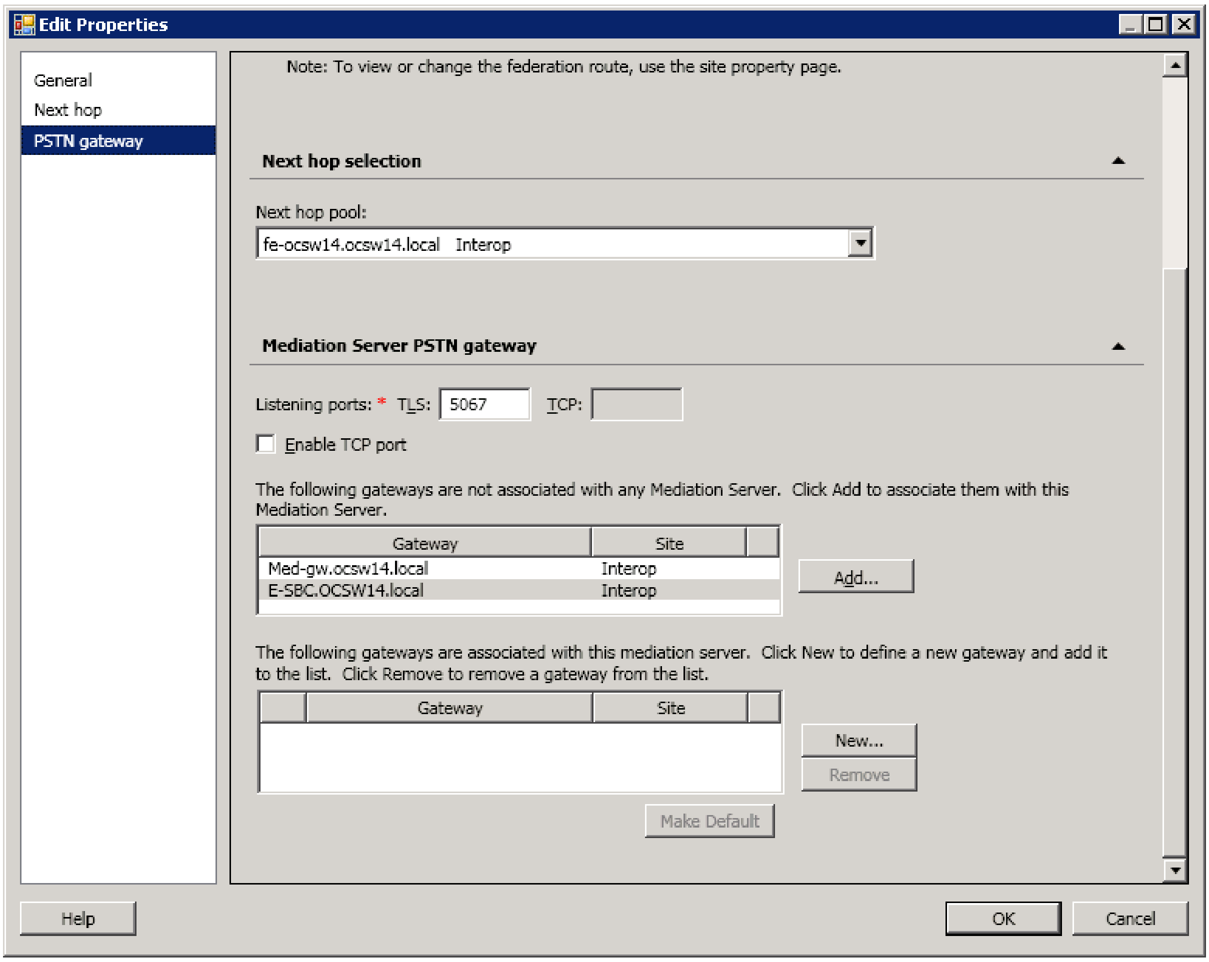
1. Right-click on the **Mediation Server** to use with the IP/PSTN Gateway (i.e., Mediation2.OCSW14.local) and choose **Edit Properties**.

Figure 4-8: Associating Mediation Server with IP/PSTN Gateway



The following screen is displayed:

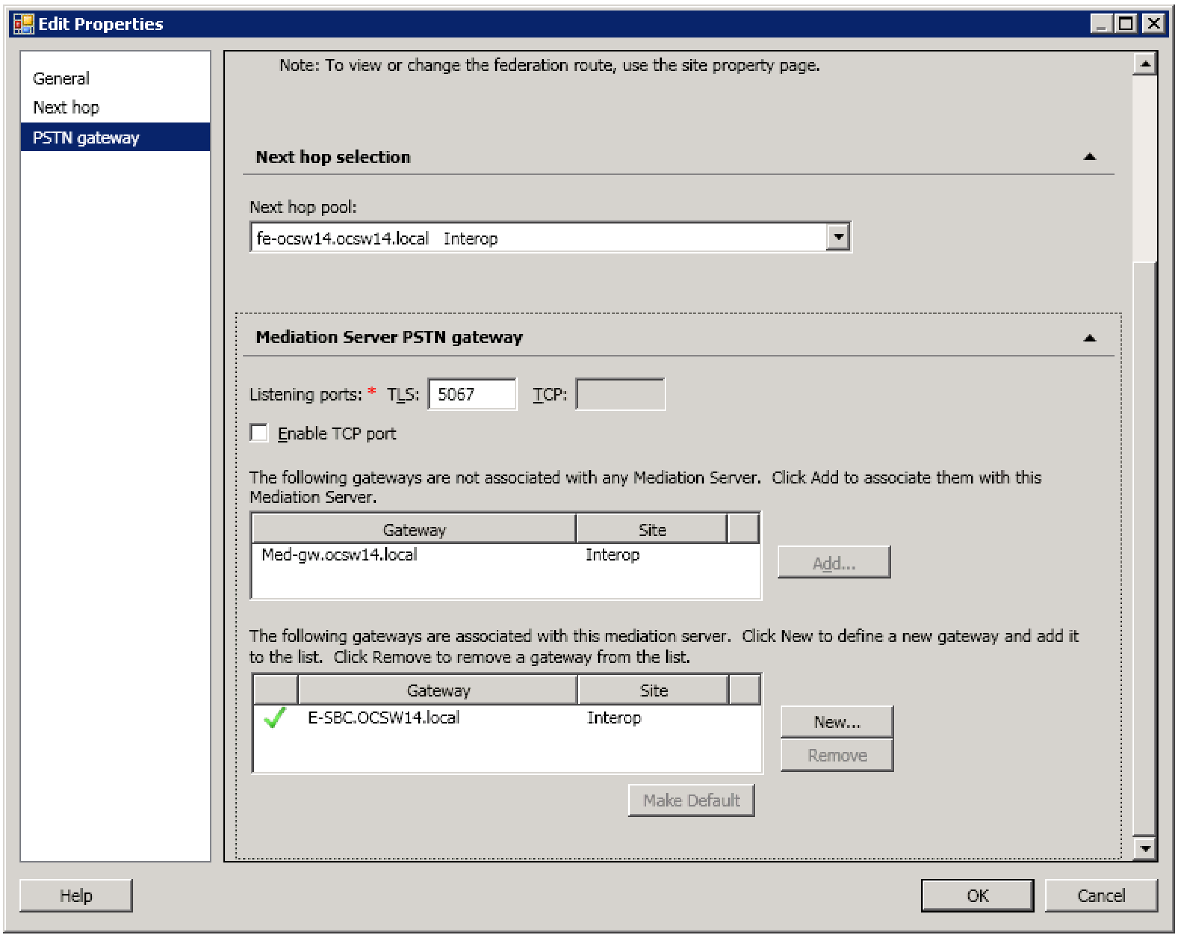
Figure 4-9: Before Associating IP/PSTN Gateway to a Mediation Server



2. In the left pane, choose **PSTN gateway** and in the Mediation Server PSTN gateway pane, select the E-SBC device that is designated as the IP/PSTN gateway (i.e., ‘E-SBC.OCSW14.local’) and click **Add** to associate it with this Mediation Server.

   Note that there are two sub-panes, one including a list of IP/PSTN gateways not associated with the Mediation Server and one including a list of IP/PSTN gateways associated with the Mediation Server.

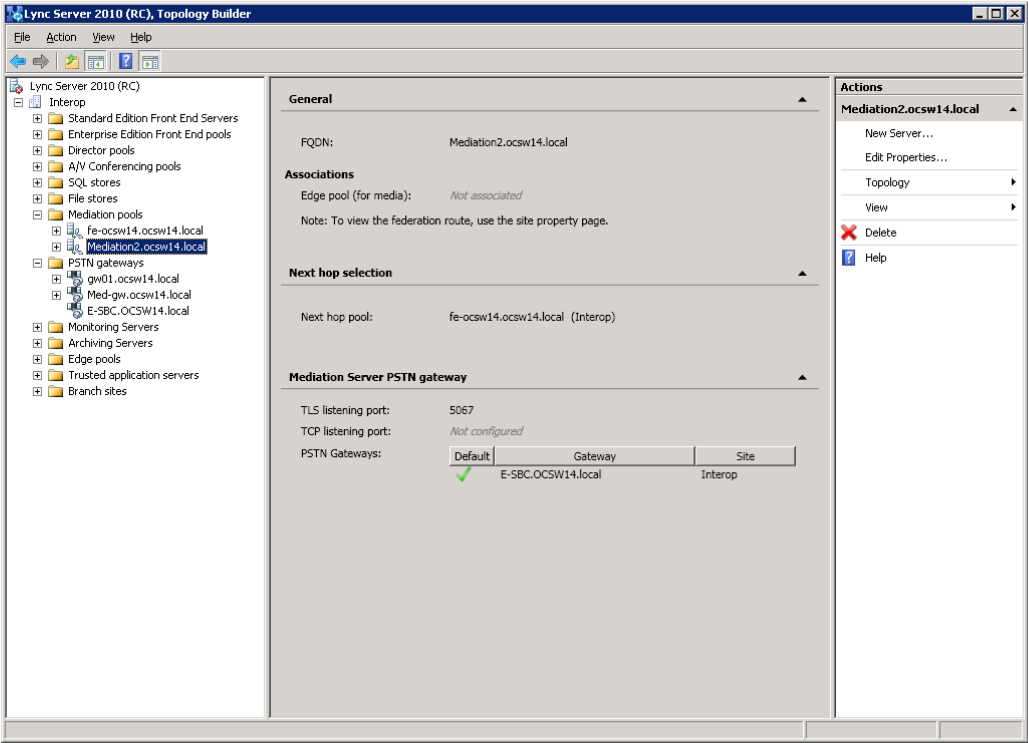
Figure 4-10: After Associating IP/PSTN Gateway to Mediation Server



In the Mediation Server PSTN gateway pane, the IP/PSTN Gateway that you associated with the Mediation Server is displayed with an adjacent Green ✓.

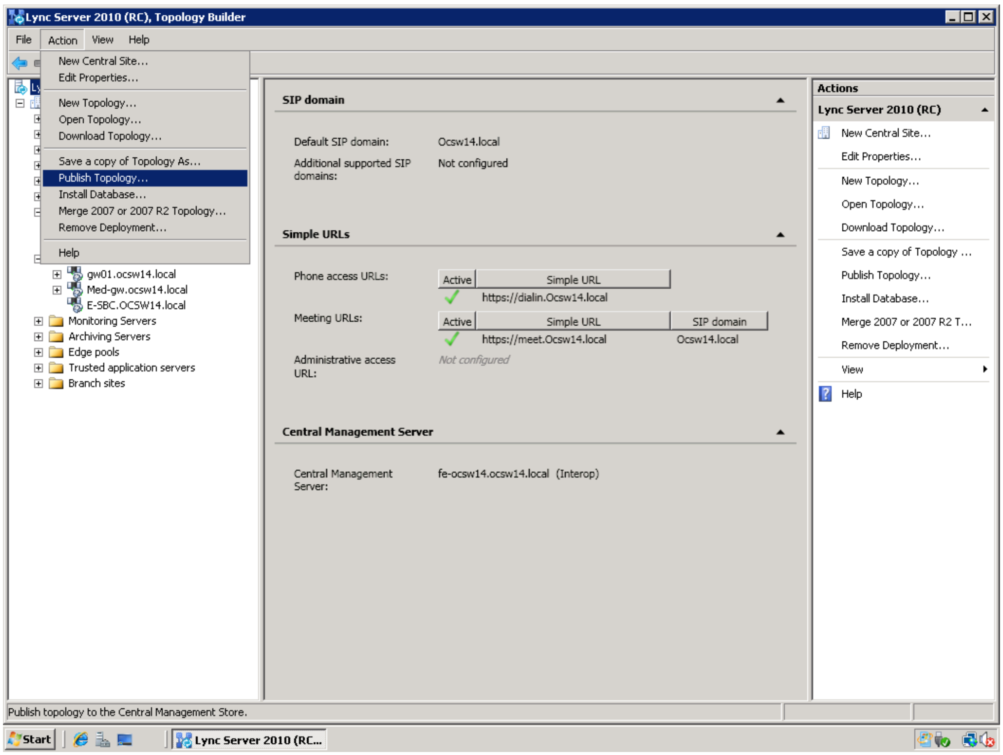
3. Click **OK**.

Figure 4-11: Media Server PSTN Gateway Association Properties



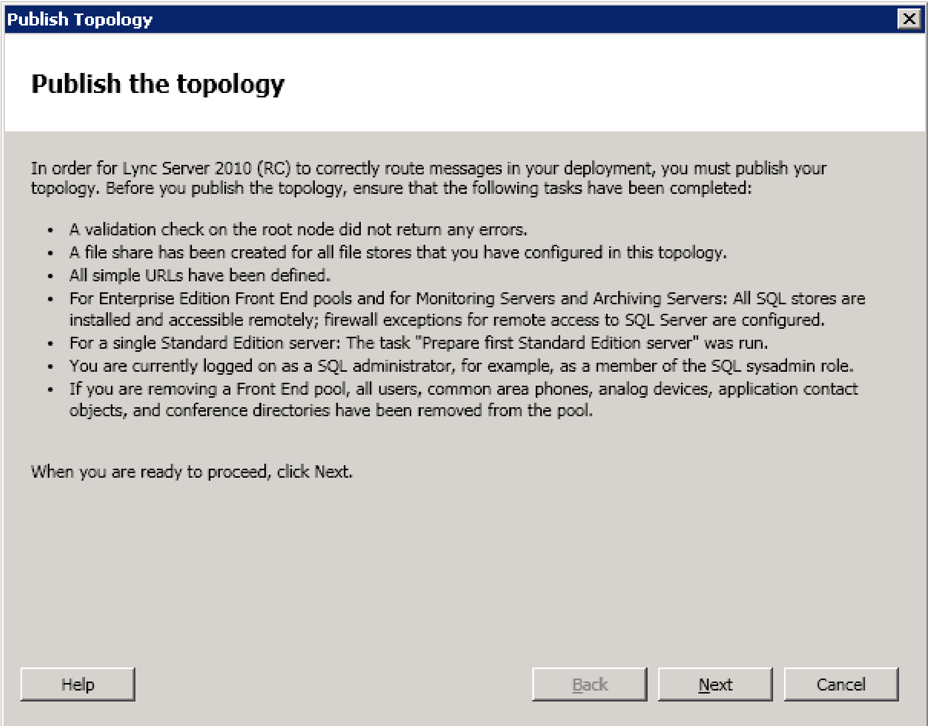
4. In the Lync Server main menu, choose **Action > Publish Topology**.

Figure 4-12: Publishing Topology



The Publish Topology screen is displayed.

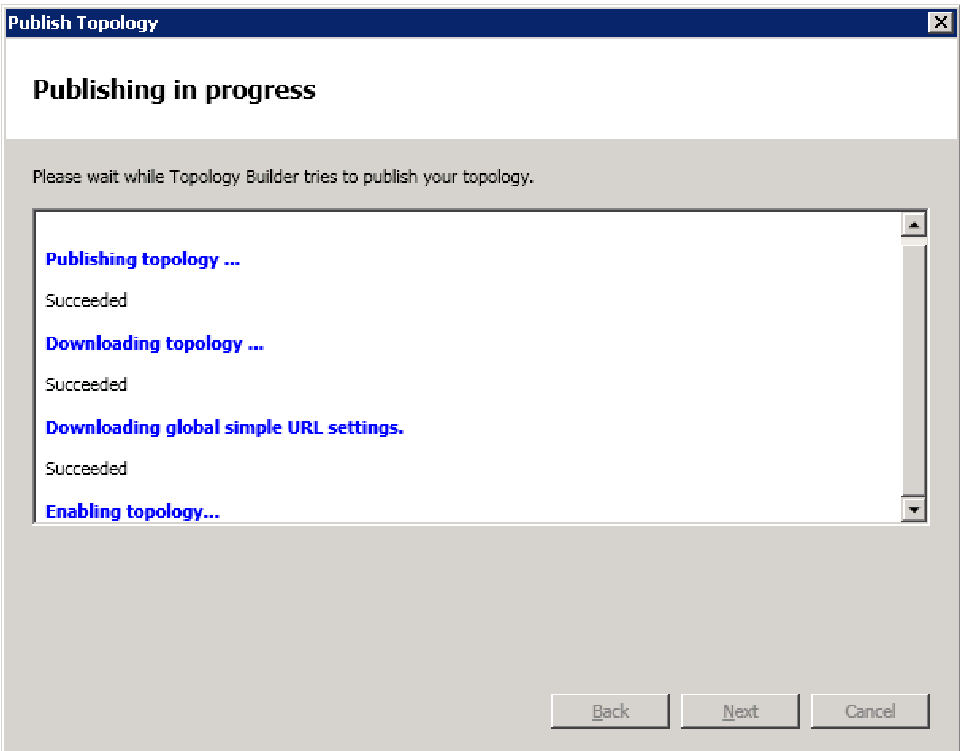
Figure 4-13: Publish Topology Confirmation



5. Click **Next**.

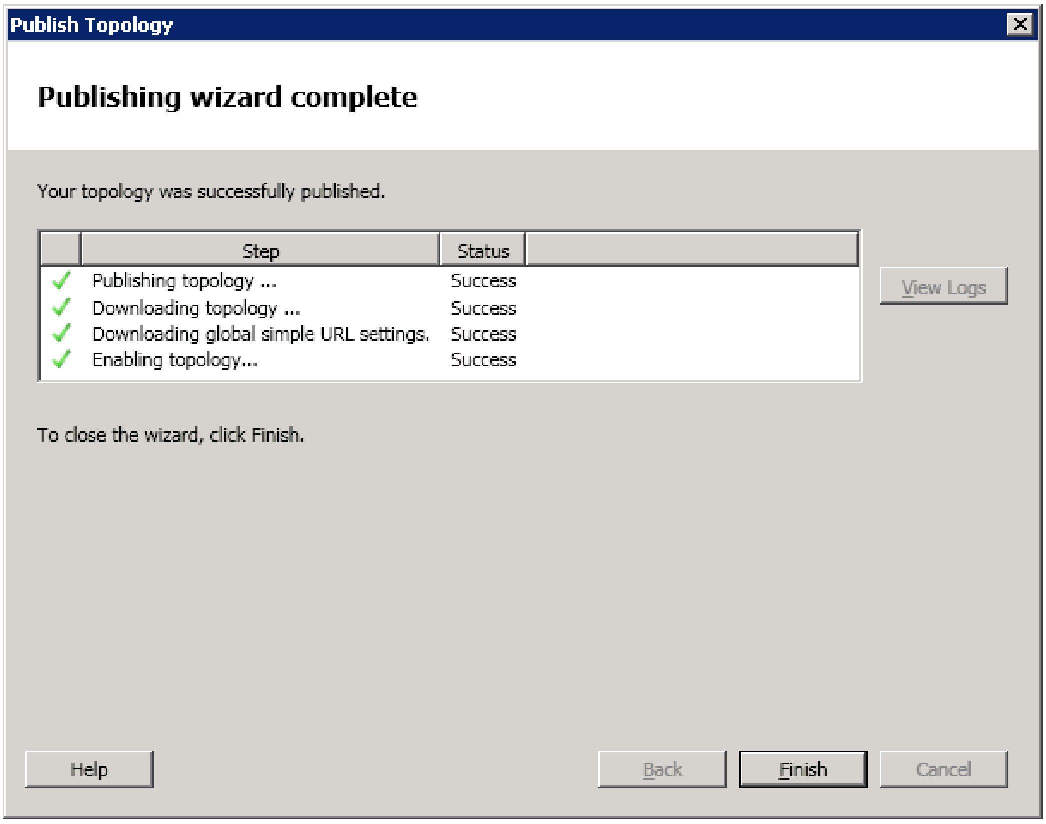
   The Topology Builder attempts to publish your topology.

Figure 4-14: Publish Topology Confirmation



Wait until the publish topology process ends successfully.

Figure 4-15: Publish Topology Successfully Completed



6. Click **Finish**.

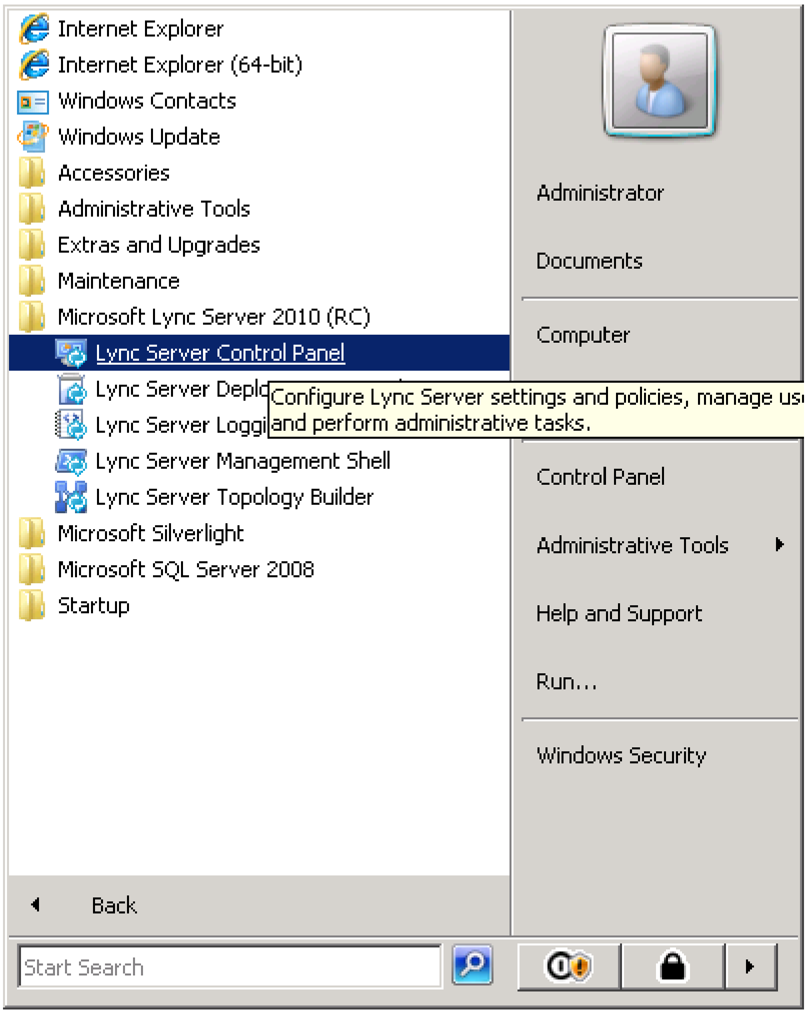
4.3 Configuring the 'Route' on the Lync Server 2010

This section describes how to configure a 'Route' on the Lync server and associates it with the IP/PSTN gateway.

➢ To configure the 'route' on the Lync server:

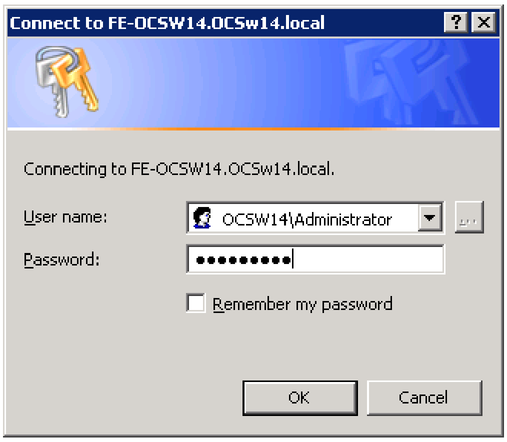
1. Open the Communication Server Control Panel (CSCP), click **Start**, select **All Programs**, and select **Lync Server Control Panel**.

Figure 4-16: Lync Server Control Panel



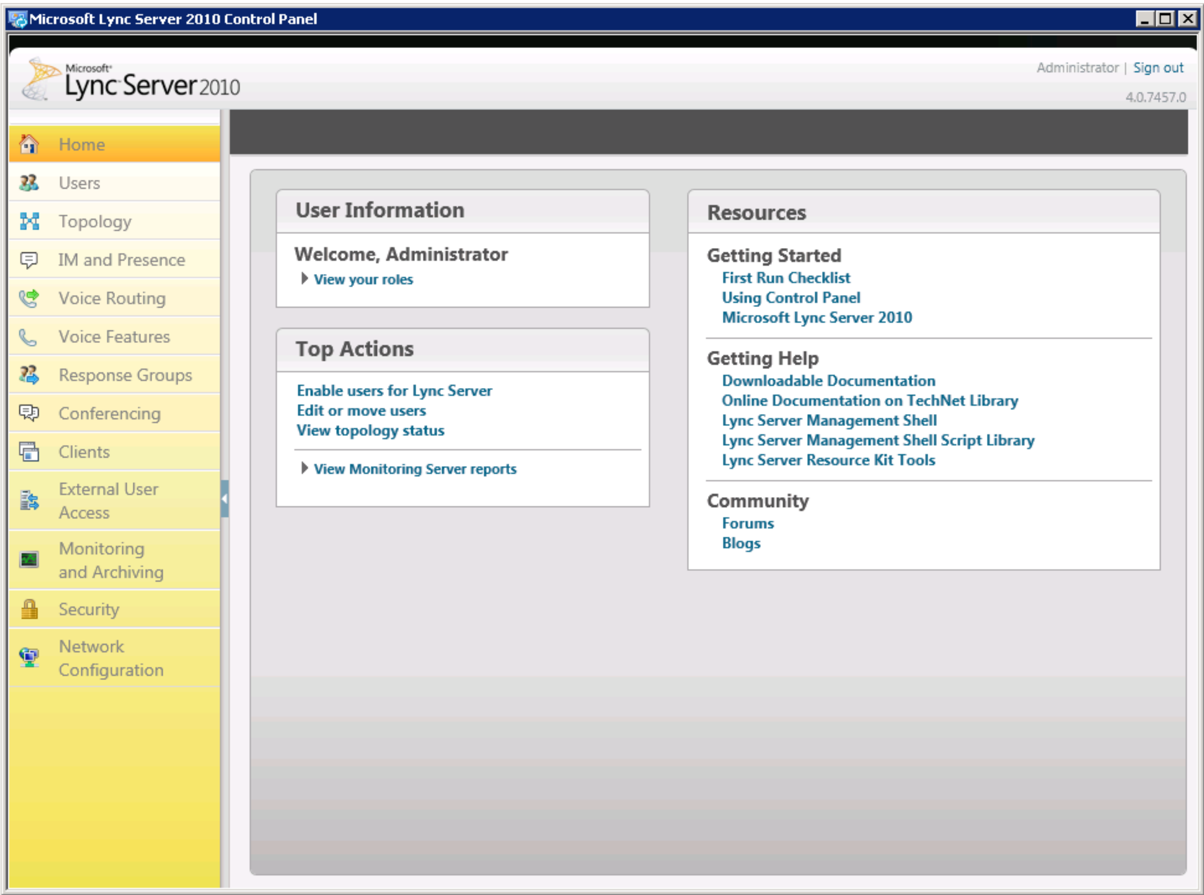
2. You are prompted for credentials; enter your domain username and password.

Figure 4-17: Lync Server Credentials



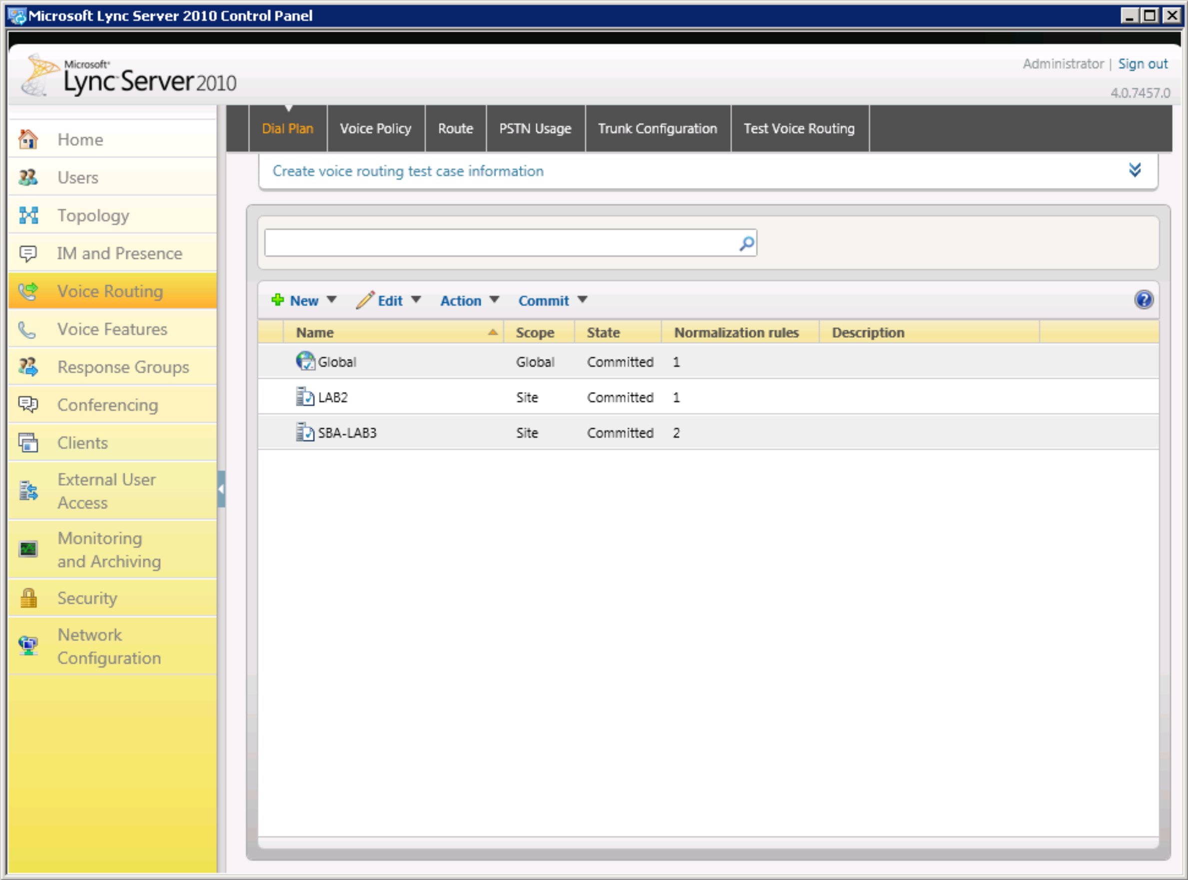
The CSCP Home page is displayed.

Figure 4-18: CSCP Home page



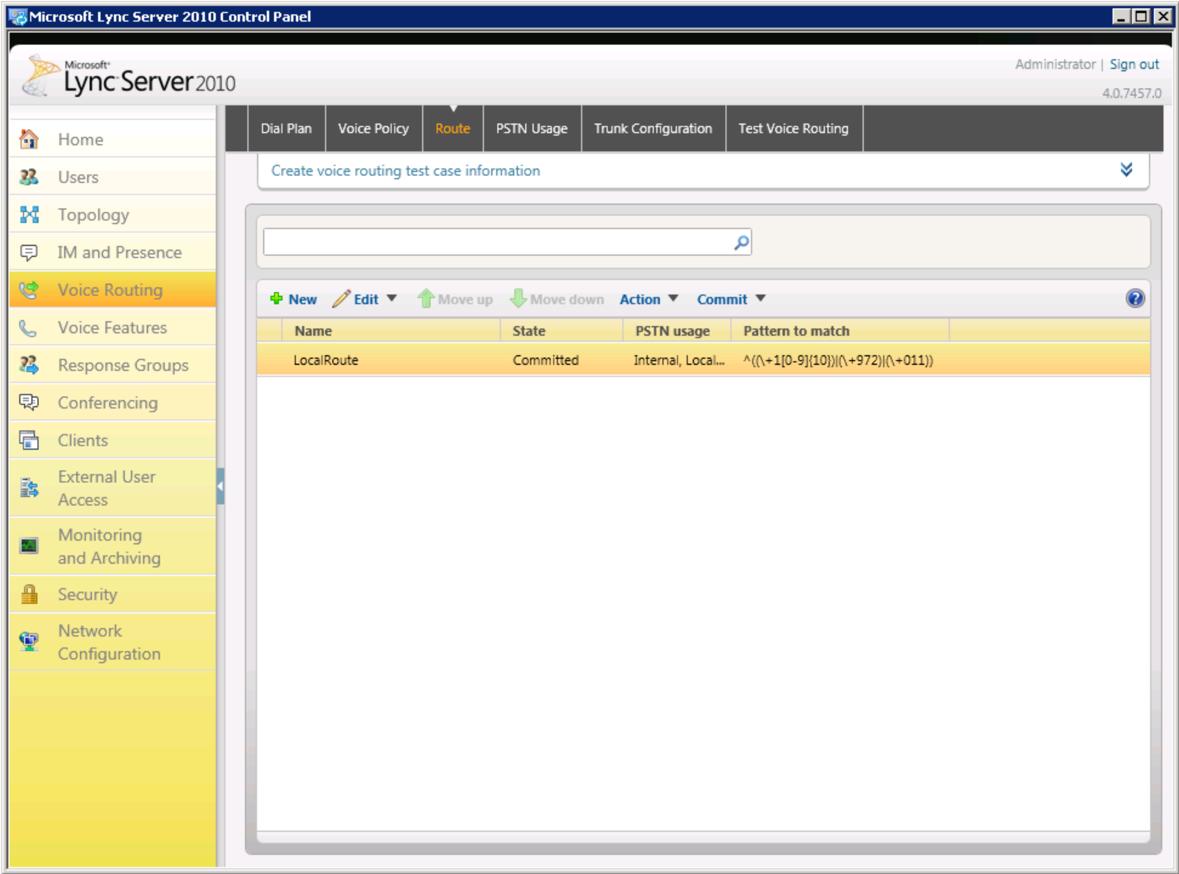
3. In the Navigation pane, select the **Voice Routing** option.

Figure 4-19: Voice Routing



4. In the Voice Routing menu at the top of the page, select the **Route** option.

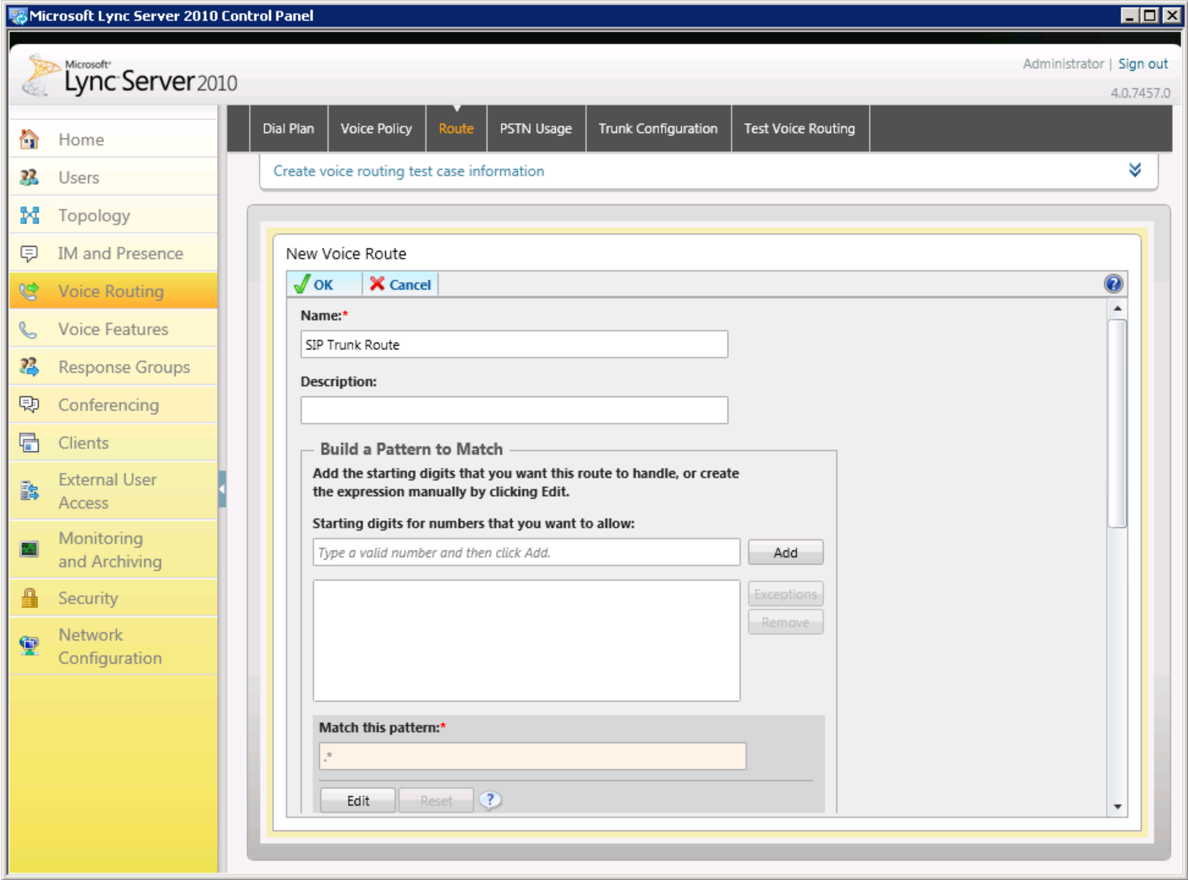
Figure 4-20: Route Option



5. In the content area toolbar, click 

.
6. In the Build a Pattern to Match pane, fill in a Name for this route (i.e., SIP Trunk Route) and a Pattern to Match for the phone numbers you want this route to handle. In this example, the pattern to match is '*', which implies "to match all numbers".
7. Click Add.

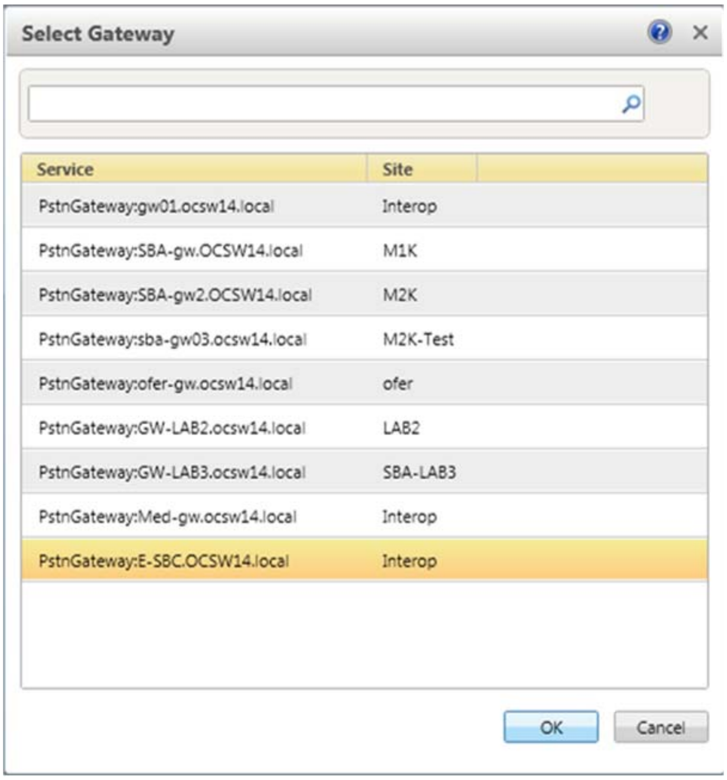
Figure 4-21: Adding New Voice Route



8. Associate the route with the IP/PSTN gateway you created above; scroll down to the Associated Gateways pane and click **Add**.

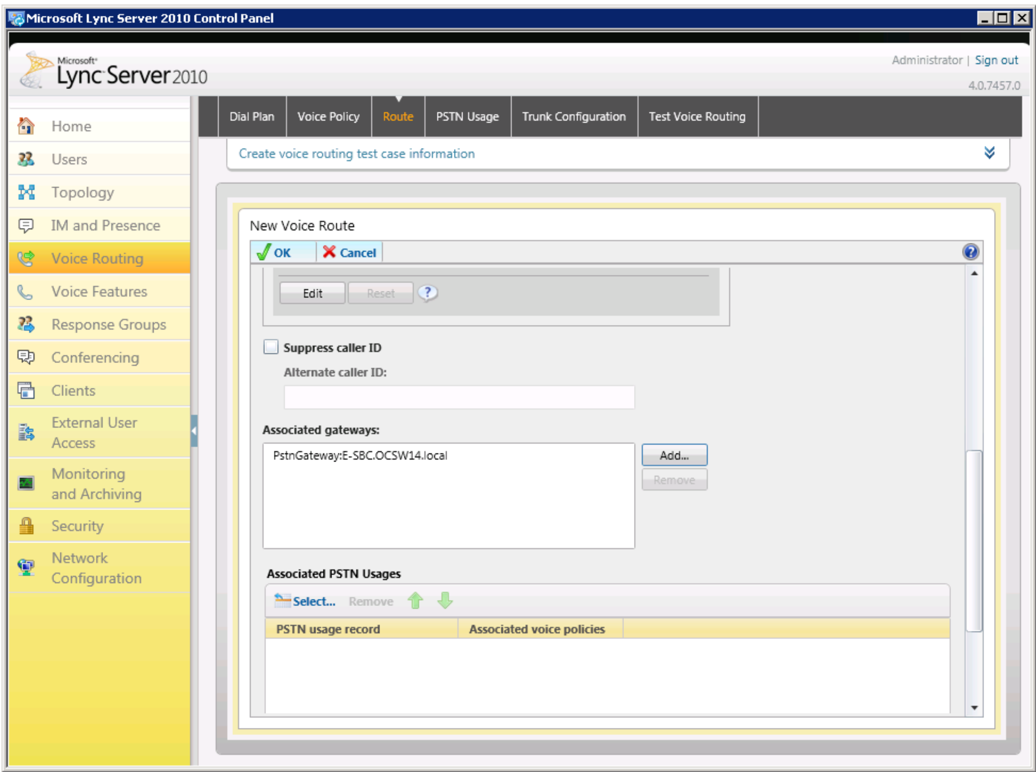
   A list of all the deployed Gateways is displayed.

Figure 4-22: List of Deployed Gateways



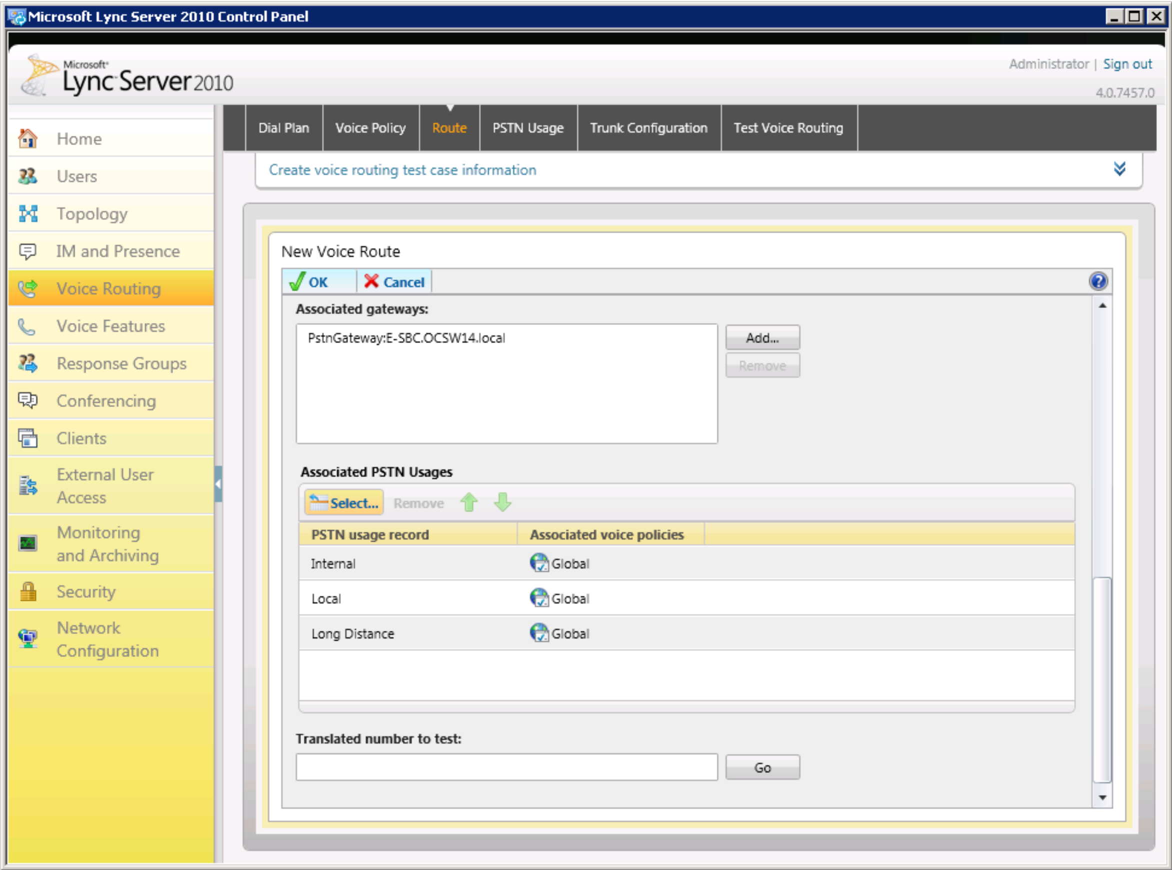
9. Select the IP/PSTN Gateway you created above and click **OK**.

Figure 4-23: Selecting the PSTN Gateway



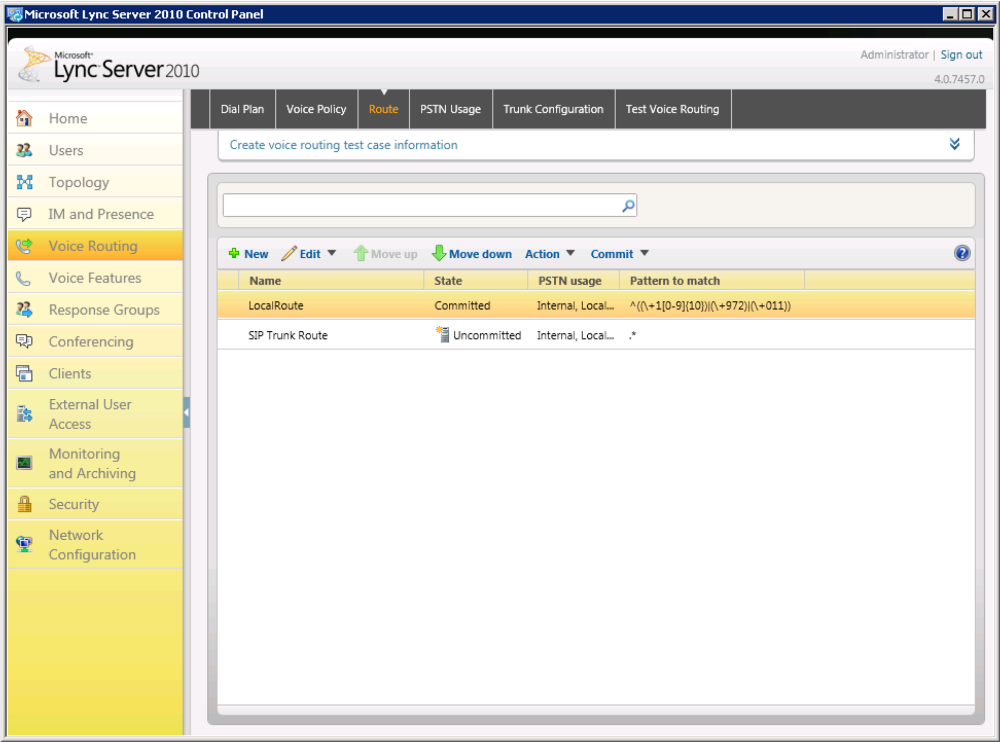
10. Associate a PSTN Usage with this route. In the **Associated PSTN Usages** toolbar, click **Select** and add the associated PSTN Usage.

Figure 4-24: Associating PSTN Usage with a PSTN Gateway



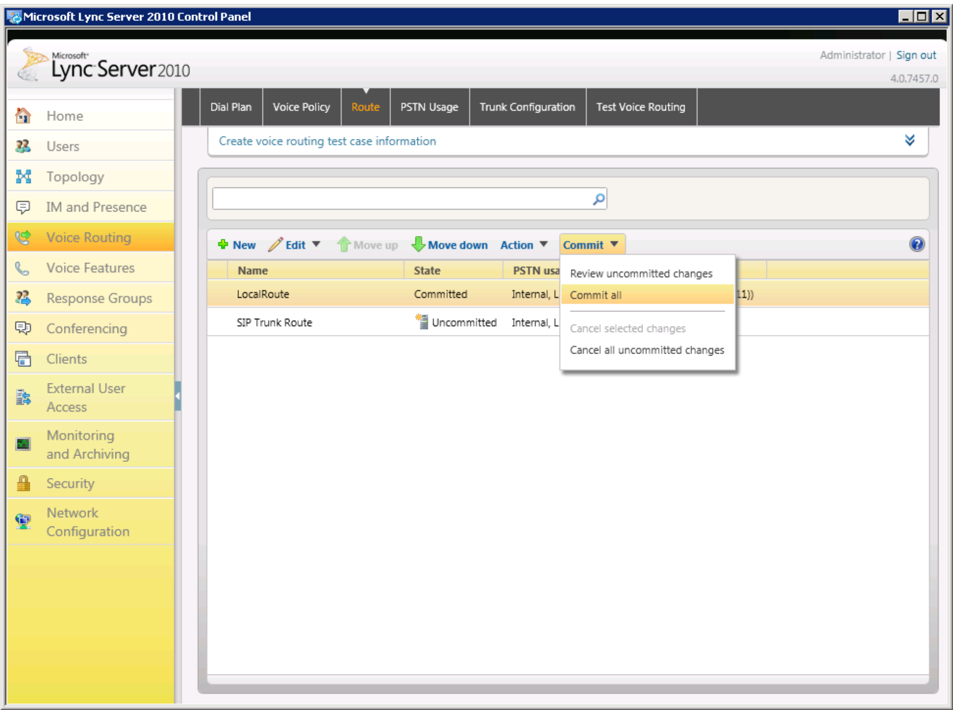
11. Click the **OK** button in the toolbar at the top of the New Voice Route pane.

Figure 4-25: Confirmation of New Voice Route



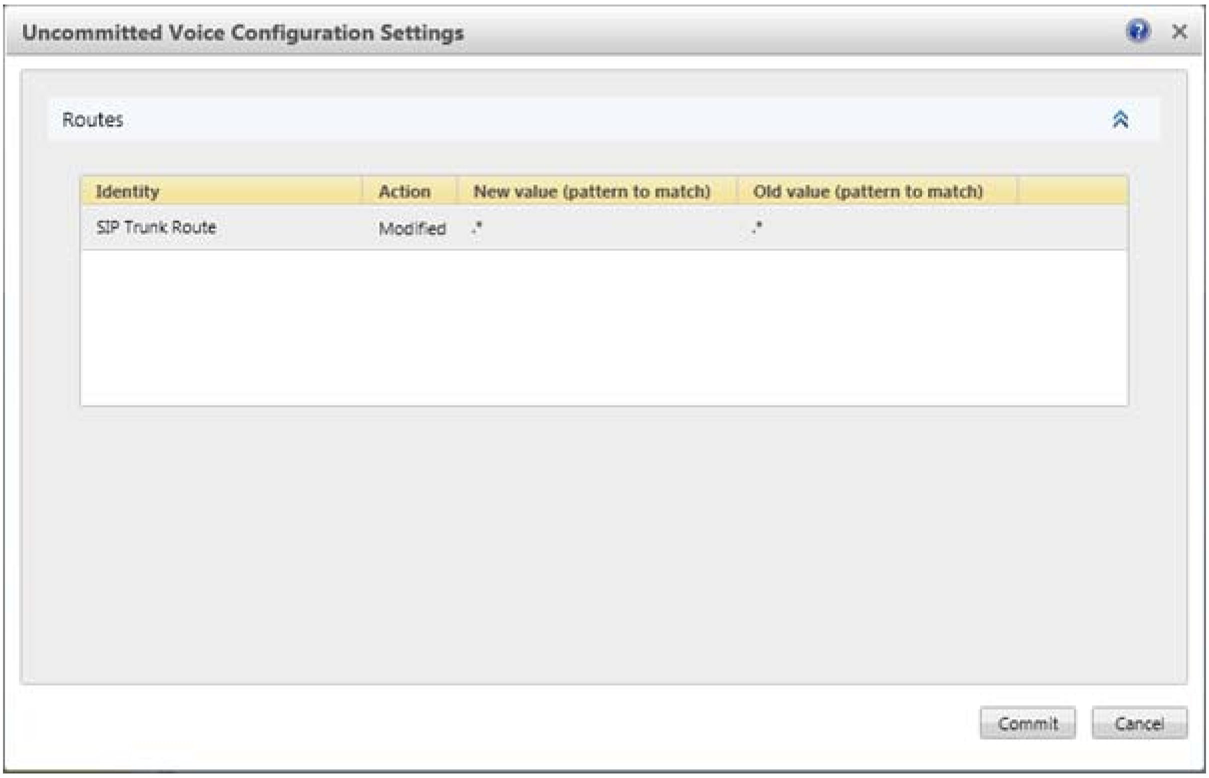
12. In the Content area toolbar, click the arrow adjacent to the **Commit** button; a drop-down menu is displayed; select the 'Commit All' option.

Figure 4-26: Committing Voice Routes



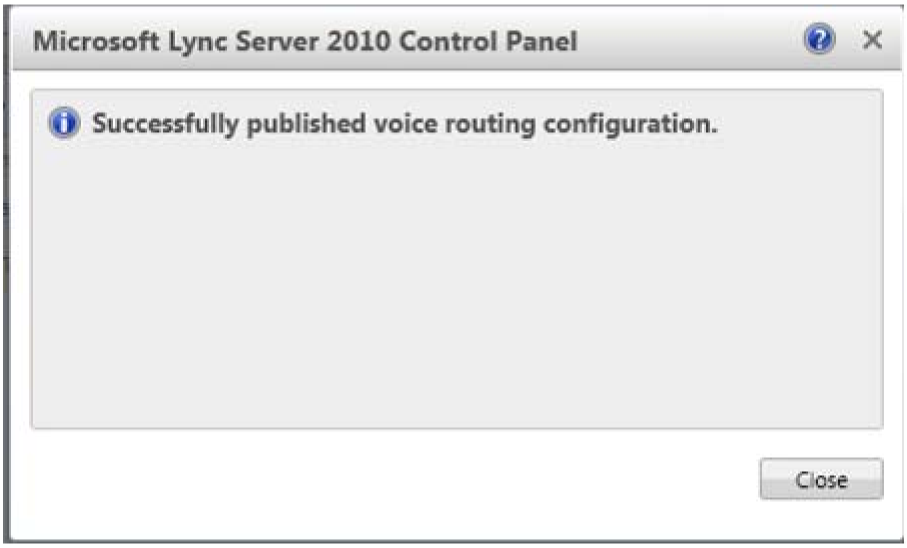
13. In the Uncommitted Voice Configuration Settings window, click **Commit**.

Figure 4-27: Uncommitted Voice Configuration Settings



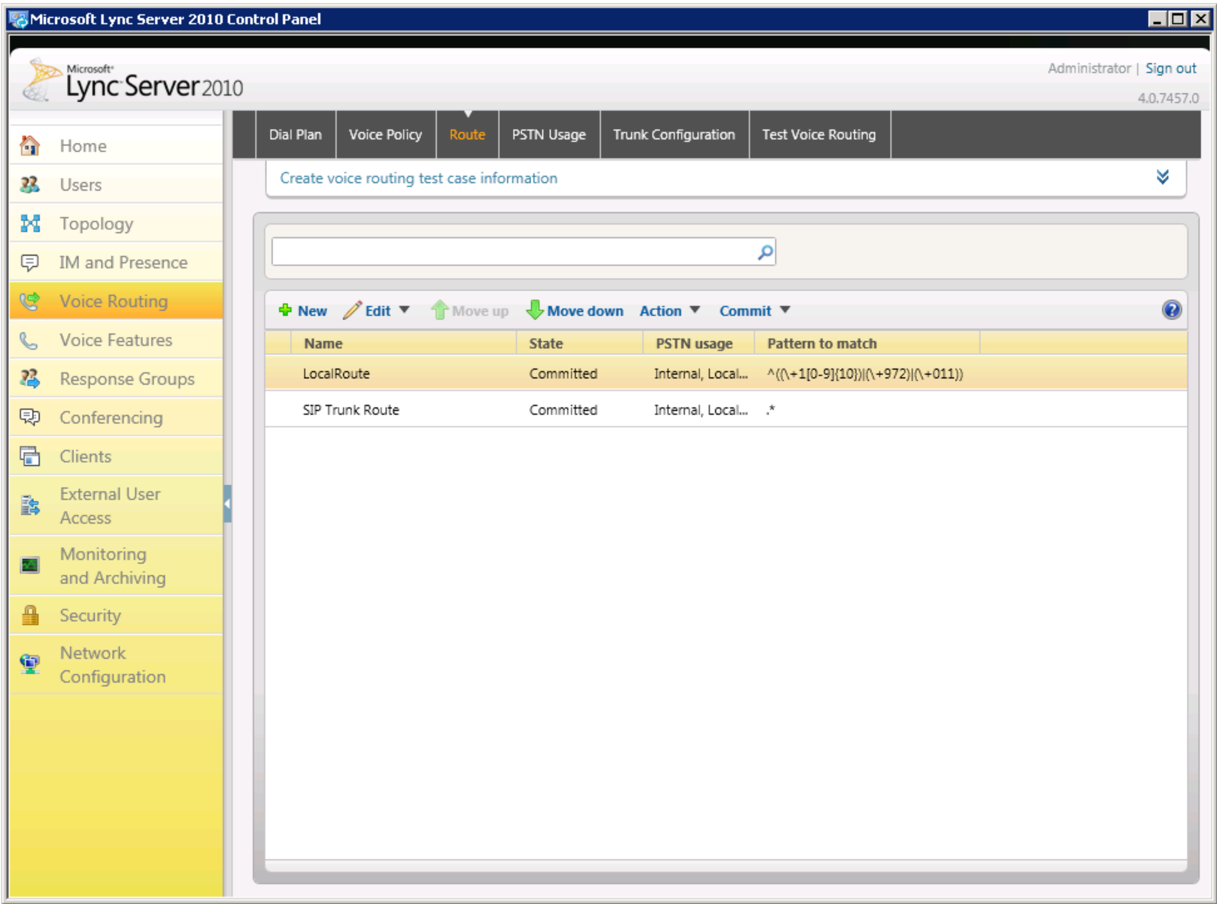
14. A message is displayed confirming a successful voice routing configuration; in the **Microsoft Lync Server 2010 Control Panel** prompt, click **Close**.

Figure 4-28: Voice Routing Configuration Confirmation



The new committed Route is now displayed in the Voice Routing screen.

Figure 4-29: Voice Routing Screen Displaying Committed Routes



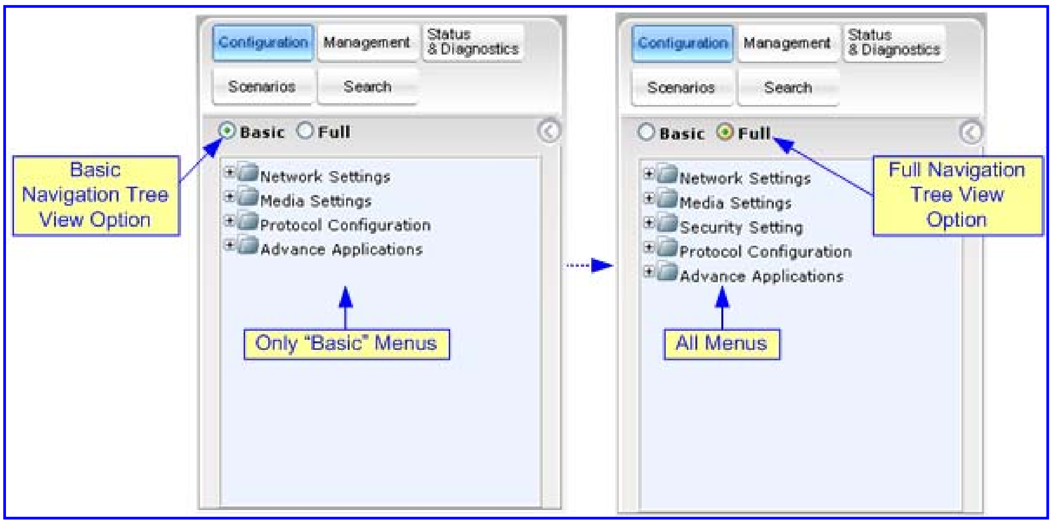
5 Configuring the E-SBC Device

This section provides a step-by-step procedure to configure the E-SBC device. These are the steps:

- **Step 1**: Configure IP Addresses - see Section 5.1 on page 38.
- **Step 2**: Enable the SBC Capabilities. See Section 5.2 on page 43.
- **Step 3**: Configure the Number of Media Channels. See Section 5.3 on page 44.
- **Step 4**: Configure the Proxy Sets. See Section 5.4 on page 45.
- **Step 5**: Configure the IP Groups. See Section 5.5 on page 47.
- **Step 6**: Configure the Voice Coders. See Section 5.6 on page 49.
- **Step 7**: Define Silence Suppression and Comfort Noise. See Section 5.6.1 on page 51.
- **Step 8**: Configure IP Profile Settings. See Section 5.7 on page 52.
- **Step 9**: Configure IP-to-IP Routing Setup. See Section 5.8 on page 54.
- **Step 10**: Configure Number Manipulation. See Section 5.9 on page 57.
- **Step 11**: Configuring IP Profile for Call Forwarding. See Section 5.10 on page 62.
- **Step 12**: Configuring SIP General Parameters. See Section 5.11 on page 65.
- **Step 13**: Configuring Supplementary Services.
- **Step 14**: Defining Reasons for Alternative Routing. See Section 5.13 on page 67.
- **Step 15**: Configuring SIP Proxy & Registration for Gateway Registration. See Section 5.14 on page 70.

The procedures described in this section are performed using the E-SBC device's' Web-based management tool (i.e., embedded Web server). Before configuring the E-SBC device, ensure that the Web interface's Navigation tree is in full menu display mode (i.e., the **Full** option on the Navigation bar is selected), as displayed below:

Figure 5-1: Web Interface Showing Basic/Full Navigation Tree Display



5.1 Step 1: Configure IP Addresses

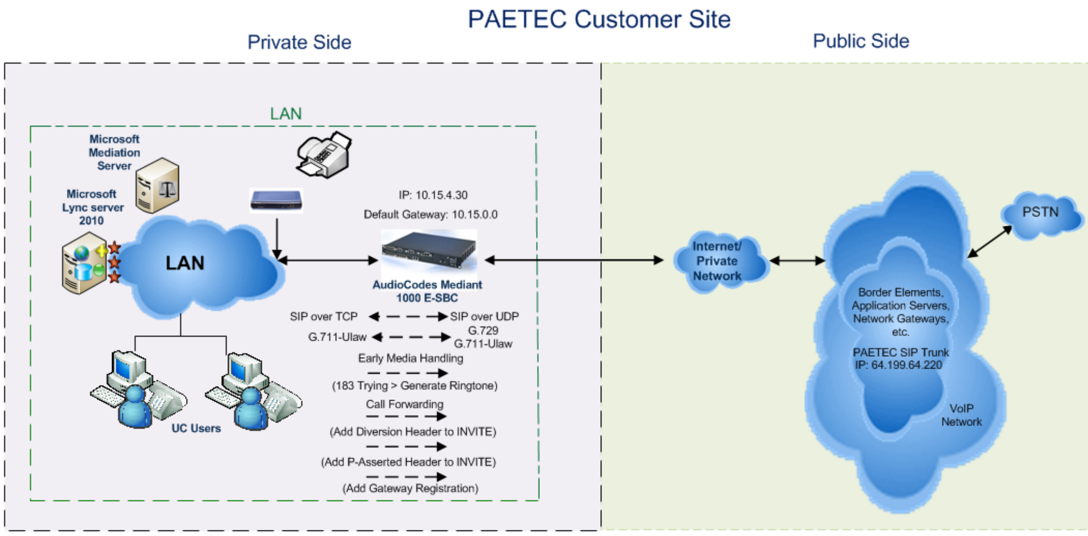
5.1.1 LAN and WAN Interface Separation

This section describes how to configure IP addresses when the internal data-routing capabilities of the MSBG device are used to connect to the PAETEC SIP Trunk. In this case, configure a separate WAN interface, as shown in the figure below.

Notes:

- The VoIP and Management interface must be in the same subnet as the data-routing interface, as shown in the figure below.
- When operating with both VoIP and data-routing functionalities, it's recommended to define the Default Gateway IP address for the VoIP network interface in the same subnet and with the same VLAN ID as the IP address for the data-routing LAN interface, as shown in the figure below.

Figure 5-2: Physical Interface Separation



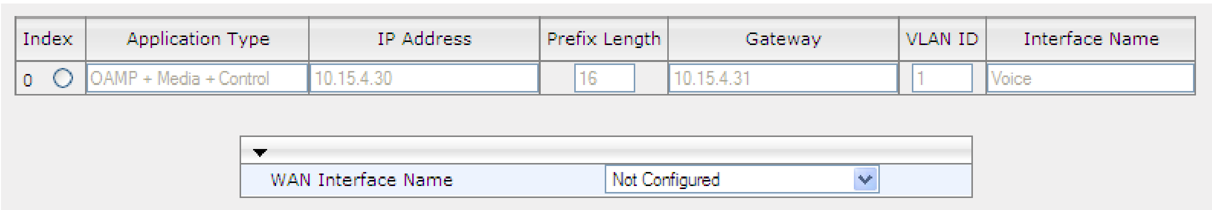
5.1.1.1 Configuring the LAN IP Addresses

This section describes how to assign the LAN IP addresses.

➢ **To assign a LAN VoIP and Management IP address using the Web interface:**

1. Open the ‘IP Settings’ page (**Configuration** tab > **VoIP** menu > **Network** sub-menu > **IP Settings**).
2. Select the 'Index' radio button corresponding to the Application Type **OAMP + Media + Control** (i.e., VoIP and management interface), and then click **Edit**.
3. Configure the new IP address and prefix length so that it corresponds to your network IP addressing scheme (e.g., 10.15.4.30).
4. Configure additional IP interfaces, if required.

Figure 5-3: Multiple Interface Table Page



5. Click **Apply**, and then **Done** to apply and validate settings. If validation fails, the MSBG device does not reboot.
6. Save your settings to flash memory and reset the MSBG device.

➢ **To define the MSBG device's LAN data-routing IP address:**

1. Access the MSBG device's Web interface with the IP address that you assigned to the VoIP and Management interface.
2. Access the 'Connections' page (**Configuration** tab > **Data** menu > **Data System** > **Connections**).

Figure 5-4: Connections Page

3. Click the **Edit** icon corresponding to the "LAN Switch VLAN 1" connection, and then click the **Settings** tab.
4. In the 'IP Address' and 'Subnet Mask' fields, enter the required IP address (e.g., 10.15.4.31) and subnet respectively, and then click **OK**.

Figure 5-5: Defining LAN Data-Routing IP Address

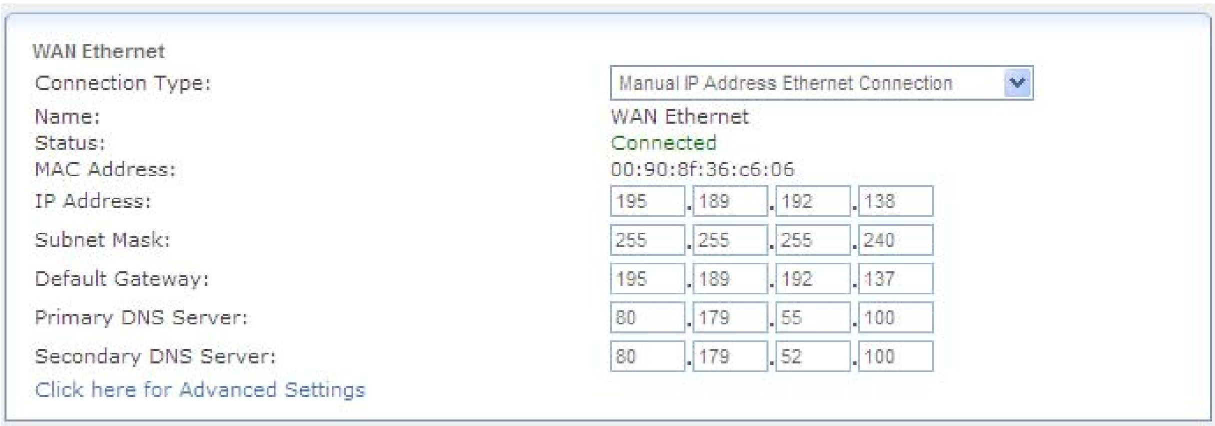
5.1.1.2 Assigning WAN IP Addresses

This section describes how to assign the WAN IP addresses.

➢ **To assign a WAN IP address:**

1. Cable the MSBG device to the WAN network (i.e., ADSL or Cable modem), using the WAN port.
2. Access the MSBG device's Web interface with the Voice and Management IP address.
3. Access the 'Settings' page (**Configuration** tab > **Data** menu > **WAN Access** > **Settings** tab).

Figure 5-6: Configuring the WAN IP Address

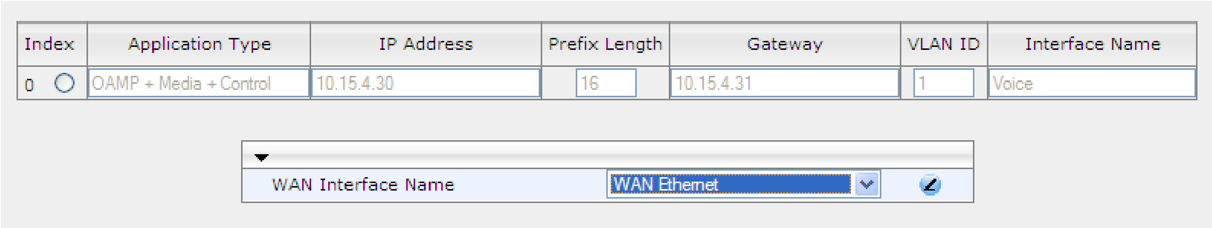


4. From the 'Connection Type' drop-down list, select the required connection type for the WAN, and then configure the IP address (e.g., 195.189.192.138).

➢ **To assign a WAN interface for VoIP traffic:**

1. Select the WAN interface.
2. Open the 'Multiple Interface Table' page (**Configuration** tab > **VoIP** menu > **Network** sub-menu > **IP Settings**).

Figure 5-7: Selecting WAN Interface for VoIP Traffic in Multiple Interface Table Page



3. From the 'WAN Interface Name' drop-down list, select the WAN interface for VoIP traffic.
4. Click **Done** and then reset the MSBG device for your setting to take effect.

5.1.2 LAN Only Interface

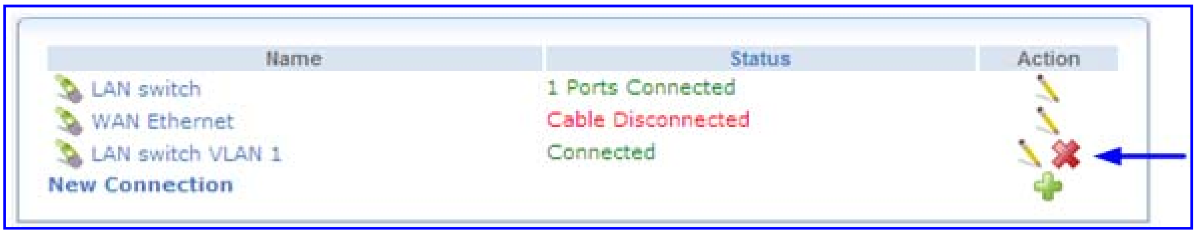
This section describes how to configure IP addresses when a single LAN interface is used to connect to the PAETEC SIP Trunk. In this configuration, the internal data-routing capabilities of the E-SBC device are not used. Consequently, disable the internal data-routing interface as described in the procedure below.

Note: When operating in LAN VoIP-only mode, do not use the E-SBC device's WAN port.

➢ **To operate the E-SBC device as a LAN VoIP gateway only:**

1. Disconnect the network cable from the WAN port and then connect one of the E-SBC device's LAN ports to the network.
2. Disable or remove the data-routing IP network interface:
   - Access the 'Connections' page (Configuration tab > Data menu > Data System > Connections).
   - Delete the "LAN Switch VLAN 1" connection by clicking the corresponding Remove button, and then clicking OK to confirm deletion.

Figure 5-8: Removing Data-Routing Connection Interface



3. Configure VoIP IP network interfaces in the 'Multiple Interface' table (**Configuration** tab > **VoIP** menu > **Network** > **IP Settings**).
   - In the 'Multiple Interface' table, define a single IP network interface for application types "OAMP + Media + Control".

Figure 5-9: Multiple Interface Table

Index	Application Type	IP Address	Prefix Length	Gateway	VLAN ID	Interface Name	Primary DNS Server IP Address	Secondary DN IP Addre
0	OAMP + Media + Control	10.15.9.118	16	10.15.0.0	1	Voice	0.0.0.0	0.0.0.0

- Click **OK** to save settings.

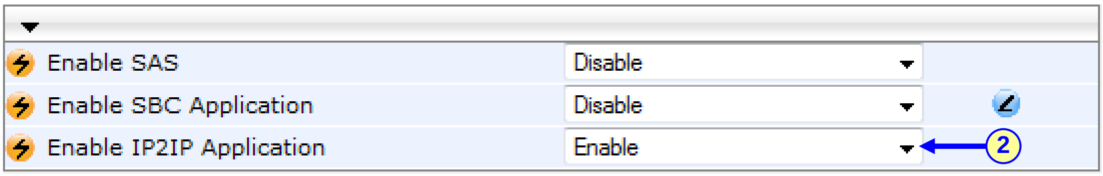
5.2 Step 2: Enable the SIP SBC Application Mode

This step describes how to enable the gateway-SBC devices' SIP SBC application mode.

➢ **To enable the SIP SBC application mode:**

1. Open the 'Application Enabling' page (**Configuration** tab > **VoIP** menu > **Applications Enabling** > **Applications Enabling**).

Figure 5-10: Application Enabling



2. From the 'Enable IP2IP Application' drop-down list, select "Enable".

   Reset with BURN to FLASH is required.

Note: To enable IP2IP capabilities on the AudioCodes E-SBC device, your device must be loaded with the feature key that includes the IP2IP feature and must be running SIP version 6.2 or later.

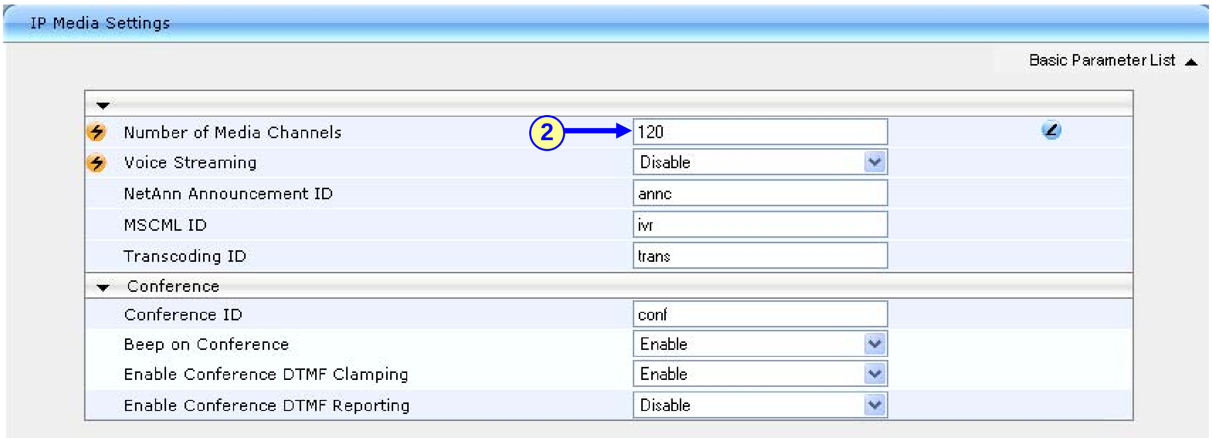
5.3 Step 3: Configure the Number of Media Channels

To perform the coder transcoding capabilities, define DSP channels. The number of media channels represents the number of digital signaling processors (DSP) channels that the device allocates to IP-to-IP calls (the remaining DSP channels can be used for PSTN calls). Two IP media channels are used per IP-to-IP call. The maximum number of media channels available on the Mediant 1000 E-SBC device is 120 (i.e., up to 60 IP-to-IP calls). The maximum number of media channels available on the Mediant 3000 E-SBC Media Gateway device is 2016 (i.e., up to 1008 IP-to-IP calls).

➢ **To configure the number of media channels:**

1. Open the 'IP Media Settings' page (**Configuration** tab > **VoIP** menu > **IP Media** > **IP Media Settings**).

Figure 5-11: IP Media Channels Settings



2. In the 'Number of Media Channels’, enter “120” to enable up to 60 IP-to-IP calls with transcoding. Click **Apply New Value**.

5.4 Step 4: Configure the Proxy Sets

This step describes how to configure the Proxy Sets. The Proxy Sets represent the IP addresses (or FQDN), which are required for communicating with the entities in the network:

- Proxy Set ID #1 is assigned with the IP address of PAETEC SIP Trunk.
- Proxy Set ID #2 is assigned with the IP address of Lync Mediation server.

These Proxy Sets are later assigned to IP Groups (see Section 5.5 on page 47).

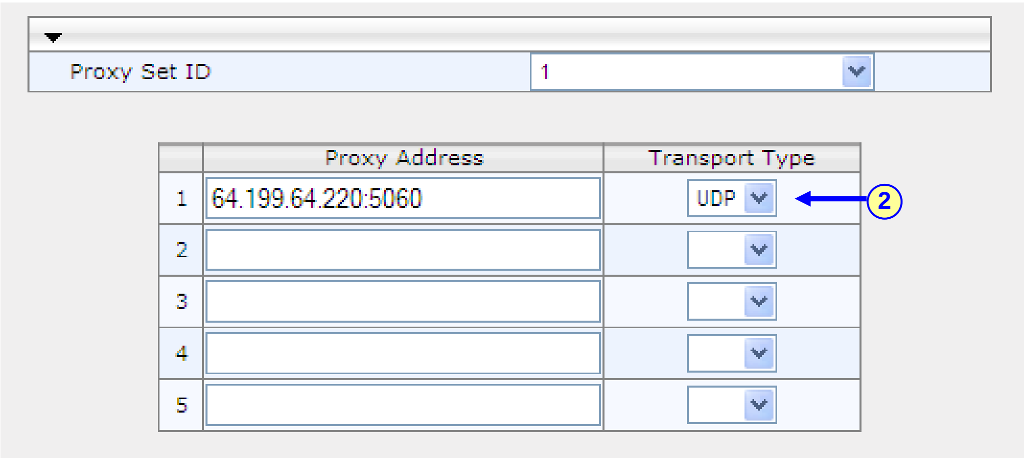
➢ **To configure proxy sets:**

1. Open the 'Proxy Sets Table' page (**Configuration** tab > **VoIP** menu > **Control Network** > **Proxy Sets Table**).
2. Configure the Proxy Set for PAETEC SIP Trunk:

   From the 'Proxy Set ID' drop-down list, select "1".

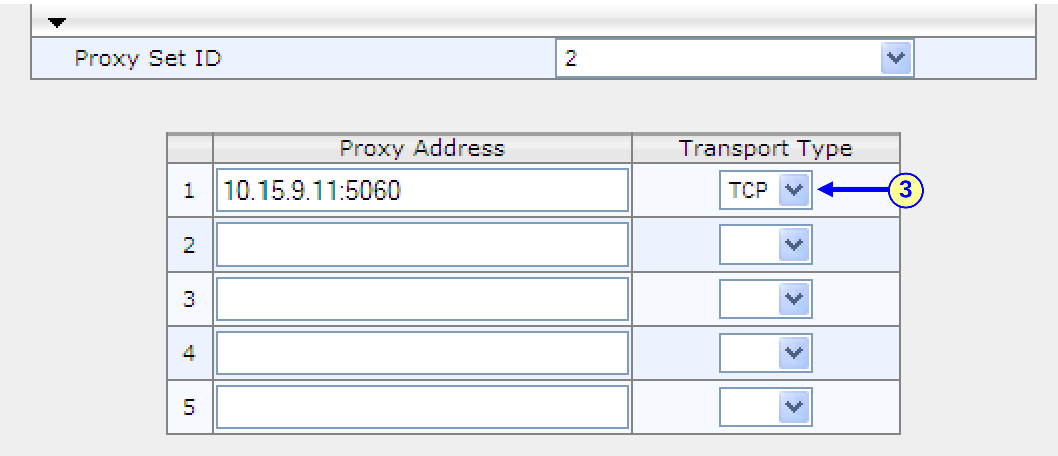
   a. In the 'Proxy Address' column, enter the IP address or FQDN of the PAETEC SIP Trunk and the listening port of the PAETEC SIP Trunk.
   b. From the 'Transport Type' drop-down list, corresponding to the IP address entered above, select "UDP".
   c. Repeat steps ‘a’ and ‘b’ as required for alternate PAETEC IP Border Element (if used).

Figure 5-12: Proxy Set ID 1 for PAETEC SIP Trunk



3. Configure the Proxy Set for the Lync Mediation Server:
   a. From the 'Proxy Set ID' drop-down list, select "2".
   b. In the 'Proxy Address' column, enter the IP address or the FQDN and the listening port of the Lync Mediation Server.
   c. From the 'Transport Type' drop-down list corresponding to the IP address entered above, select "TCP" Transport Type.

Figure 5-13: Proxy Set ID 2 for Lync Mediation Server



5.5 Step 5: Configure the IP Groups

This step describes how to create IP groups. Each IP group represents a SIP entity in the device's network. Create IP groups for:

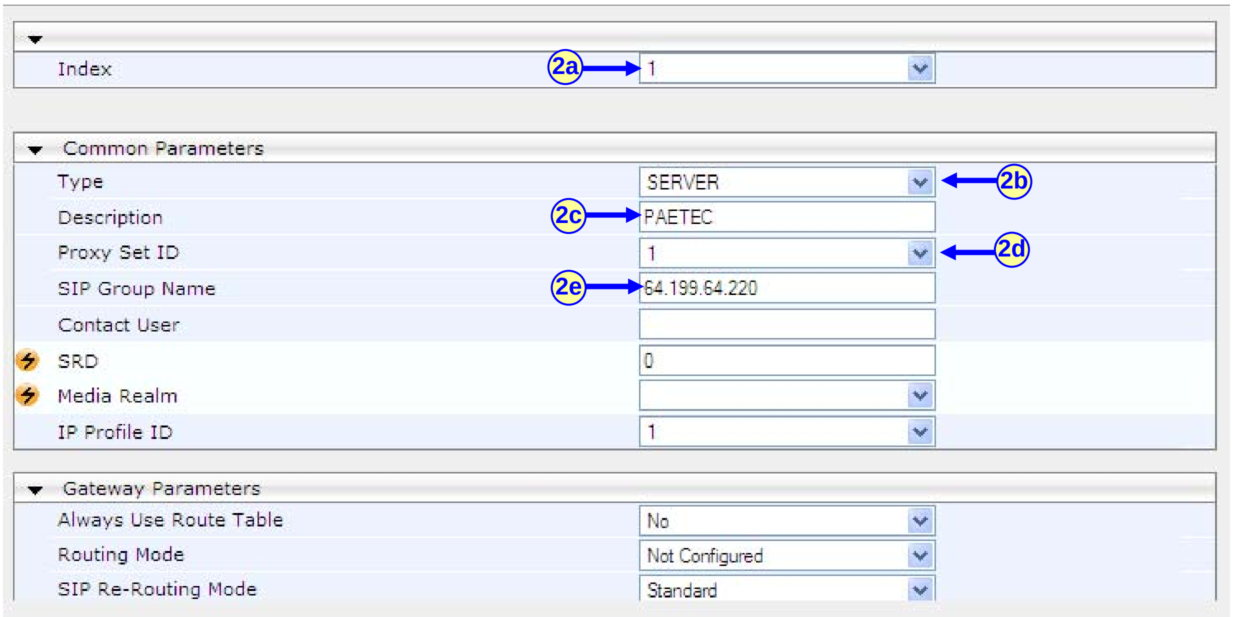
1. PAETEC SIP Trunk
2. Lync Server 2010 – Mediation Server

These IP groups are later used by the IP2IP application for routing calls.

➢ To configure IP Groups:

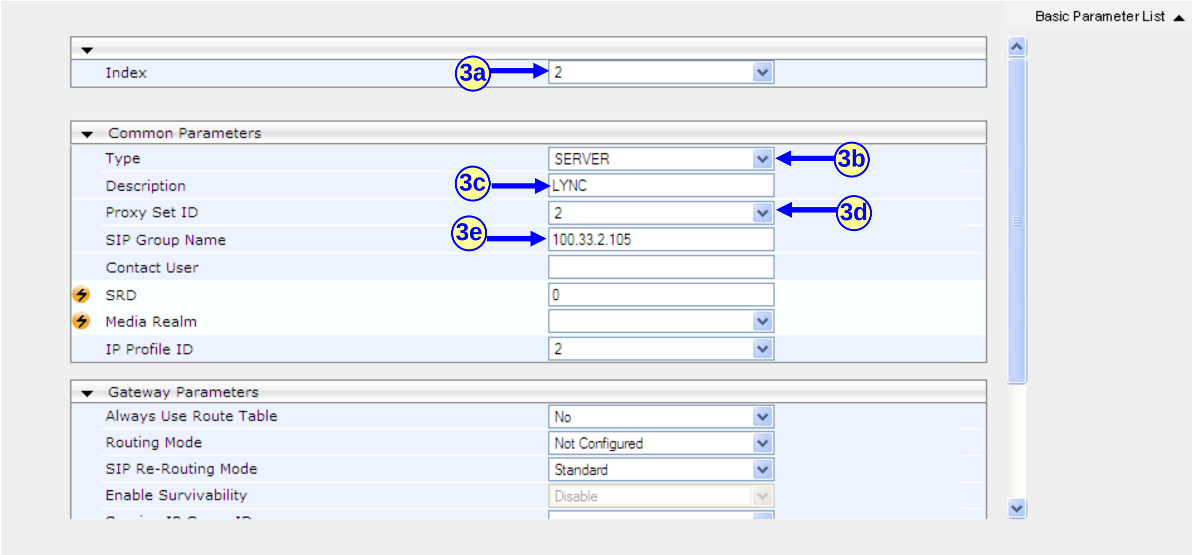
1. Open the 'IP Group Table' page (**Configuration** tab > **VoIP** menu > **Control Network**> **IP Group Table**).

Figure 5-14: IP Group – for PAETEC Server



2. Define an IP Group for the PAETEC SIP Trunk as follows:
   a. IP Group Index ‘1’
   b. Type: “SERVER”
   c. Description: arbitrary name. (e.g., “PAETEC”)
   d. Proxy Set ID: “1” (represents the IP address, configured in Section 5.4 on page 45, for communicating with this IP Group).
   e. SIP Group Name: The SIP Request-URI host name used in INVITE messages sent to the IP Group, or the host name in the From header of INVITE messages received from the IP Group. Enter the WAN IP address.

Figure 5-15: IP Group - for LYNC Mediation Server



3. Define an IP Group for the LYNC Mediation Server as follows:
   a. Select IP Group Index ‘2’:
   b. **Type:** “SERVER”
   c. **Description:** <Free Description> (e.g., “Lync Mediation Server”)
   d. Proxy Set ID: “2”
   e. **SIP Group Name:** The SIP Request-URI host name used in INVITE messages sent to the IP Group, or the host name in the From header of INVITE messages received from the IP Group. Enter the Gateway Name.

5.6 Step 6: Configure the Voice Coders

Since the LYNC Mediation Server supports both the G.711A-law and G.711U-law voice coders, while the PAETEC SIP Trunk requires the G.711U-law coder, you can configure a single coder table reference for both services by utilizing the G711U-law coder or you can create a more dynamic servicing interworking based on commonality of supported vocoders via the default Coders and Coders Group tables.

The Coder table and Coders Group table are associated and referenced within each IP Profile index. Both IP Profile indices 2 & 3 referenced in this document, reference 'Default Coder Table' which is associated with the routing tables. IP Profile index 1 referenced in this document, references "Coder Group 1"

The Coder Group table is associated within an IP Profile. Both IP Profile indices 2 & 3 referenced in this document, reference 'Default Coder Table', which is associated with the IP Groups 1 and 3 respectively. IP Profile index 1 referenced in this document, references 'Coder Group 1'.

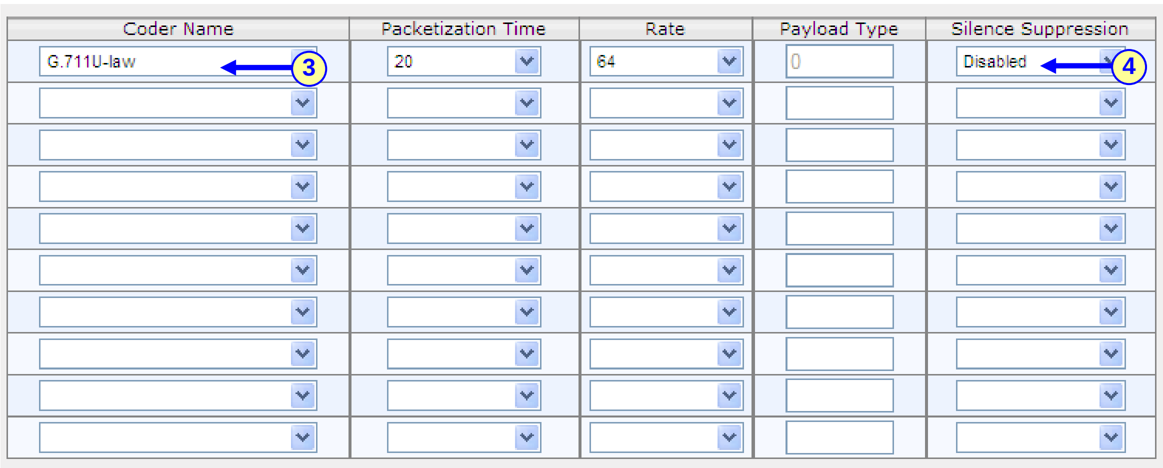
- IP Profile index 2 and 3 reference **Coder Group** setting 'Default Coder Group' which is associated with the IP Groups 1 and 3.
- IP Profile index 1 references **Coder Group** setting 'Coder Group 1', which is associated with IP Group 2.

The referenced usage of the IP Profiles is based on the routing tables for **IP to Trunk Group Routing** and **Tel to IP Routing**. Within the routing settings it will pick up the attributes for the vocoder usage based on the respective setting for the IP Profile used based on its respective **Coder Group** setting for the respective leg of the call session.

➢ To configure the Coder Table for LYNC Mediation Server and PAETEC SIP Trunk:

1. Open the 'Coders Table' page (**Configuration** tab > **VoIP** menu > **Coders And Profiles** > **Coders**).

Figure 5-16: Coder Group Table - Mediation Server



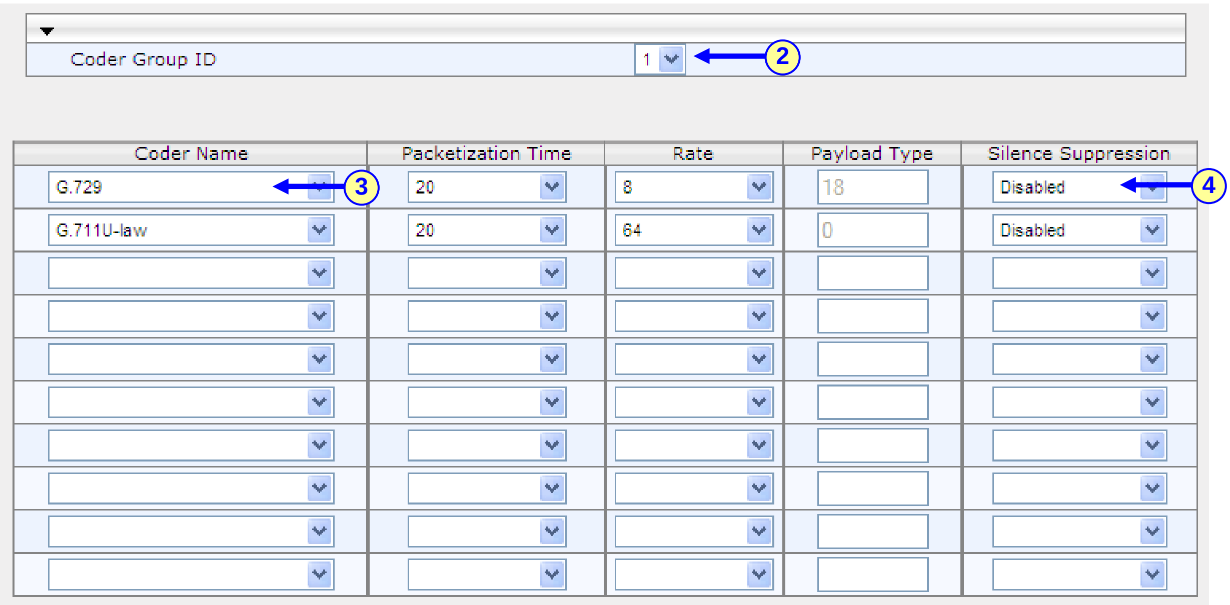
2. From the 'Coders Table' prepare to select via drop-down list, coder and attributes.
3. Select the G.711U-law coder, as shown in Figure 5-10.

4. From ‘Silence Suppression’ drop-down list, select ‘Enable’ or ‘Disabled’ as shown in Figure 5-10.
5. This is now the board default coder table. It is referenced by the datafill of parameter **Coder Group** setting **'Default Coder Group**' of IP Profile index **2** and **3**. This allows a user to list the allowed vocoders in a supported group to be referenced and utilized. This points to table 'Coder Group' for IP Profile index 2 and 3, where the usage of the Default Coder Group is explicitly referenced. As shown above, Coders Table is declared to support G.729 and G.711U-law.

➢ To configure the Coder Table for PAETEC SIP Trunk usage:

1. Open the 'Coders Table' page (**Configuration** tab > **VoIP** menu > **Coders And Profiles** > **Coders Group Settings**).

Figure 5-17: Coder Group Table 1 – PAETEC SIP Trunk



2. From the ‘Coder Group Setting’ table, select via drop-down list, index **1**. This index, **1** is referenced by the datafill of parameter 'Coders Group index' of IP Profile index **1**. This allows a user to list the allowed vocoders in a supported group to be referenced and utilized. This points to table 'Coder Group' for IP Profile index 1, where Coder Group 1 is explicitly referenced. As shown above, Coder Group 1 is declared to support G.729 and G.711U-law.
3. Select the **G.729** and **G.711U-law** coder, in the specific order as shown in the figure above. This will be the preference as advertised in the SDP for proper ‘Offer/Answer’ interworking.
4. From the ‘Silence Suppression’ drop-down list, select **Enabled** or **Disabled**.

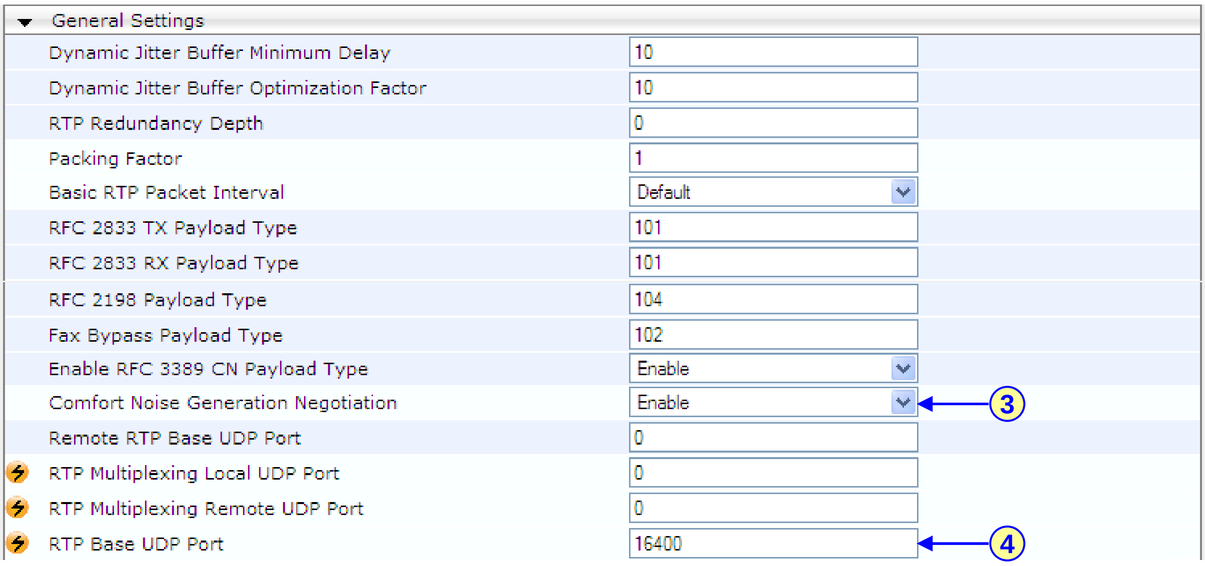
5.6.1 Step 7: Define Silence Suppression and Comfort Noise

Overall voice quality has been significantly improved for the Microsoft Lync 2010 environment. These improvements include suppression of typing noise during calls and improved generation of 'comfort noise' which reduces hissing and smoothes over the discontinuous flow of audio packets. You may need to modify the Silence Suppression and Comfort Noise parameters to achieve this goal. Note that the Echo Canceller is enabled by default.

➢ **To configure silence suppression parameters:**

1. Silence Suppression is configured per coder type (see Section 5.6 on page 49 to enable Silence Suppression per coder.)
2. Open the 'RTP/RTCP Settings' page (**Configuration** tab > **Media** menu > **RTP / RTCP Settings**).

Figure 5-18: RTP/RTCP Settings Page



3. From the 'Comfort Noise Generation Negotiation' drop-down list, select 'Enable'. This action enables negotiation and usage of Comfort Noise (CN).
4. Default 'RTP Base UDP Port' is '6000'. As seen here, it is set to '16400' to demonstrate that the RTP port range can be changed to support specific network requirements for managed traffic.
5. Click **Submit**.

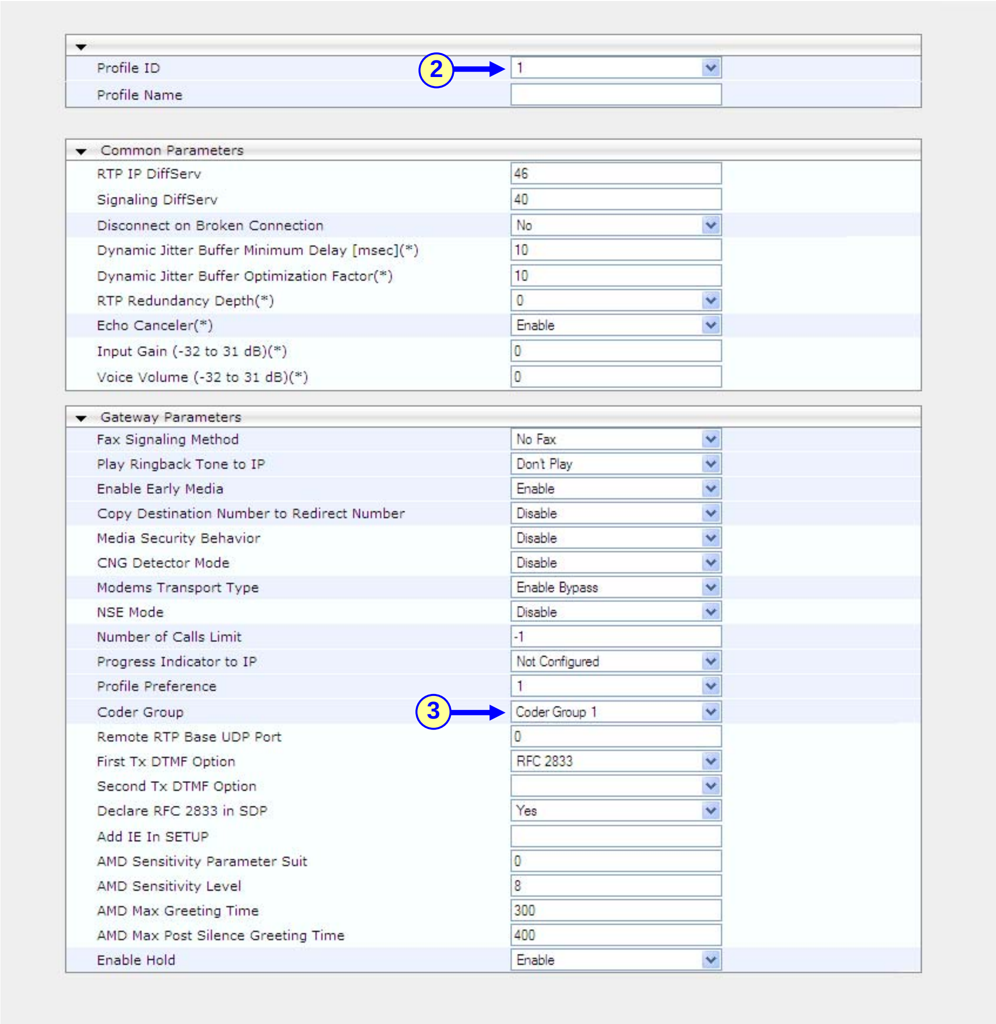
5.7 Step 8: Configure IP Profile Settings

This section describes how to configure the IP Profile Settings.

➢ **To configure IP Profile Settings for PAETEC:**

1. Open the 'IP Profile Settings' **page (Configuration** tab > **VoIP** menu > **Coders and Profiles** > **IP Profile Settings**).

Figure 5-19: IP Profile Settings - PAETEC Server

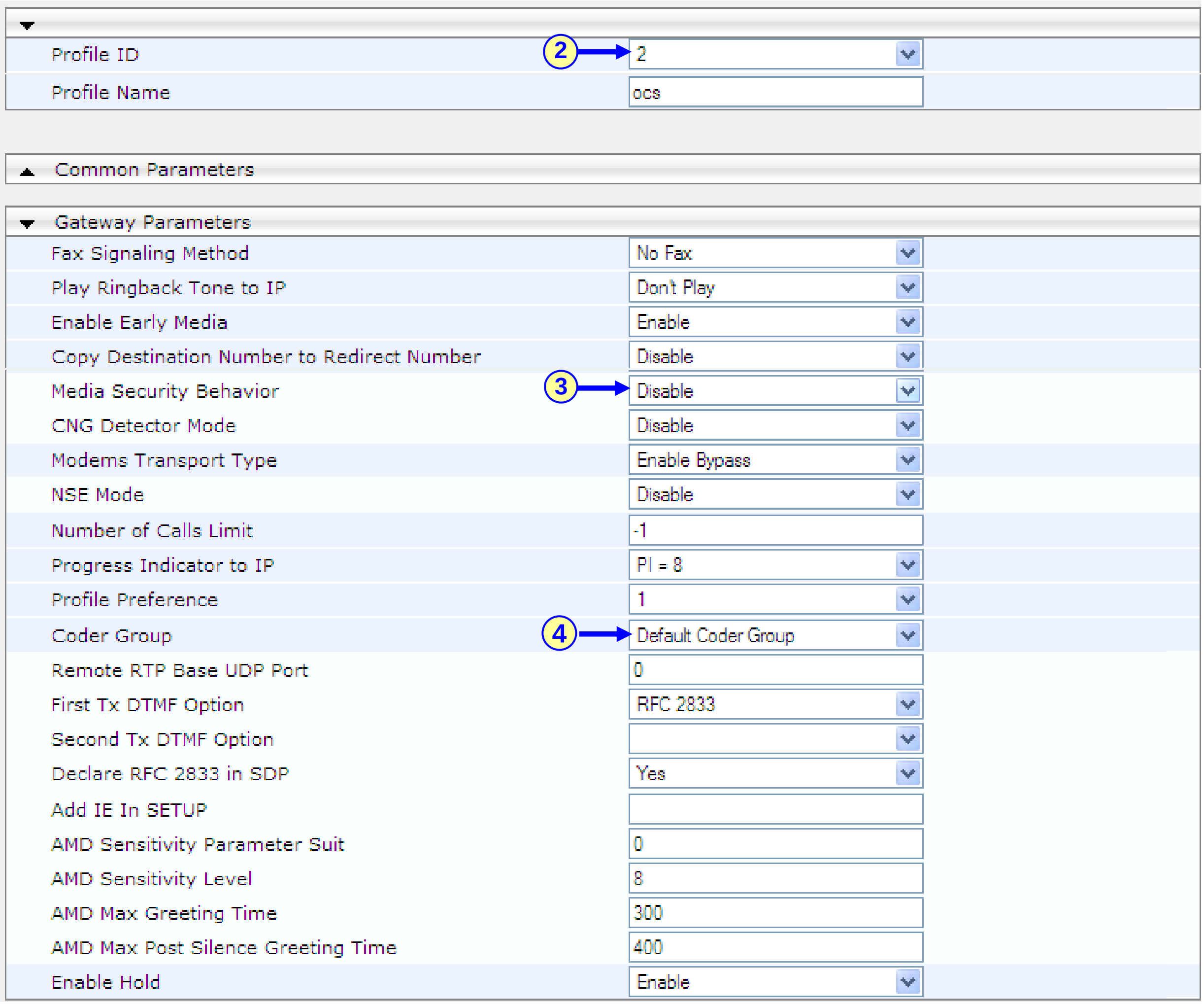


2. From the 'Profile ID' drop-down list, select '1'.
3. From the 'Coder Group' drop-down list, select 'Coder Group 1' (see Figure 5-17 on page 50 to which it refers)

➢ To configure IP Profile settings for LYNC Mediation Server:

1. Open the 'IP Profile Settings' page (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **IP Profile Settings**).

Figure 5-20: IP Profile Settings – LYNC Mediation Server



2. From the 'Profile ID' drop-down list, select '2'.
3. From the 'Media Security Behavior' drop-down list, select one of the following options:
   - "Mandatory" if Mediation Server is configured to SRTP Required
   - "Preferable-Single media" if Mediation Server is configured to SRTP Optional.
   - "Disable" if the Mediation Server is configured to SRTP disabled.
4. From the 'Coder Group' drop-down list, select 'Default Coder Group' (see Figure 5-27 on page to 62 which it refers).

5.8 Step 9: Configure IP-to-IP Routing Setup

The E-SBC devices' IP-to-IP call routing capability is performed in two stages:

1. **Inbound IP Routing:** Recognizes the received call as an IP-to-IP call, based on the call's source IP address. This step is configured in the 'Inbound IP Routing Table'

2. **Outbound IP Routing:** Once recognized as an IP-to-IP call in the first stage (see above), the call is routed to the appropriate destination (i.e., IP address). This step is configured in the 'Outbound IP Routing Table'.

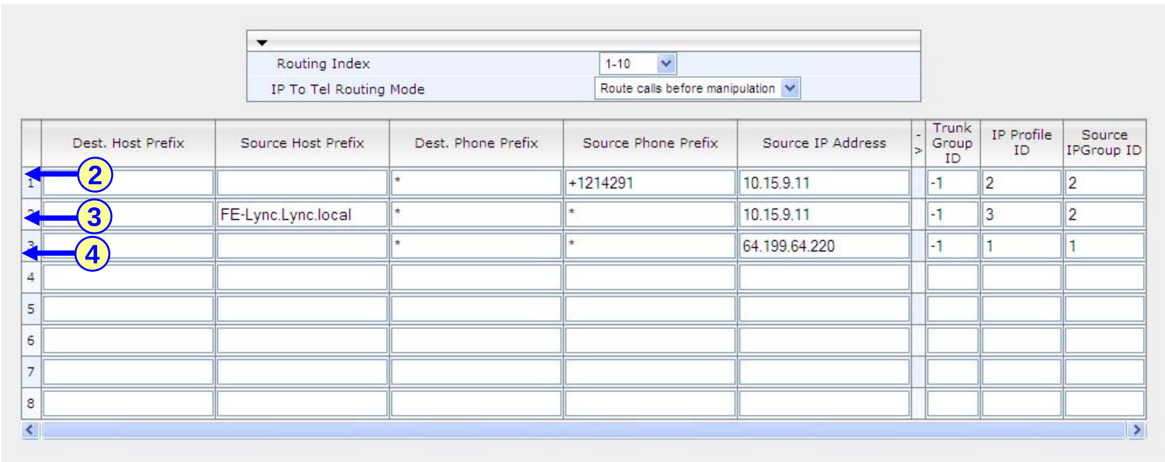
5.8.1 Configure Inbound IP Routing

This step defines how to configure the E-SBC device for routing inbound (i.e., received) IP-to-IP calls. Figure 5-21 illustrates three different call scenarios, corresponding to Index #1, Index #2 and Index#3 (described below).

➢ **To configure inbound IP routing:**

1. Open the 'Inbound IP Routing Table' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Routing** submenu > **IP to Trunk Group Routing**).

Figure 5-21: Inbound IP Routing Table Page



	Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Trunk Group ID	IP Profile ID	Source IPGroup ID
1			*	+1214291	10.15.9.11		-1	2	2
2		FE-Lync.Lync.local	*	*	10.15.9.11		-1	3	2
3			*	*	64.199.64.220		-1	1	1
4									
5									
6									
7									
8									

2. **Index #1** configuration identifies all IP calls received from the Mediation Server as IP-to-IP calls and assigns them to the IP Group ID configured for the Lync Mediation Server as verified Lync assigned telephone numbers within a prefix range:
   - 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
   - ‘Source Phone Prefix': Enter the Lync assigned telephone prefix to screen for valid direct IP-to-IP calls of PAETEC DIDs assigned to the location.
   - 'Source IP Address': Enter the IP address of the Mediation Server.
   - 'Trunk Group ID': Enter "-1" to indicate that these calls are IP-to-IP calls.
   - ‘IP Profile ID: Enter ‘2’ indicate the IP Profile for Mediation Server.
   - 'Source IP Group ID': Enter "2" to assign these calls to the IP Group pertaining to the Mediation Server.
3. **Index #2** configuration identifies all IP calls received from the Mediation Server in the event of a call forwarding scenario (see Section 5.10 on page 62) as IP-to-IP calls and assigns them to the IP Group ID configured for the Mediation Server:
   - ‘Source Host Prefix: Enter the Lync Front end FQDN – in case of call forwarding, the Source host in the incoming INVITE from the Mediation Server is the Lync Front End server FQDN, while for regular calls, the Source host is the Mediation Server FQDN.
   - 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
   - 'Source IP Address': Enter the IP address of the Mediation Server.
   - 'Trunk Group ID': Enter "-1" to indicate that these calls are IP-to-IP calls.
   - ‘IP Profile ID: Enter ‘3’ to indicate that the IP Profile supports the call forwarding scenario.
   - 'Source IP Group ID': Enter "2" to assign these calls to the IP Group pertaining to the Mediation Server.
4. **Index #3** configuration identifies all IP calls received from PAETEC SIP Trunk as IP-to-IP calls and assigns them to the IP Group ID configured for the PAETEC SIP Trunk:
   - 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
   - 'Source IP Address': Enter the IP address of PAETEC SIP Trunk.
   - 'Trunk Group ID': Enter "-1" to indicate that these calls are IP-to-IP calls.
   - ‘IP Profile ID: Enter ‘1’ indicate the IP Profile for PAETEC SIP Trunk.
   - 'Source IP Group ID': Enter "1" to assign these calls to the IP Group pertaining to PAETEC SIP Trunk.

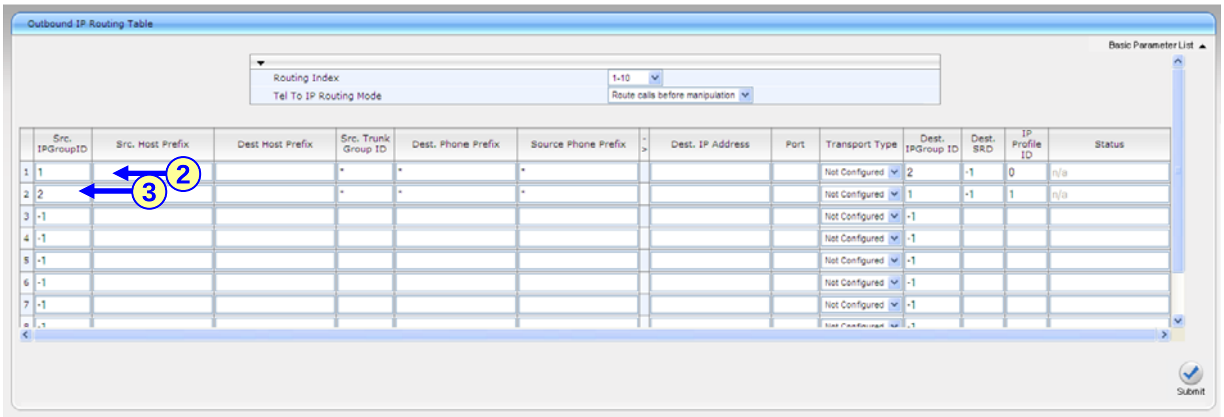
5.8.2 Configure Outbound IP Routing

This step defines how to configure the E-SBC device for outbound routing (i.e., sent) IP-to-IP calls. Figure 5-22 illustrates two different call scenarios, corresponding to Index #1 and Index #2 (described below).

➢ **To configure outbound IP routing:**

1. Open the 'Outbound IP Routing Table' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Routing** submenu > **Tel to IP Routing**).

Figure 5-22: Outbound IP Routing Table Page



2. **Index #1** defines routing of IP calls to the Lync 2010 Mediation server. All calls received from Source IP Group ID 1 (i.e., from the PAETEC SIP trunk) are routed to Destination IP Group ID 2 (i.e., to Lync 2010 Mediation Server):
   - 'Source IP Group ID': Select "1" to indicate received (inbound) calls identified as belonging to the IP Group configured for the PAETEC SIP Trunk.
   - 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
   - 'Source Phone Prefix': Enter the asterisk (*) symbol to indicate all callers.
   - 'Dest IP Group ID': Select "2" to indicate the destination IP Group to where the calls must be sent, i.e., to Lync 2010 Mediation Server.
   - Set IP Profile ID to 0
3. **Index #2** defines the routing of IP calls to the PAETEC SIP Trunk. All calls received from IP Group ID 2 (i.e., Lync 2010 Mediation Server) are routed to Destination IP Group ID 1 (i.e., PAETEC SIP Trunk):
   - 'Source IP Group ID': Select "2" to indicate received (inbound) calls identified as belonging to the IP Group configured for the Lync 2010 Mediation Server 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
   - 'Source Phone Prefix': Enter the asterisk (*) symbol to indicate all callers.
   - 'Dest IP Group ID': Select "1" to indicate the destination IP Group to where the calls must be sent, i.e., to the PAETEC SIP Trunk.
   - Set IP Profile to 1

5.9 Step 10: Configure Number Manipulation

The Manipulation Tables submenu allows you to configure number manipulation and mapping of NPI/TON to SIP messages. This submenu includes the following options:

- Dest Number IP->Tel. See Section 5.9.1 on page 58.
- Dest Number Tel->IP. See Section 5.9.1 on page 58.
- Source Number IP->Tel. See Section 5.9.2 on page 60.
- Source Number Tel->IP. See Section 5.9.2 on page 60.

5.9.1 Configure Destination Phone Number Manipulation

This section describes how to configure the destination phone number manipulation.

➢ **To configure Destination Phone Number Manipulation Table for IP -> Tel Calls Table:**

- Open the ‘Destination Phone Number Manipulation Table for IP -> Tel calls’ page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Manipulations** submenu > **Dest Number IP** >**Tel**).

Figure 5-23: Destination Phone Number Manipulation Table for IP -> Tel Calls Page

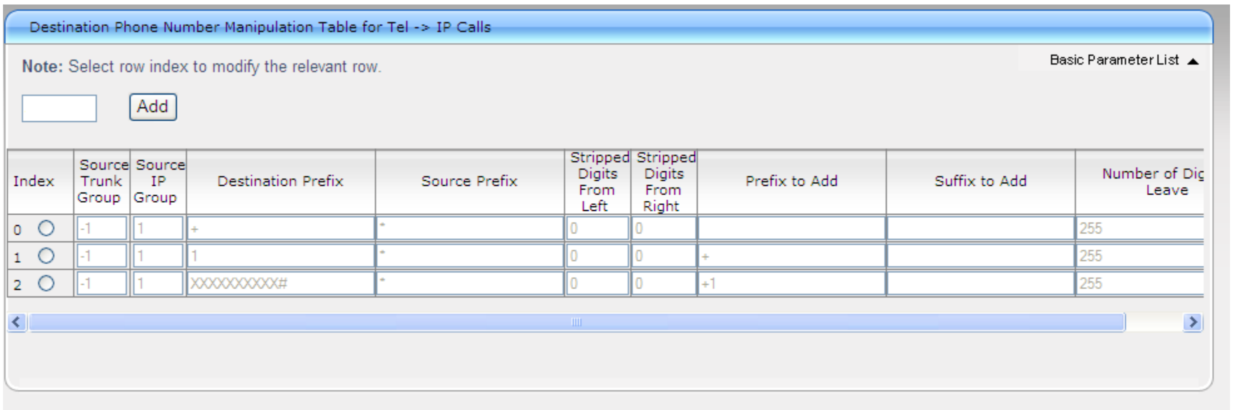
Index	Destination Prefix	Source Prefix	Source IP Address	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Num
1	+1	*	10.15.9.11	2	0	1		255
2	+0	*	10.15.9.11	1	0			255
3	+	*	10.15.9.11	1	0	1		255
4	1	*	10.15.9.11	1	0	1		255

- **Index #1** defines destination number manipulation of IP calls from Lync 2010 Mediation Server. All calls received from IP address 10.15.9.11 (i.e., from Mediation Server) and the destination number prefix begins with '+1', Remove the ‘+1’ from the Number, and add the prefix ‘1’.
- **Index #2** defines destination number manipulation of IP calls from Lync 2010 Mediation Server. All calls received from IP address 10.15.9.11 (i.e., from Mediation Server) and the destination number prefix begins with '+0', Remove the ‘+’ from the Number.
- **Index #3** defines destination number manipulation of IP calls from Lync 2010 Mediation Server. All calls received from IP address 10.15.9.11 (i.e., from Mediation Server) and the destination number prefix begins with '+', Remove the ‘+’ from the Number, and add the prefix ‘1’.
- **Index #4** defines destination number manipulation of IP calls from Lync 2010 Mediation Server. All calls received from IP address 10.15.9.11 (i.e., from Mediation Server) and the destination number prefix begins with '1', Remove the ‘1’ from the Number, and add the prefix ‘1’.

➢ **To configure Destination Phone Number Manipulation Table for Tel -> IP Calls Table:**

1. Open the ‘Destination Phone Number Manipulation Table for Tel -> IP calls’ page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Manipulations** submenu > **Dest Number Tel->IP**).

Figure 5-24: Destination Phone Number Manipulation Table for Tel -> IP Calls Page



- **Index #0** defines destination number manipulation of IP calls from PAETEC SIP Trunk. All calls received from Source IP Group 1 (i.e., from PAETEC SIP Trunk) and the destination number prefix begins with '+', do not perform any changes to the number.
- **Index #1** defines destination number manipulation of IP calls from PAETEC SIP Trunk. All calls received from Source IP Group 1 (i.e., from PAETEC SIP Trunk) and the destination number prefix begins with '1', add the '+' prefix to the number.
- **Index #2** defines destination number manipulation of IP calls from PAETEC SIP Trunk. All calls received from Source IP Group 1 (i.e., from PAETEC SIP Trunk) and the destination number length is 10 digit number, add the '+1' prefix to the number.

5.9.2 Configure Source Phone Number Manipulation

➢ **To configure Source Phone Number Manipulation Table for IP -> Tel Calls Table:**

1. Open the 'Source Phone Number Manipulation Table for IP -> Tel calls' page (**Configuration** tab > **VoIP menu** > **GW and IP to IP** > **Manipulations** submenu > **Source Number IP** >**Tel**).

Figure 5-25: Source Phone Number Manipulation Table for IP -> Tel Calls Page

Index	Destination Prefix	Source Prefix	Source IP Address	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Num
1	*	+1	10.15.9.11	2	0			255
2	*	+	10.15.9.11	1	0			255
3	*	1	10.15.9.11	1	0			255
4	*	anonymous	10.15.9.11	20	0	a7192083390		255

Number of Digits to Leave	NPI	TON	Presentation
255	Not Configured	Not Configured	Allowed
255	Not Configured	Not Configured	Allowed
255	Not Configured	Not Configured	Allowed
255	Not Configured	Not Configured	Not Configured

- **Index #1** defines Source number manipulation of IP calls from Lync Mediation Server. All calls received from IP address 10.15.9.11 (i.e., from Lync Mediation Server) and the source number prefix begins with '+1', remove the '+1' from the number.
- **Index #2** defines Source number manipulation of IP calls from Lync Mediation Server. All calls received from IP address 10.15.9.11 (i.e., from Lync Mediation Server) and the source number prefix begins with '+', remove the '+' from the number.
- **Index #3** defines Source number manipulation of IP calls from Lync Mediation Server. All calls received from IP address 10.15.9.11 (i.e., from Lync Mediation Server) and the source number prefix begins with '1', remove the '1' from the number.
- **Index #4** defines Source number manipulation of anonymous calls from Lync Mediation Server. Anonymous calls received from IP address 10.15.9.11 (i.e., from Lync Mediation Server) replace the 'anonymous' caller ID with a modified well-known number, i.e., a7192083390. This manipulation is performed to create a well-known number in the P-Asserted-Identity header. Without this number, the PAETEC SIP Trunk could potentially reject the call. See below (in the Source Phone Number Manipulation Table for Tel -> IP index #4) for the Source Number manipulation Tel->IP manipulation rule that restricts the caller ID for an anonymous call.

➢ To configure Source Phone Number Manipulation Table for Tel -> IP Calls Table:

1. O pen the ‘Source Phone Number Manipulation Table for Tel -> IP calls’ page (**Configuration** tab > **VoIP menu** > **GW and IP to IP** > **Manipulations** submenu > **Source Number Tel** > **IP**).

Figure 5-26: Source Phone Number Manipulation Table for Tel -> IP Calls Page

Index	Source Trunk Group	Source IP Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digit Leave	Presentation
0	-1	1	*	+	0	0			255	Allowed
1	-1	1	*	1	0	0	+		255	Allowed
2	-1	1	*	Restricted	0	0			255	Not Configured
3	-1	1	*	XXXXXXXXXX#	0	0	+1		255	Allowed
4	-1	2	*	a7192083390	1	0			255	Restricted

- **Index #0** defines Source number manipulation of IP calls from PAETEC SIP Trunk. All calls received from Source IP Group 1 (i.e., from PAETEC SIP Trunk) and the Source number prefix begins with '+', do not perform any changes to the number.
- **Index #1** defines Source number manipulation of IP calls from PAETEC SIP Trunk. All calls received from Source IP Group 1 (i.e., from PAETEC SIP Trunk) and the Source number prefix begins with '1', Add a ‘+’ as a prefix to the number.
- **Index #2** defines Source number manipulation of IP calls from PAETEC SIP Trunk. All calls received from Source IP Group 1 (i.e., from PAETEC SIP Trunk) and the Source number prefix is set as 'Restricted', do not perform any changes to the call.
- **Index #3** defines Source number manipulation of IP calls from PAETEC SIP Trunk. All calls received from Source IP Group 1 (i.e., from PAETEC SIP Trunk) and the Source number length is 10 digit number, add the ‘+1’ prefix to the number.
- **Index #4** defines Source number manipulation of anonymous calls from Microsoft Lync environment. All calls received from Source IP Group 2 (and the Source number was modified to a7192083390 (which is a specially modified well known number that was inserted for the anonymous caller ID on the source number manipulation IP->Tel above), the ‘a’ is removed and the presentation should be set to ‘restricted’. To simply mark all calls as Private calls, use an asterisk ‘*’ in the Source Prefix field. Individual Telephone Numbers or ranges can also be set in this manner as well. This number will be presented in the P-Asserted Identity.

5.10 Step 11: Configure IP Profile for Call Forwarding

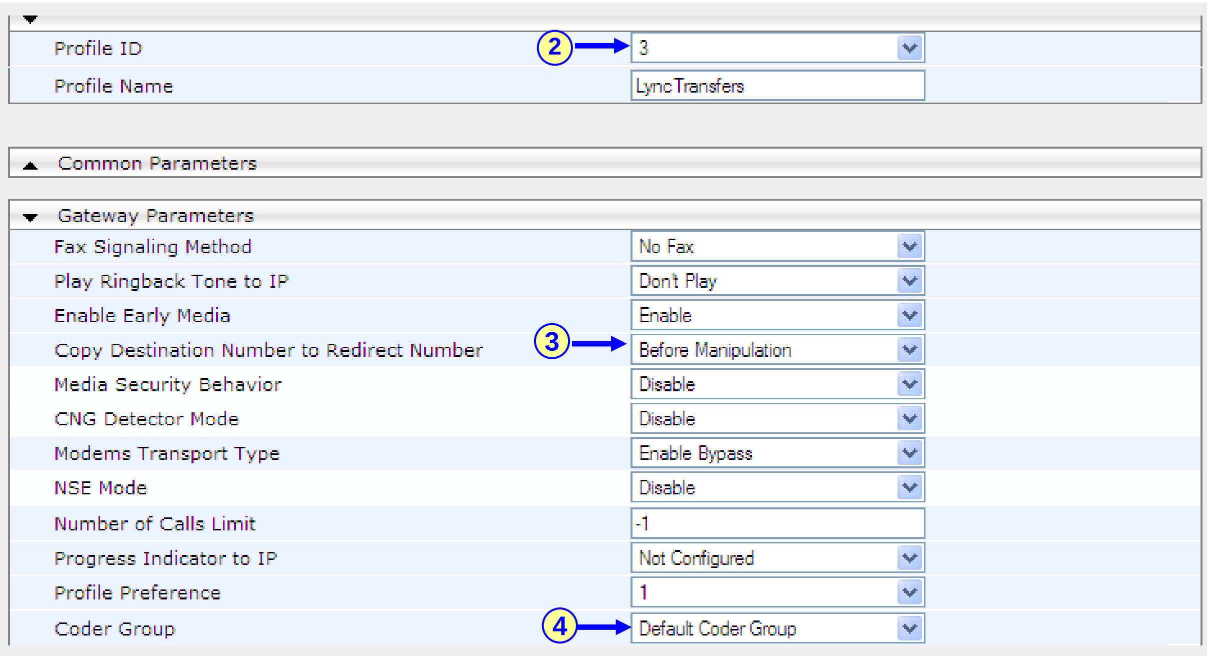
Call forwarding must be implemented as part of integrating the Microsoft Lync 2010 server and the PAETEC SIP Trunk. Since the Microsoft Lync client forwards the call back to the SIP Trunk, it does not provide any information in the forwarded INVITE (such as Diversion Header) informing that this call has been forwarded. Consequently, it's necessary to configure a special IP Profile that adds the diversion header toward the SIP trunk in the event of a call forwarding scenario.

This profile is later associated to the routing table in the event of a call forwarding scenario (see Section 5.8.1 on page 54).

➢ **To configure an IP Profile for call forwarding:**

1. Open the 'IP Profile Settings' page (**Configuration tab** > **VoIP menu** > **Coders and Profiles** > **IP Profile Settings**).

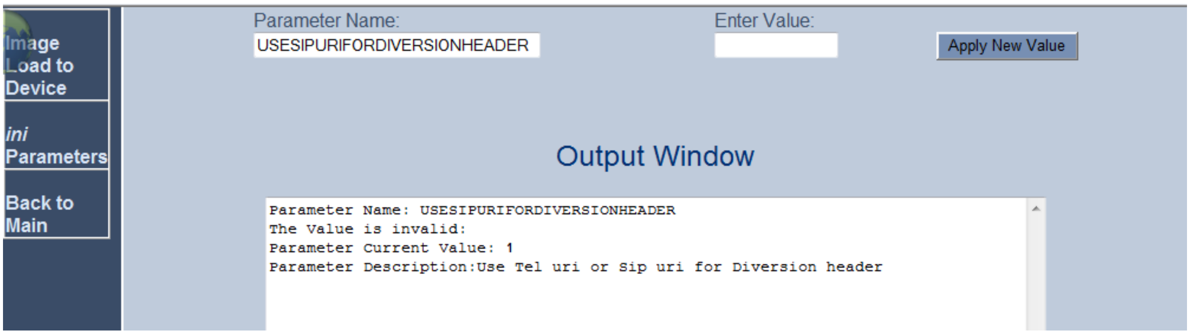
Figure 5-27: IP Profile Settings for Call Forwarding "Numbers"



2. From the 'Profile ID' drop-down list, select '3'.
3. From the 'Copy Destination Number to Redirect Number' drop-down list, select 'Before Manipulation'; this parameter adds the Diversion Header to the INVITE in event of a call forwarding scenario.
4. From the 'Coder Group' drop-down list, select 'Default Coder Group'.

5. Open the 'Admin" page by appending the case-sensitive suffix 'AdminPage' to the Media Gateway's IP address in your Web browser's URL field (e.g., http://10.15.4.15/AdminPage).
6. In the left pane, click ***ini* Parameters**.

Figure 5-28: Output Window



7. In the 'Parameter Name' field, enter **USESIPURIFORDIVERSIONHEADER**. In the 'Enter Value' field, enter "1".
8. Click **Apply New Value**.

5.10.1 Configure Redirect Number Manipulation

In the event of a call forwarding scenario, a Diversion header must be added to the INVITE towards the PAETEC SIP Trunk (as configured in Section 5.10 above). In this case, the E-SBC copies the Destination number to the Redirect number and adds this number to the Diversion header. To have a well-known number in the Diversion header (for PAETEC SIP Trunk), a manipulation rule should be defined to replace the Redirect number with a well-known number.

➢ **To configure the Redirect Number Tel -> IP table:**

1. Open the 'Redirect Number Tel -> IP' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Manipulations** submenu > **Redirect Number Tel** > **IP**).

Figure 5-29: Redirect Number Tel -> IP Page

Index	Source Trunk Group	Source IP Group	Destination Prefix	Redirect Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digit Leave
1	-1	-1	*	*	20	0	7133439307		255

- Index **#1** defines redirect number manipulation for the call forwarding scenario.

The Redirect number is changed to a well-known number, i.e., 7133439307.

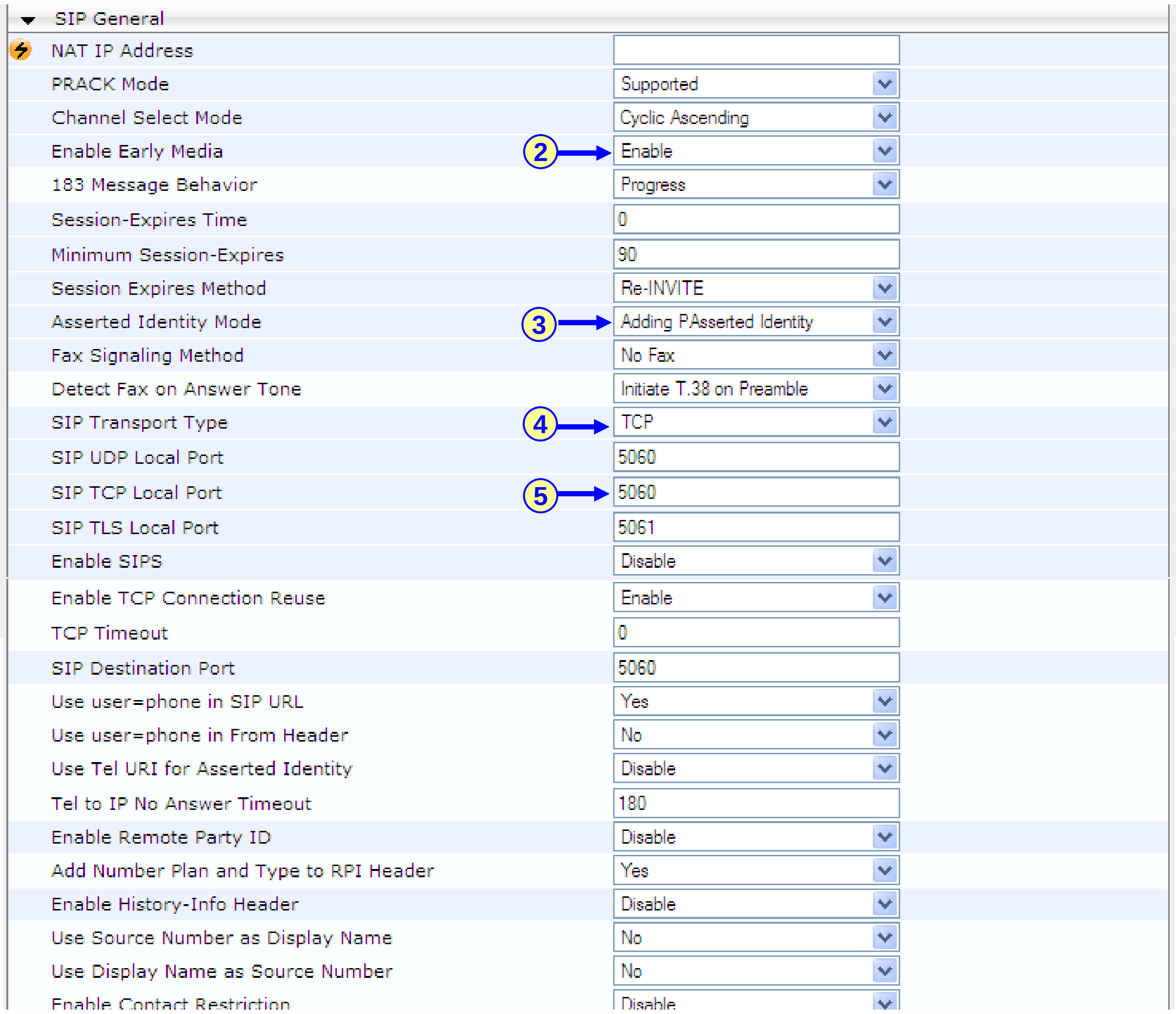
5.11 Step 12: Configuring General SIP Parameters

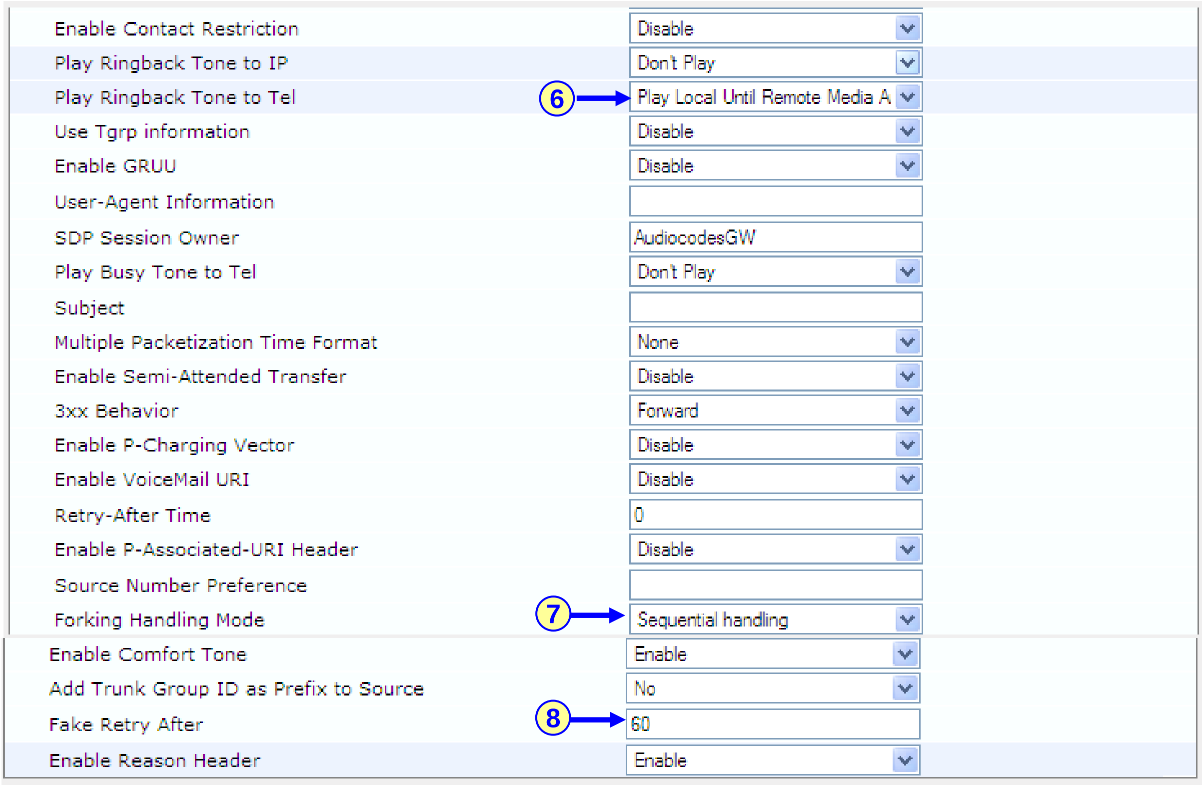
This section describes how to configure the general SIP parameters.

➢ **To configure the general SIP parameters:**

1. Open the 'SIP General' page (**Configuration** tab > **VoIP** menu > **SIP Definitions** submenu > **General Parameters**).

Figure 5-30: General SIP Parameters Page





2. From the ‘Enable Early Media’ drop-down list, select ‘Enable’ to enable early media.
3. From the ‘Asserted Identity Mode’ drop-down list, select ‘Adding PAsserted Identity’.
4. From the ‘SIP Transport Type’ drop-down list, select ‘TCP’ in case the Mediation Server is configured to use TCP transport type.
5. In the ‘SIP TCP Local Port’ field, enter ‘5060’; this port is the listening E-SBC device port for TCP transport type. This port must match the transmitting port of the Mediation Server.
6. From ‘Play Ringback Tone to Tel’ drop-down list, select ‘Play Local Until Remote Media Arrive’. Plays the RBT according to the received media. If a SIP 180 response is received and the voice channel is already open (due to a previous 183 early media response or due to an SDP in the current 180 response), the E-SBC device plays a local RBT if there are no prior received RTP packets. The E-SBC device stops playing the local RBT as soon as it starts receiving RTP packets. At this stage, if the E-SBC device receives additional 18x responses, it does not resume playing the local RBT.
7. From the ‘Forking Handling Mode’ drop-down list, select ‘Sequential handling’; this parameter determines whether18x with SDP is received. In this case, the E-SBC device opens a voice stream according to the received SDP. The E-SBC device re-opens the stream according to subsequently received 18x responses with SDP.
8. In the ‘Fake Retry After’ field, enter ‘60’ seconds. This parameter determines whether the E-SBC device, on receipt of a SIP 503 response without a Retry-After header, behaves as if the 503 response included a Retry-After header and with the period (in seconds) specified by this parameter.

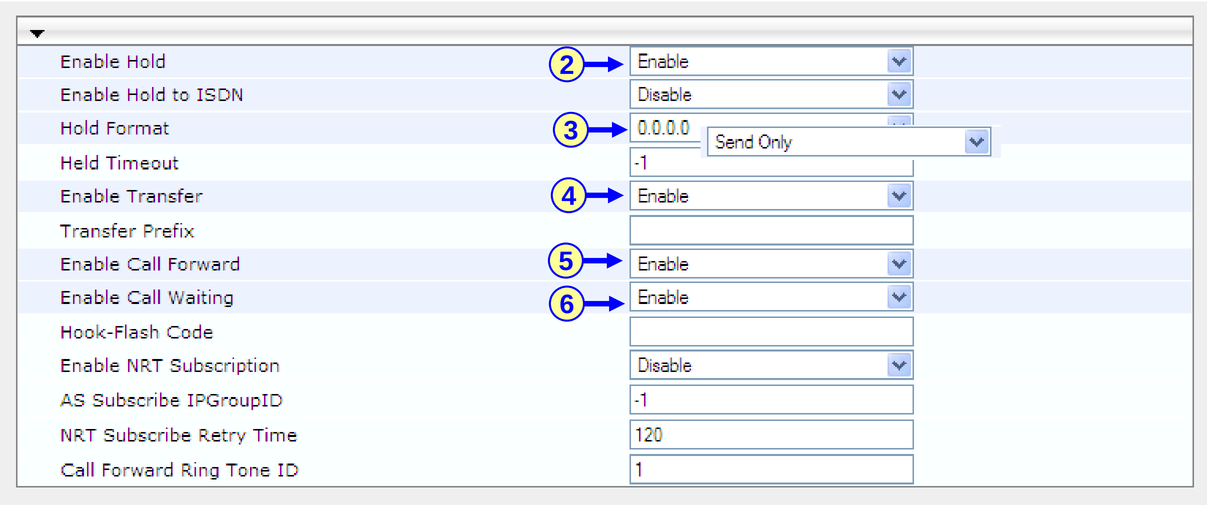
5.12 Step 13: Configuring SIP Supplementary Services

This section describes how to configure the SIP Supplementary Services parameters.

➢ **To configure SIP Supplementary Services parameters:**

1. Open the 'SIP Supplementary Services' page (**Configuration** tab > **VoIP** menu > **GW and IP 2 IP** sub-menu > **DTMF and Supplementary** sub-menu > **Supplementary Services**).

Figure 5-31: SIP Supplementary Services Page



2. In the 'Enable Hold' drop-down list, select **Enable**.
3. From the 'Hold Format' drop-down list, select **0.0.0.0** to enable Hold in the no-media-in-either-direction method, or select **Send Only** to enable Hold to support one-way audio for Music on Hold type service support. PAETEC supports the interworking to support Music on Hold. From the drop down-list, select **Send Only**.
4. From the 'Enable Transfer' drop-down list, select **Enable**.
5. From the 'Enable Call Forward' drop-down list, select **Enable**.
6. In the 'Enable Call Waiting' drop-down list, select **Enable**.

5.13 Step 14: Defining Reasons for Alternative Routing

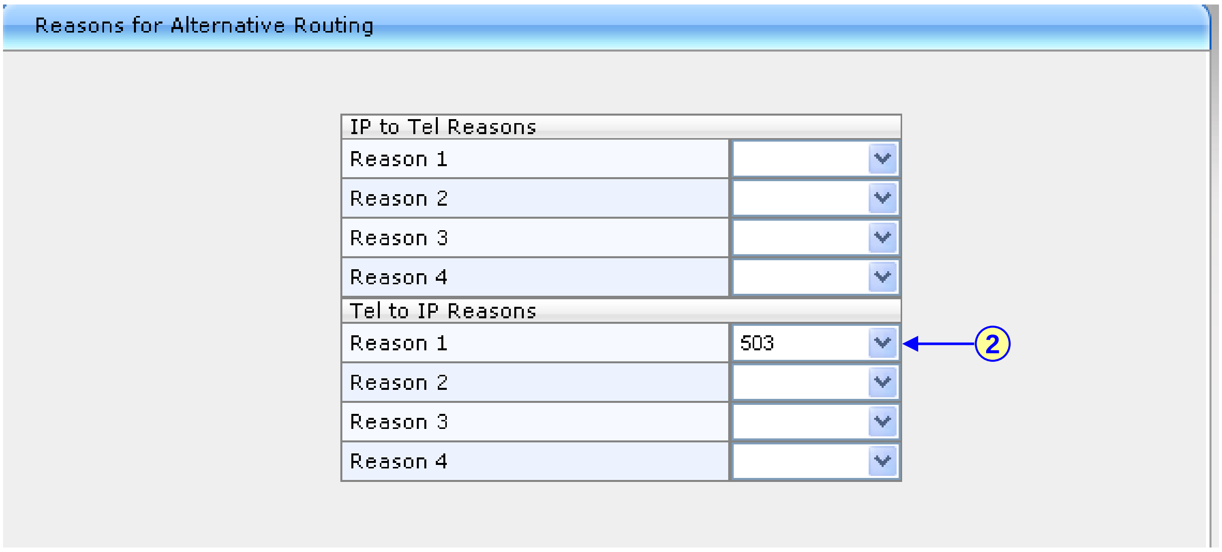
A 503 SIP response from the Mediation Server to an INVITE must cause the E-SBC device to perform a failover. In other words, if the Lync Mediation Server primary proxy server is not responding, an attempt is made to establish communication with the secondary proxy server. For this event to occur, you need to perform the following actions:

- Configure the Reasons for Alternative Routing for Tel-to-IP calls to '503 SIP response'.
- Configure the Lync Mediation Proxy Set for redundancy purposes.

➢ To define SIP Reason for Alternative Routing:

1. Open the 'Reasons for Alternative Routing' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Routing** submenu > **Alternative Routing Reasons**).

Figure 5-32: Reasons for Alternative Routing Page



2. Under the Tel to IP Reasons group, for Reason 1, select '503'.
3. Click **Submit**.
4. Open the 'Proxy & Registration' page (**Configuration** > **VoIP** > **SIP Definitions** > **Proxy & Registration**) and configure the 'Redundant Routing Mode' parameter to 'Proxy' as shown below in Figure 5-33. This will allow entry back into the Proxy Set table for the next available route. This redundant route is configured in the next step (on Proxy Set ID 2, see Figure 5-34 below).
5. Open the 'Proxy Sets Table' page (**Configuration** tab > **VoIP** menu > **Control Network** > **Proxy Sets Table**). Configure the Proxy Set for the Lync Mediation Server:

   From the 'Proxy Set ID' drop-down list, select "2".

   a. In the 'Proxy Address' column, enter a second IP address or the FQDN and the listening port of the secondary Lync Mediation Server.
   b. From the 'Is Proxy Hot Swap' drop-down list, select "Yes".
6. Click **Submit**.

Figure 5-33: 'Proxy & Registration' Page

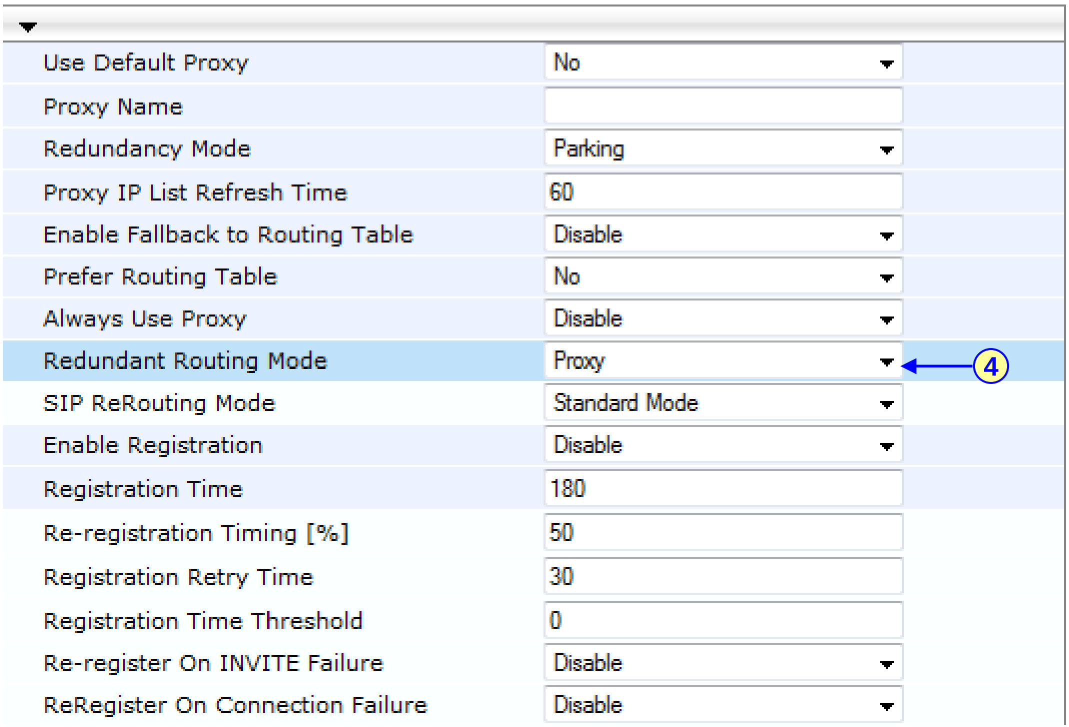
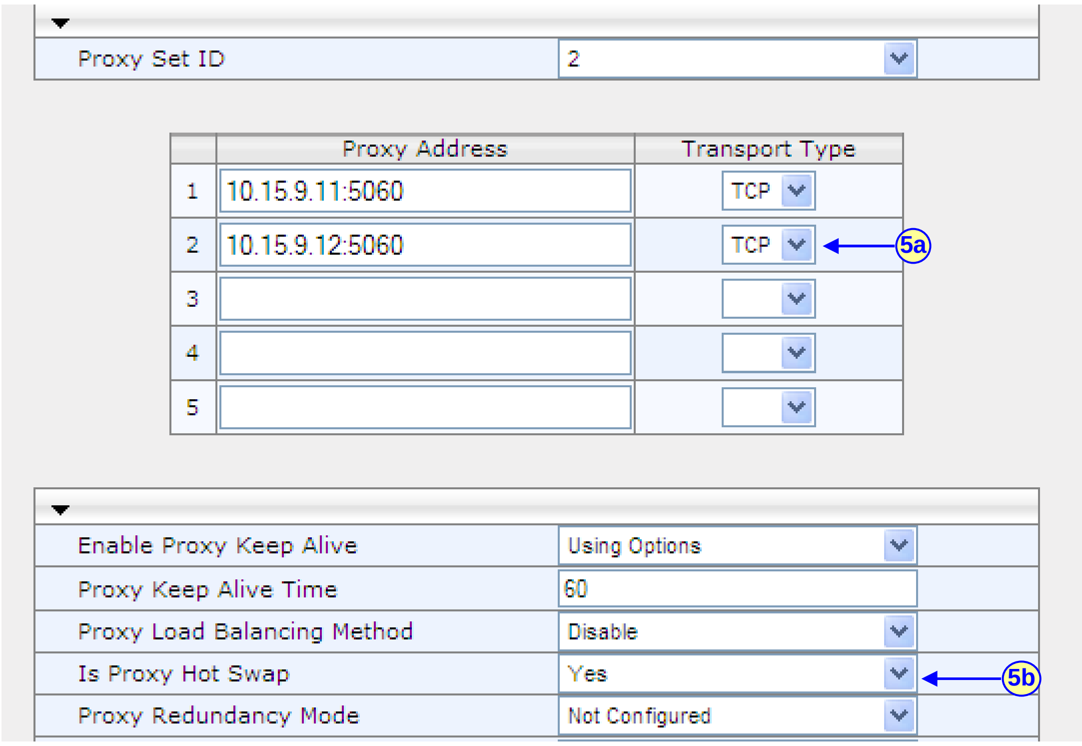


Figure 5-34: Proxy Set ID 2 for Lync Mediation Server



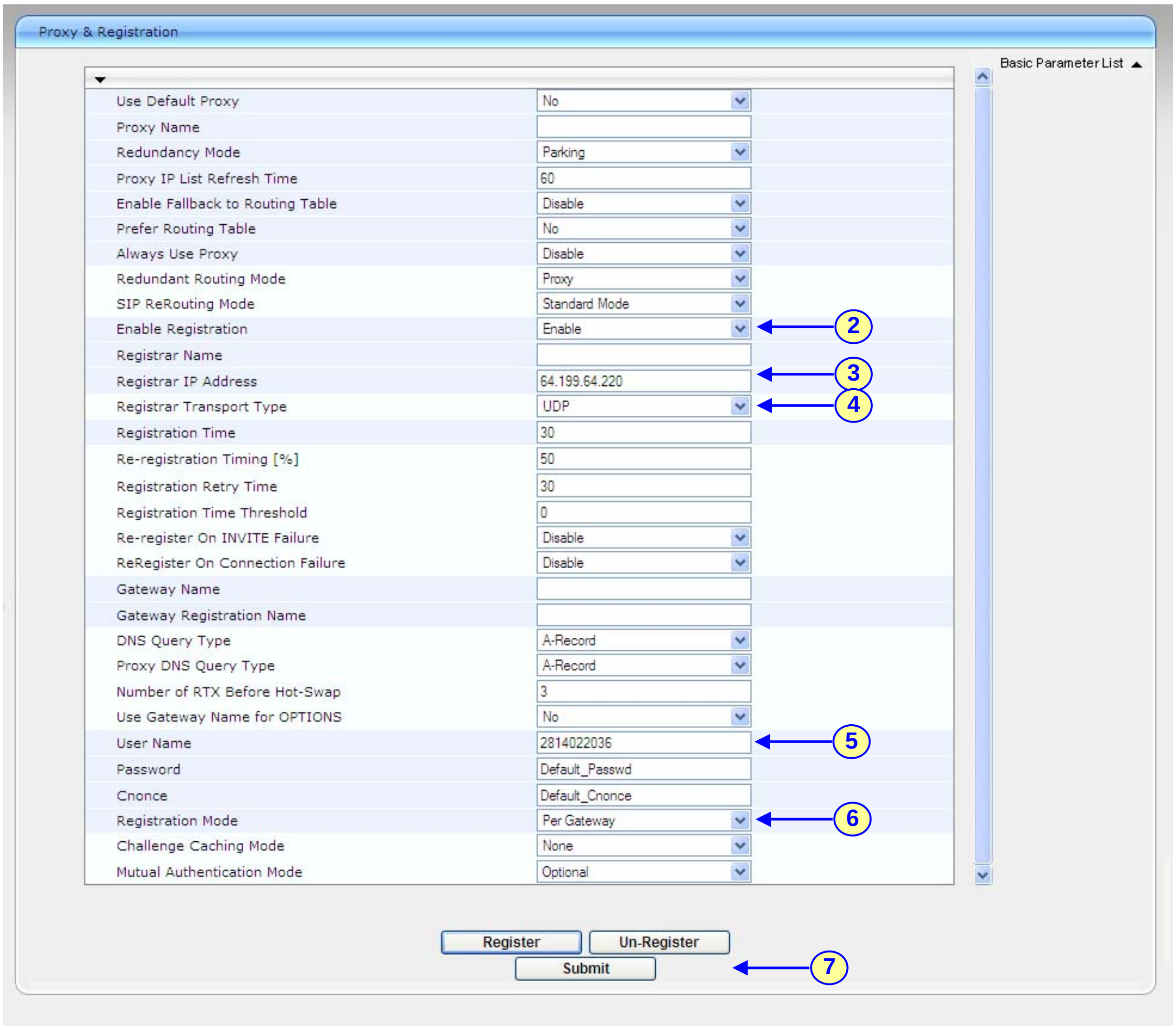
5.14 Step 15: Configuring for Gateway Registration

This section describes how to configure the SIP Proxy and Registration parameters to support Gateway Registration.

➢ To configure the SIP Proxy and Registration parameters:

1. Open the 'SIP General Parameters' page (**Configuration** tab > **VoIP** menu > **SIP Definitions** submenu > **Proxy & Registration Parameters**).

Figure 5-35: SIP Proxy & Registration Parameters Page

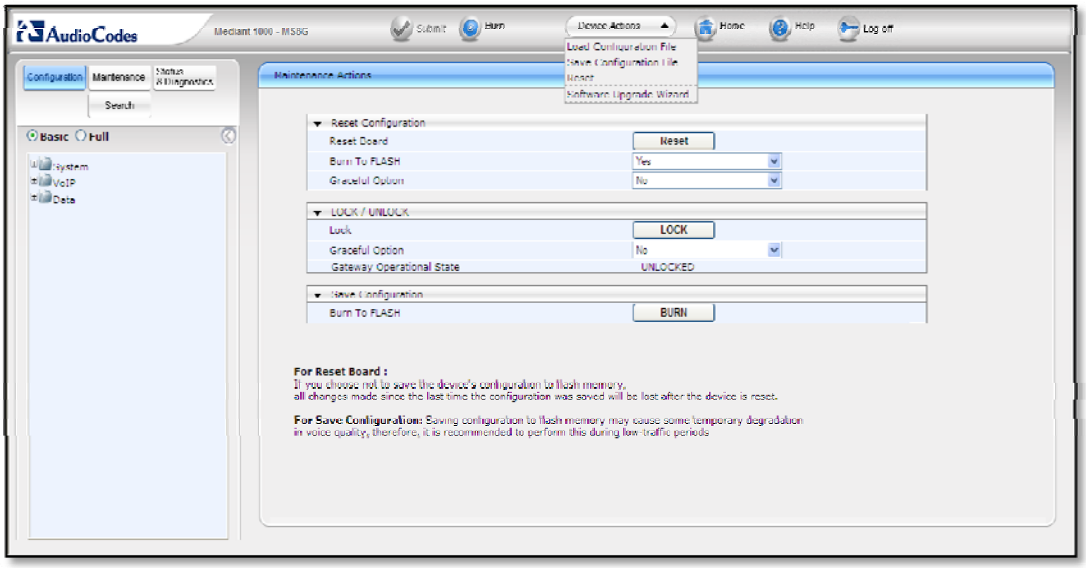


2. In the 'Enable Registration' drop-down list, select **Enable**.
3. In the 'Registrar IP Address', enter the address supporting the registrar. **[64.199.64.220]**
4. From the 'Registrar Transport Type' drop-down list, select **UDP**.
5. From the 'User Name', enter the valid number for registration of the gateway device. **[2814022036]**.
6. From the 'Registration Mode' drop-down list, select **Per Gateway**.
7. Once completed, select **Submit** and then **Register**.
8. **Reset** the device with the **Burn** option to ensure that it maintains its datafill after restart.

6 Saving the MSBG Configuration

It's advisable to periodically save the configuration when completing the following steps of this configuration guide. Return to the Home page and select the **Maintenance** tab in the left pane. Under **Maintenance Actions** click the **BURN** button. You can also use the **Device Action** drop-down menu and the **Burn** button to access the same page.

Figure 6-1: Saving the MSBG Configuration



Reader's Notes

7 Troubleshooting

This section provides some tips for troubleshooting problems, including troubleshooting commands and general suggestions to assist with trouble escalations.

7.1 Debugging Procedures

This section discusses the following debugging procedures:

- Case Reporting Procedures. See Section 7.1.1 below.
- Syslog. See Section 7.1.2 on page 74.
- Wireshark Network Sniffer. See Section 7.1.3 on page 76.

7.1.1 Case Reporting Procedures

When reporting a problem to AudioCodes' Technical Support department, the following information should be provided:

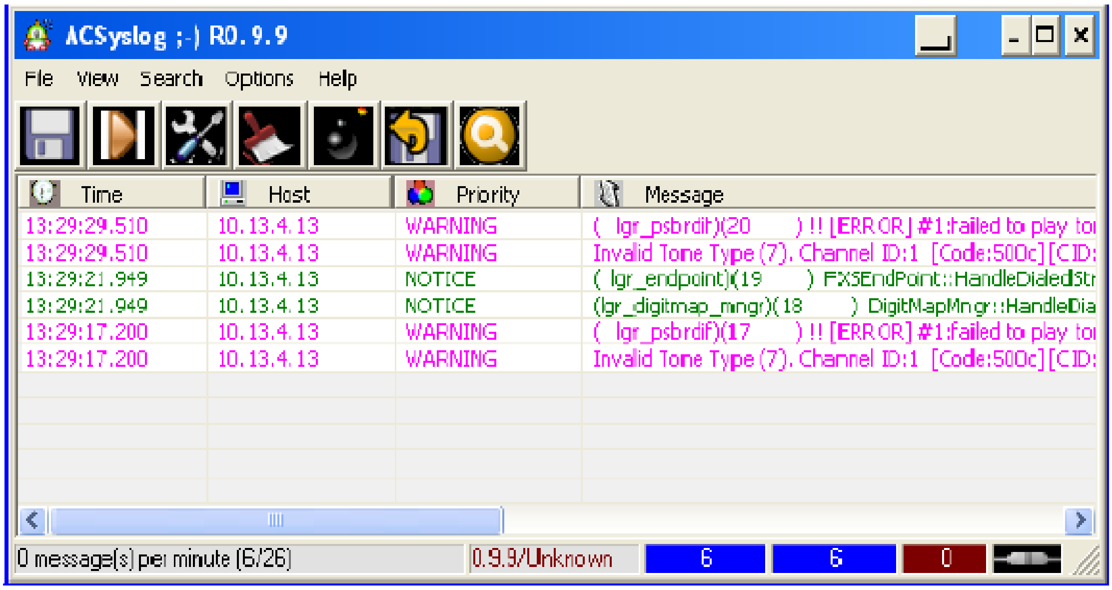
- Basic information (required for all types of problems):
  - Problem description (nature of failure, symptoms, call direction, etc.)
  - Network diagram
  - *ini* configuration file (downloaded to your PC from the device using the Web interface)
  - Syslog trace (without missing messages)
  - Unfiltered IP network trace using the Wireshark application

  (Note: If you are unable to collect all the network traffic, then at least collect the mandatory protocols SIP, RTP and T38.)
- Advanced information (if required on request):
  - PSTN message traces - for PSTN problems
  - Media stream traces - for problems related to voice quality, modem\fax, DTMF detection, etc.

7.1.2 Syslog

Syslog is a standard for forwarding log messages in an IP network. A Syslog client, embedded in the device sends error reports/events generated by the device to a remote Syslog server using IP/UDP protocol. This information is a collection of error, warning and system messages that record every internal operation of the device. You can use the supplied AudioCodes proprietary Syslog server "ACSyslog" (shown in Figure 7-1) or any other third-party Syslog server for receiving Syslog messages.

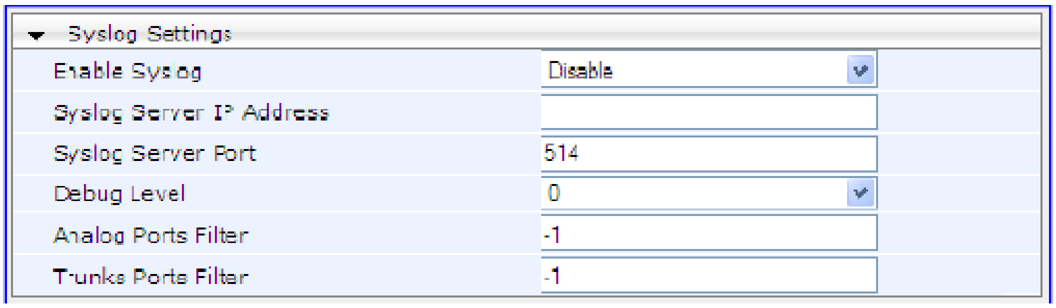
Figure 7-1: AudioCodes' Proprietary Syslog Server



➢ **To activate the Syslog client on the device using the Web interface:**

1. Open the 'Syslog Settings' page (**Configuration** tab > **System** menu > **Syslog Settings**).
2. In the 'Syslog Server IP Address' field, enter the IP address of the Syslog server (*ini* file parameter SyslogServerIP).
3. From the 'Enable Syslog' drop-down list, select 'Enable' to enable the device to send Syslog messages to a Syslog server (defined in Step 2).

Figure 7-2: Enabling Syslog



4. From the 'Debug Level' drop-down list, select '5' if debug traces are required.

To enable Syslog reporting, using the *ini* file, load an *ini* file to the device with the following settings:

```
[Syslog]
SyslogServerIP = 192.168.2.35
EnableSyslog = 1
SyslogServerPort = 514
GWDebugLevel = 5
```

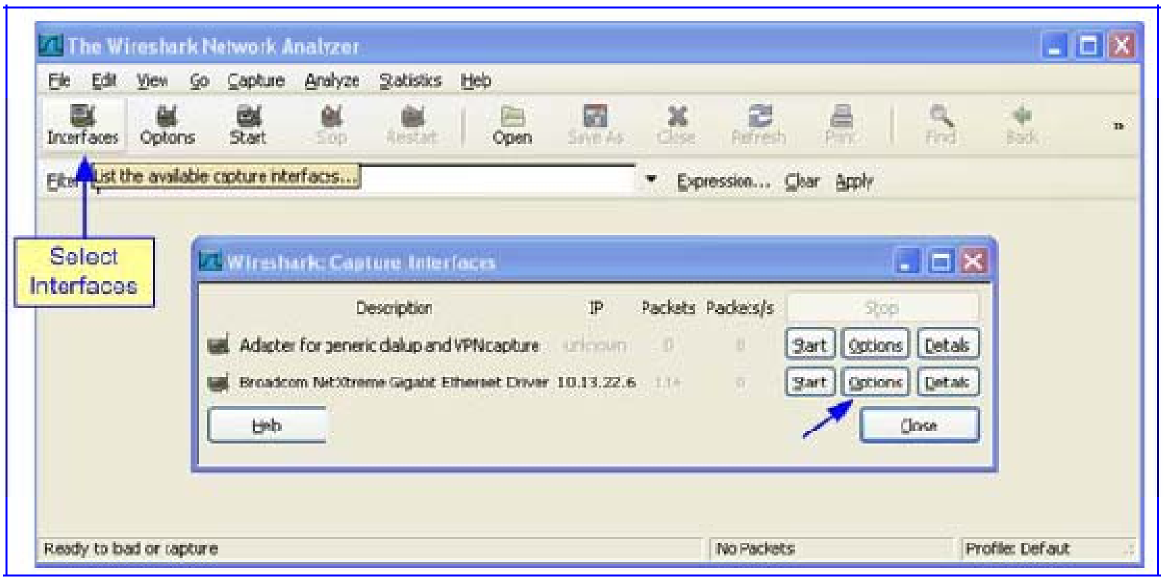
7.1.3 Wireshark Network Sniffer

Wireshark is a freeware packet sniffer application that allows you to view the traffic that is being passed over the network. Wireshark can be used to analyze any network packets. Wireshark can also be used to analyze RTP data streams and extract the audio from the data packets (only for G.711). The audio can be saved as a *.pcm file.

➢ **To record traffic that is sent to / from the device:**

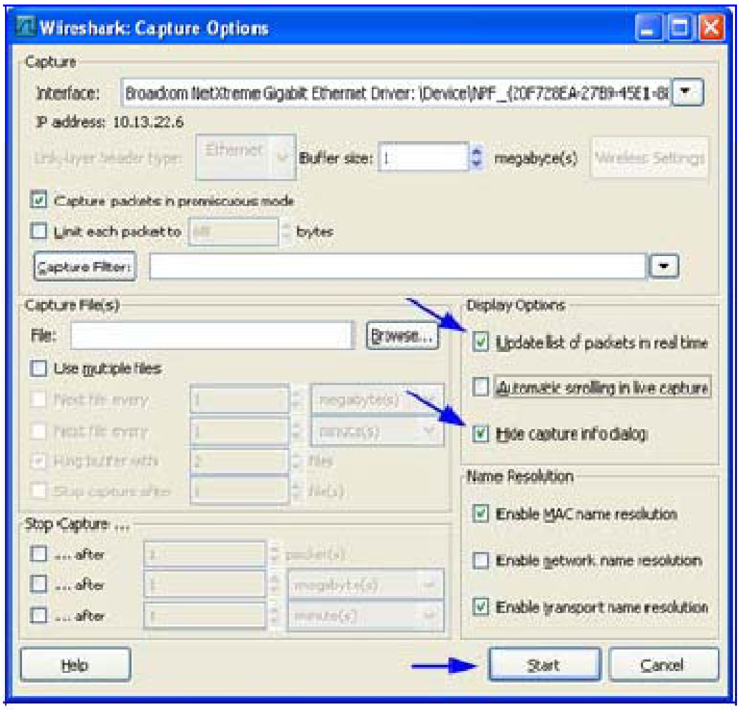
1. Install Wireshark on your PC. (You can download it from the following Web site: http://www.wireshark.org/
2. Connect the PC and the device to the same hub.
3. If you are using a switch, use a switch with port mirroring for the port to which the Wireshark is connected.
4. Start Wireshark.
5. Select the network interface that is currently being used by the PC - on the toolbar, click **Interfaces**, and then in the 'Capture Interfaces' dialog, click the **Options** button corresponding to the network interface:

Figure 7-3: Selecting the Network Interface Currently Used by the PC



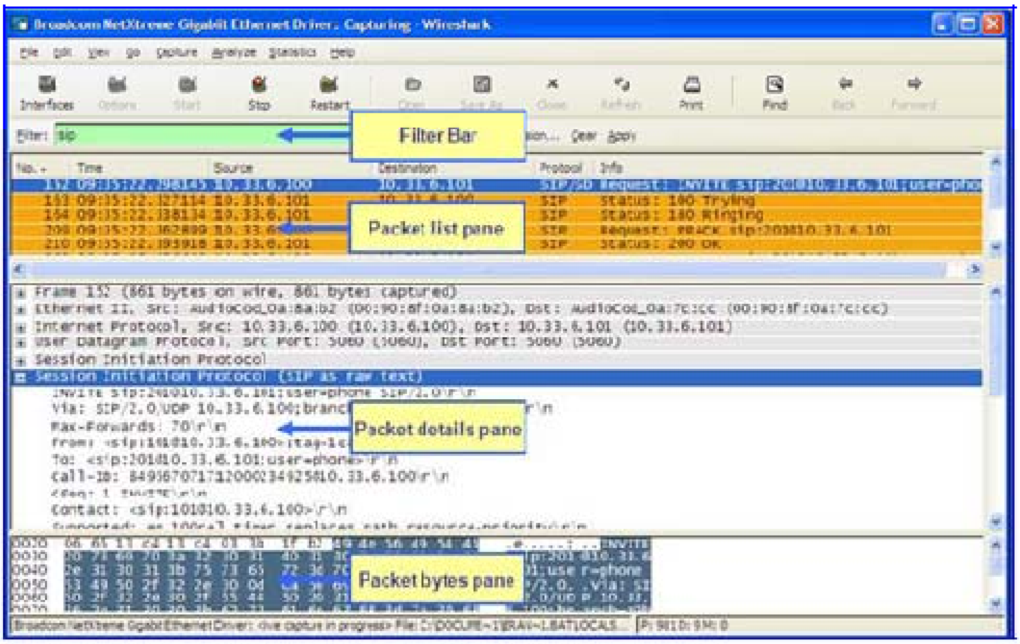
6. In the 'Capture Options' dialog, select the required display options:

Figure 7-4: Configuring Wireshark Display Options



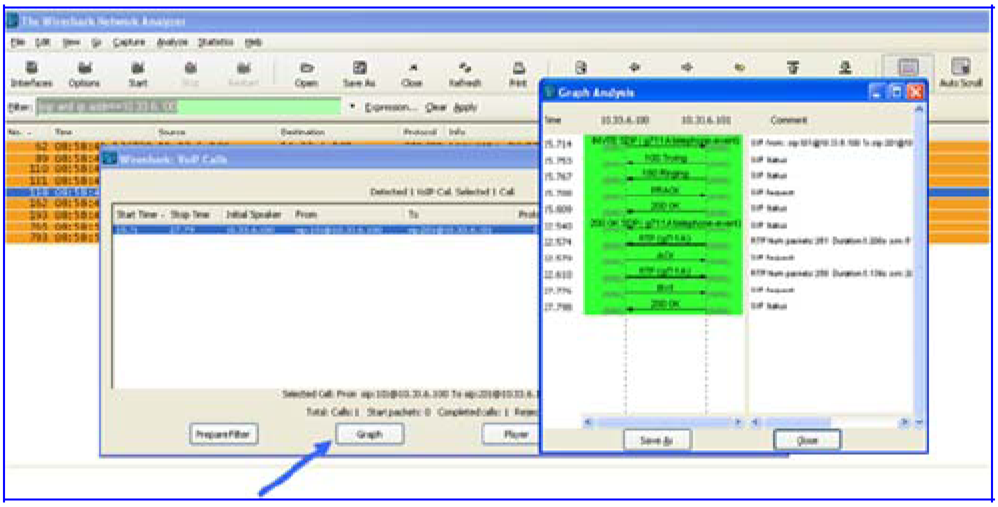
7. Click **Start**.

Figure 7-5: Captures Packets



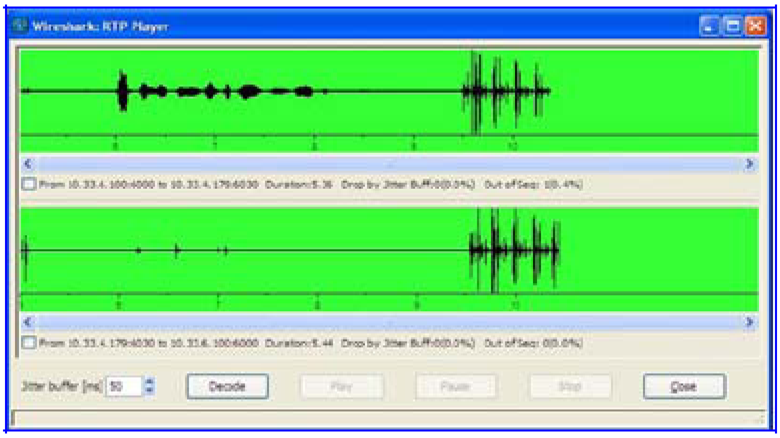
8. To view VoIP call flows, choose **VoIP Calls** from the **Statistics** menu. You can view the statistics in graph format by clicking **Graph**.

Figure 7-6: Viewing VoIP Call Flows



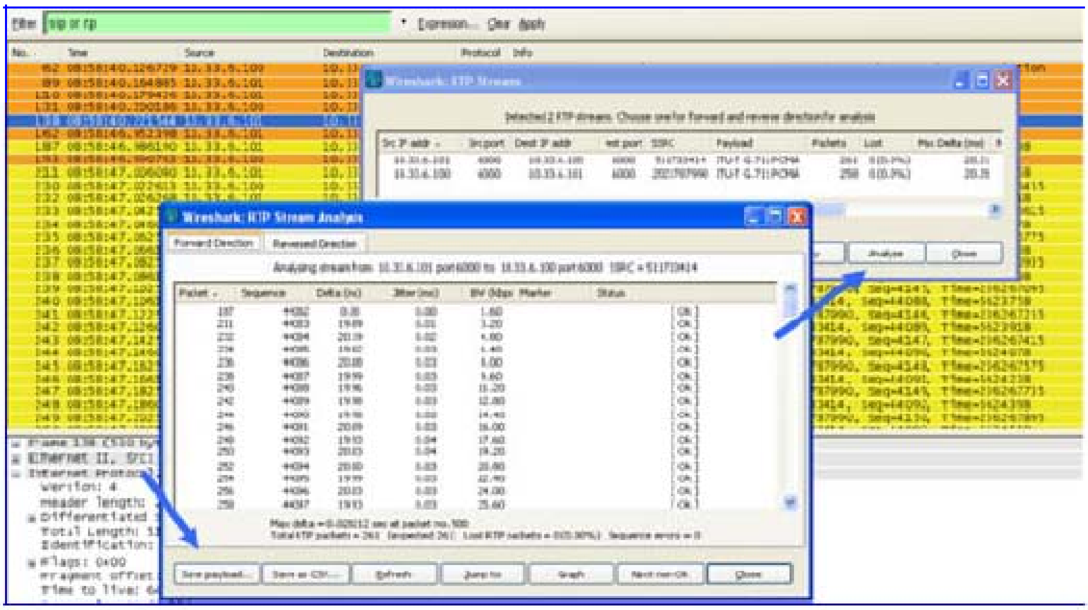
9. To play G.711 RTP streams, click the **Player** button.

Figure 7-7: Playing G.711 RTP Streams



10. To analyze the RTP data stream and extract the audio (which can be played using programs such as CoolEdit) from the data packets (only for G.711), point from the **Statistics** menu to **RTP** and then choose **Stream Analysis**.

Figure 7-8: Analyzing the RTP Data



a. Save the audio payload of the RTP stream to a file.

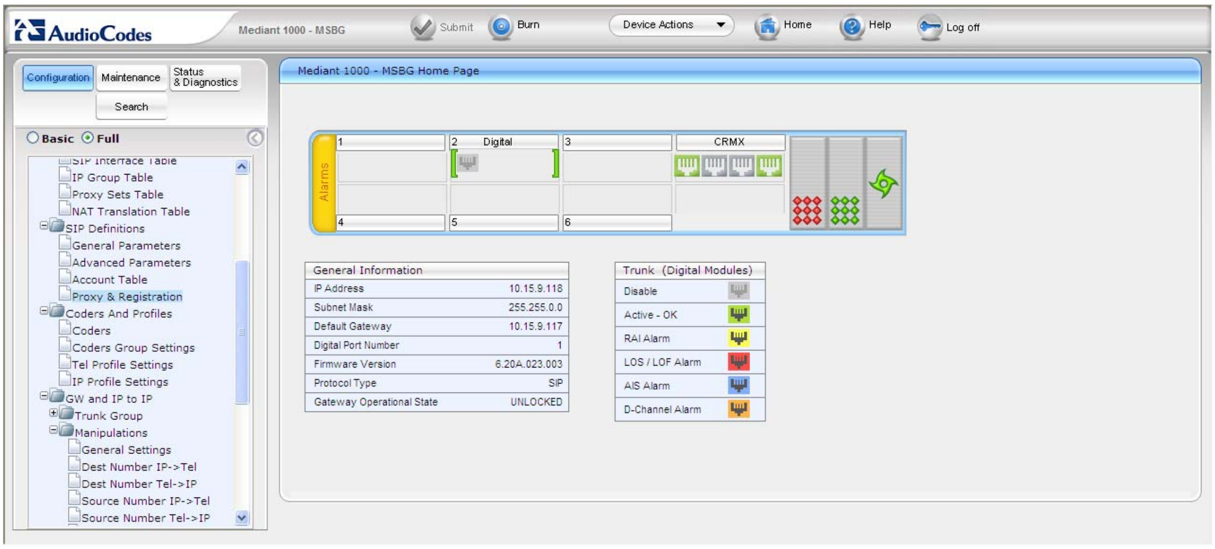
b. Save the Payload as a *.pcm file.

c. Select the ‘forward’ option.

7.2 Verifying Firmware

To verify the firmware load actively running on the device, log into the device and view the firmware version on the product homepage as shown in Figure 7-9. In this instance, firmware load 6.20A.023.003 was used for certification testing.

Figure 7-9: Viewing the Active Firmware Version



Configuration Note

www.audiocodes.com