Mediant 1000 E-SBC

Configuration Note

Connecting Verizon Business SIP Trunk
to
Microsoft® Lync Server
Using
AudioCodes Mediant 1000 E-SBC

Document #: LTRT-31900

Table of Contents

Table of Figures

Reader's Notes

Notice

This document describes the procedure for integrating the Verizon Internet Telephony Service Provider (ITSP) SIP Trunk with Microsoft® Lync Server using AudioCodes Mediant 1000 E-SBC.

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Abbreviations and Terminology

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

Note: Throughout this guide, the term *gateway* refers to AudioCodes' Mediant 1000 E-SBC device.

Reader's Notes

1 Introduction

This document is intended for incumbent IP telephony customers who wish to successfully integrate their Microsoft® Lync environments with a Verizon Business SIP Trunk, using the AudioCodes' E-SBC device.

Microsoft® Unified Communications implements the Microsoft® Lync platforms to combine Enterprise voice, instant messaging, enhanced presence, audio-video conferencing, and e-mail into a familiar, integrated communications experience. Enterprise voice is a Microsoft® Lync server fully featured VoIP solution, which includes connectivity with the telephony network.

For Enterprises wishing to fully utilize their deployed Microsoft® Lync server and not only communicate over IP within the Enterprise, but also outside the Enterprise, a SIP trunk provided by an Internet Telephony Service Provider (ITSP) (such as Verizon IP Trunking service) for connection to the traditional PSTN network, is one of the best possible solutions. Unlike traditional telephony, where bundles of physical wires are delivered from the PSTN service provider to a business, a SIP trunk allows a company to replace these traditional fixed PSTN lines with PSTN connectivity using a SIP Trunking service provider on the Internet.

Connecting Microsoft® Lync directly to a SIP trunk poses various implementation issues. Some ITSPs require SIP user registration, which Microsoft® Lync does not support. In addition, some ITSPs implement SIP over UDP, while Microsoft® Lync implements SIP over TCP or TLS. However, AudioCodes E-SBC devices are capable of providing the interface between the Microsoft® Lync and the ITSP's network, thereby allowing the support for all required functionalities.

Verizon Business has introduced new Internet protocol-based capabilities for its Contact Center Services and VoIP portfolio to help businesses enhance customer-service operations and leverage the benefits of VoIP. IP Trunking enables companies who have already invested in the Microsoft Lync system, to now connect on as few as one converged access lines for both internal and external traffic.

Note: This document is relevant for the Microsoft® Lync environment, as well as for the Microsoft Office Communication Server R2 environment.

Note: The scope of this document does not cover security aspects for connecting the SIP Trunk to the Microsoft Lync environment. Security measures should be implemented in accordance with your organization’s security policies. For basic security guidelines, see the ‘AudioCodes Security Guidelines’.

To facilitate the understanding of the configuration setup, the procedures described in this document are based on an example scenario as described in Section 2 on page 11.

Reader's Notes

2 Example Scenario

The procedures described in this document describe the following example scenario:

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

The setup requirements are as follows:

- While the Microsoft® Lync Server 2010 environment is located on the Enterprise's Local Area Network (LAN), the Verizon Business SIP Trunks are located on the WAN.
- Microsoft® Lync Server 2010 works with the TLS transport type, while the Verizon Business SIP trunk works on the SIP over UDP transport type.
- Microsoft® Lync Server 2010 supports G.711a/ulaw, while Verizon Business SIP Trunk supports also G.729 coder type.
- Support for call forwarding
- Support for early media handling

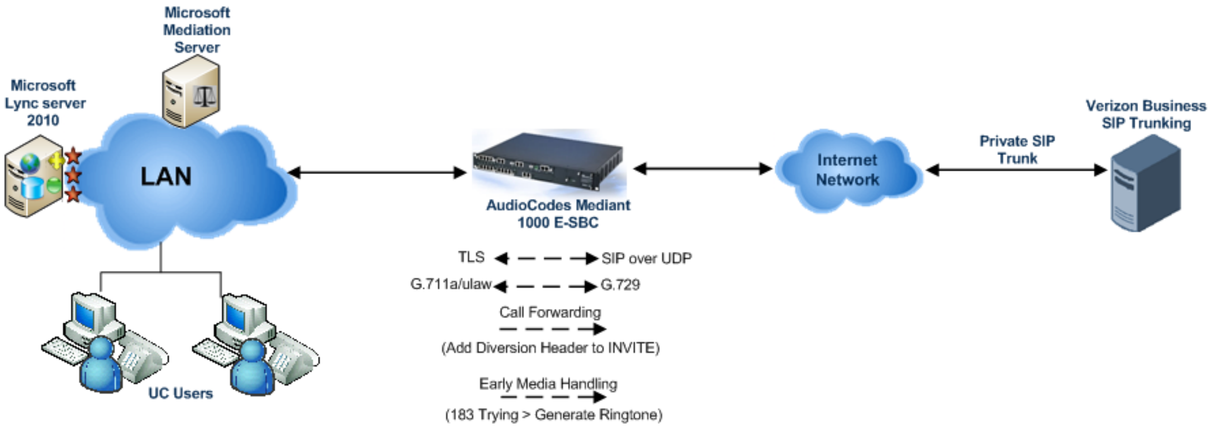
The AudioCodes **Mediant 1000B** E-SBC media gateway, which enables smooth integration for supporting all the required functionalities, was used to implement this solution.

The Mediant 1000 E-SBC is a networking E-SBC 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 E-SBC device offers enhanced dialing plans and voice routing capabilities along with SIP-to-SIP mediation, allowing enterprises to implement SIP Trunking services (IP-to-IP call routing) and IP-based Unified Communications, as well as flexible PSTN and legacy PBX connectivity.

A similar configuration is implemented in the case where the customer uses the Microsoft® Office Communication Server 2007 R2.

Figure 2-1 below illustrates the above example scenario.

Figure 2-1: Microsoft Lync and Verizon Business SIP Trunking



Reader's Notes

3 Configuring Lync Server 2010

This section describes how to configure the Lync Server 2010 to operate with the Mediant 1000 E-SBC. This section describes the following procedures:

1. Configuring the Mediant 1000 E-SBC as a 'PSTN Gateway'. See Section 3.1 on page 13.
2. Associating the 'PSTN Gateway' with the Mediation Server. See Section 3.2 on page 18.
3. Configuring a 'Route' to utilize the SIP trunk connected to the Mediant 1000 E-SBC. See Section 3.3 on page 24.

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

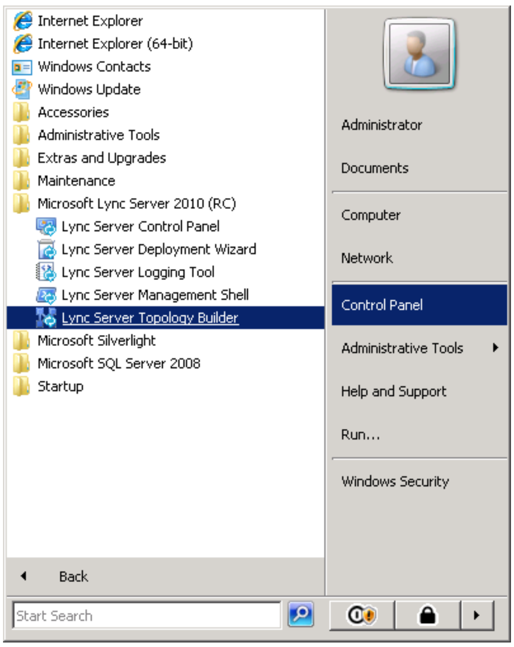
3.1 Configuring the Mediant 1000 E-SBC as a 'PSTN Gateway'

This section describes how to configure the Mediant 1000 E-SBC as a PSTN Gateway.

➢ **To configure the Mediant 1000 E-SBC as a PSTN Gateway and associating it with the Mediation Server:**

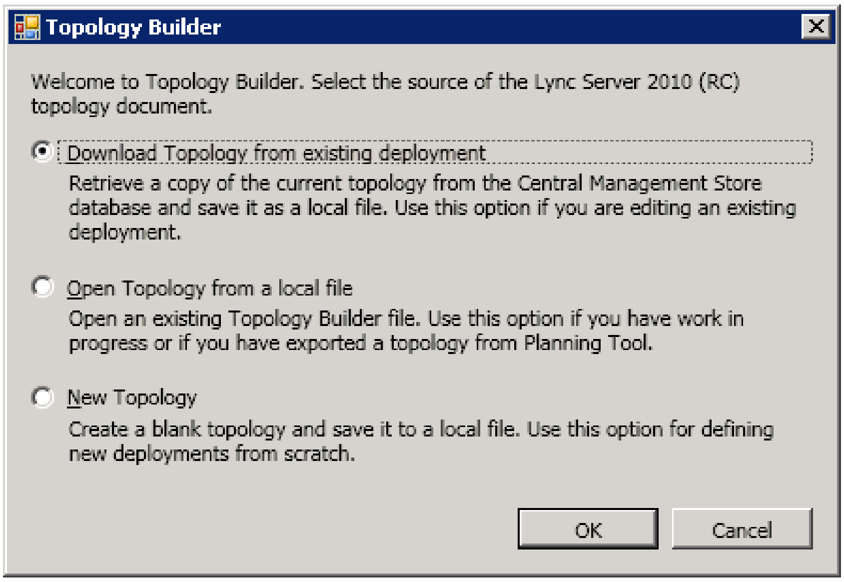
1. On the server where the Topology Builder is located, start the Lync Server 2010 **Topology Builder**: Click **Start**, select **All Programs**, then select **Lync Server Topology Builder**.

Figure 3-1: Starting the Lync Server Topology Builder



The following screen is displayed:

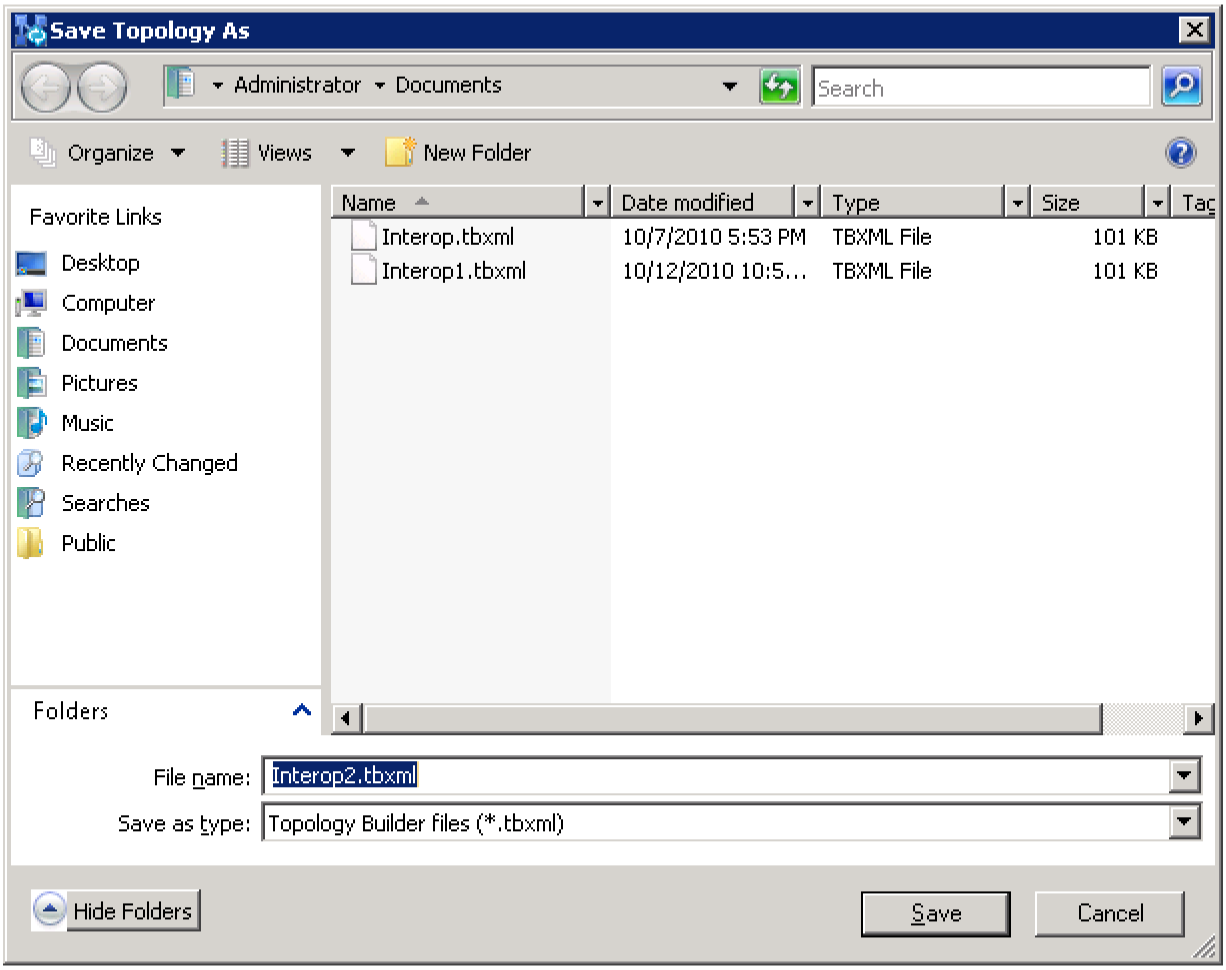
Figure 3-2: Topology Builder Options



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

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

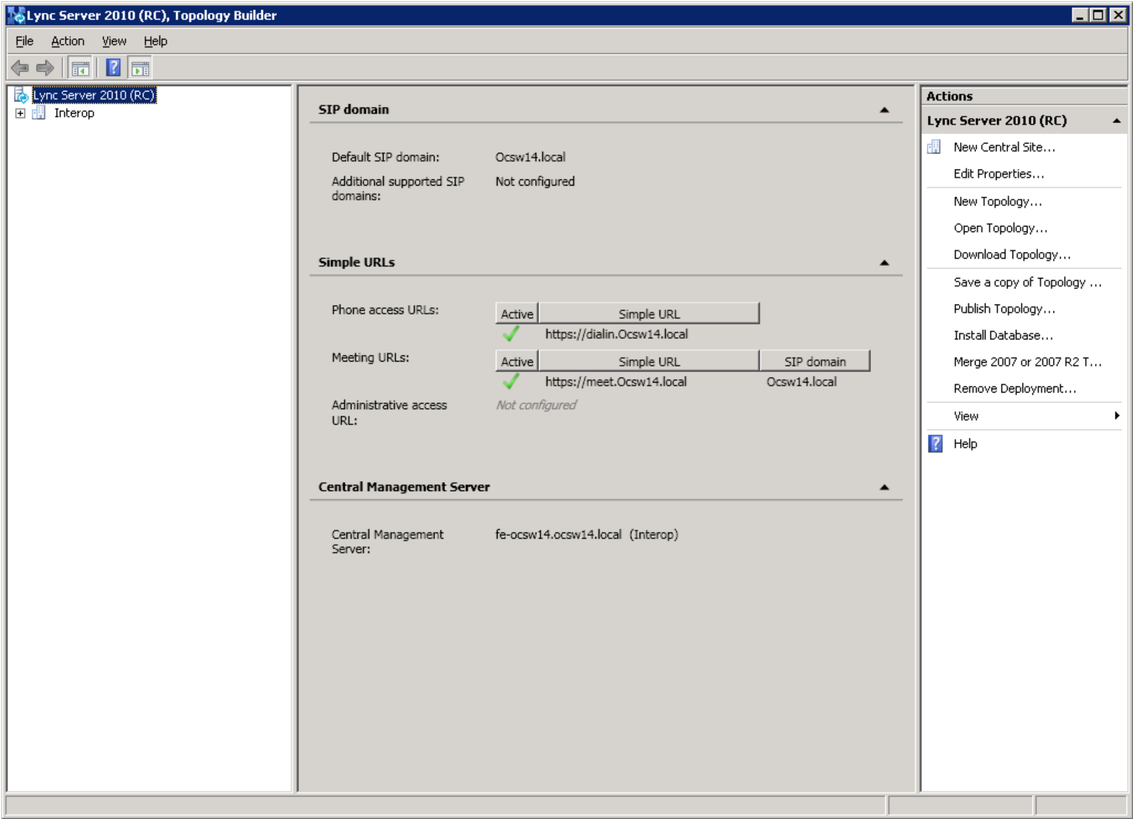
Figure 3-3: Save Topology



3. Enter new **File Name** and **Save** – this action enables you to rollback from any changes you make during the installation.

   The Topology Builder screen with the topology downloaded is displayed.

Figure 3-4: Downloaded Topology



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

Figure 3-5: New IP/PSTN Gateway

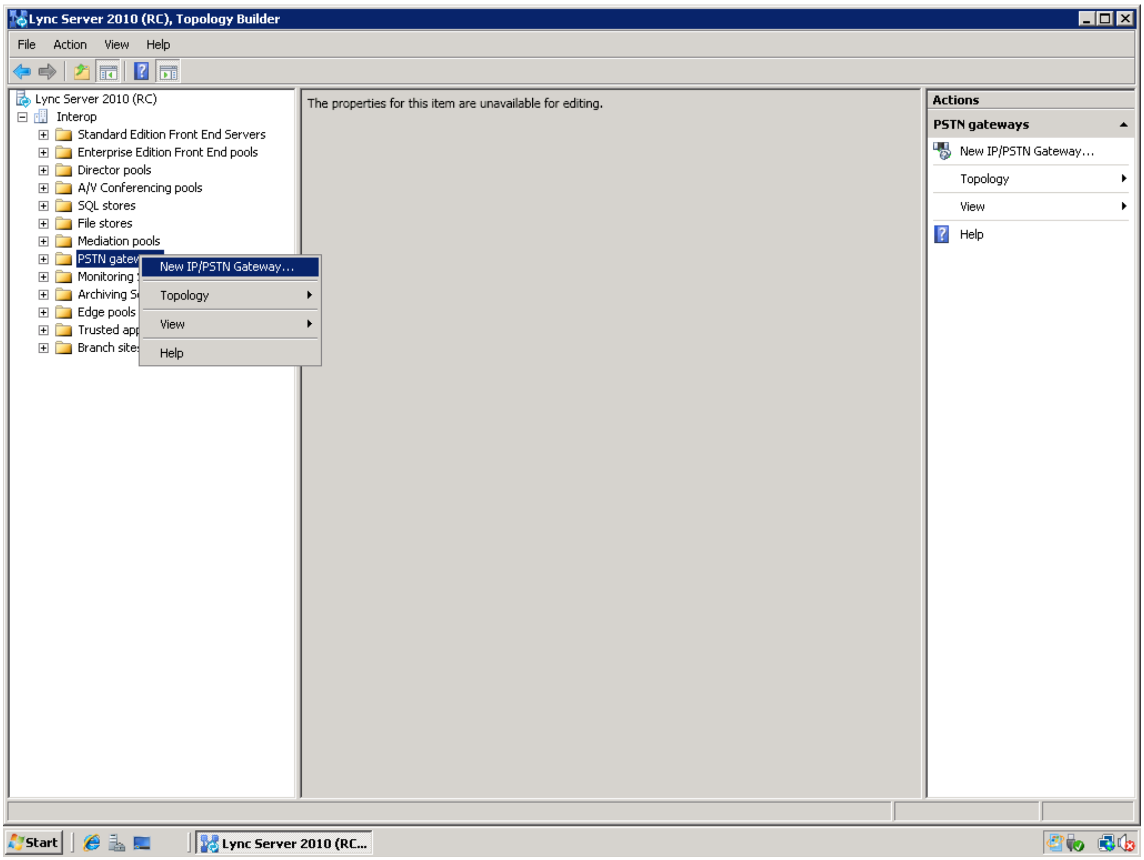
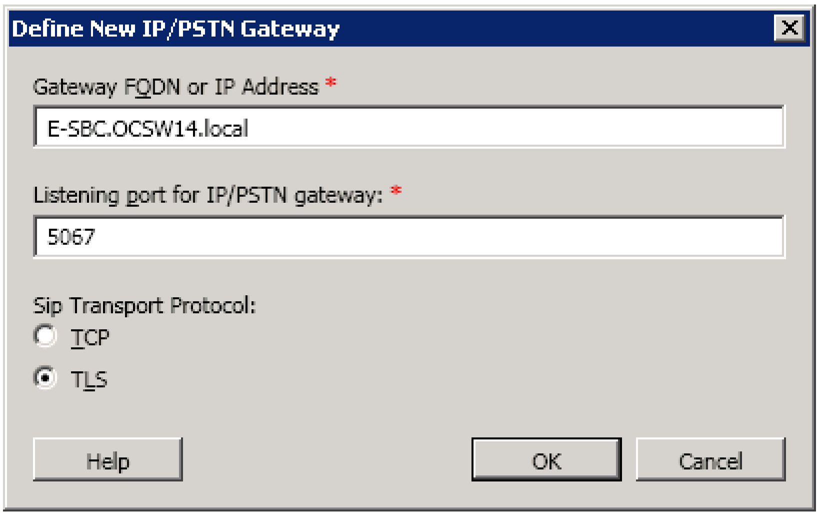


Figure 3-6: Define New IP/PSTN Gateway

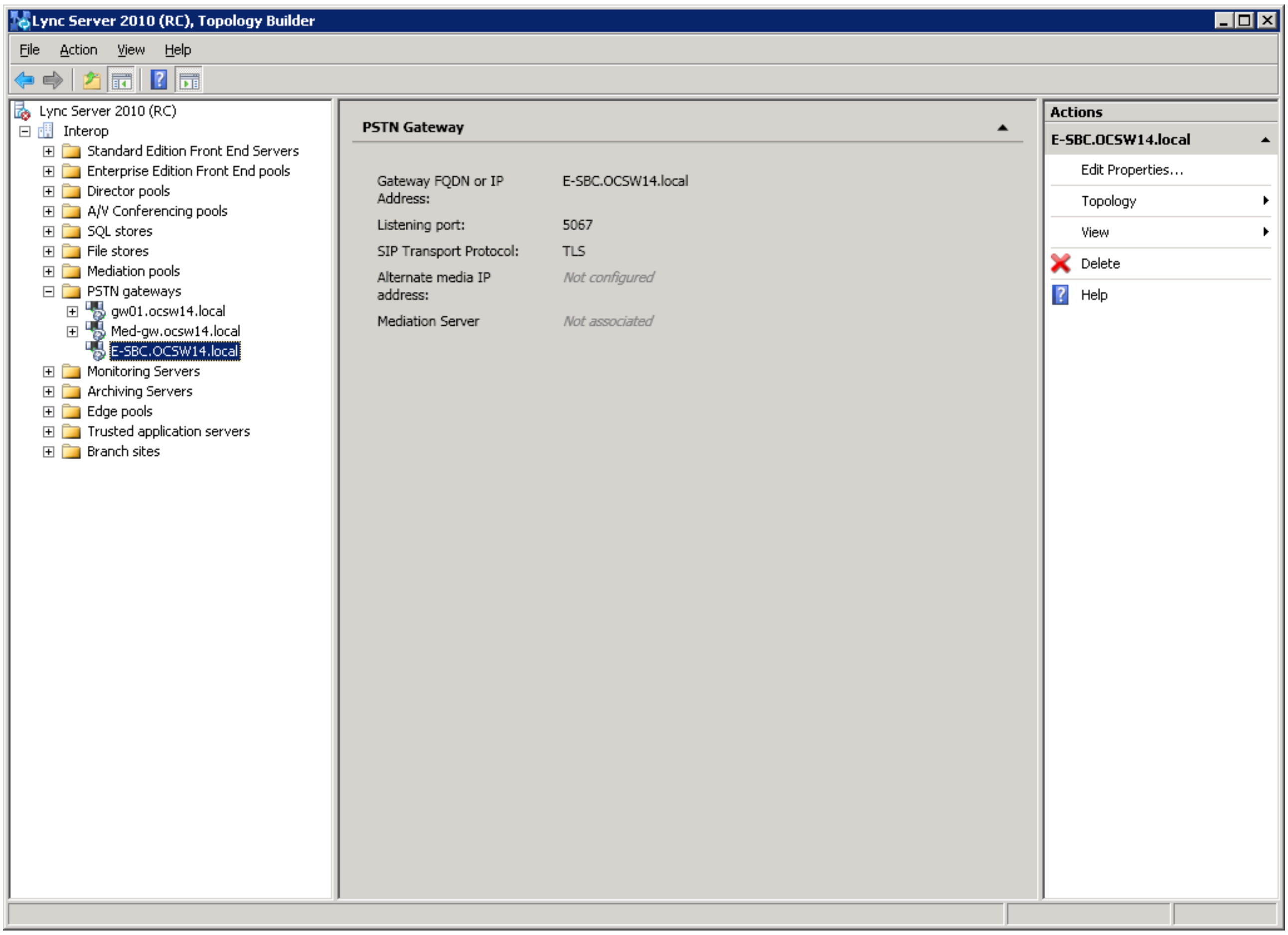


5. Enter the FQDN of the Mediant 1000 E-SBC (i.e. 'E-SBC.OCSW14.local') and click **OK**.

   Note that the listening port for the Gateway is '5067' and the transport type is 'TLS'.

   The Mediant 1000 E-SBC is now added as a 'PSTN Gateway'.

Figure 3-7: PSTN Gateway



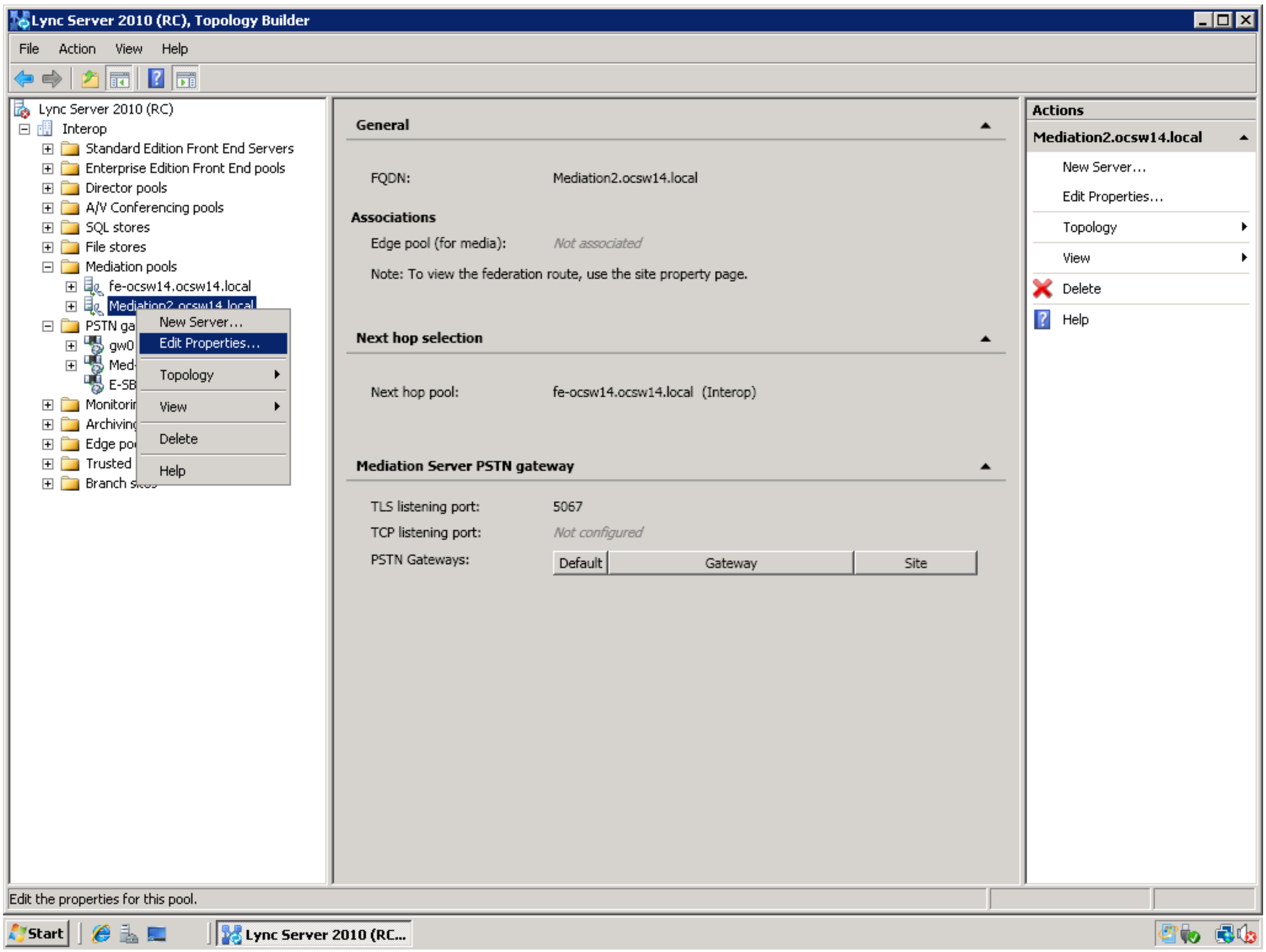
3.2 Associating the 'PSTN Gateway' with the Mediation Server

This section describes how to associate the 'PSTN Gateway' with the Mediation Server.

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

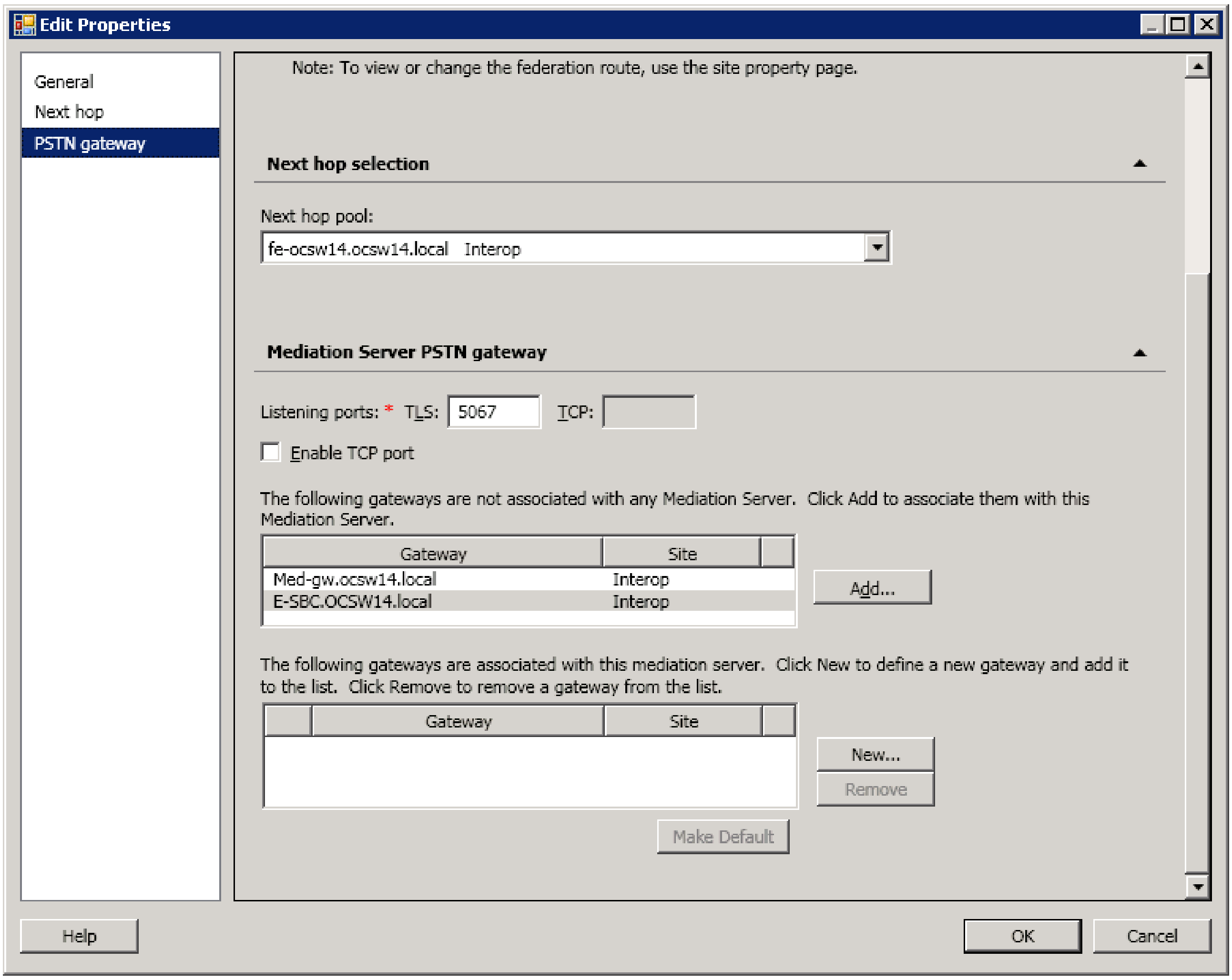
1. Right-click on the **Mediation Server** to use with the E-SBC (i.e. Mediation2.OCSW14.local) and choose **Edit Properties**.

Figure 3-8: Associating Mediation Server with PSTN Gateway



The following screen is displayed:

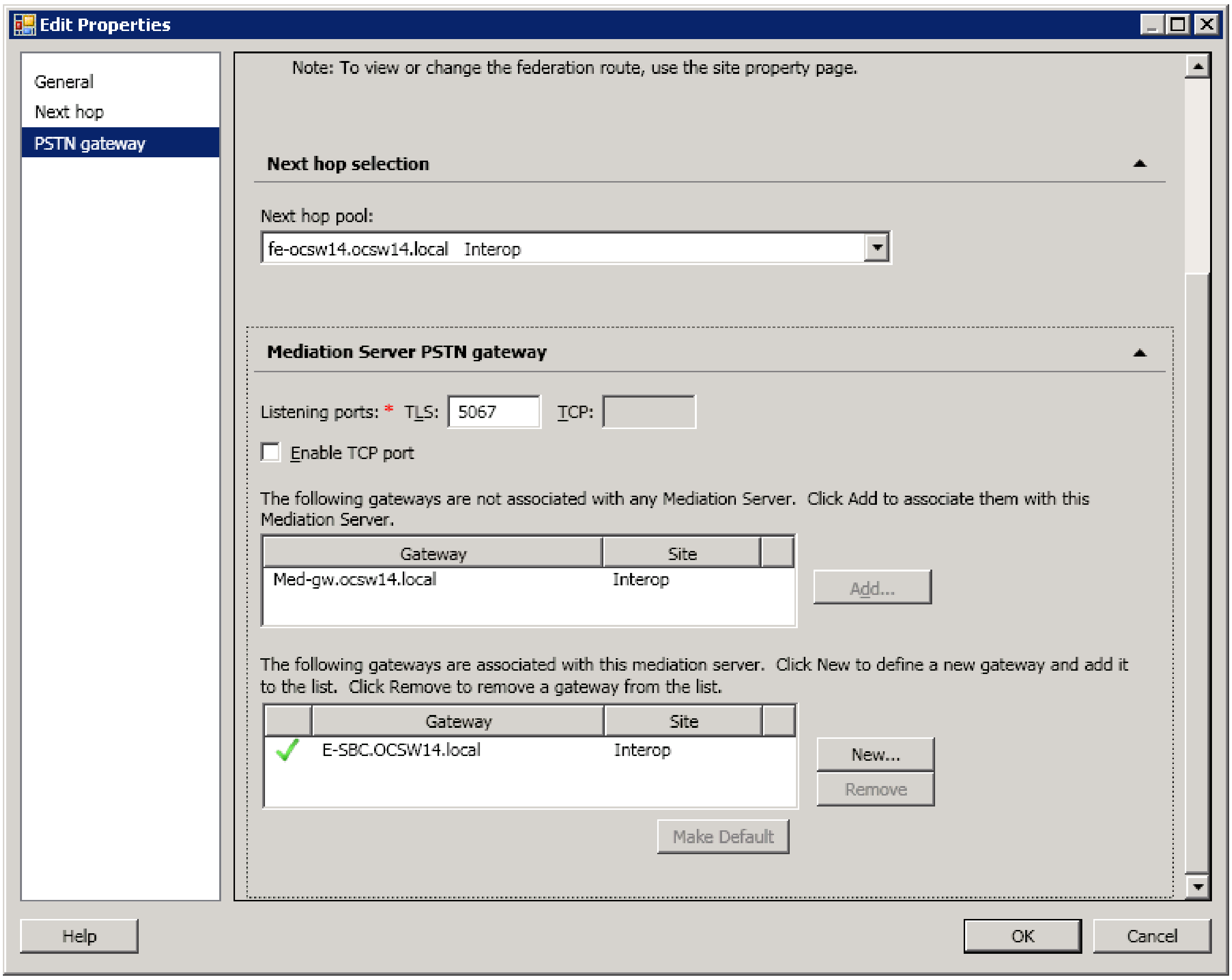
Figure 3-9: Before Associating PSTN Gateway to a Mediation Server



2. In the top-left corner, choose **PSTN gateway** and in the Mediation Server PSTN gateway pane, mark the E-SBC 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 gateways not associated with the Mediation Server and one including a list of gateways associated with the Mediation server.

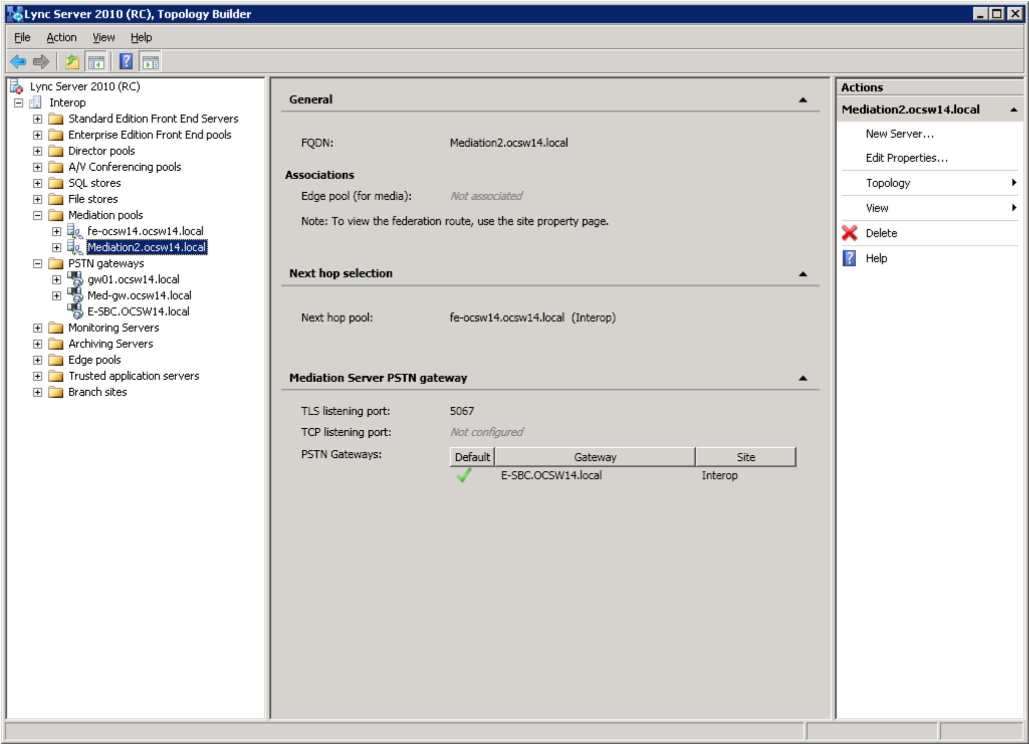
Figure 3-10: After Associating PSTN Gateway to Mediation Server



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

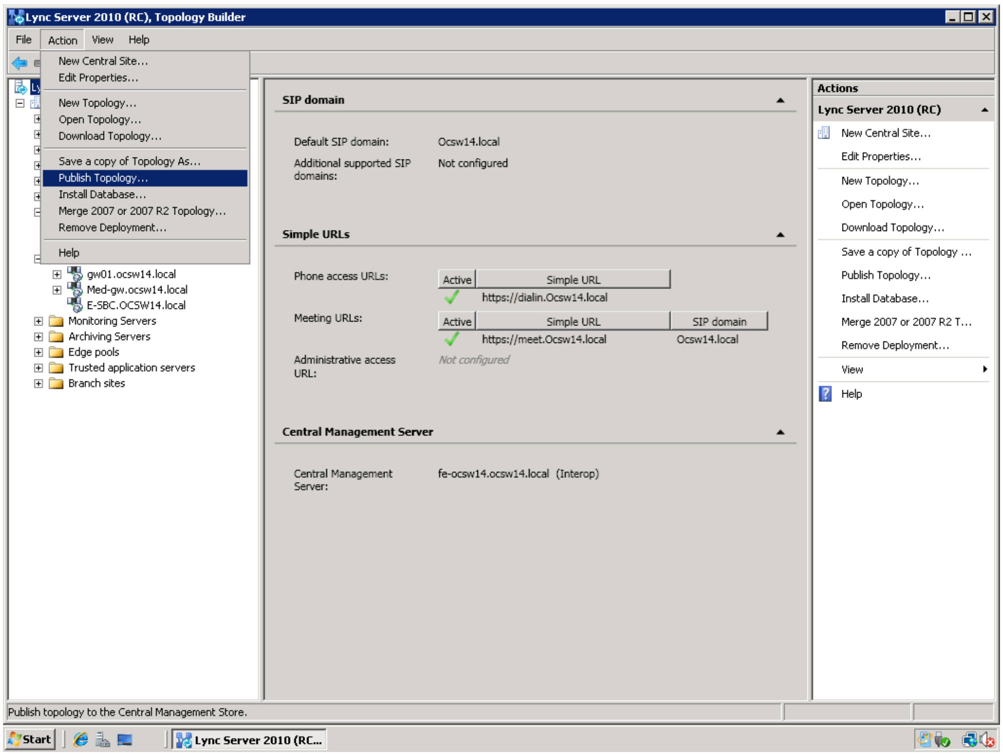
3. Click **OK**.

Figure 3-11: Media Server PSTN Gateway Association Properties



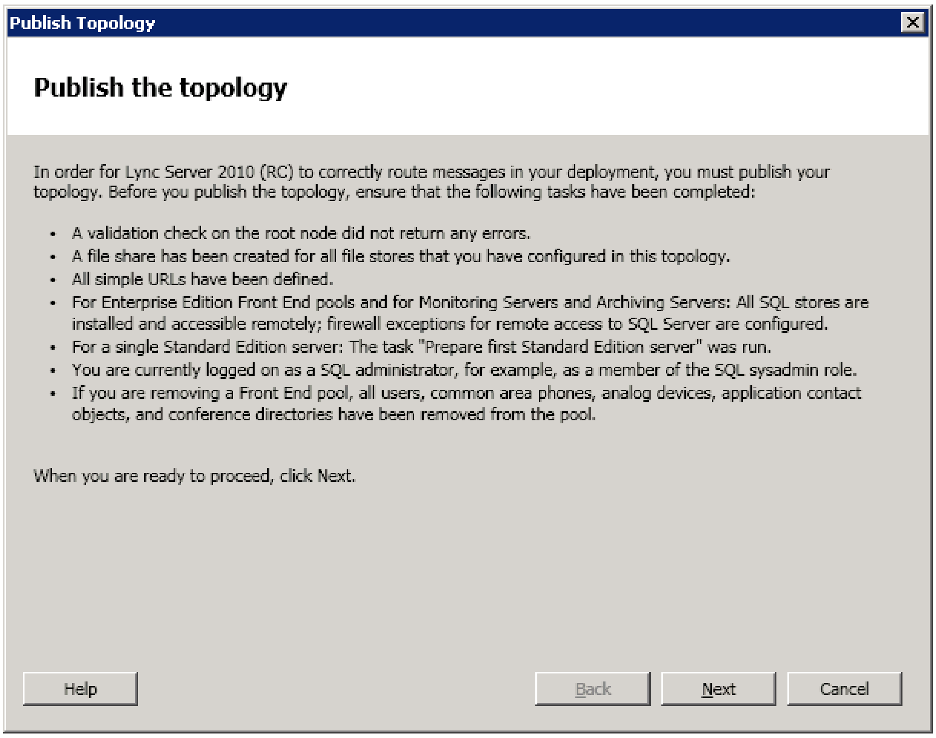
4. In the Lync Server main menu, choose **Action > Publish Topology**.

Figure 3-12: Publishing Topology



The Publish Topology screen is displayed.

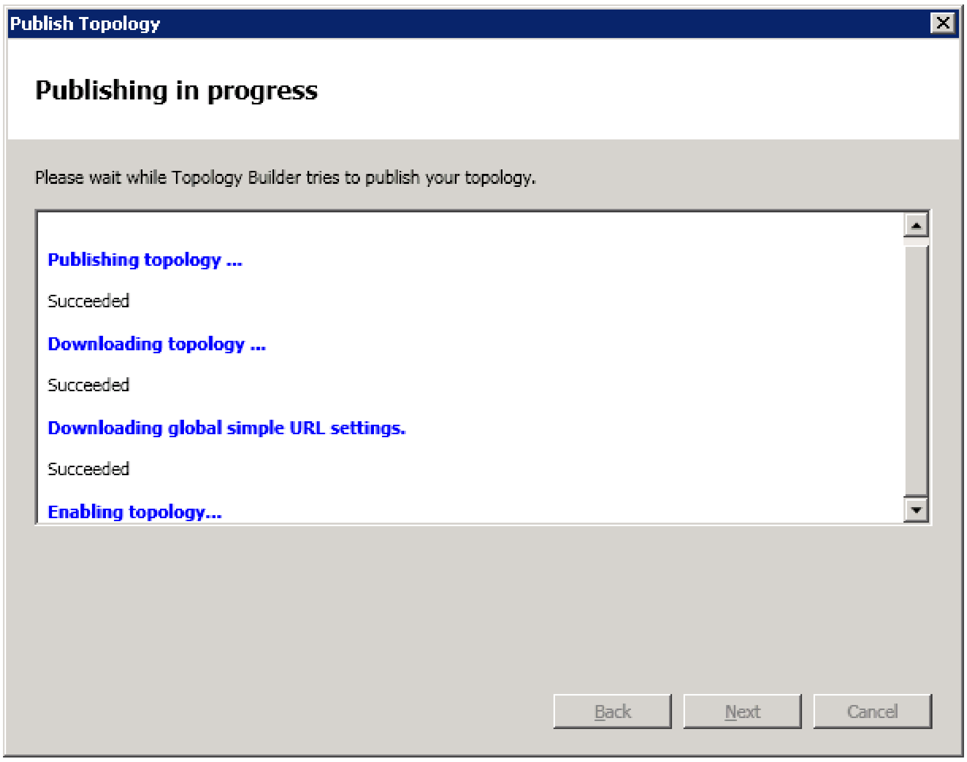
Figure 3-13:Publish Topology Confirmation



5. Click **Next**.

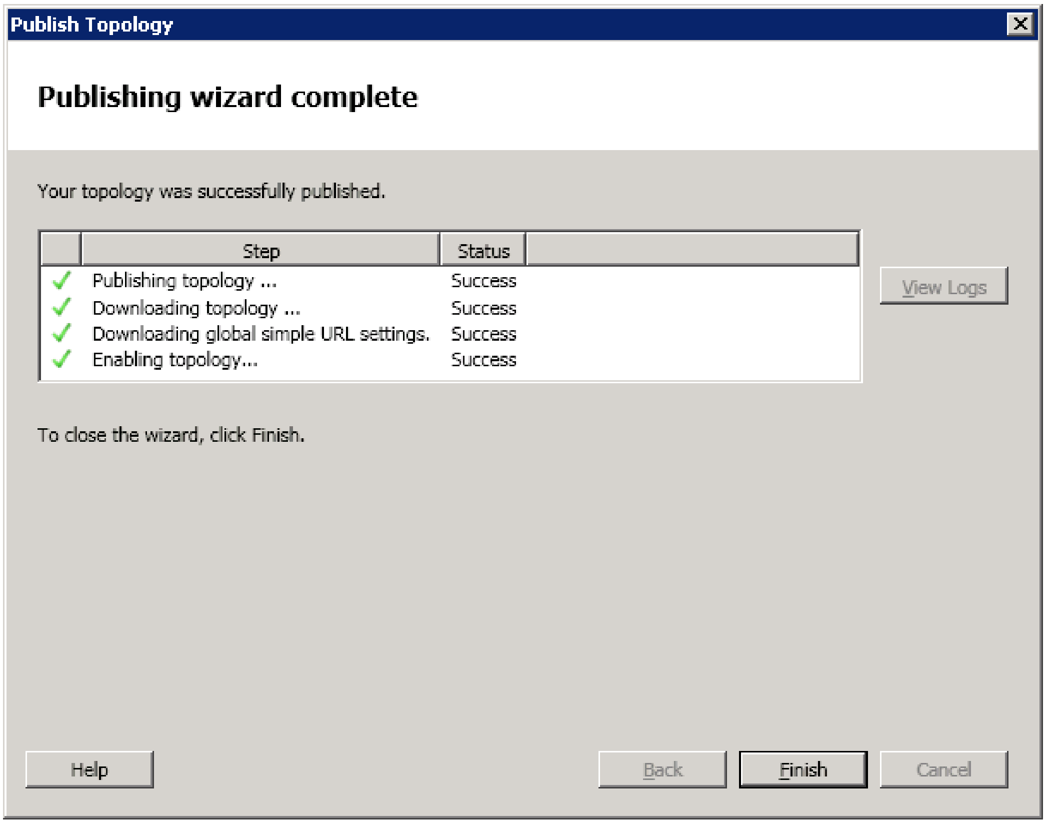
   The Topology Builder attempts to publish your topology.

Figure 3-14: Publish Topology Confirmation screen



Wait until the publish topology process has ended successfully.

Figure 3-15: Publish Topology Successfully Completed



6. Click **Finish**.

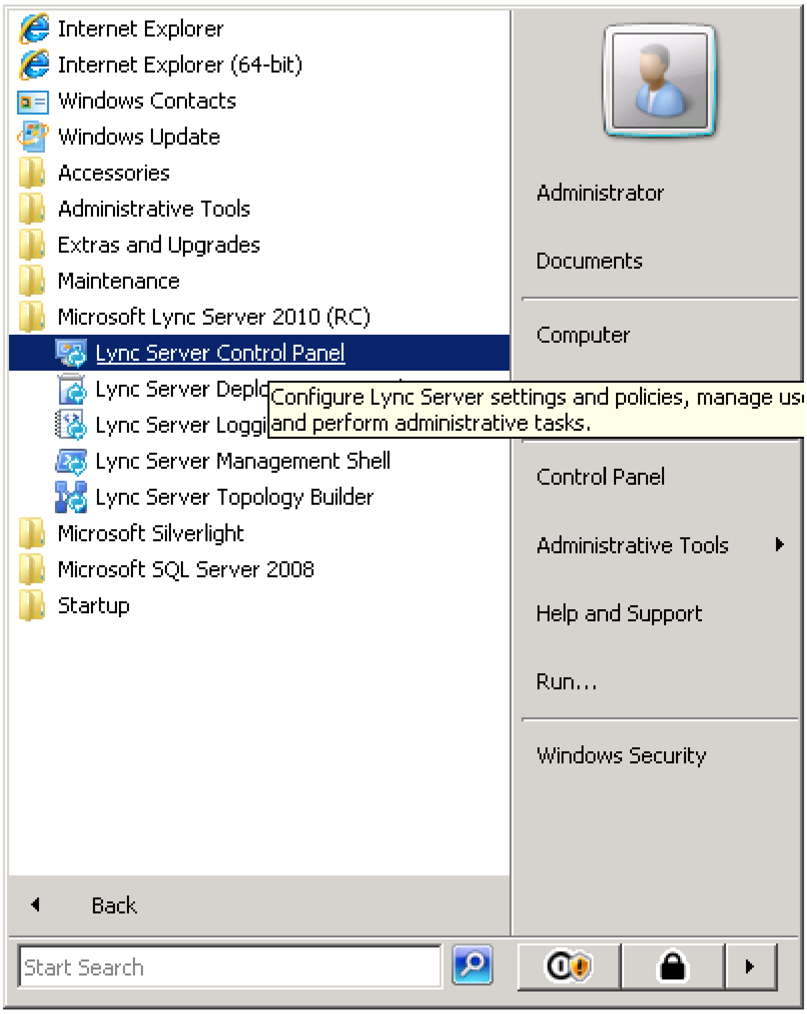
3.3 Configuring the 'Route' on the Lync Server 2010

This section describes how to configure a 'Route' on the Lync server and associate it with the E-SBC PSTN gateway.

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

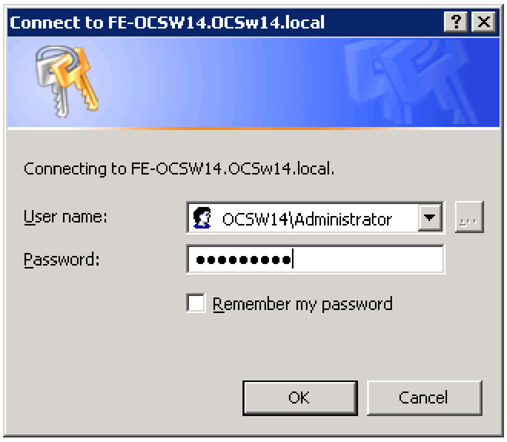
1. Open the Communication Server Control Panel (CSCP), click **Start**, select **All Programs**, and select **Lync Server Control Panel**.

Figure 3-16: Lync Server Control Panel



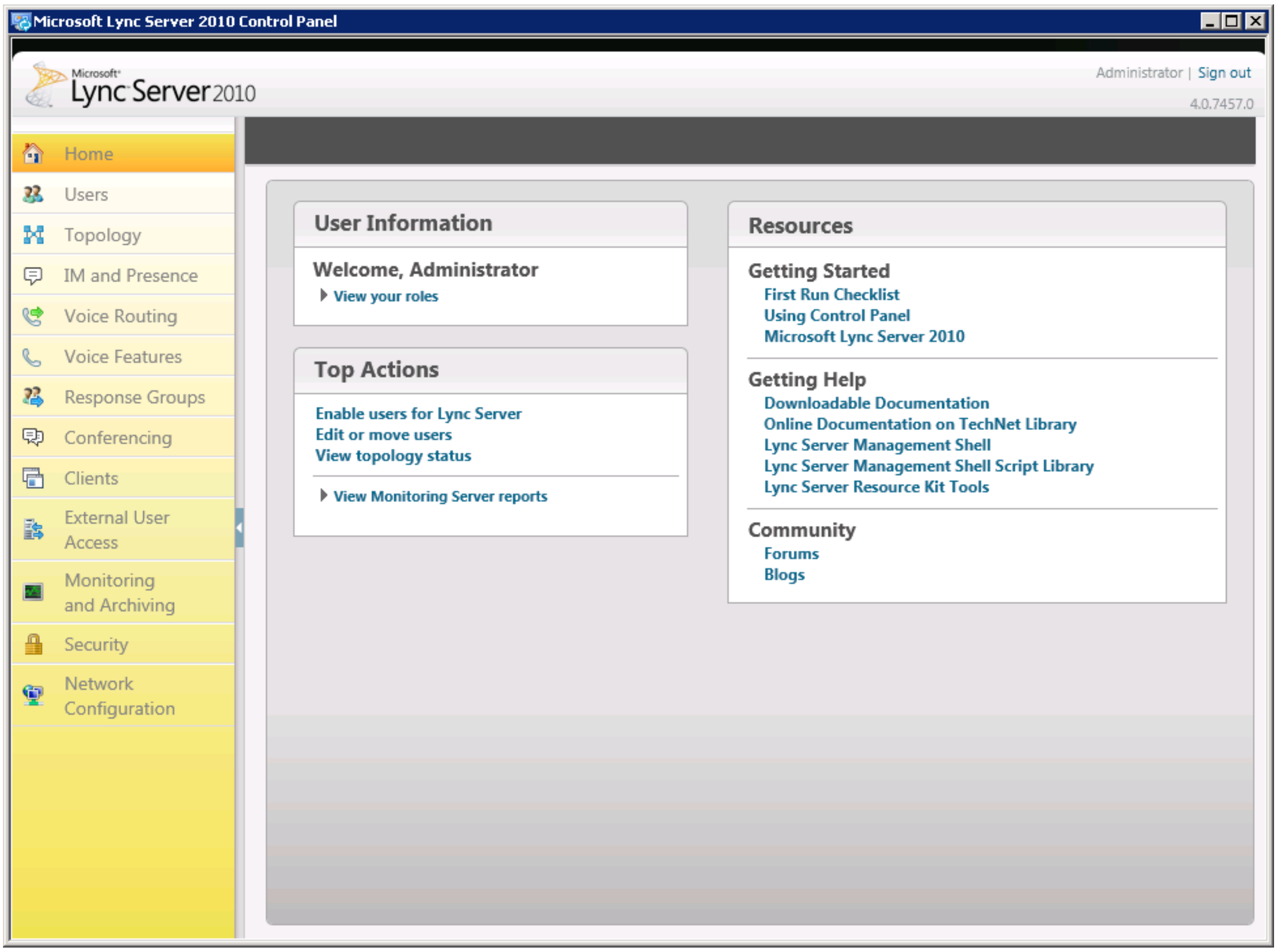
2. You are prompted for credentials; enter your domain username and password.

Figure 3-17: Lync Server Credentials



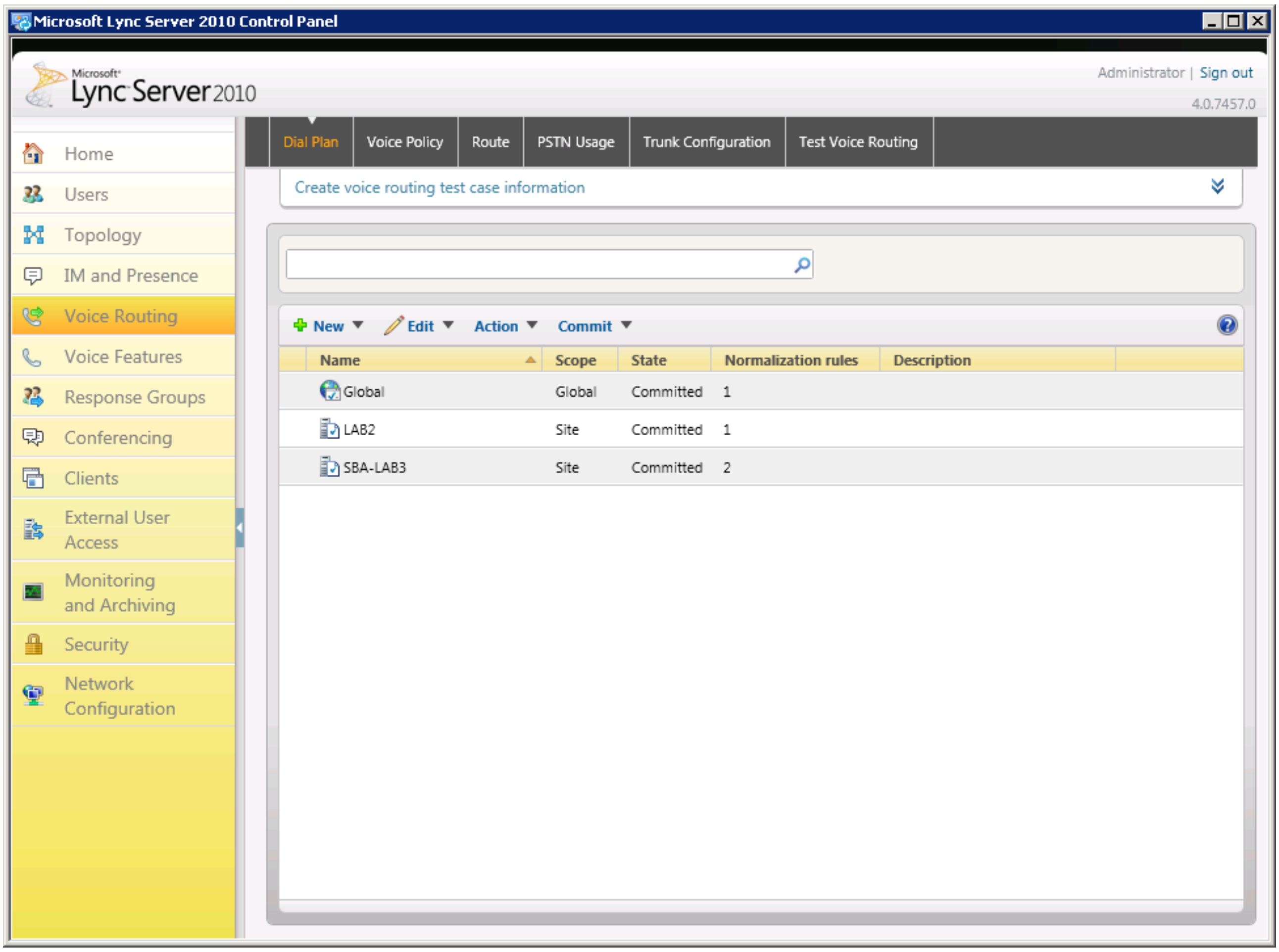
The CSCP Home page is displayed.

Figure 3-18: CSCP Home page



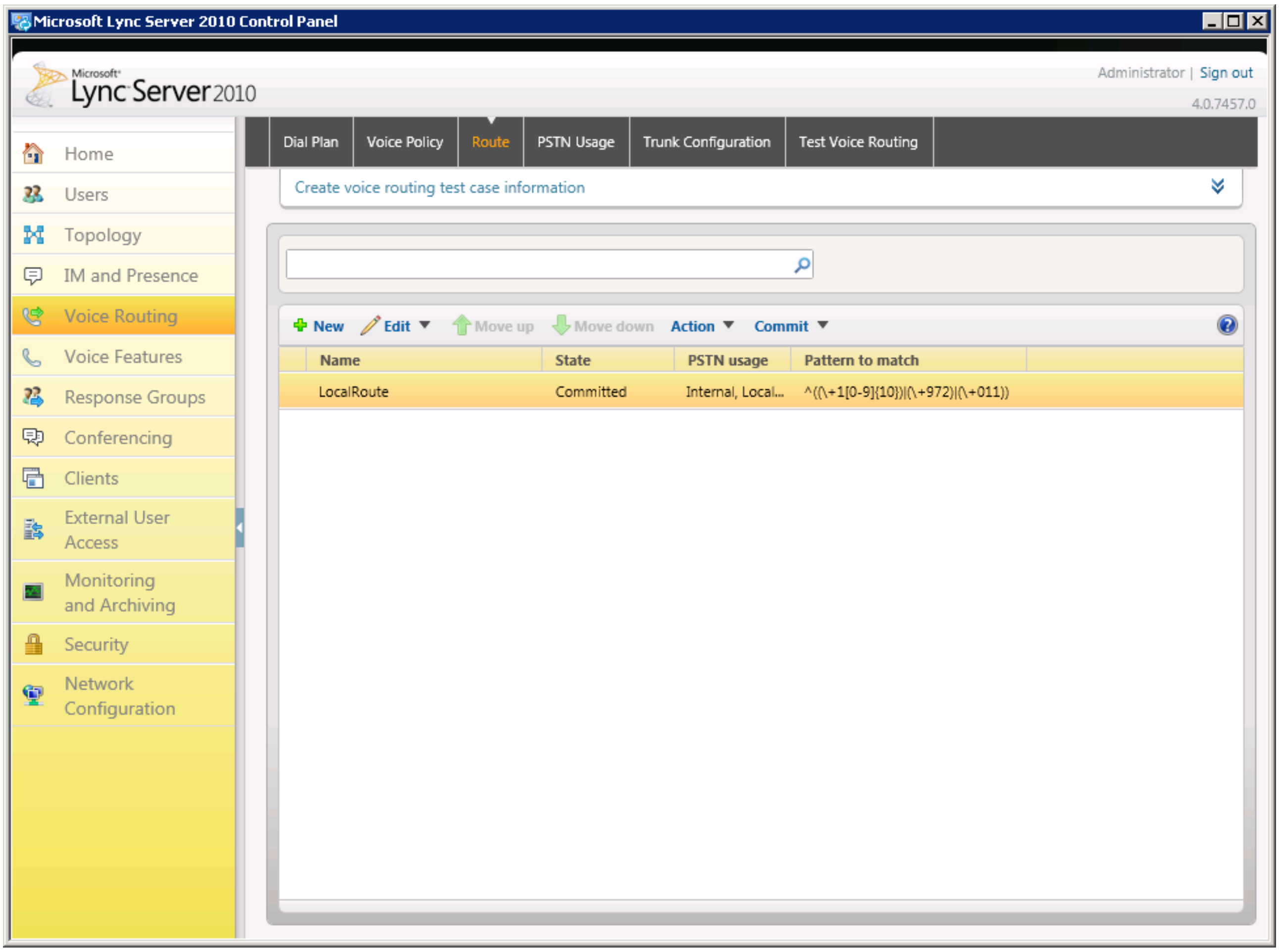
3. In the Navigation pane, select the **'Voice Routing'** option.

Figure 3-19: Voice Routing Option



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

Figure 3-20: Route Option

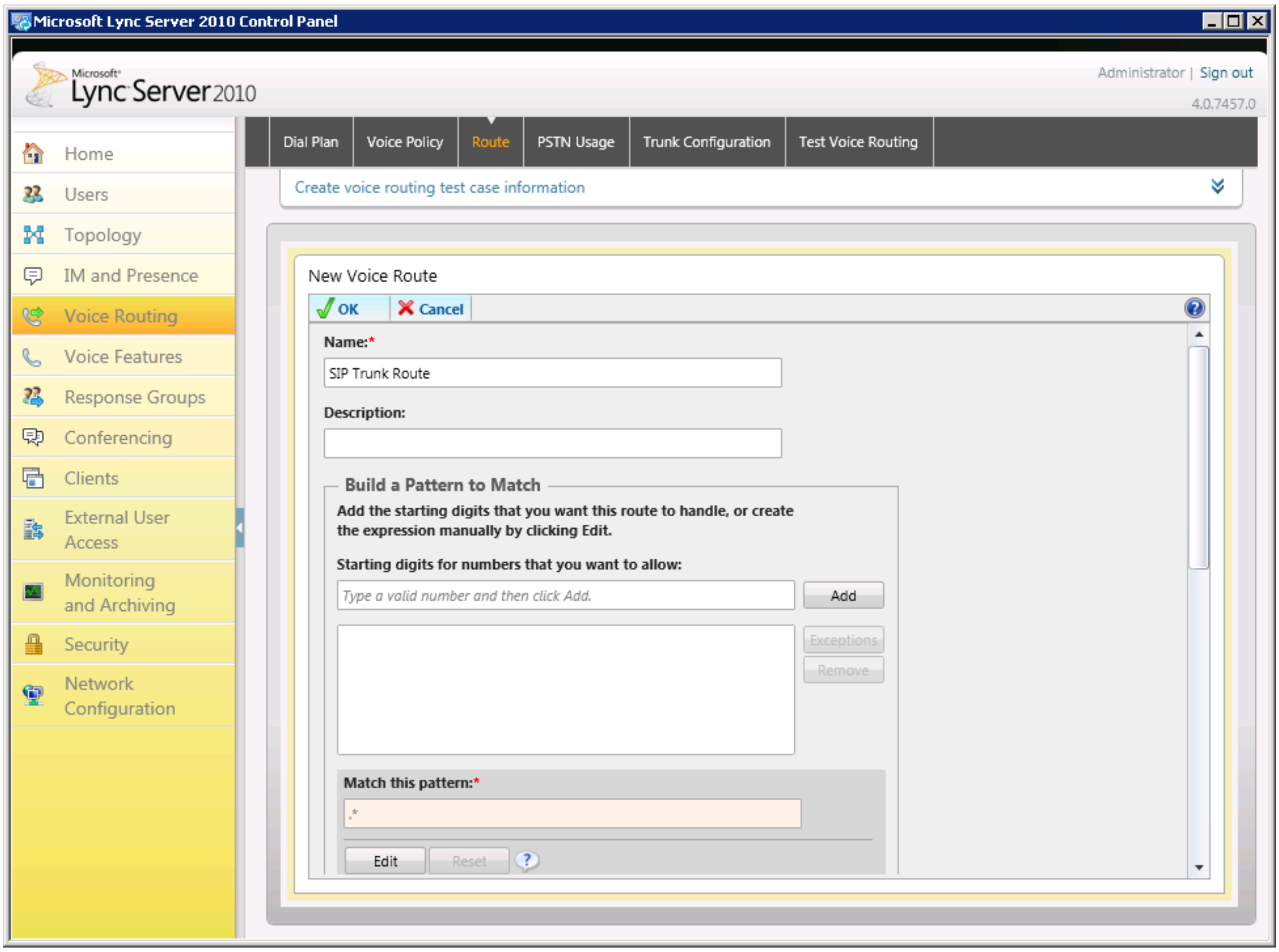


5. In the content area toolbar, click  .

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 wish this route to handle. In this example, the pattern to match is '*', which implies "to match all numbers".

7. Click **Add**.

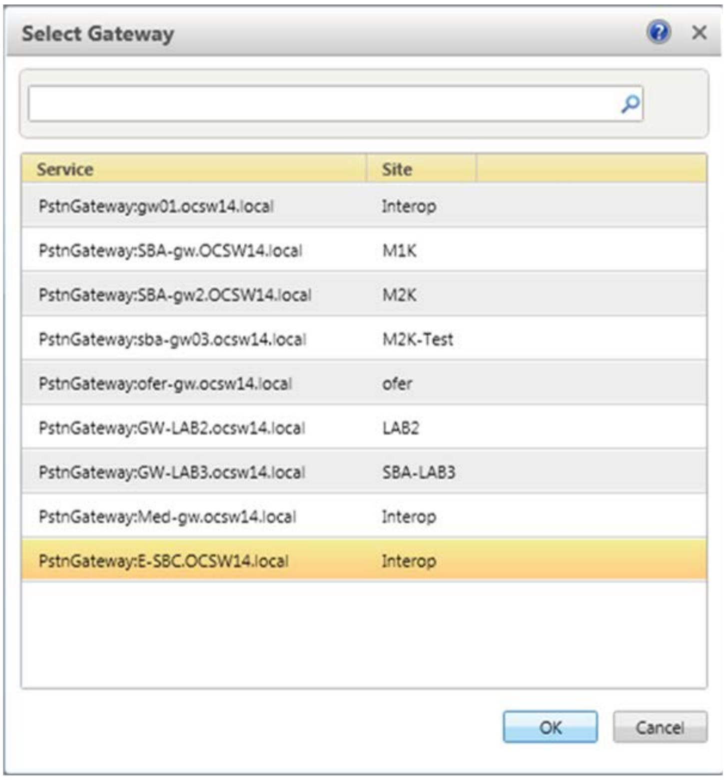
Figure 3-21: Adding New Voice Route



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

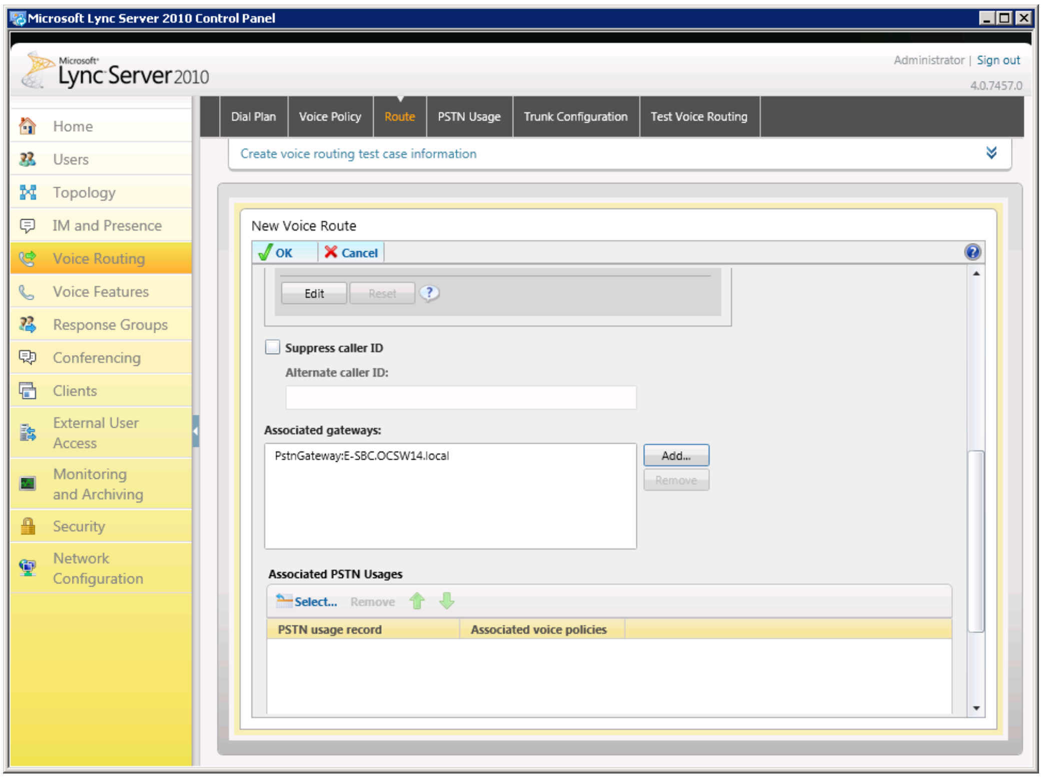
   A list of all the deployed Gateways is displayed.

Figure 3-22: List of Deployed Gateways



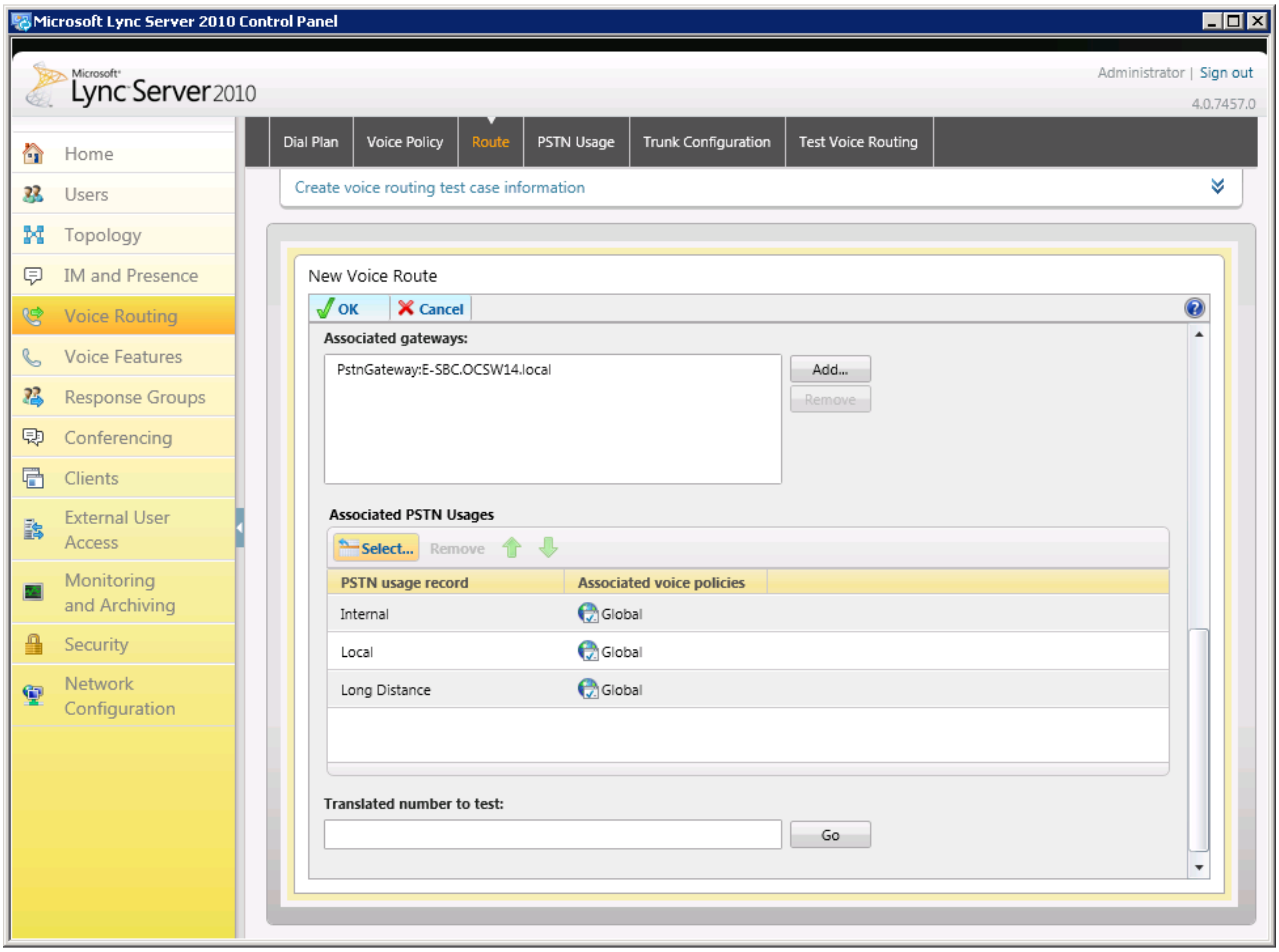
9. Select the E-SBC Gateway you created above and click **OK**.

Figure 3-23: Selecting the E-SBC Gateway



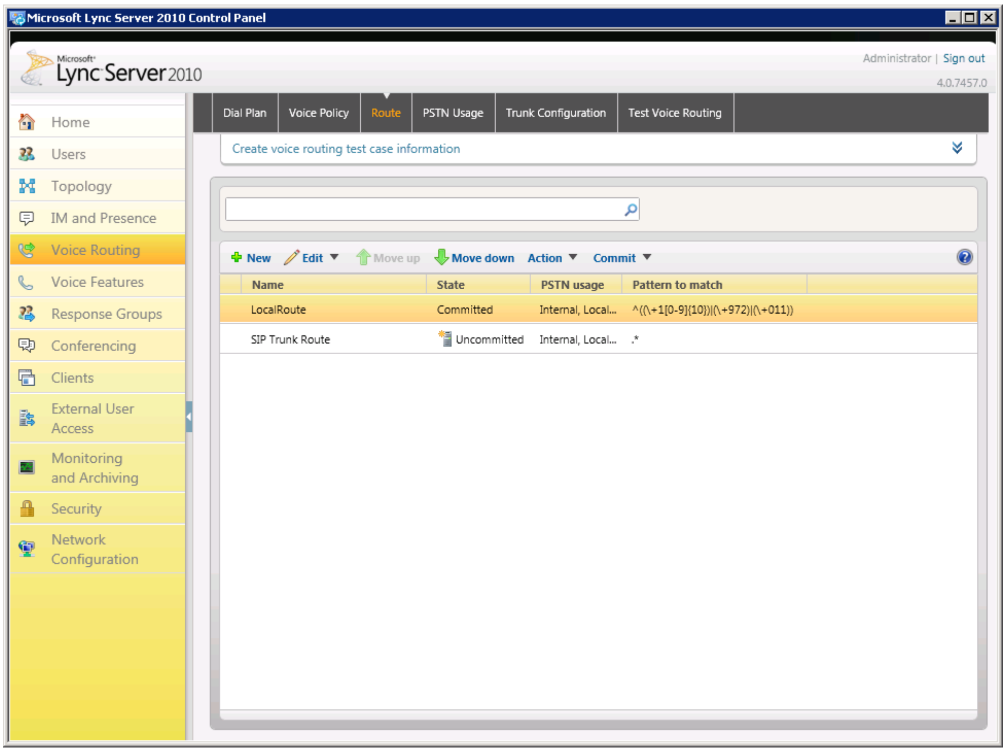
10. Associate a PSTN Usage to this route. In the **Associated PSTN Usages** toolbar, click **Select** and add the associated PSTN Usage.

Figure 3-24: Associating PSTN Usage to E-SBC Gateway



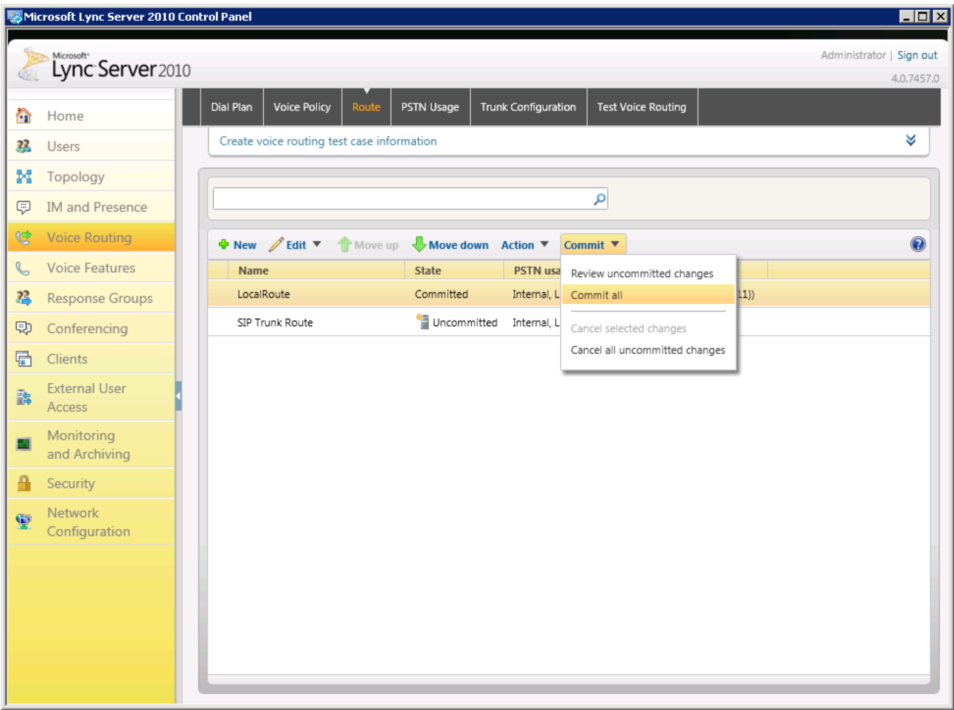
11. Click the **OK** button in the toolbar at the top of the New Voice Route pane.

Figure 3-25: Confirmation of New Voice Route



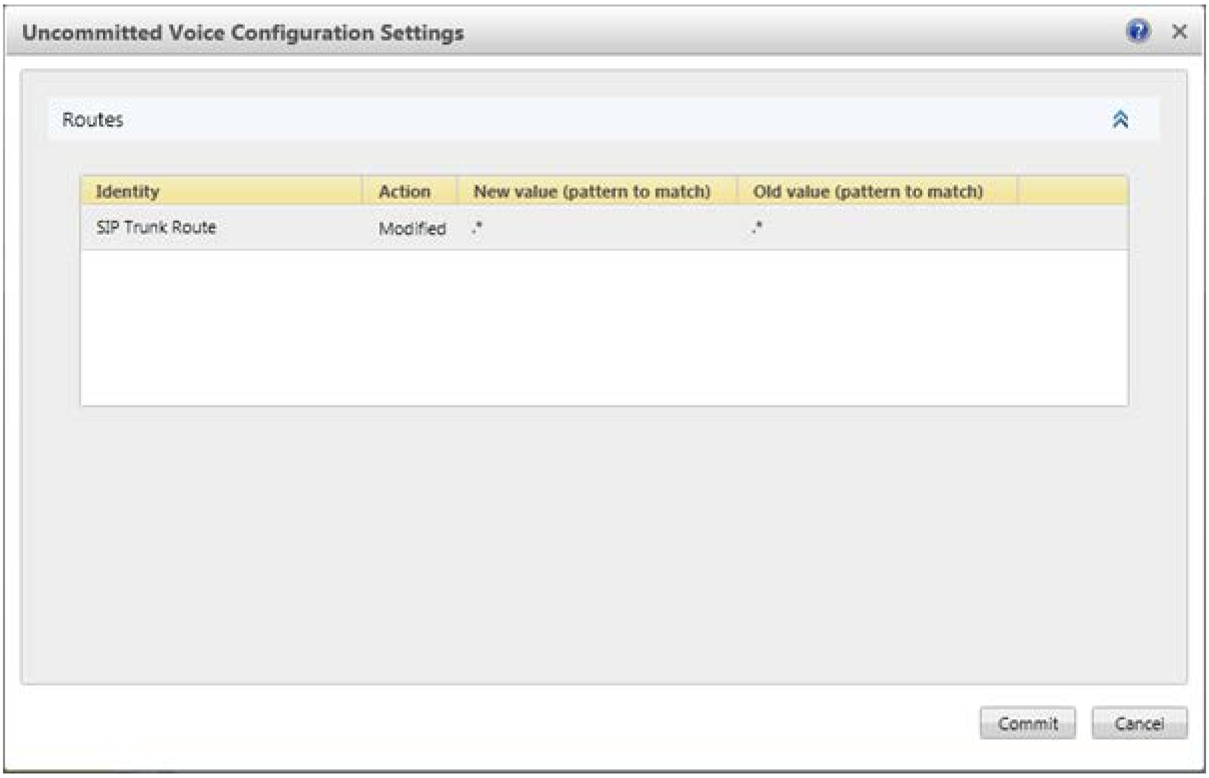
12. In the Content area Toolbar, click on the arrow adjacent to the **Commit** button; a drop-down menu is displayed; select the 'Commit All' option.

Figure 3-26: Committing Voice Routes



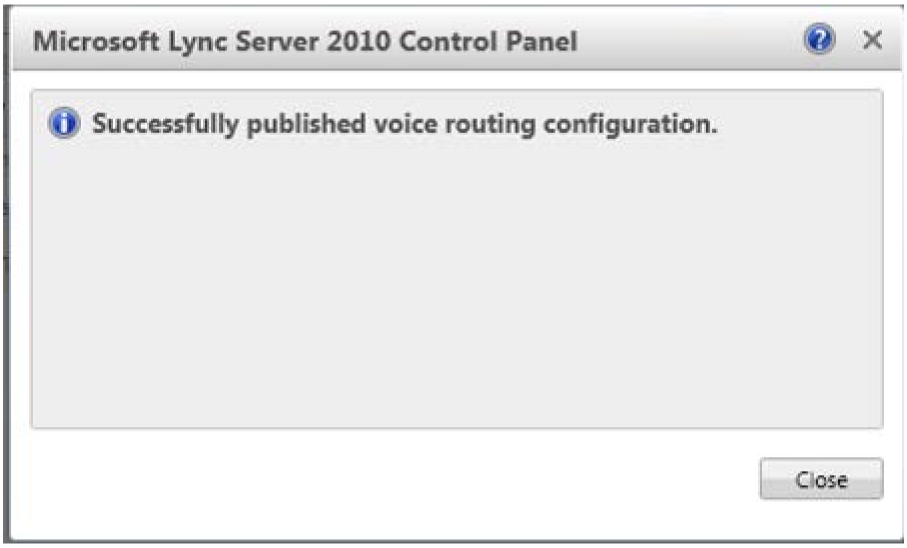
13. In the Uncommitted Voice Configuration Settings window, click **Commit**.

Figure 3-27: Uncommitted Voice Configuration Settings



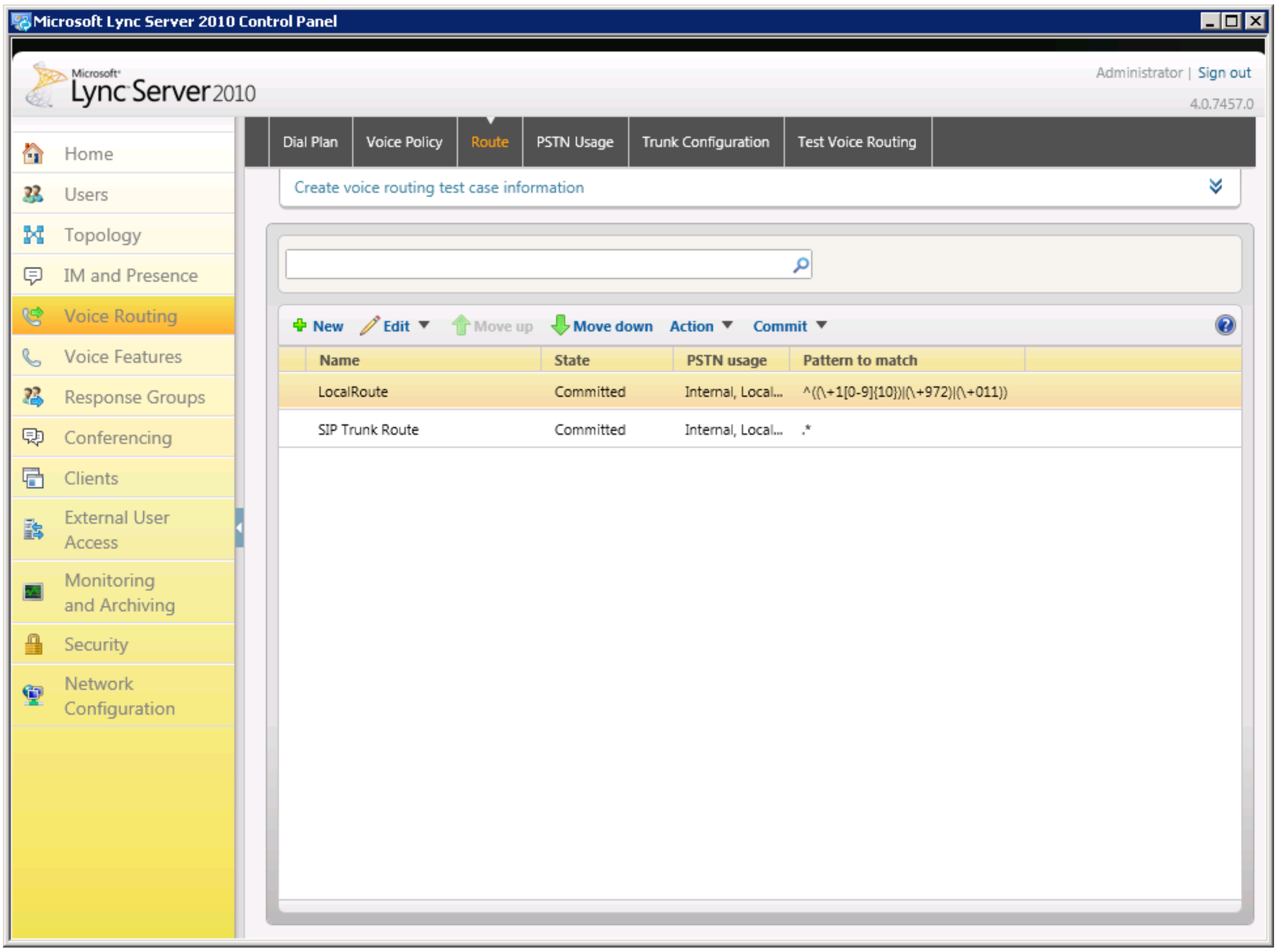
14. A message is displayed, confirming a successful voice routing configuration; in the **Microsoft Lync Server 2010 Control Panel** prompt, click **Close**.

Figure 3-28: Voice Routing Configuration Confirmation



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

Figure 3-29: Voice Routing Screen Displaying Committed Routes



Reader's Notes

4 Configuring AudioCodes Gateway

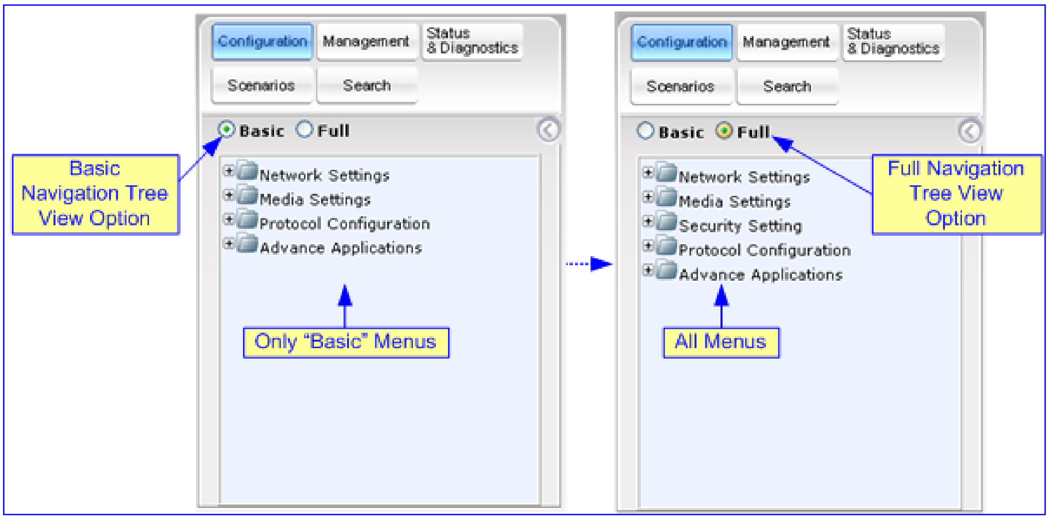
This section provides step-by-step procedures for configuring AudioCodes' gateway. These procedures are based on the setup example described in Section 2 on page 11.

The steps for configuring the gateway can be summarized as follows:

- **Step 1**: Configure IP Addresses (see Section 4.1 on page 36).
- **Step 2**: Enable the SBC Capabilities (see Section 4.2 on page 42).
- **Step 3**: Configure the Number of Media Channels (see Section 4.3 on page 43).
- **Step 4**: Configure the Proxy Sets (see Section 4.4 on page 44).
- **Step 5**: Configure the IP Groups (see Section 4.5 on page 46)
- **Step 6**: Define SIP TLS Transport Type (see Section 4.6 on page 48).
- **Step 7**: Configure Secure Real-Time Transport Protocol (SRTP) (see Section 4.7 on page 56).
- **Step 8**: Configure the Voice Coders (see Section 4.8 on page 57)
- **Step 9**: Define Silence Suppression and Comfort Noise (see Section 4.8.1 on page 58).
- **Step 10**: Configure IP Profile Settings (see Section 4.9 on page 59).
- **Step 11:** Configure IP Profile for Call Forwarding (see Section 4.9.1 on page 61).
- **Step 12**: Configure IP-to-IP Routing Setup (see Section 4.10 on page 63).
- **Step 13**: Configure Number Manipulation (see Section 4.11 on page 66).
- **Step 14**: Configuring SIP General Parameters (see Section 4.12 on page 72).
- **Step 15**: Defining Reasons for Alternative Routing (see Section 4.13 on page 75).

The procedures described in this section are performed using the gateway's Web-based management tool (i.e., embedded Web server). Before you begin configuring the gateway, 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 4-1: Web Interface Showing Basic/Full Navigation Tree Display



4.1 Step 1: Configure IP Addresses

This section describes how to configure the IP addresses of the E-SBC devices' LAN and WAN interfaces.

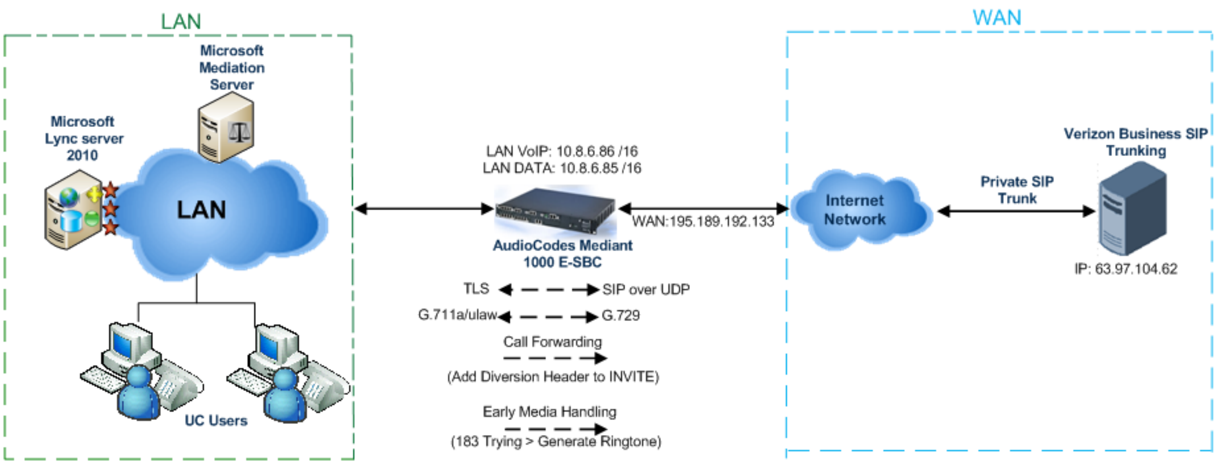
4.1.1 LAN and WAN Interface Separation

This section describes how to configure IP addresses when the internal data-routing capabilities of the E-SBC device are used in order to connect to the Verizon Business SIP Trunk. In this case, you must 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 is 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 4-2: Physical Interface Separation



4.1.1.1 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** submenu > **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., 10.8.6.86).
4. Configure additional IP interfaces, if required.

Figure 4-3: Multiple Interface Table Page

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

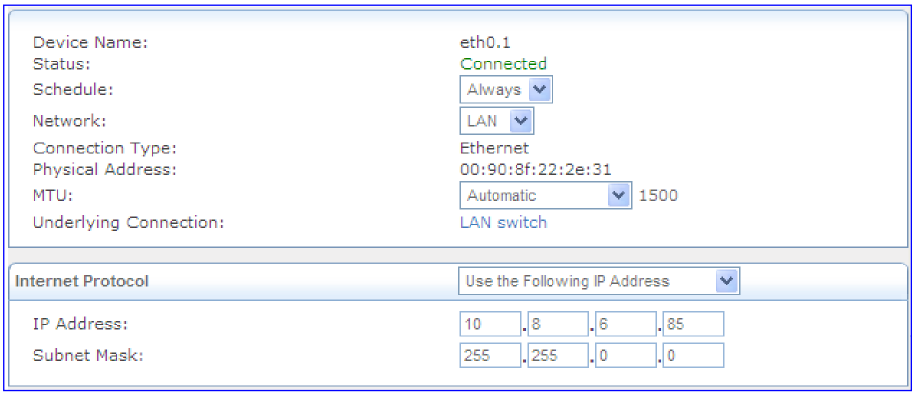
➢ To define the E-SBC device's LAN data-routing IP address:

1. Access the E-SBC 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 4-4: 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., 10.8.6.85) and subnet respectively, and then click **OK**.

Figure 4-5: Defining LAN Data-Routing IP Address



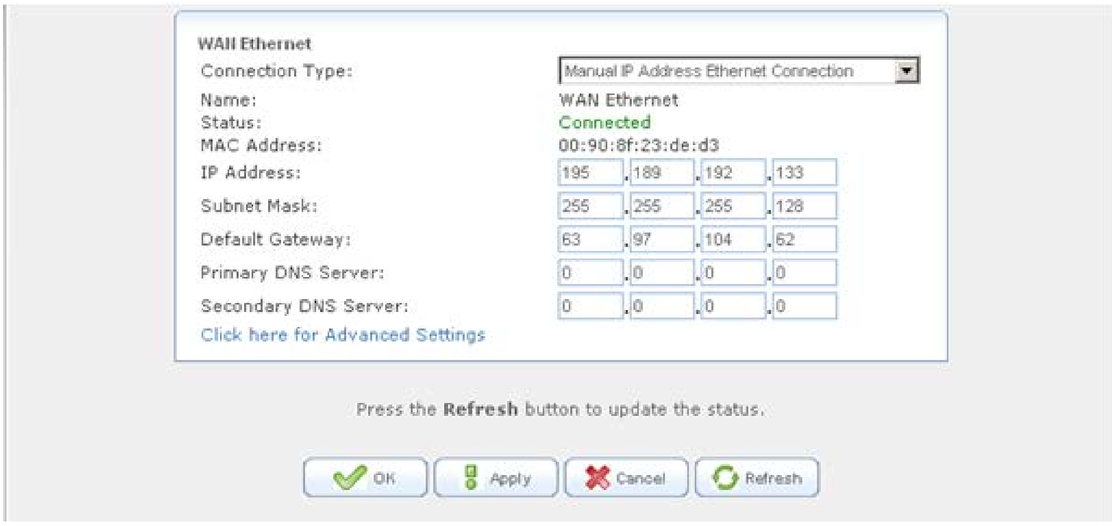
4.1.1.2 Assigning WAN IP Addresses

This section describes how to assign the WAN IP addresses.

➢ **To assign a WAN IP address:**

1. Cable the E-SBC device to the WAN network (i.e., ADSL or Cable modem), using the WAN port.
2. Access the E-SBC 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 4-6: Configuring the WAN IP Address

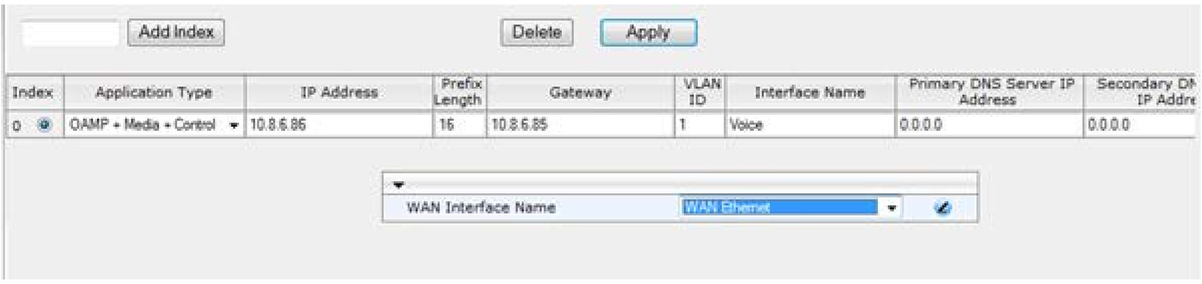


4. From the 'Connection Type' drop-down list, select the required connection type for the WAN, and then configure the IP address (e.g., 195.189.192.133).

➢ **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** submenu > **IP Settings**).

Figure 4-7: Selecting WAN Interface for VoIP Traffic in Multiple Interface Table Page

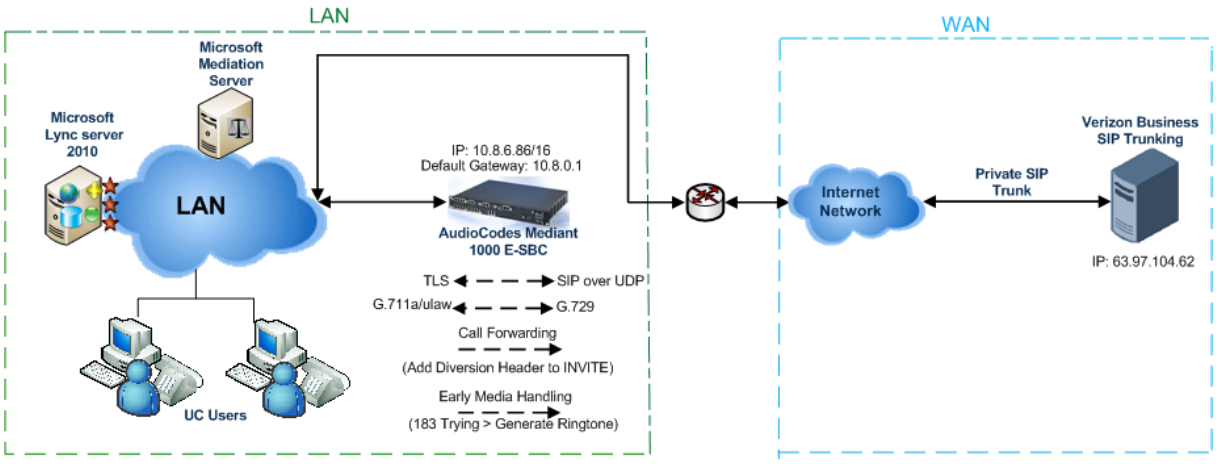


3. From the 'WAN Interface Name' drop-down list, select the WAN interface for VoIP traffic.
4. Click **Done**, and then reset the E-SBC device for your setting to take effect.

4.1.2 Single LAN Interface

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

Figure 4-8: Single LAN Interface

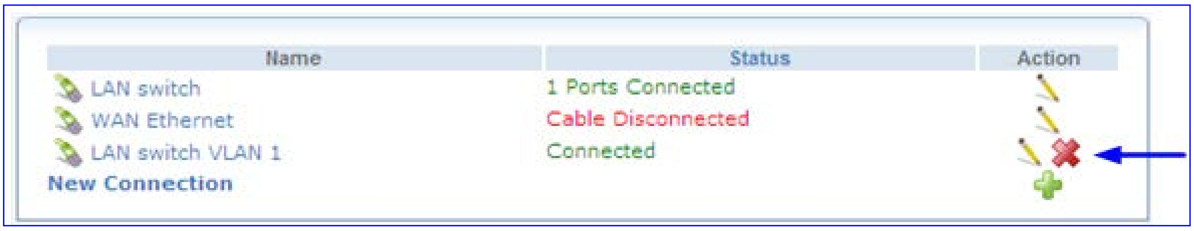


Note: When operating in LAN VoIP-only mode, do not use the E-SBC device's WAN port.

➢ **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 4-9: Removing Data-Routing Connection Interface



Name	Status	Action
LAN switch	1 Ports Connected	
WAN Ethernet	Cable Disconnected	
LAN switch VLAN 1	Connected	
New Connection		

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 4-10: Multiple Interface Table

Index	Application Type	IP Address	Prefix Length	Gateway	VLAN ID	Interface Name
0	OAMP + Media + Control	10.8.6.86	16	10.8.0.1	1	Voice

   - Click **OK** to save settings.

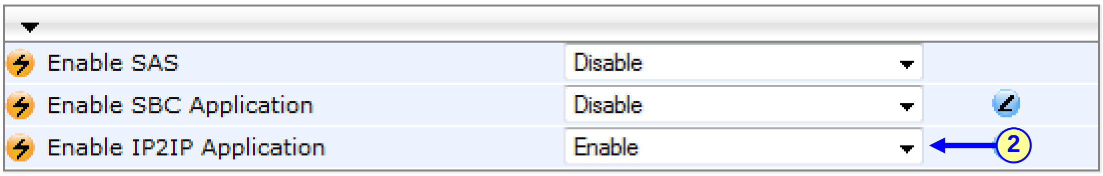
4.2 Step 2: Enable the SIP IP2IP Application

This step describes how to enable the gateway's SIP IP2IP application.

To enable the SIP IP2IP application:

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

Figure 4-11: Application Enabling



2. From the ‘Enable IP2IP Application’ drop-down list, select “Enable”.
   Reset with BURN to FLASH is required.

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

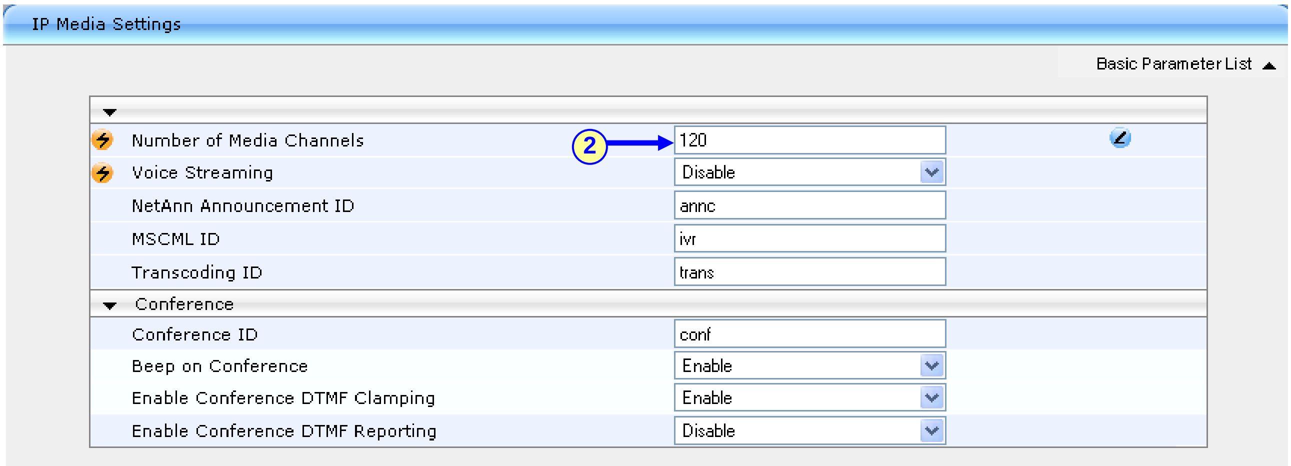
4.3 Step 3: Configure the Number of Media Channels

In order to reform the coder transcoding, you need to define DSP channels. The number of media channels represents the number of digital signaling processors (DSP) channels that the gateway 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 gateway is 120 (i.e., up to 60 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 4-12: IP Media Channels Settings



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

4.4 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 Verizon Business 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 4.5 on page 46).

➢ To configure proxy sets:

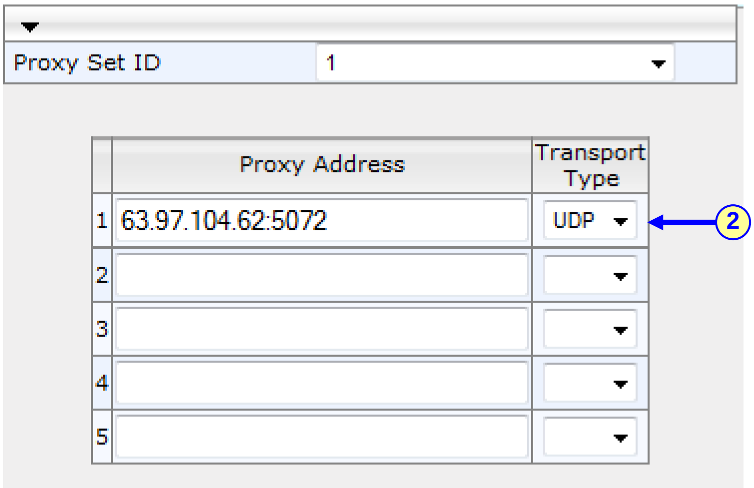
1. Open the 'Proxy Sets Table' page (**Configuration** tab > **VoIP** menu > **Control Network** > **Proxy Sets Table**).
2. Configure the Proxy Set for Verizon Business SIP Trunk:

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

   a. In the 'Proxy Address' column, enter the IP address or FQDN of the Verizon Business SIP Trunk and the listening port of the Verizon Business SIP Trunk.

   b. From the 'Transport Type' drop-down list, corresponding to the IP address entered above, select "UDP".

Figure 4-13: Proxy Set ID 1 for Verizon Business SIP Trunk



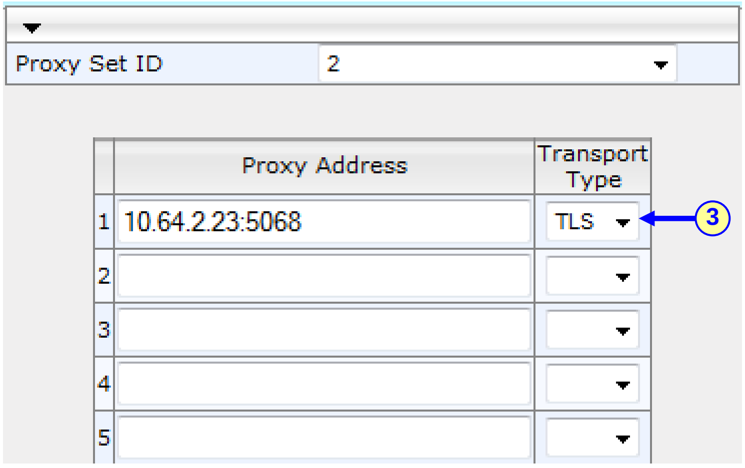
3. Configure the Proxy Set for the Lync Mediation Server:

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

   a. In the 'Proxy Address' column, enter the IP address or the FQDN and the listening port of the Lync Mediation Server.

   b. From the 'Transport Type' drop-down list corresponding to the IP address entered above, select "TCP" or “TLS” depending on the deployed Mediation Server Transport Type.

Figure 4-14: Proxy Set ID 2 for Lync Mediation Server



4.5 Step 5: Configure the IP Groups

This step describes how to create IP groups. Each IP group represents a SIP entity in the gateway's network. You need to create IP groups for the following entities:

1. Verizon Business 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**).
2. Define IP Group #**1** for the Verizon ITSP as follows:
   a. IP Group Index '1'
   b. Type: "SERVER"
   c. Description: arbitrary name. (e.g., "Verizon")
   d. Proxy Set ID: "1" (represents the IP address, configured in Section 4.4 on page 44, 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.
   f. IP Profile ID: "1": Different IP profile is used for the Verizon Business SIP Trunk and the Mediation Server. See Section 4.9 on page 59.

Figure 4-15: IP Group 1 Table

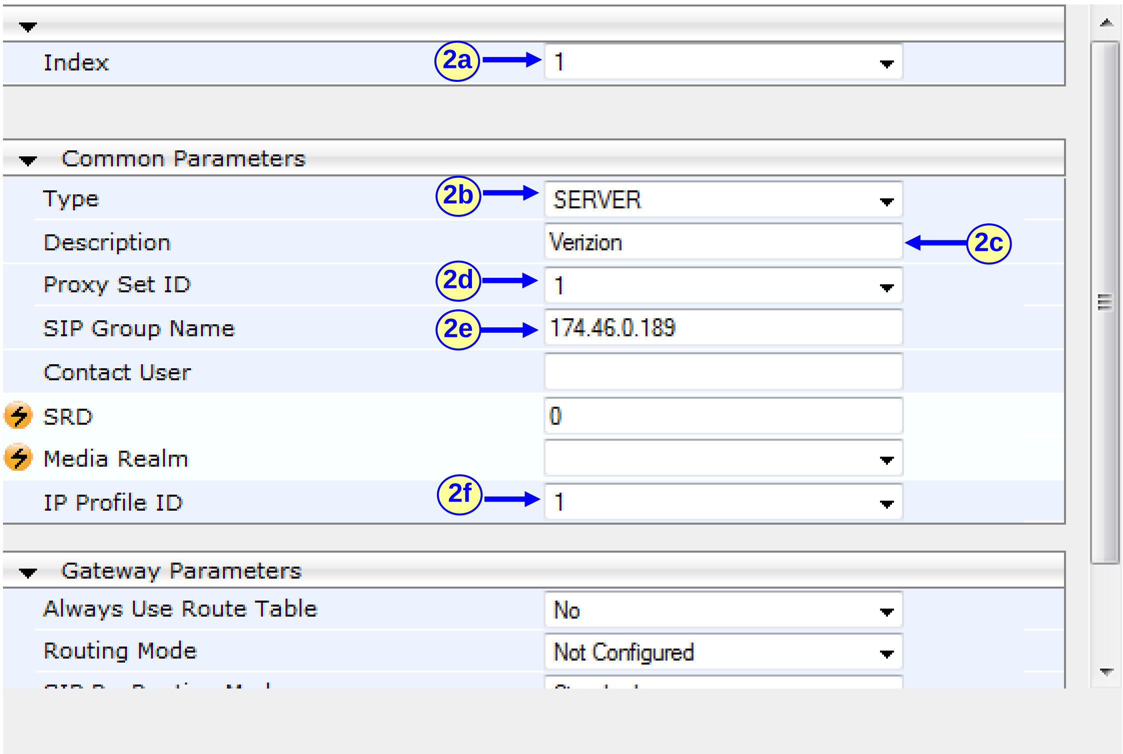
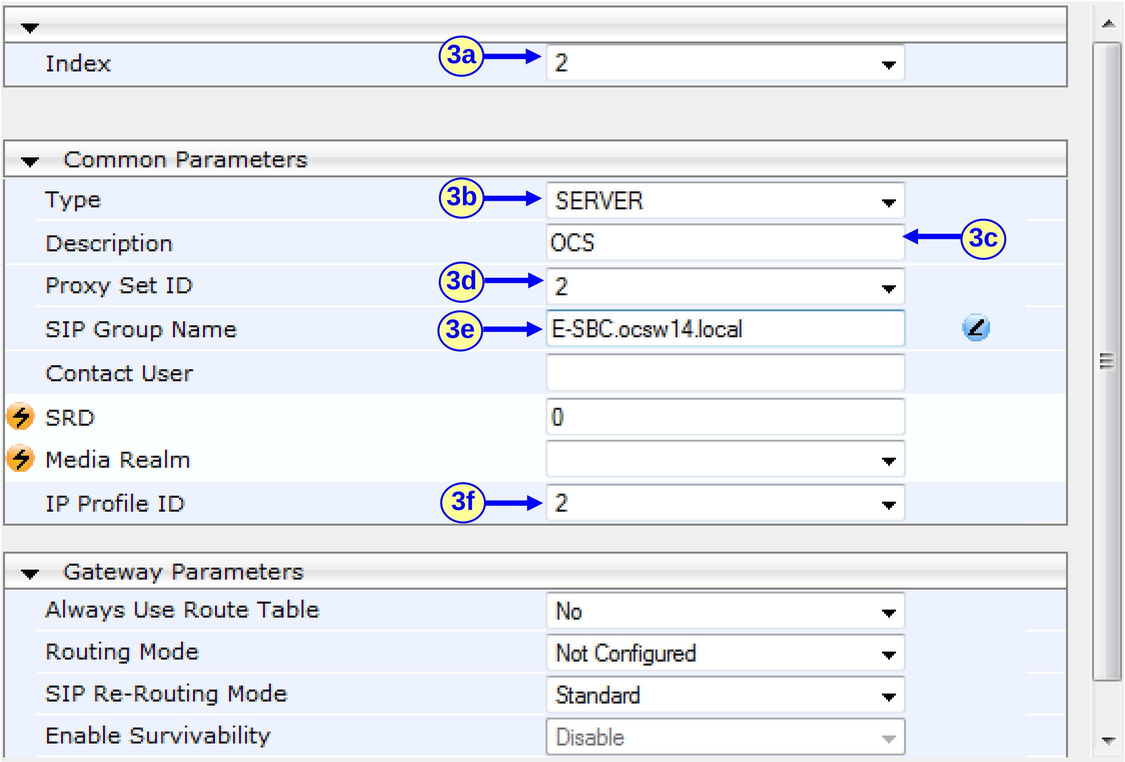


Figure 4-16: IP Group 2 Table Page



3. Define IP Group #**2** for 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.
   f. **IP Profile ID:** “2” (see Section 4.9 on page 59).

4.6 Step 6: Define SIP TLS Transport Type

This section describes how to configure AudioCodes gateways for implementing a TLS connection with the Mediation Server.

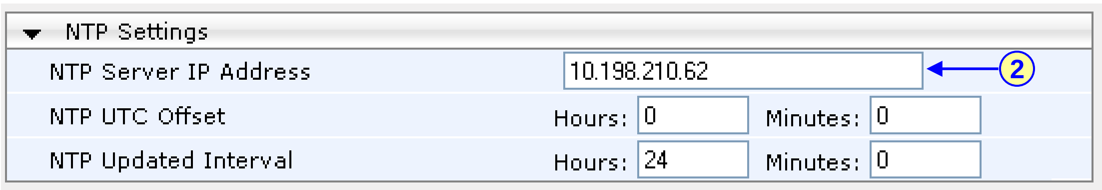
4.6.1 Configure NTP and DNS Server

The procedure below describes how to configure the NTP Server IP address or FQDN and the Domain Name System (DNS) servers.

➢ **To configure NTP servers:**

1. Open the 'Application Settings' page (**Configuration** tab > **System** menu > **Application Settings**).

Figure 4-17: Application Settings

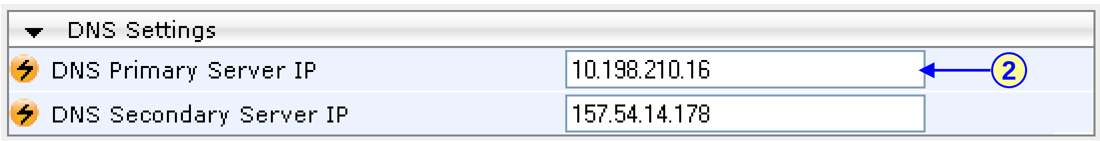


2. Define the NTP server's IP address so that it corresponds to your network environment.

➢ **To configure DNS servers:**

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

Figure 4-18:DNS Settings



2. In the 'DNS Primary Server IP' and 'DNS Secondary Server IP' fields, set the primary and secondary DNS server's IP addresses with the IP address of your DNS server.
3. Click the **Submit** button to save your changes.
4. Save the changes to flash memory, by clicking **Burn** button on the toolbar. The changes take effect after the restart operation.

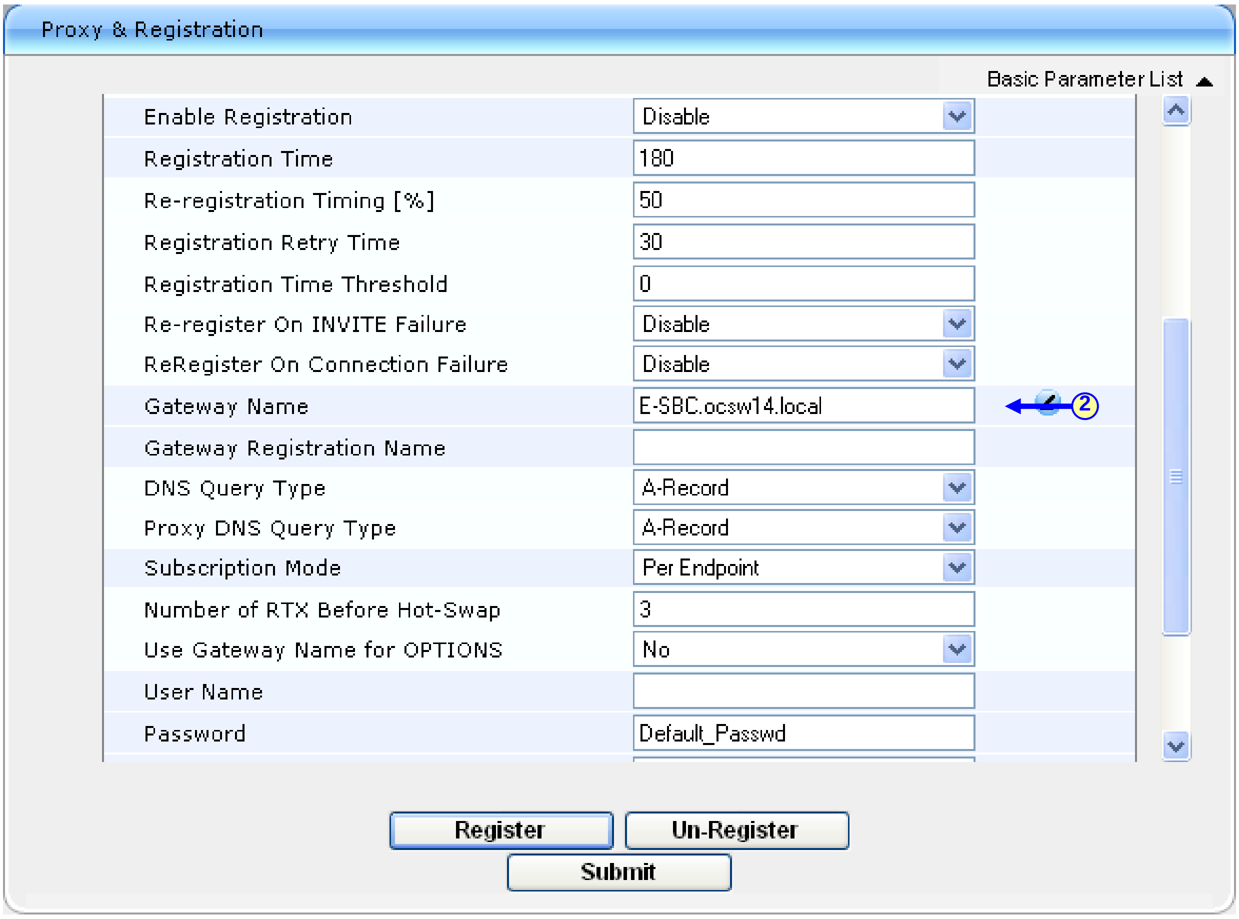
4.6.2 Configure the Gateway Name

The procedure below describes how to configure the Media Gateway name.

➢ **To configure the Media Gateway name:**

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

Figure 4-19: Proxy & Registration



2. In the 'Gateway Name' field, assign a unique FQDN name to the Media Gateway within the domain, for example, 'E-SBC.OCSW14.local'.

4.6.3 Configure a Certificate

The procedure below describes how to exchange a certificate with the Microsoft Certificate Authority. The certificate is used by the E-SBC device to authenticate the connection with the management PC (the PC used to manage the E-SBC using the embedded Web server).

➢ **To configure a certificate:**

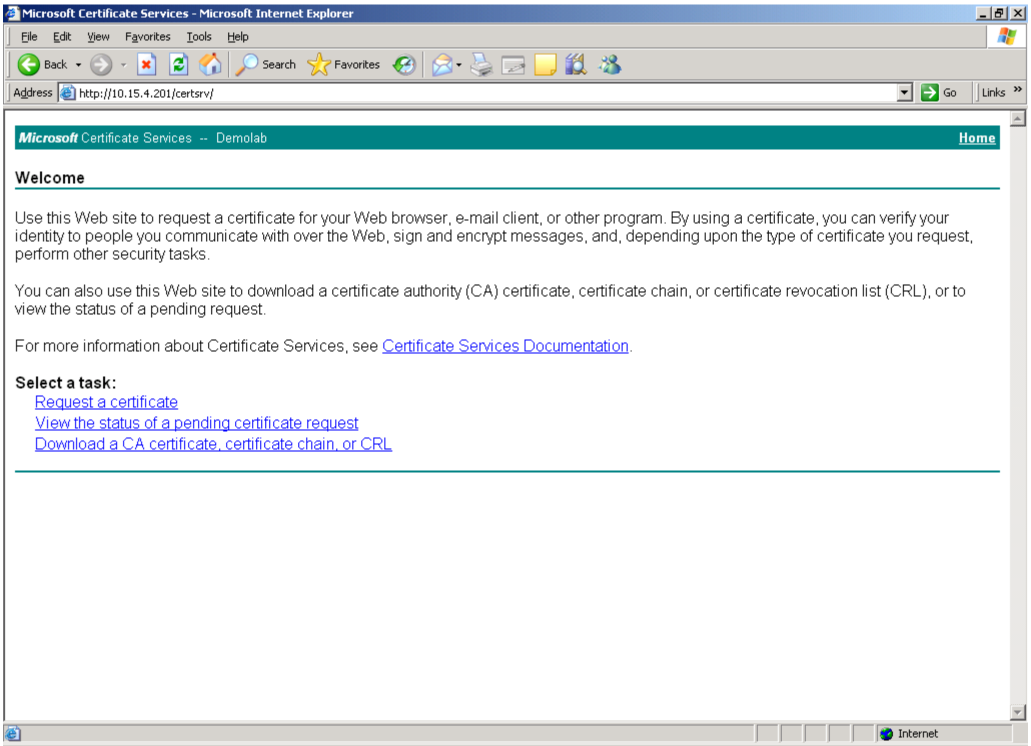
1. Open the ‘Certificates’ page (**Configuration** tab > **System** menu > **Certificates**).

Figure 4-20: Certificates Page

2. In the ‘Subject Name’ field, enter the Media Gateway name as configured in the previous section (see Section 4.6.2 on page 49), and then click **Generate CSR**; a Certificate request is generated.
3. Copy the CSR (from the line “----BEGIN CERTIFICATE” to “END CERTIFICATE REQUEST----”) to a text file (such as Notepad), and then save it to a folder on your PC as *certreq.txt*.

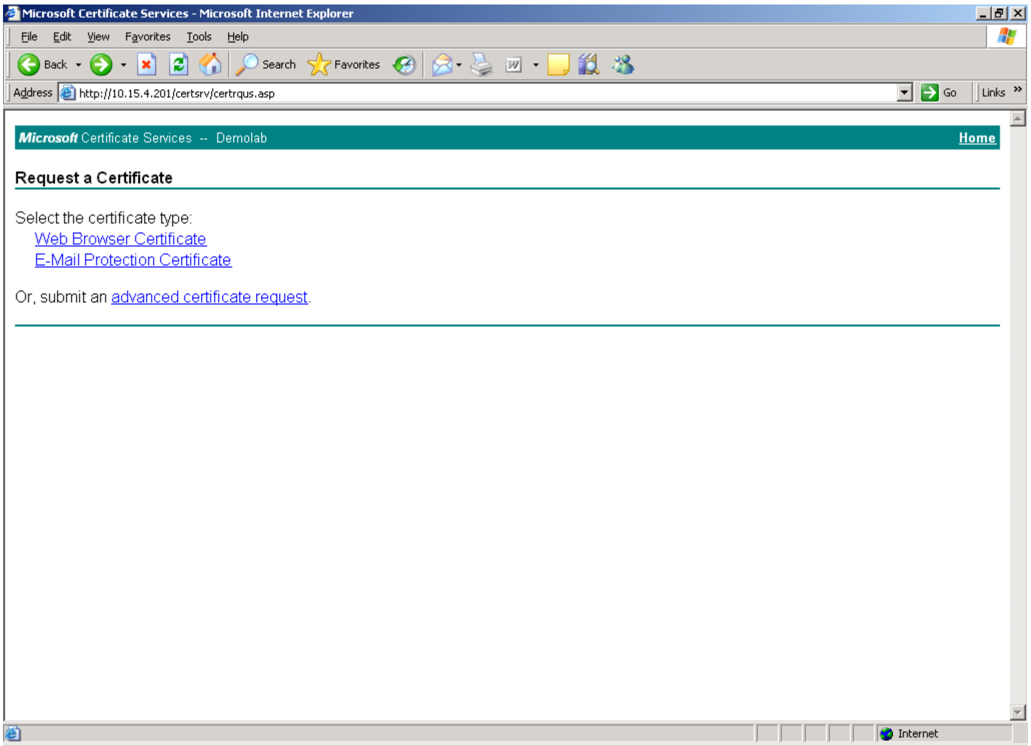
4. Navigate to the certificate 'Server http://<Certificate Server>/CertSrv'.

Figure 4-21: Microsoft Certificate Services Web Page



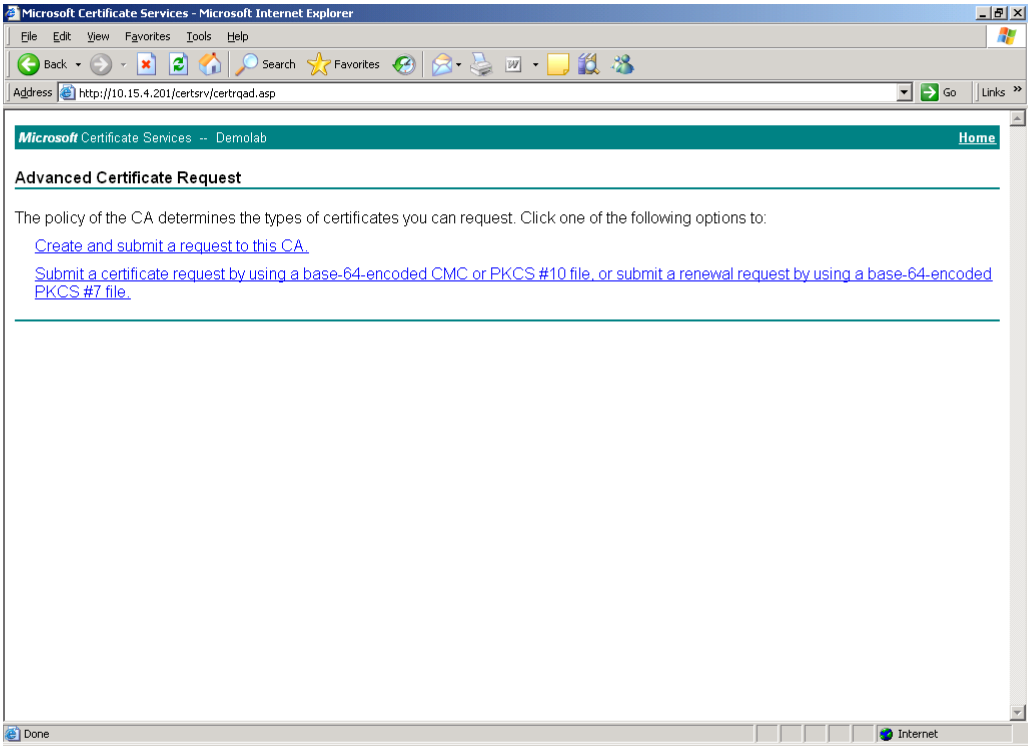
5. Click the link **Request a Certificate**.

Figure 4-22: Request a Certificate Page



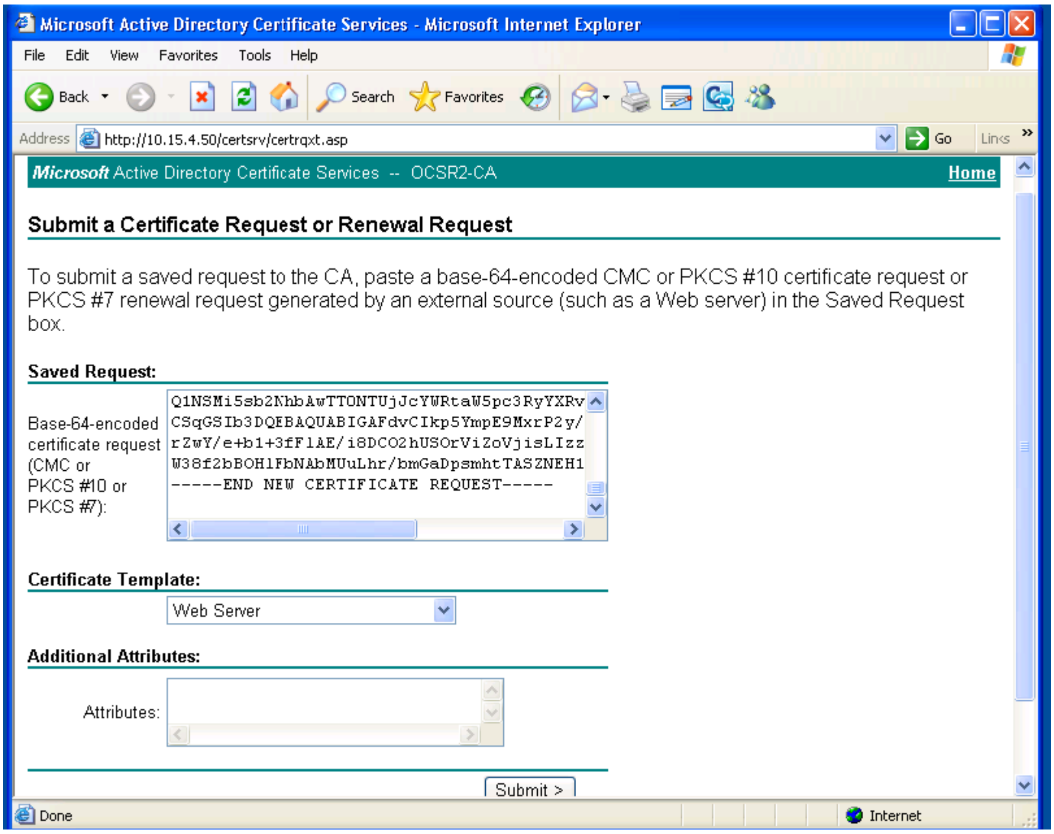
6. Click the link **Advanced Certificate Request**, and then click **Next**.

Figure 4-23: Advanced Certificate Request Page



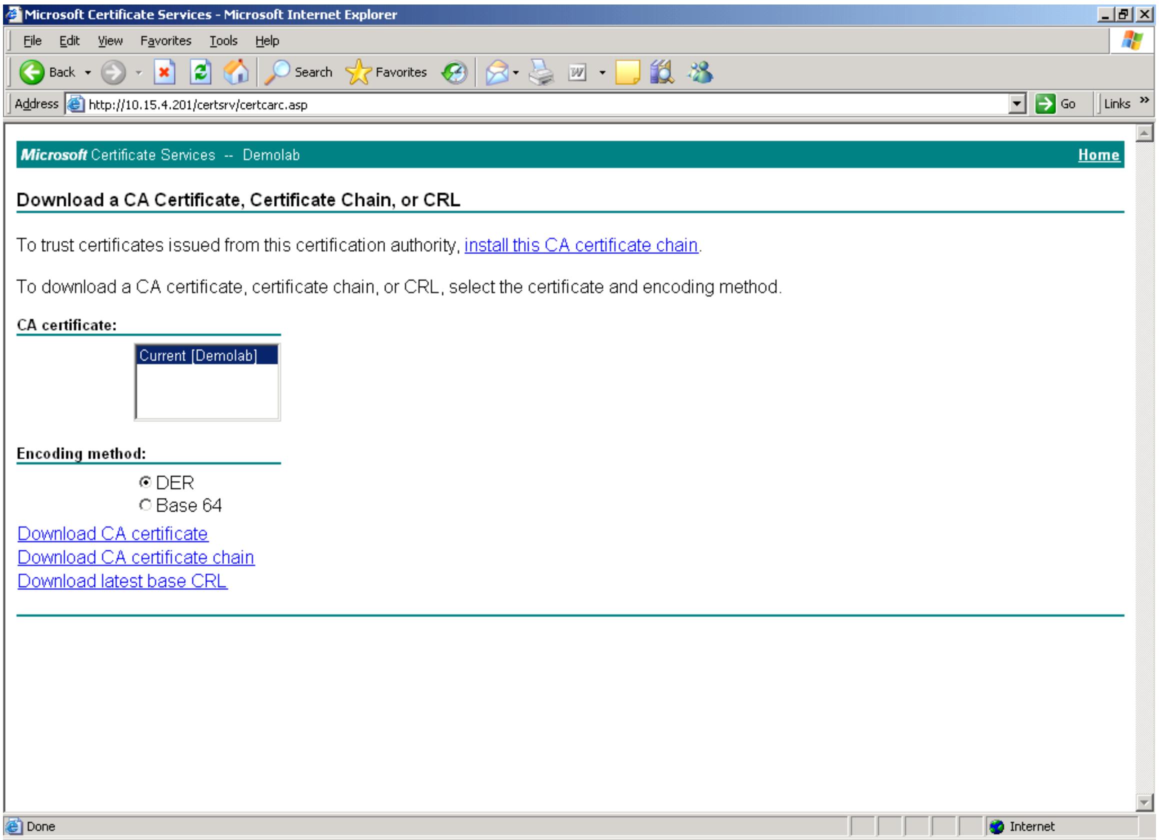
7. Click the link **Submit a Certificate request by using base64 encoded...**, and then click **Next**.

Figure 4-24: Submit a Certificate Request or Renewal Request Page



8. Open the *certreq.txt* file that you created and saved (see Step 3), and then copy its contents to the 'Base64 Encoded Certificate Request' text box.
9. Select "Web Server" from the **Certificate Template** drop-down box.
10. Click **Submit**.
11. Choose the 'Base 64' encoding option, and then click the link **Download CA certificate**.
12. Save the file as '*gateway.cer'* in a folder on your PC.
13. Navigate to the certificate Server http://<Certificate Server>/CertSrv.
14. Click the link **Download a CA Certificate**, **Certificate Chain or CRL**.

Figure 4-25: Download a CA Certificate, Certificate Chain, or CRL Page



15. Under the Encoding method group, perform the following:
    a. Select the 'Base 64' encoding method option.
    b. Click the link Download CA certificate.
16. Save the file as '*certroot.cer'* in a folder on your PC.
17. Navigate back to the 'Certificates' page.
18. In the 'Certificates' page, in the 'Server Certificate' field, click **Browse** and select the '*Gateway.cer'* certificate file that you saved on your local disk (see Step 12), and then click **Send File** to upload the certificate.

19. In the 'Certificates' page, in the 'Trusted Root Certificate Store' field, click **Browse** and select the '*Certroot.cer*' certificate file that you saved on your local disk (see Step 16), and then click **Send File** to upload the certificate.

Figure 4-26: Certificates Page

20. Save (burn) the Media Gateway configuration and reset the Media Gateway, using the Web interface's 'Maintenance Actions' page (On the Navigation bar, click the **Management** tab, and then in the Navigation tree, select the **Management Configuration** menu, and then choose the **Maintenance Actions**).

4.7 Step 7: Configure Secure Real-Time Transport Protocol (SRTP)

If you configure TLS for the SIP transport link between the E-SBC and the Mediation Server, you must specify Secure RTP (SRTP) encryption with one of the following options:

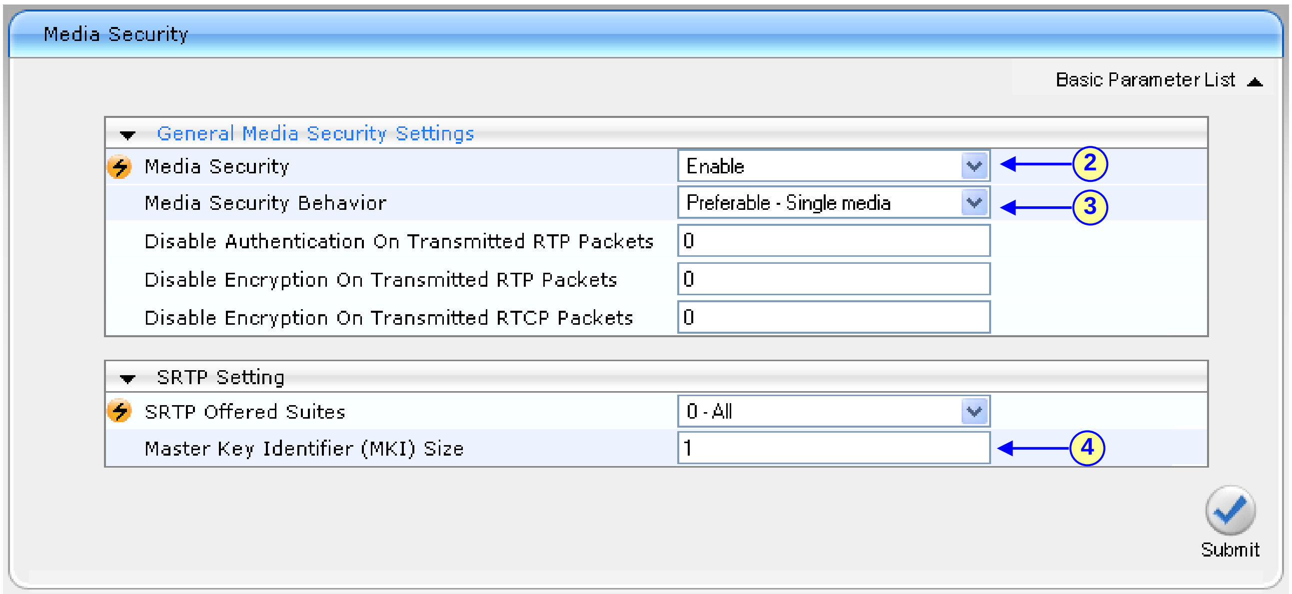
- **Required:** SRTP should be attempted, but do not use encryption if negotiation for SRTP is unsuccessful.
- **Optional:** Attempt to negotiate the use of SRTP to secure media packets. Use RTP if SRTP cannot be negotiated.
- **Not used:** Send media packets using RTP.

If you choose to configure the Mediation Server to use SRTP (Required or Optional), you need to configure the Media Gateway to operate in the same manner.

➢ **To configure the media security:**

1. Open the 'Media Security' page (**Configuration** tab > **Media** menu > **Media Security**).

Figure 4-27: Media Security Page



2. From the ‘Media Security’ drop-down list, select “Enable”, to enable SRTP.
3. From the ‘Media Security Behavior’ drop-down list, select:
   - “Mandatory” if Mediation Server is configured to SRTP Required
   - ”Preferable-Single media” if Mediation Server is configured to SRTP Optional.
4. In the ‘Master Key Identifier (MKI) Size’ field, enter ‘1’.
5. Click **Submit**.
6. Save (burn) the E-SBC configuration and reset the Gateway.

Note: In order to set the ‘Media Security Behavior’ to the IP Profile of the Mediation Server, see the IP Profile Settings (see Section 4.9 on page 59).

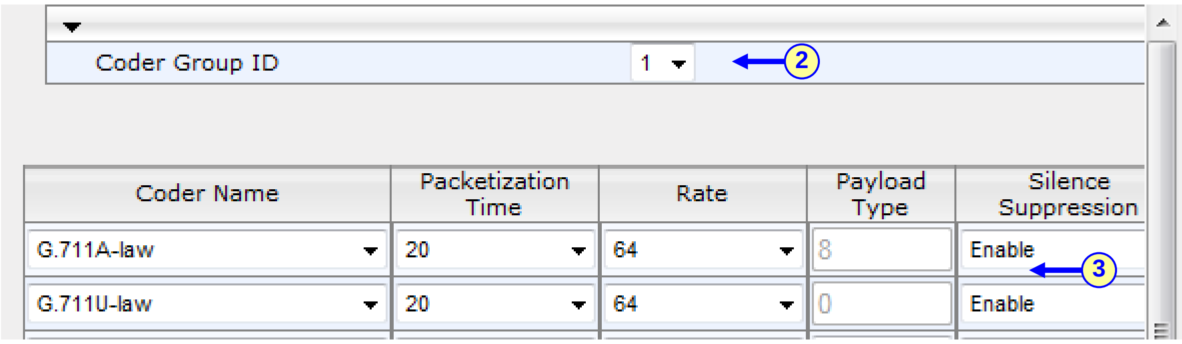
4.8 Step 8: Configure the Voice Coders

Since the Mediation Server support only G.711 a/ulaw voice coders, while the ITSP SIP trunk requires G.729 coder, you need to configure two coder tables for each entity. The Coder table is associated with an IP Profile (see Section 4.9 on page 59) which is associated with the IP Group (see Section 4.5 on page 46).

➢ To configure Coder Table for Mediation server:

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

Figure 4-28: Coder Group Table - Mediation Server

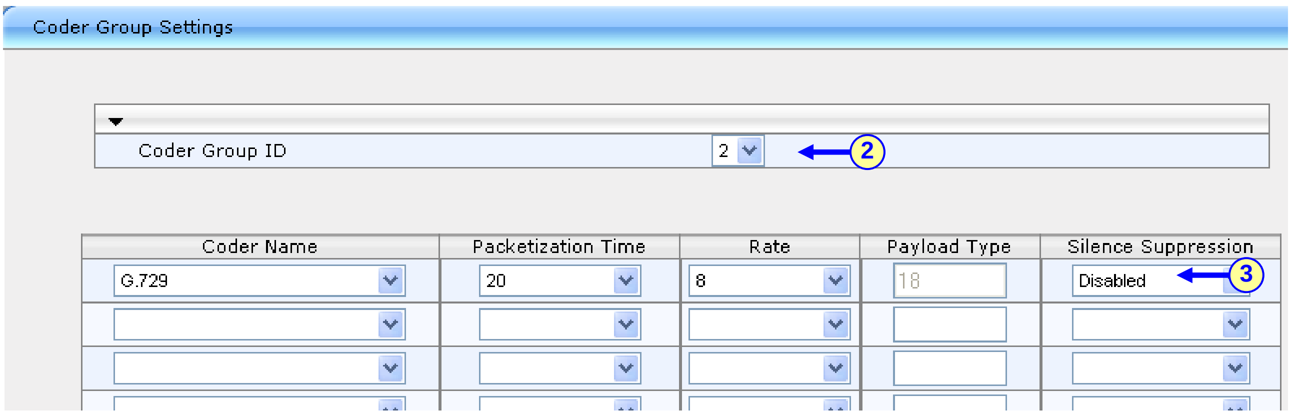


2. From the 'Coder Group ID drop-down list, select '1'.
3. Select the G.711A-law and G.711U-law coders, as shown in the figure above.
4. From 'Silence Suppression' drop-down list, select 'Enable' as shown in the figure above.

➢ To configure Coder Table for Verizon Business SIP Trunk:

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

Figure 4-29: Coder Group Table - ITSP SIP Trunk



2. From the 'Coder Group ID drop-down list, select '2'.
3. Select the G.729 coder, as shown in the figure above.

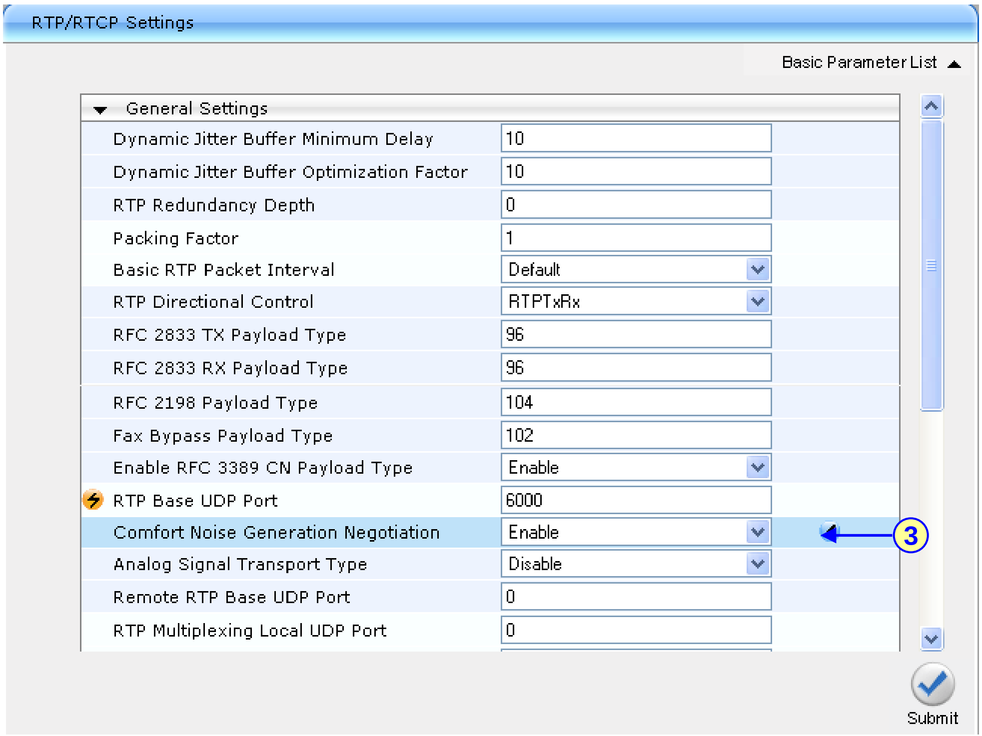
4.8.1 Step 9: Define Silence Suppression and Comfort Noise

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 change the E-SBC 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 4.8 on page 57 above to enable Silence Suppression per coder.)
2. Open the 'RTP/RTCP Settings' page (**Configuration** tab > **Media** menu > **RTP / RTCP Settings**).

Figure 4-30: RTP/RTCP Settings Page



3. From the 'Comfort Noise Generation Negotiation' drop-down list, select 'Enable'. This enables negotiation and usage of Comfort Noise (CN).
4. Click Submit.

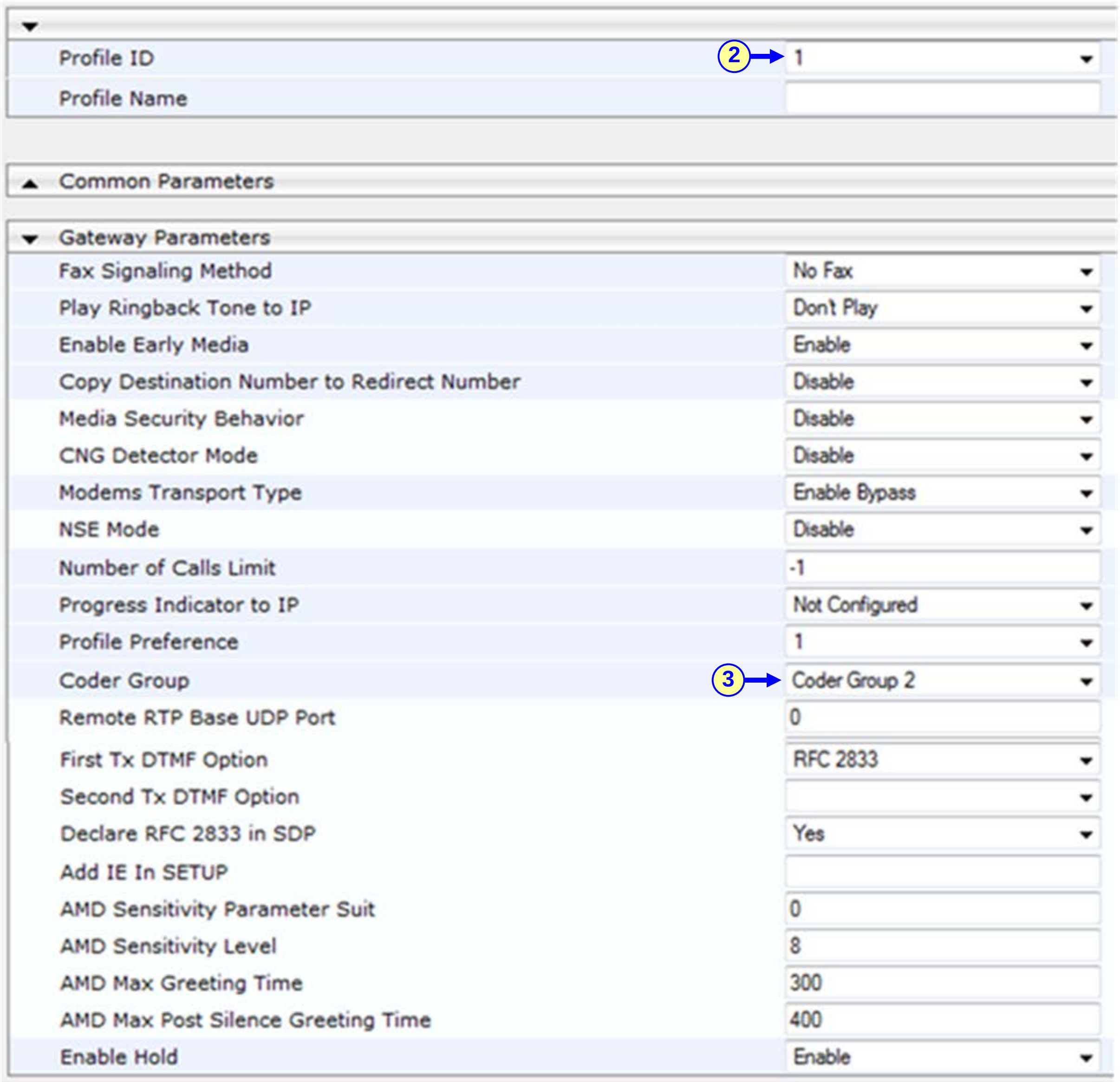
4.9 Step 10: Configure IP Profile Settings

This section describes how to configure the IP Profile Settings.

➢ **To configure IP Profile for Verizon :**

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

Figure 4-31: IP Profile Page-Verizon Server

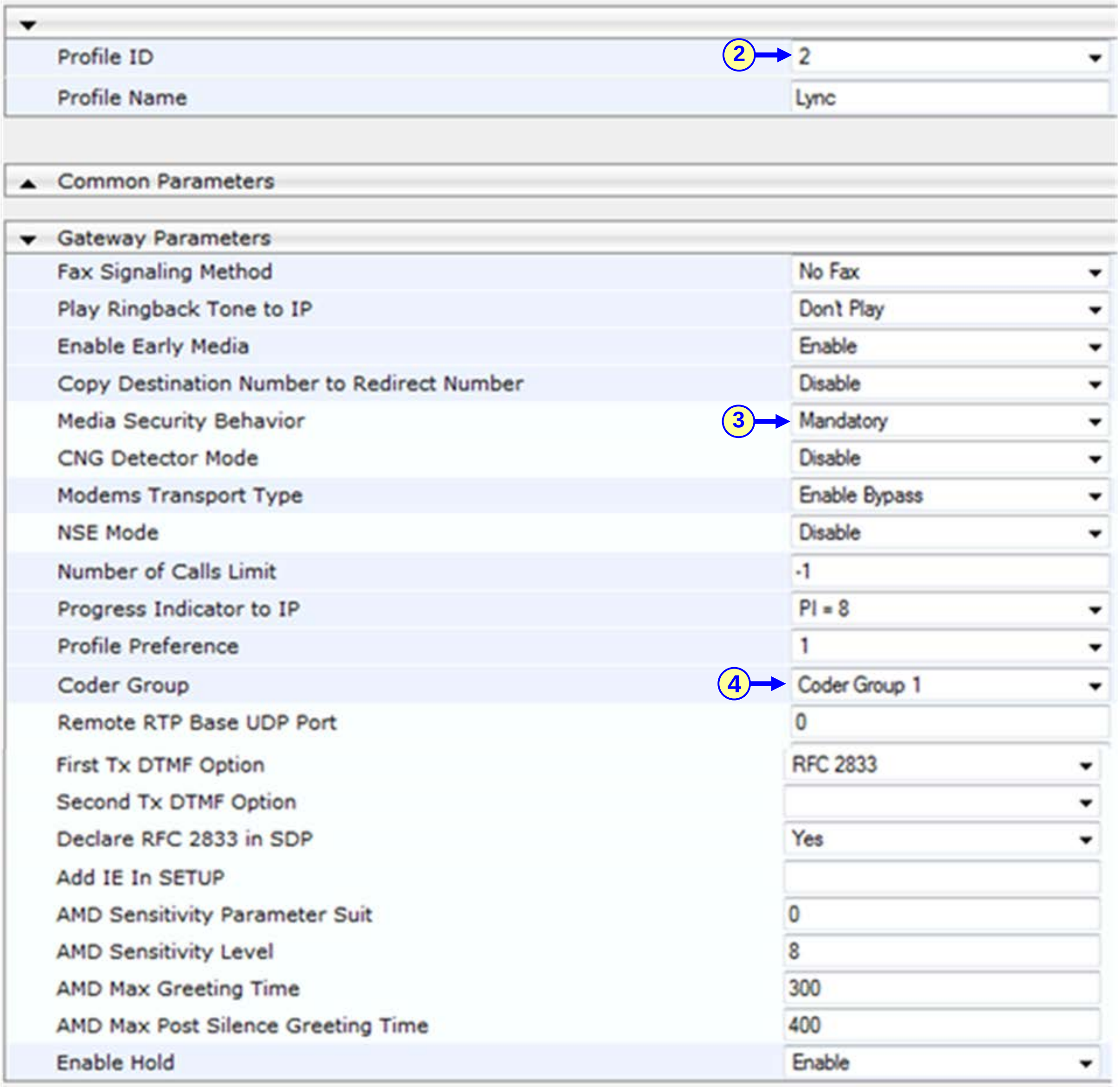


2. From the 'Profile ID' drop-down list, select '1'.
3. From the 'Coder Group' drop-down list, select 'Coder Group 2'.

➢ **To configure IP Profile for Mediation Server:**

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

Figure 4-32: IP Profile Page-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 ‘Coder Group 1’.

4.9.1 Step 11: Configure IP Profile for Call Forwarding

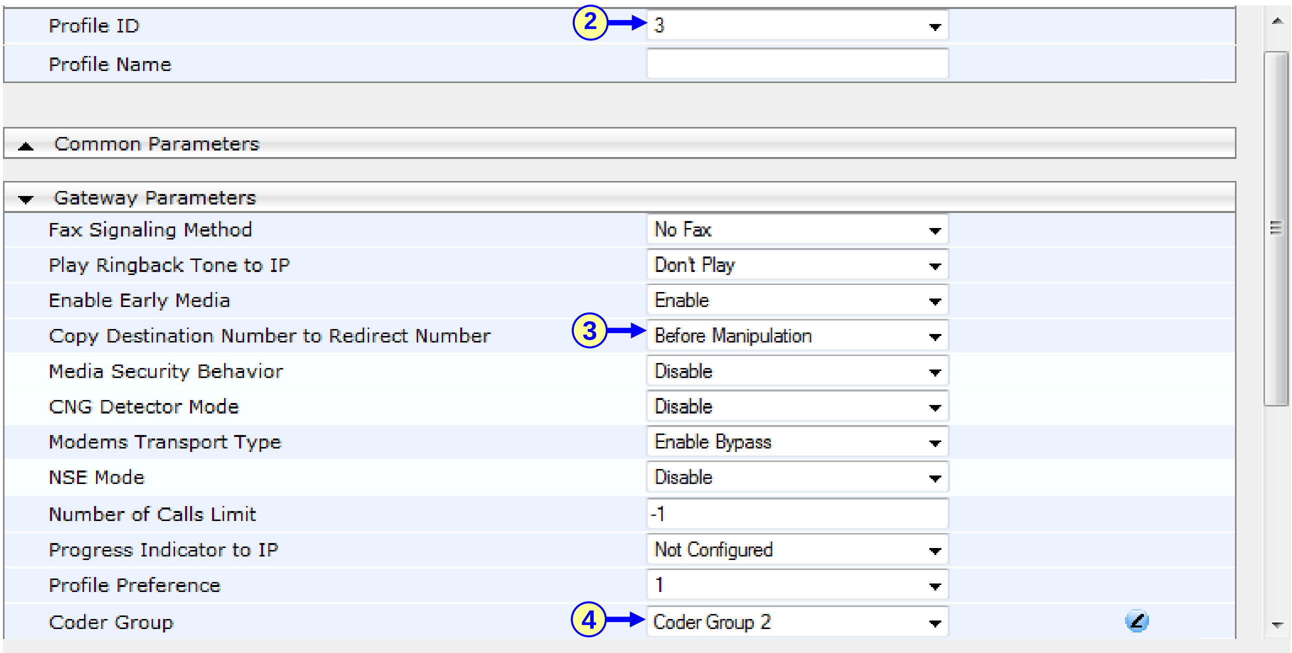
One of the challenges with the integration of the Microsoft Lync 2010 server and the Verizon Business SIP Trunk is the implementation of call forwarding. Since the Microsoft Lync client forwards the call back to the SIP Trunk, it does not provide any information in the forwarded INVITE (such as Diversion header) informing that this call has been forwarded. Consequently, it is 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.

➤ **To configure IP Profile for call forwarding:**

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

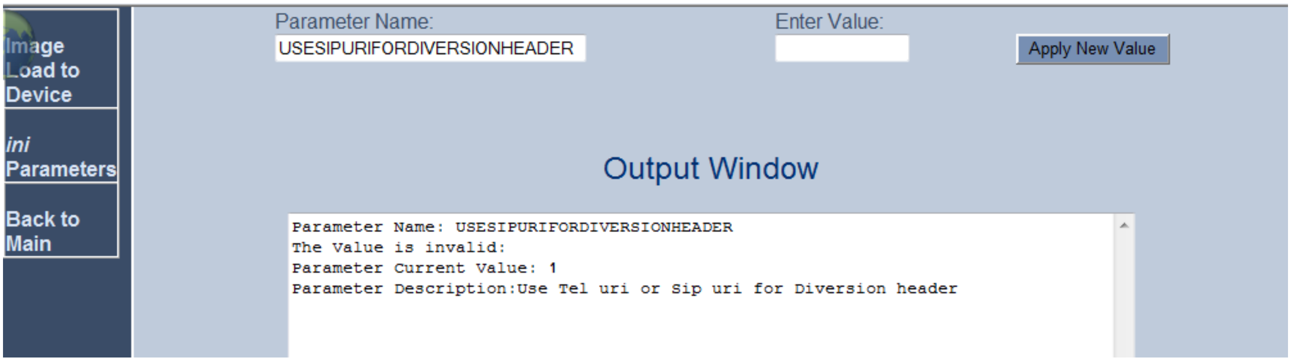
Figure 4-33: IP Profile Settings for Call Forwarding "numbers"



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 'Coder Group 2'.

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. On the left pane, click ***ini*** Parameters.

Figure 4-34: Output Window



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

4.10 Step 12: Configure IP-to-IP Routing Setup

The E-SBC's IP-to-IP call routing capabilities 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 stage 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 stage is configured in the 'Outbound IP Routing Table'.

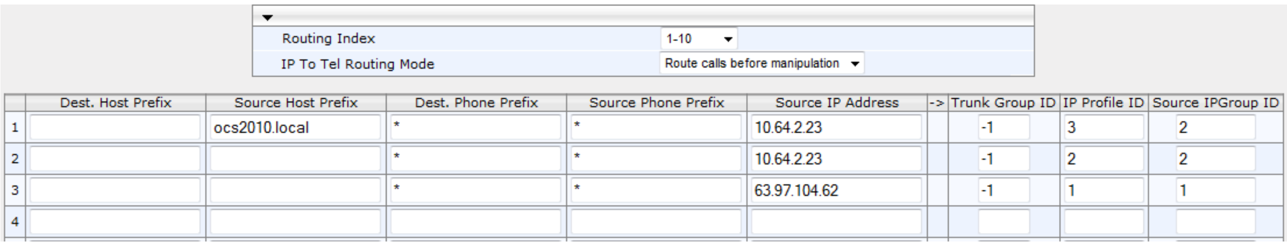
4.10.1 Configure Inbound IP Routing

This step defines how to configure the E-SBC for routing inbound (i.e., received) IP-to-IP calls.

➢ **To configure in bound 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 4-35: Inbound IP Routing Table Page



Routing Index	1-10
IP To Tel Routing Mode	Route calls before manipulation

	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		ocs2010.local	*	*	10.64.2.23		-1	3	2
2			*	*	10.64.2.23		-1	2	2
3			*	*	63.97.104.62		-1	1	1
4									

2. **Index #1** configuration identifies all IP calls received from the Mediation Server in the event of a call forwarding Scenario 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 ServerFQDN.
   - 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
   - 'Source IP Address': Enter the IP address of 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.

3. **Index #2** 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:
   - '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 ‘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.
4. **Index #3** configuration identifies all IP calls received from Verizon Business SIP Trunk as IP-to-IP calls and assigns them to the IP Group ID configured for the Verizon Business SIP Trunk:
   - 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
   - 'Source IP Address': Enter the IP address of Verizon Business 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 Verizon Business SIP Trunk.
   - 'Source IP Group ID': Enter "1" to assign these calls to the IP Group pertaining to Verizon Business SIP Trunk.

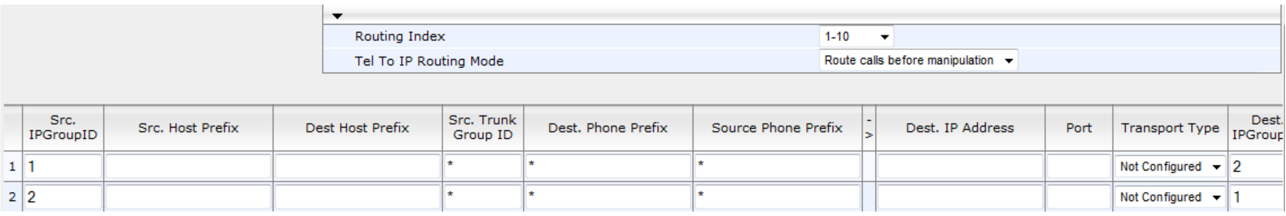
4.10.2 Configure Outbound IP Routing

This step defines how to configure the gateway for outbound routing (i.e., sent) IP-to-IP calls.

➢ **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 4-36: 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 Verizon Business 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 Verizon Business 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.
3. Index **#2** defines the routing of IP calls to the Verizon Business 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., Verizon Business 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 Verizon Business SIP Trunk.

4.11 Step 13: 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 4.11.1 on page 67.
- Dest Number Tel->IP. See Section 4.11.1 on page 67.
- Source Number IP->Tel. See Section 4.11.2 on page 69.
- Source Number Tel->IP. See Section 4.11.2 on page 69.
- Redirect Number IP->Tel. See Section 4.11.3 on page 71.
- Redirect Number Tel->IP. See Section 4.11.3 on page 71

4.11.1 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:**

1. 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 4-37: 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	Numb
1	+1	*	10.64.2.23	2	0			255
2	+	*	10.64.2.23	1	0			255
3	1	*	10.64.2.23	1	0			255

- **Index #1** defines destination number manipulation of IP calls from Lync 2010 Me4diation server. All calls received from IP address 10.64.2.23 (i.e., from Mediation Server) and the destination number prefix begins with '+1', Remove the '+1' from the Number.
- **Index #2** defines destination number manipulation of IP calls from Lync 2010 Me4diation server. All calls received from IP address 10.64.2.23 (i.e., from Mediation Server) and the destination number prefix begins with '+', 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 10.64.2.23 (i.e., from Mediation Server) and the destination number prefix begins with '1', Remove the '1' from the Number.

➢ 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 4-38: Destination 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
0	-1	1	+	*	0	0			255
1	-1	1	1	*	0	0	+		255
2	-1	1	XXXXXXXXXX#	*	0	0	+1		255

- **Index #1** defines destination number manipulation of IP calls from Verizon Business SIP Trunk. All calls received from Source IP Group 1 (i.e., from Verizon Business SIP Trunk) and the destination number prefix begins with '+', do not perform any changes to the number.
- **Index #2** defines destination number manipulation of IP calls from Verizon Business SIP Trunk. All calls received from Source IP Group 1 (i.e., from Verizon Business SIP Trunk) and the destination number prefix begins with '1', add the '+' prefix to the number.
- **Index #3** defines destination number manipulation of IP calls from Verizon Business SIP Trunk. All calls received from Source IP Group 1 (i.e., from Verizon Business SIP Trunk) and the destination number length is 10 digit number, add the '+1' prefix to the number.

4.11.2 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 4-39: 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	Numt
2	*	+1	10.64.2.23	2	0			255
3	*	+	10.64.2.23	1	0			255
4	*	1	10.64.2.23	1	0			255
5	*	anonymous	10.64.2.23	20	0	7192083390		255

NPI	TON	Presentation
Not Configured	Not Configured	Allowed
Not Configured	Not Configured	Allowed
Not Configured	Not Configured	Allowed
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 10.62.2.23 (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 10.62.2.23 (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 10.62.2.23 (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 10.62.2.23 (i.e., from Lync Mediation Server) replace the 'anonymous' caller ID with a well known number i.e. 7192083390. This manipulation is performed to create a well known number in the P-Asserted-Identity header. Without this number, the Verizon Business SIP Trunk rejects the call. See below for the Source Number manipulation Tel->IP manipulation rule that restricts the caller ID to an anonymous call.

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

1. Open 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 4-40: 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	*	XXXXXXXXXX#	0	0	+1		255	Allowed
4	-1	2	*	7192083390	1	0			255	Restricted

- **Index #1** defines Source number manipulation of IP calls from Verizon Business SIP Trunk. All calls received from Source IP Group 1 (i.e., from Verizon Business SIP Trunk) and the Source number prefix begins with '+', do not perform any changes to the number.
- **Index #2** defines Source number manipulation of IP calls from Verizon Business SIP Trunk. All calls received from Source IP Group 1 (i.e., from Verizon Business SIP Trunk) and the Source number prefix begins with '1', Add a ‘+’ as a prefix to the number.
- **Index #3** defines Source number manipulation of IP calls from Verizon Business SIP Trunk. All calls received from Source IP Group 1 (i.e., from Verizon Business 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 Verizon Business SIP Trunk. All calls received from Source IP Group 1 (i.e., from Verizon Business SIP Trunk) and the Source number is 7192083390 (which is a well known number that was replace the anonymous caller ID on the source number manipulation IP->Tel above), the presentation should be set to ‘restricted’.

4.11.3 Configure Redirect Number Manipulation

In the event of a call forwarding scenario, a Diversion header needs to be added to the INVITE towards the Verizon Business SIP Trunk (as configured in Section 4.9.1 on page 61). In this case, the E-SBC copies the Destination number to the Redirect number and adds this number to the Diversion header. In order to have a well known number in the Diversion header (for Verizon Business SIP Trunk), a manipulation rule should be defined to replace the redirect number to a well known number.

➢ **To configure 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 4-41: 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	7192083390		255

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

The redirect number is changed to a well known number i.e. 7192083390.

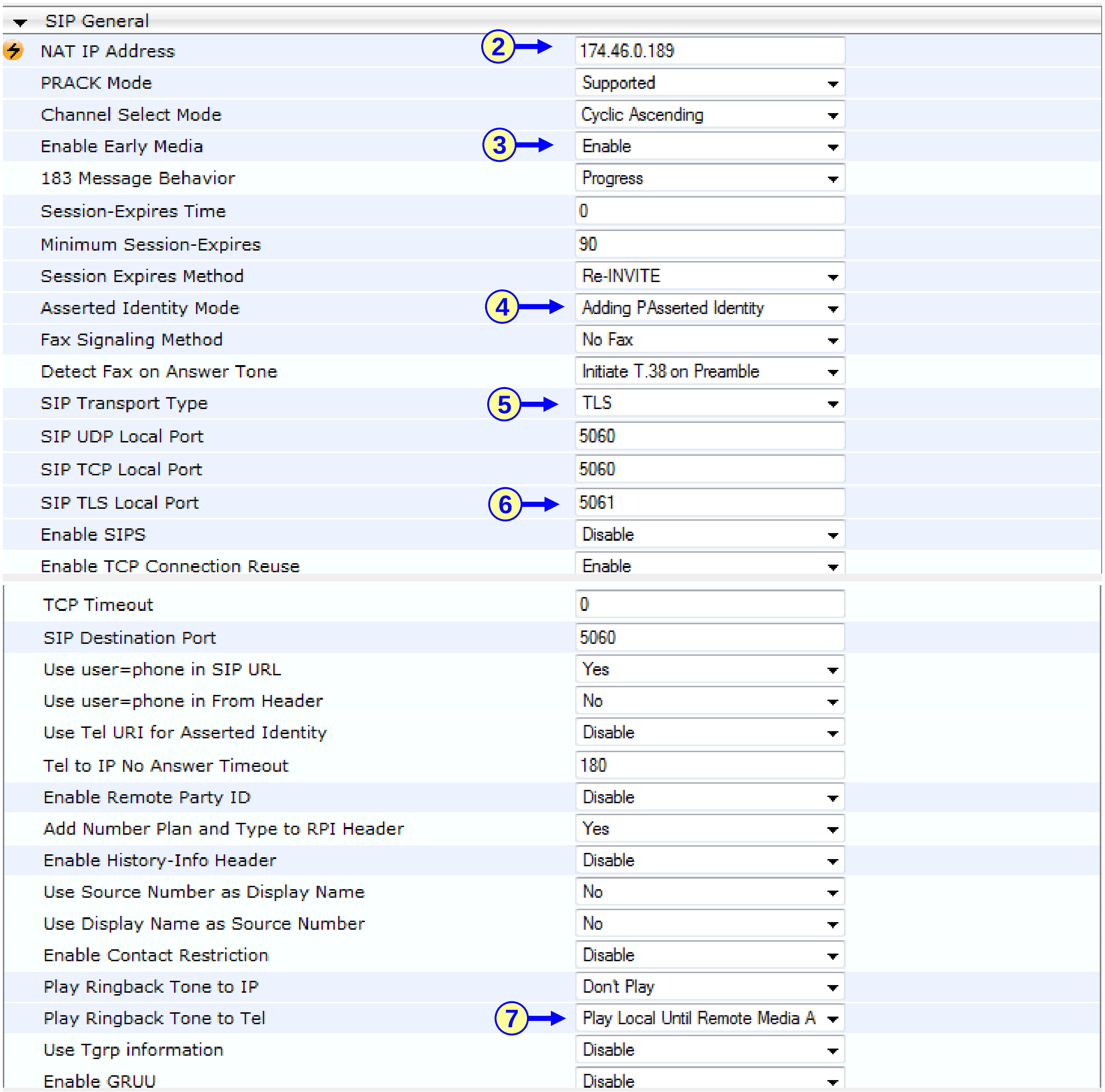
4.12 Step 14: Configuring SIP General Parameters

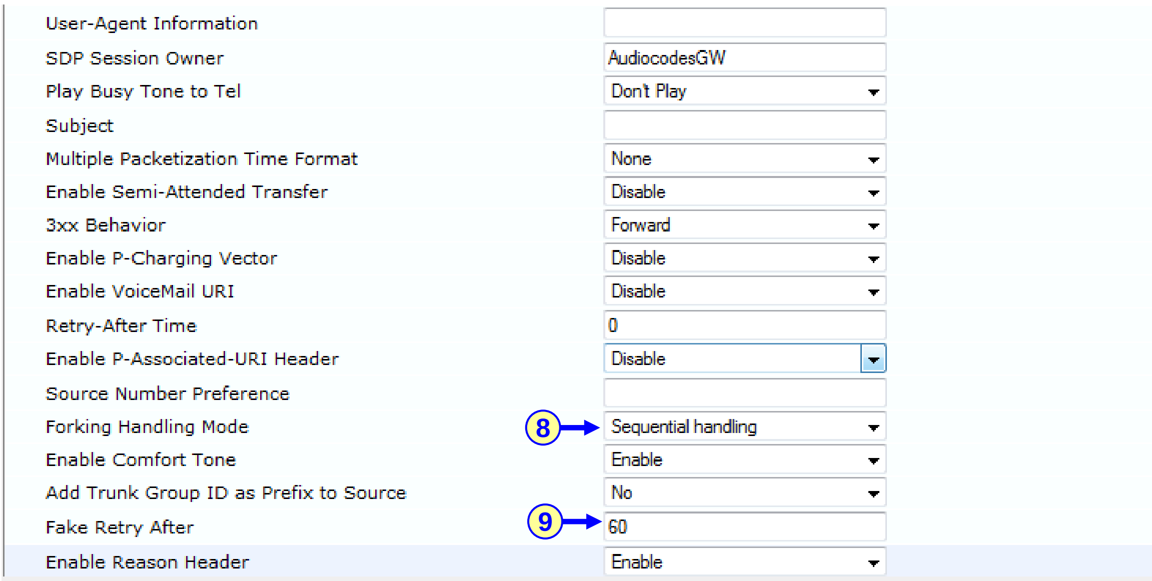
This section describes how to configure the SIP general parameters.

➢ To configure general SIP parameters:

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

Figure 4-42: SIP General Parameters Page "numbers"





2. In the 'NAT IP Ad dress' field, enter the Global (public) IP address of the E-SBC device to enable the static NAT between the E-SBC device and the Internet.
3. From the 'Enable Early Media' drop-down list, select 'Enable' to enable early media.
4. From the 'Asserted Identity Mode' drop-down list, select 'Adding PAsserted Identity'.
5. From the 'SIP Transport Type' drop-down list, select 'TLS' in case the Mediation Server is configured to use TLS transport Type.
6. In the 'SIP TLS Local Port' field, enter '5061'; this port is the listening E-SBC port for TLS transport type. This port must match the transmitting port of the Mediation Server.
7. 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.
8. 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.
9. In the 'Fake Retry After' field, enter '60' sec. This parameter determines whether the E-SBC device, upon 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.

10. 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).
11. On the left pane, click ***ini*** Parameters.
12. In the 'Parameter Name' field, enter the parameter **IGNOREALERTAFTEREARLYMEDIA**. In the 'Enter Value' field, enter “1”.
13. Click **Apply New Value**.

Figure 4-43: INI file Output Window

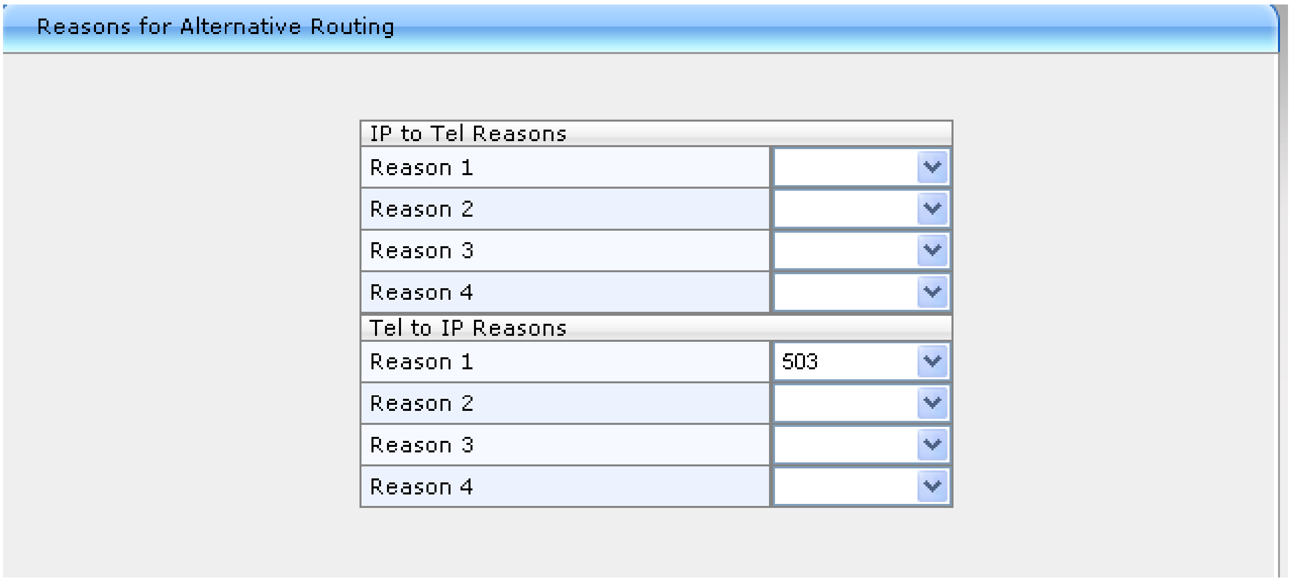
4.13 Step 15: Defining Reasons for Alternative Routing

A 503 SIP response from the Mediation Server to an INVITE must cause the Media Gateway to perform a failover. For this event to occur, you need to configure the Reasons for Alternative Routing for Tel-to-IP calls to be a 503 SIP response.

➢ To define SIP Reason for Alternative Routing:

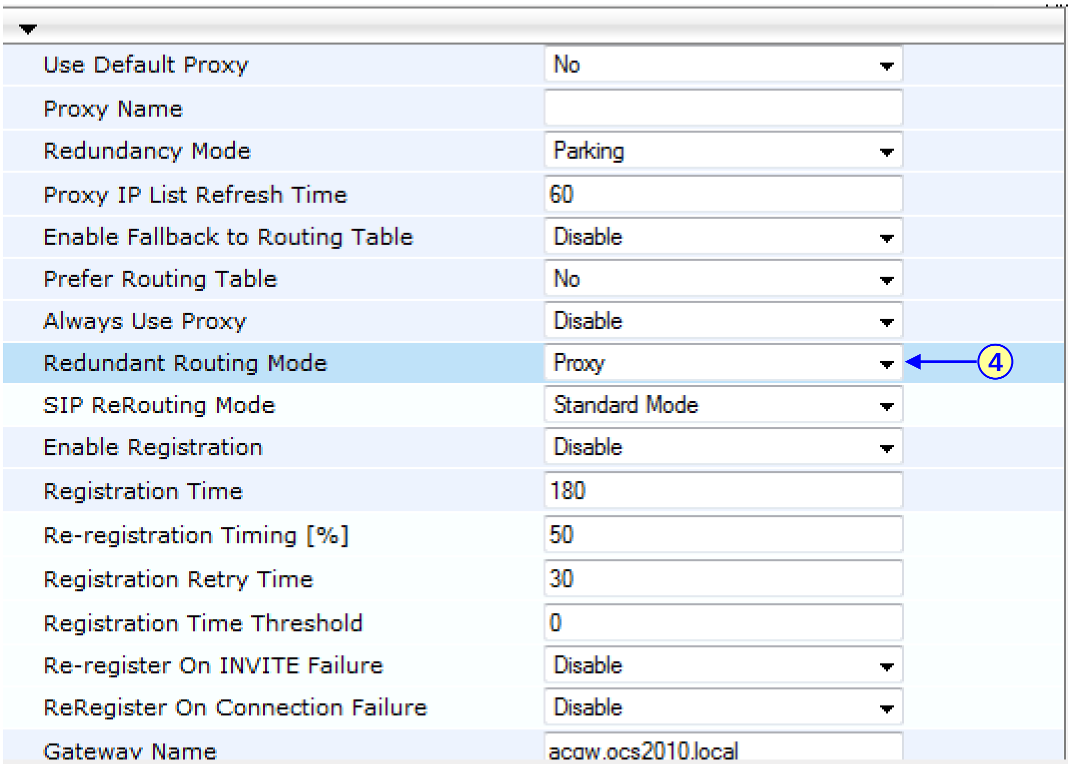
1. Open the 'Reasons for Alternative Routing' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Routing** submenu > **Alternative Routing Reasons**).

Figure 4-44: 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.

Figure 4-45: ‘Proxy & Registration’ Page



Reader's Notes

Mediant 1000 E-SBC

Configuration Note

Connecting Verizon Business SIP Trunk
to
Microsoft® Lync Server
Using
AudioCodes Mediant 1000 E-SBC

www.audiocodes.com