Lync Server 2010 and Spitfire

Mediant™ E-SBC Series

SIP Protocol

Configuration Note

Connecting Microsoft® Lync™ Server 2010 and Spitfire SIP Trunk using AudioCodes Mediant E-SBC Series

SPITFIRE®

Microsoft® Lync™

AudioCodes

Version 6.4

February 2012

Document #: LTRT-39110

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Notice

This document describes how to connect the Microsoft® Lync™ Server 2010 with Spitfire SIP Trunking using the AudioCodes Mediant E-SBC series, which includes the Mediant 800 MSBG, Mediant 800 Gateway and E-SBC, Mediant 1000 MSBG, Mediant 1000B Gateway and E-SBC, and Mediant 3000 Gateway and E-SBC.

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Date Published: February-13-2012

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Abbreviations and Terminology

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

Note: Throughout this manual, unless otherwise specified, the term *E-SBC device* refers to the Mediant 800 Gateway and E-SBC, Mediant 800 MSBG, Mediant 1000B Gateway and E-SBC, Mediant 1000 MSBG and the Mediant 3000 Gateway and E-SBC.

Table 1-1: Acronym Descriptions

Acronym	Description
Transferee	The party being transferred to the transfer target
Transferor	The party initiating the transfer
Transfer target	The new party being introduced into a call with the transferee
Blind or semi-attended transfer	The transferor having a session in hold state with the transferee and initiating the transfer by a consultation call to the target performs the transfer while the target is in ringing state
Attended transfer or transfer on conversation	The transferor waits to be in conversation state with the target before completing the transfer
CLIP	Calling Line Identification Presentation
CNIP	Calling Name Identification Presentation
CLIR	Calling Line Identification Restriction
CNIR	Calling Name Identification Restriction
COLP	Connected Line Identification Presentation
CONP	Connected Name Identification Presentation
COLR	Connected Line Identification Restriction
CONR	Connected Name Identification Restriction
CRC	Customer Relationship Centre
PG	SIP GW XXX Peripheral Gateway
ICM	SIP GW XXX Intelligent Call Manager
CCM	SIP GW XXX Call Manager
CVP	Customer voice Portal
BC	ALU Business Contact
CTI	Computer Telephony Integration

1 Introduction

AudioCodes Gateways and E-SBC have been tested and certified with Spitfire SIP Trunking.

This document describes how to setup the AudioCodes Mediant 1000 gateway to function as an E-SBC, with the Spitfire SIP Trunk and Microsoft Lync Server 2010 communication platform.

This configuration note is intended for Installation Engineers or AudioCodes and Spitfire Partners who are installing and configuring the Spitfire SIP Trunking and Lync Server 2010 Communication platform, to place VoIP calls using the AudioCodes E-SBC.

The Mediant 800 MSBG 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 800 Media Gateway and SBC enable connectivity and security between small and medium businesses (SMB) and service providers' VoIP networks. The Mediant 800 SBC functionality provides perimeter defense for protecting the enterprise from malicious VoIP attacks, mediation for allowing the connection of any PBX and/or IP-PBX to any service provider, and service assurance for service quality and manageability.

The Mediant 1000 MSBG is an all-in-one multi-service access solution product for Service Providers (SME's) offering managed services and distributed Enterprises seeking integrated services. This multi-service business gateway is designed to provide converged Voice and Data services for business customers at wire speed, while maintaining SLA parameters for superior voice quality.

The Mediant 1000B media gateway and SBC enables connectivity and security between small and medium businesses and service providers' VoIP networks. The Mediant 1000B SBC functionality provides perimeter defense for protecting the enterprise from malicious VoIP attacks, mediation for allowing the connection of any PBX and/or IP-PBX to any service provider, and service assurance for service quality and manageability. The Mediant 1000B media gateway functionality is based on field-proven VoIP services.

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

Note: The scope of this document does not cover security aspects for connecting the SIP Trunk to the Lync Server 2010 environment. Security measures should be implemented in accordance with your organization's security policies. For basic security guidelines, refer to the *AudioCodes Security Guidelines*.

Reader's Notes

2 Components Information

2.1 AudioCodes Gateway Version

Table 2-1: AudioCodes Gateway Version

Gateway Vendor	AudioCodes
Model	Mediant 800 Media Gateway and E-SBC, Mediant 800 MSBG, Mediant 1000 MSBG, Mediant 1000B Media Gateway and E-SBC, Mediant 3000 Media Gateway and E-SBC
Software Version	SIP_6.40A.027.001
Interface Type	SIP/IP
VoIP Protocol	SIP/UDP – to the Spitfire Sip Trunk SIP/TCP or TLS – to the Lync FE Server
Additional Notes	None

2.2 Spitfire SIP Trunking Version

Table 2-2: Spitfire Version

Service Vendor	Spitfire
Models	
Software Version	N/A
VoIP Protocol	SIP
Additional Notes	None

2.3 Lync Server 2010 Version

Table 2-3: Lync Server 2010 Version

PBX Vendor	Microsoft
Models	Microsoft Lync
Software Version	RTM: Release 2010 4.0.7577.0
VoIP Protocol	SIP
Additional Notes	None

2.4 Topology

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

- An enterprise has a deployed Lync Server 2010 in its private network for enhanced communication within the company.
- The enterprise decides to offer its employees enterprise voice capabilities and to connect the company to the PSTN network using the Spitfire SIP Trunking service.
- The AudioCodes Enterprise Session Border Controller (E-SBC) is used to manage the connection between the Enterprise LAN and the ITSP SIP trunk.

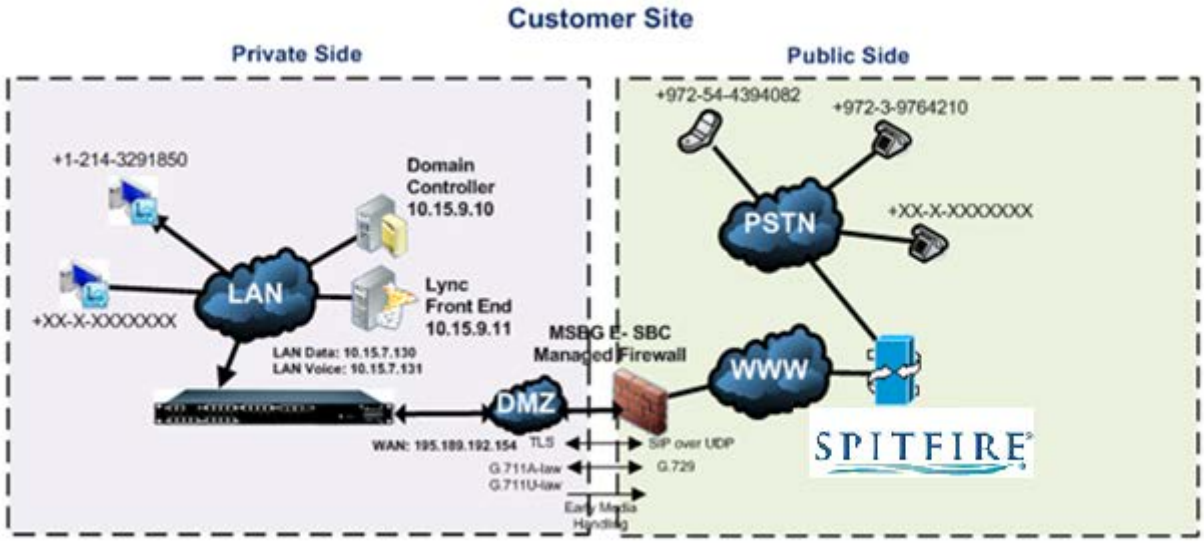
  The 'session' refers to the real-time voice session using IP SIP signaling protocol. The 'border' refers to the IP-to-IP network border between the Microsoft Lync network in the Enterprise LAN and the Spitfire SIP trunk in the public network.

Figure 2-1 below illustrates a typical topology of using the E-SBC device to connect the Lync Server 2010 LAN to the Spitfire SIP Trunking site.

The setup requirements are characterized as follows:

- While the Lync Server 2010 environment is located on the Enterprise's LAN, the Spitfire SIP Trunks are located on the WAN.
- Since the Mediant 1000 MSBG is used, the internal data routing capabilities of the device are used. Consequently, a separate WAN interface is configured in the LAN.
- Lync Server 2010 works with the TLS transport type, while the Spitfire SIP trunk works on the SIP over UDP transport type.
- Lync Server 2010 supports G.711A-law and G.711U-law coders, while the Spitfire SIP Trunk also supports the same coders' type.
- Support for early media handling.

Figure 2-1: Topology



3 Configuring Lync Server 2010

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

- Configuring the E-SBC device as an IP/PSTN Gateway. See Section 3.1 on page 14.
- Associating the IP/PSTN Gateway with the Mediation Server. See Section 3.2 on page 18.
- Configuring a Route to utilize the SIP trunk connected to the E-SBC device. See Section 3.3 on page 22.

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

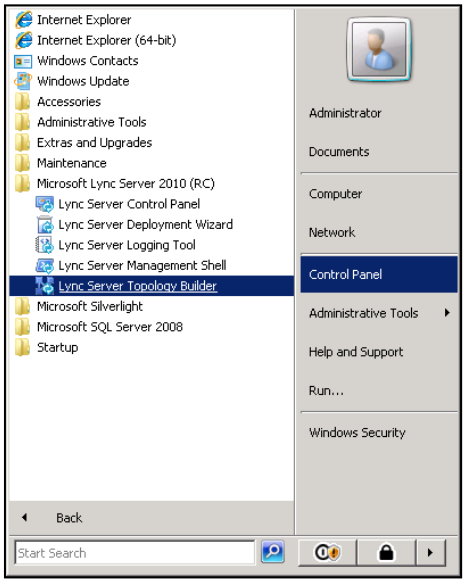
3.1 Configuring the E-SBC device as IP/PSTN Gateway

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

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

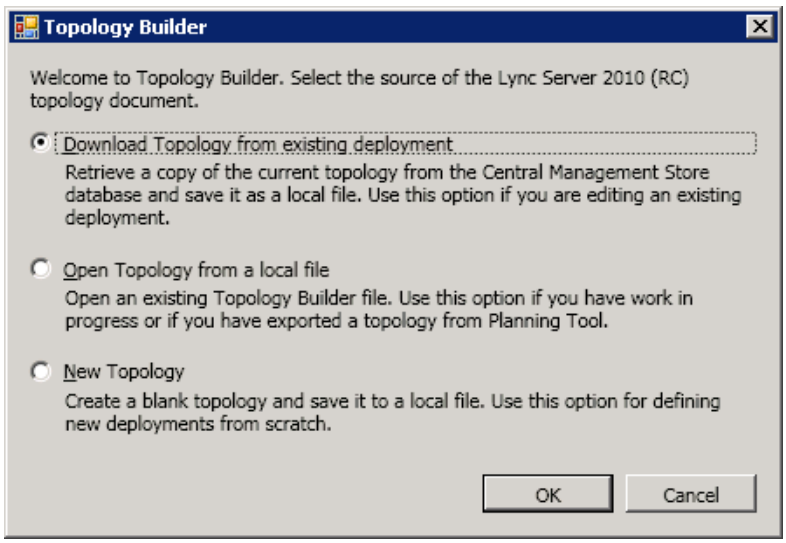
1. On the server where the Topology Builder is located, start the Lync Server 2010 Topology Builder (**Start** > **All Programs** > **Lync Server Topology Builder**).

Figure 3-1: Opening the Lync Server Topology Builder



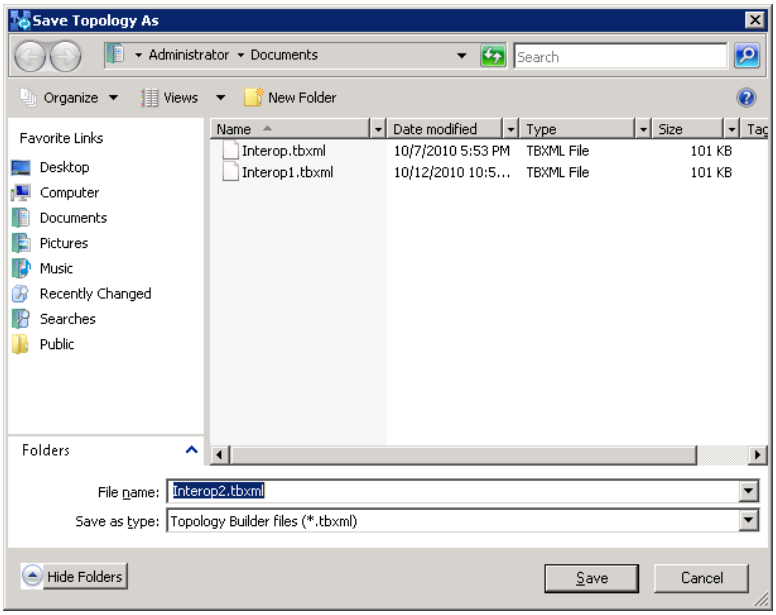
The following screen is displayed:

Figure 3-2: Topology Builder Options



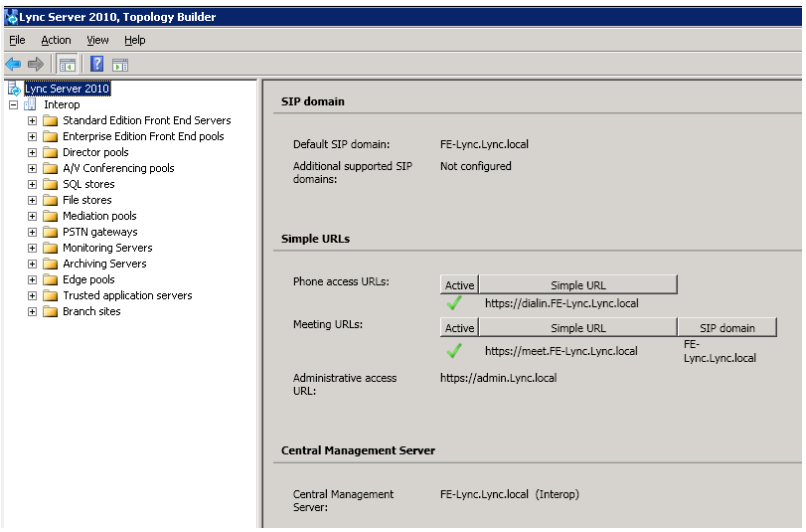
2. Click the **Download Topology from the existing deployment** option, and then click **OK**; you are prompted to save the Topology which you have downloaded.

Figure 3-3: Save Topology



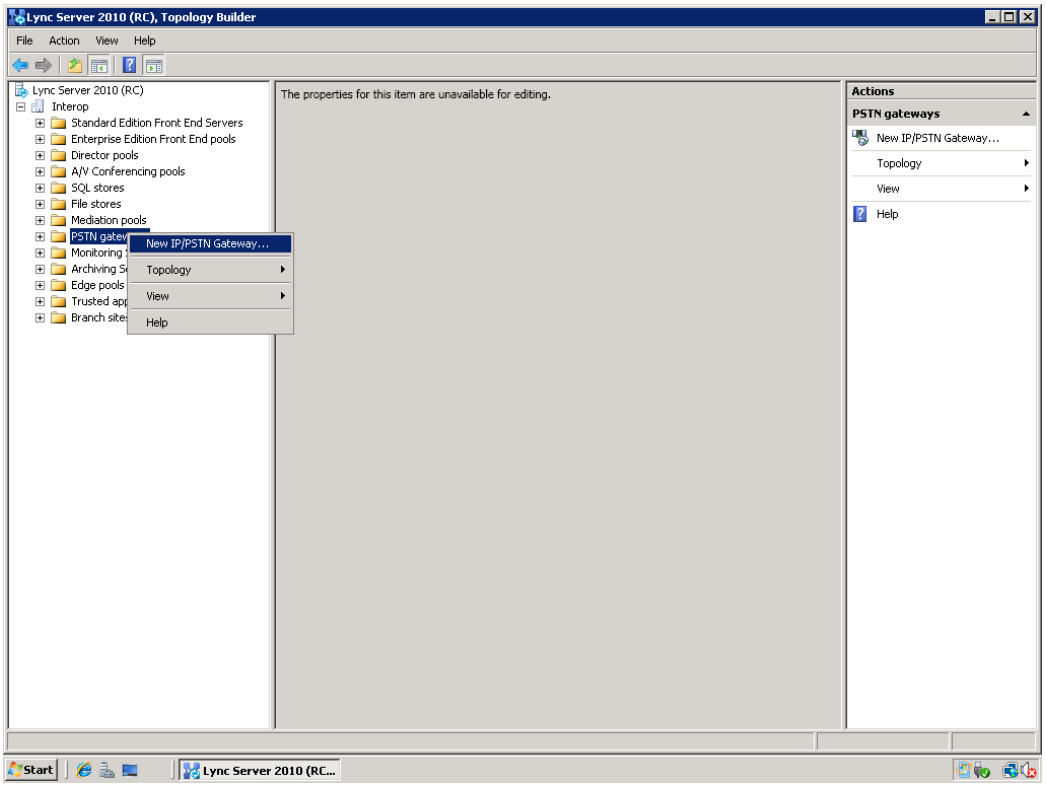
3. In the 'File name' field, enter the new filename, and then click **Save**. This enables you to rollback from any changes you made during the installation. The Topology Builder screen with the topology downloaded is displayed.

Figure 3-4: Downloaded Topology



4. Expand the 'PSTN Gateway' folder, and then choose **New IP/PSTN Gateway**.

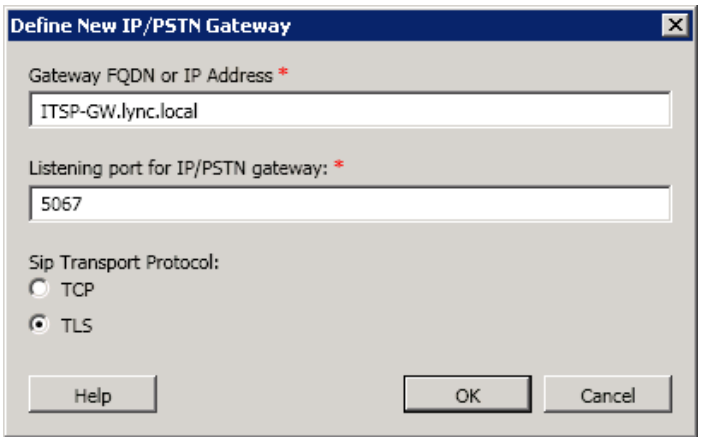
Figure 3-5: New IP/PSTN Gateway



5. In the 'Gateway FQDN or IP Address' field, enter the FQDN of the E-SBC (i.e. 'ITSP-GW.lync.local'), and then click **OK**.

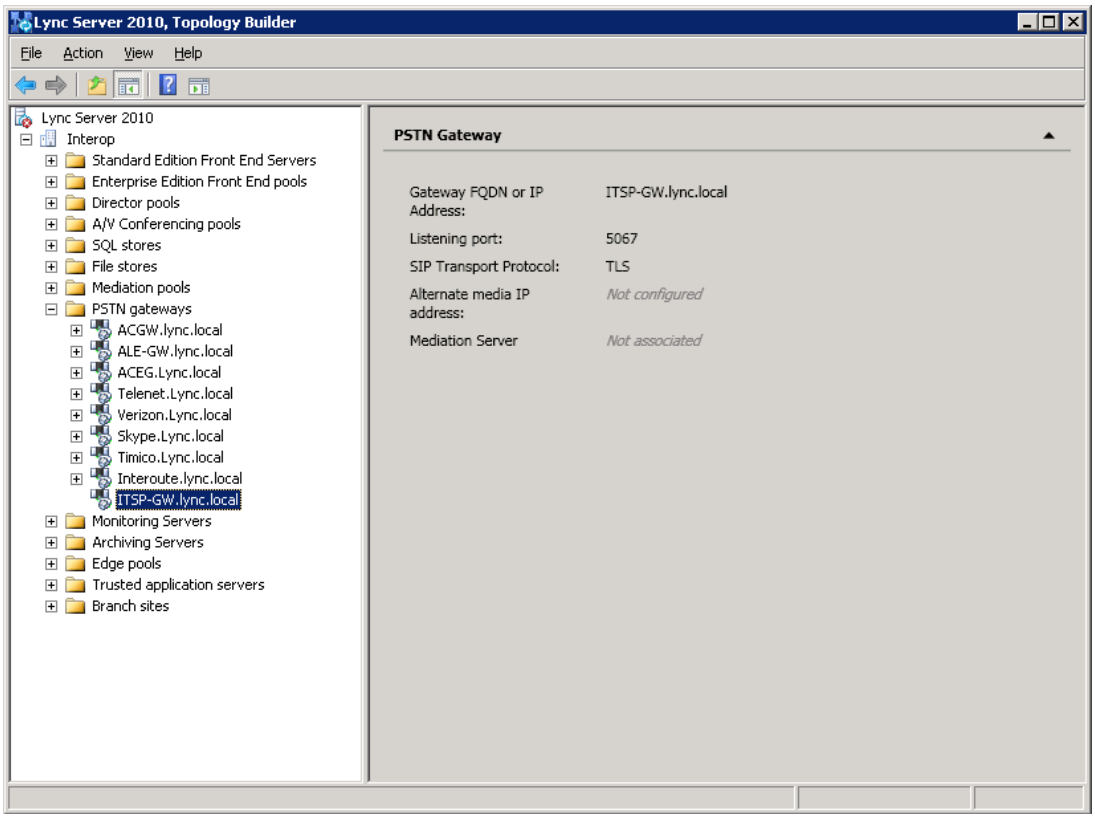
Note: The listening port for the Gateway is **5067** and the SIP Transport Protocol is **TLS**.

Figure 3-6: Define New IP/PSTN Gateway



The E-SBC device has now been added as an IP/PSTN Gateway.

Figure 3-7: IP/PSTN Gateway



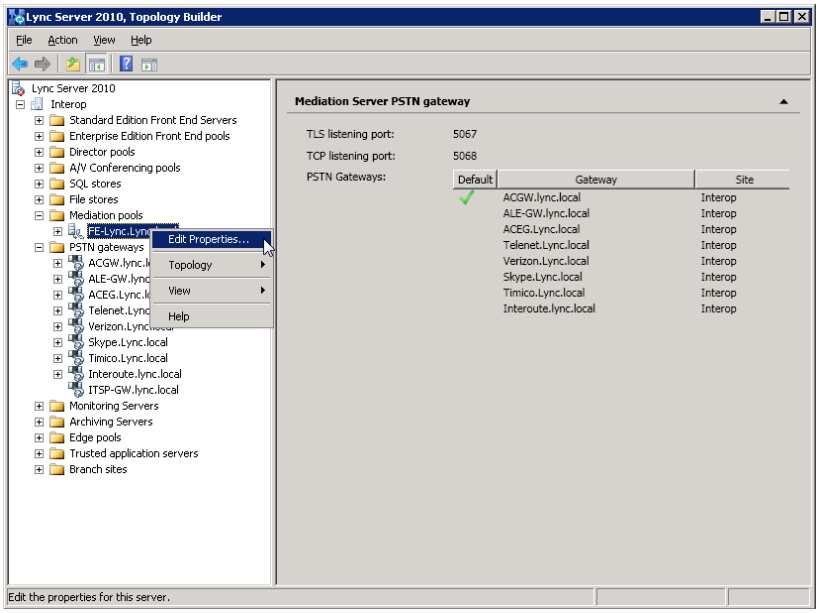
3.2 Associating the IP/PSTN Gateway with the Mediation Server

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

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

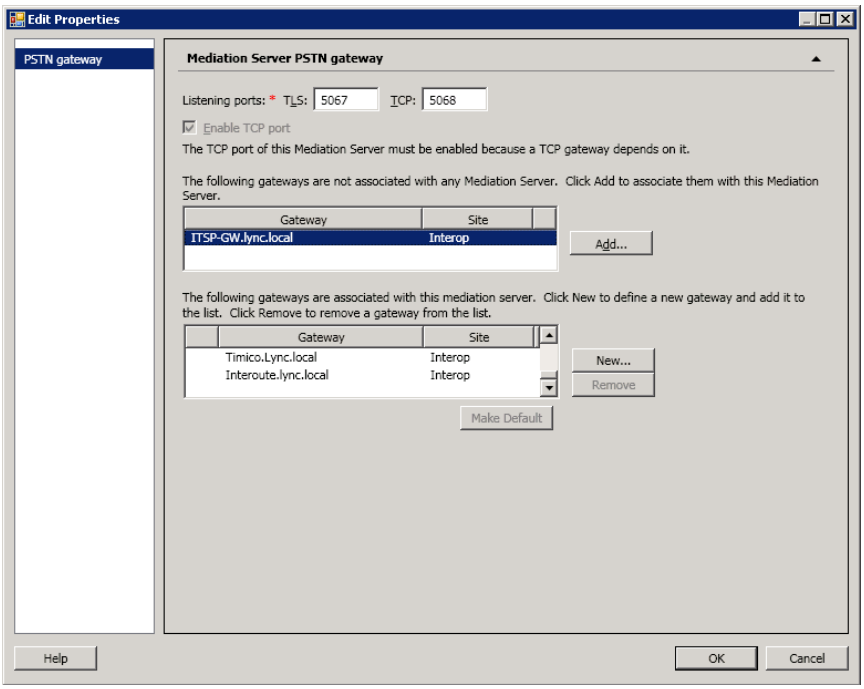
1. Expand the 'Mediation pools' folder.
2. Expand the 'FE-Lync.Lync.local' folder and then choose **Edit Properties**.

Figure 3-8: Associating Mediation Server with IP/PSTN Gateway



The following screen is displayed:

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



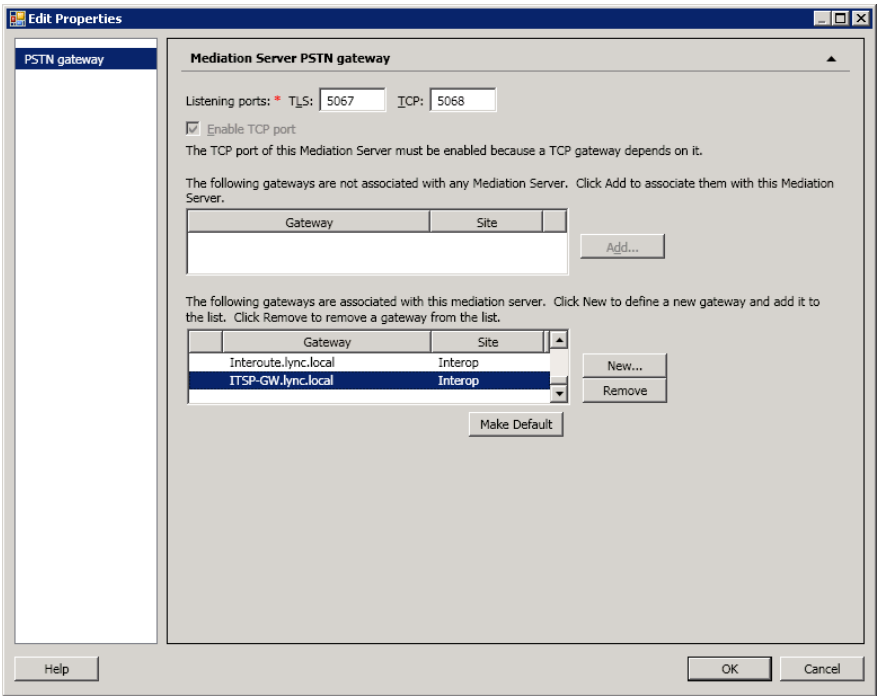
3. In the top-left menu pane, choose **PSTN gateway.**
4. In the Mediation Server PSTN gateway pane, click the E-SBC gateway (ITSP-GW.lync.local).

5. Click **Add** to associate it with this Mediation Server.

Note: 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.

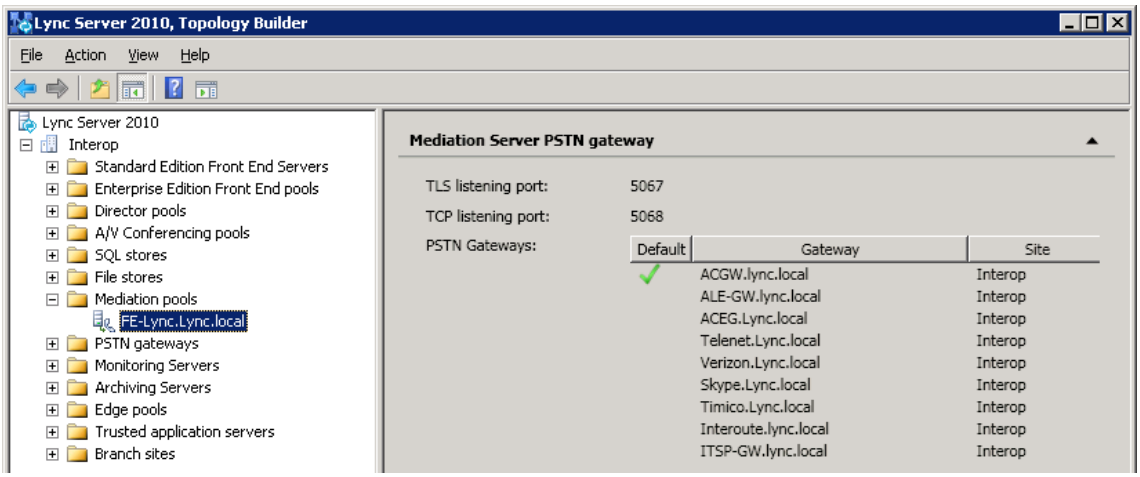
The following screen appears:

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



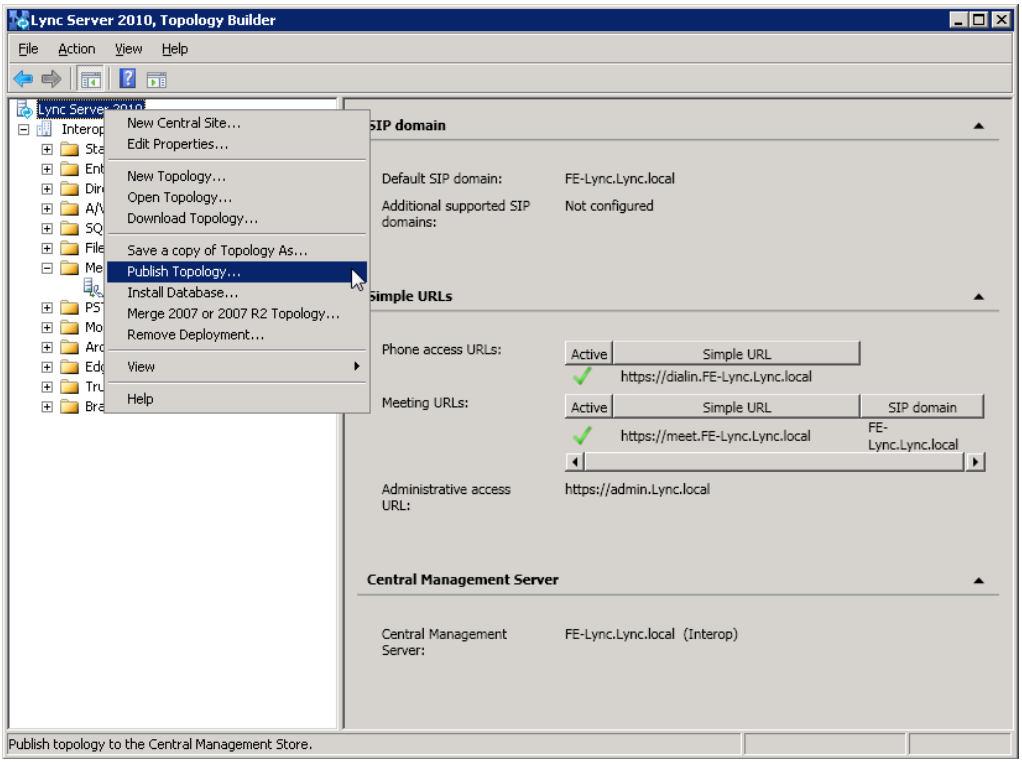
6. Click **OK**; the following screen appears:

Figure 3-11: Media Server PSTN Gateway Association Properties



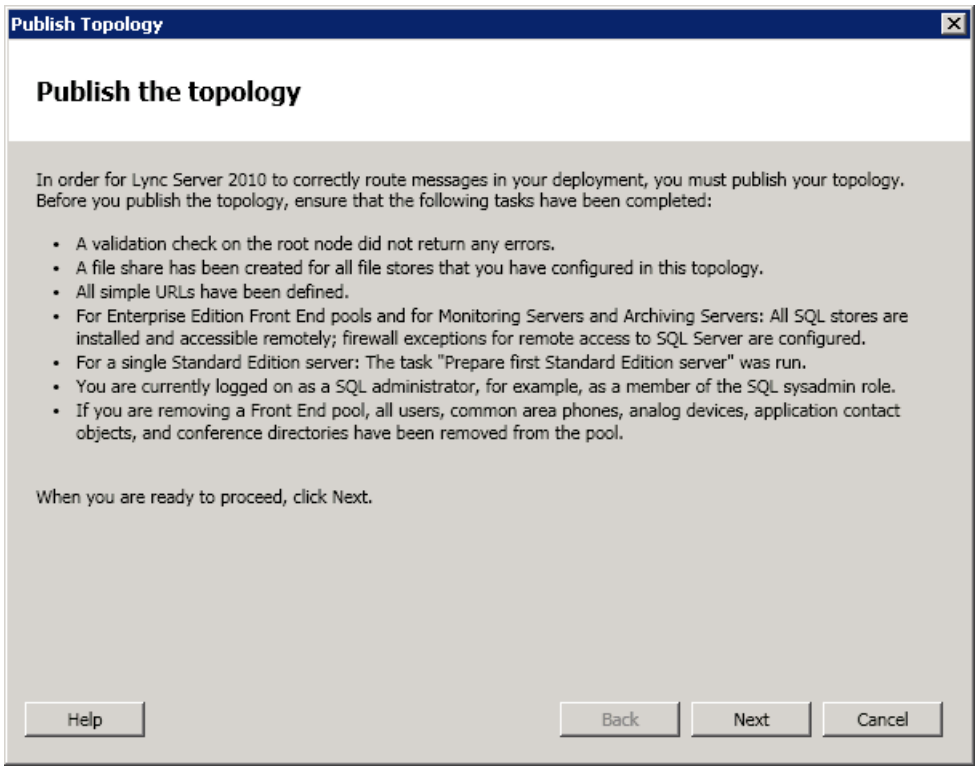
7. In the Lync Server 2010 main menu, choose **Publish Topology**.

Figure 3-12: Publishing Topology



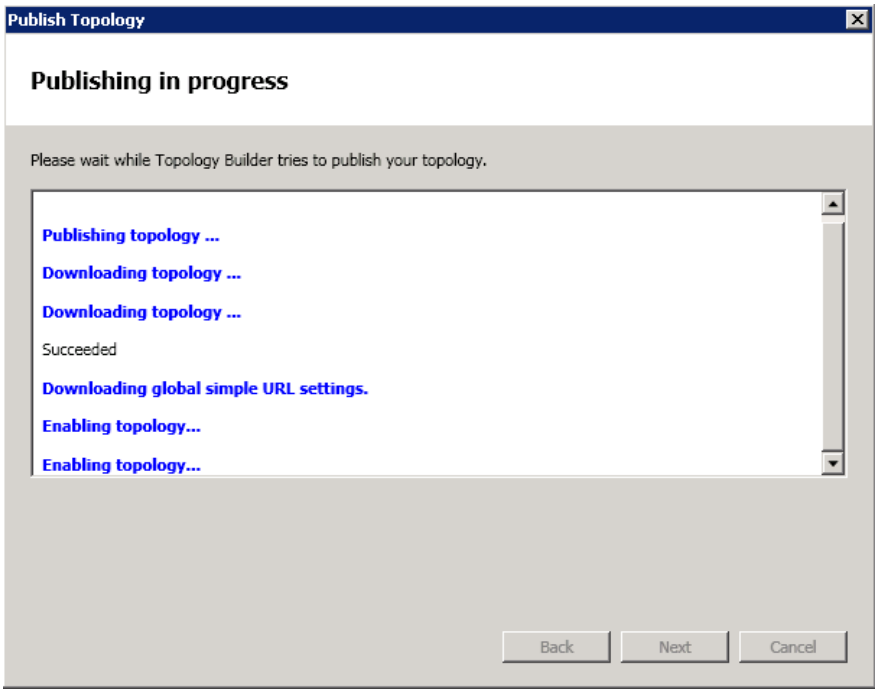
The following screen appears.

Figure 3-13: Publish Topology Confirmation



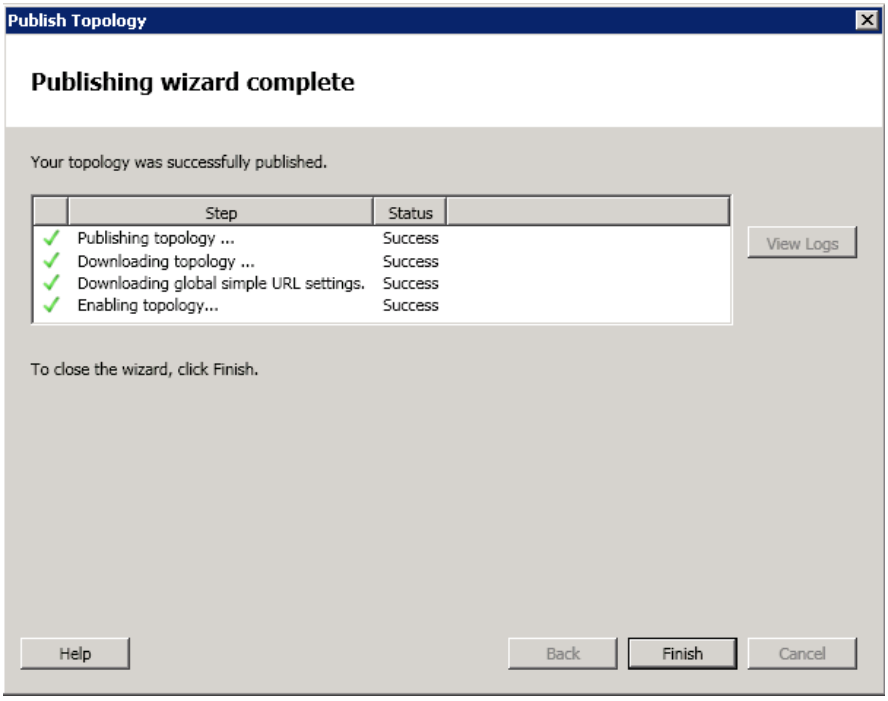
8. Click **Next**; the Topology Builder attempts to publish your topology.

Figure 3-14: Publish Topology Progress screen



9. Wait until the publish topology process has ended successfully.

Figure 3-15: Publish Topology Successfully Completed



10. Click **Finish**.

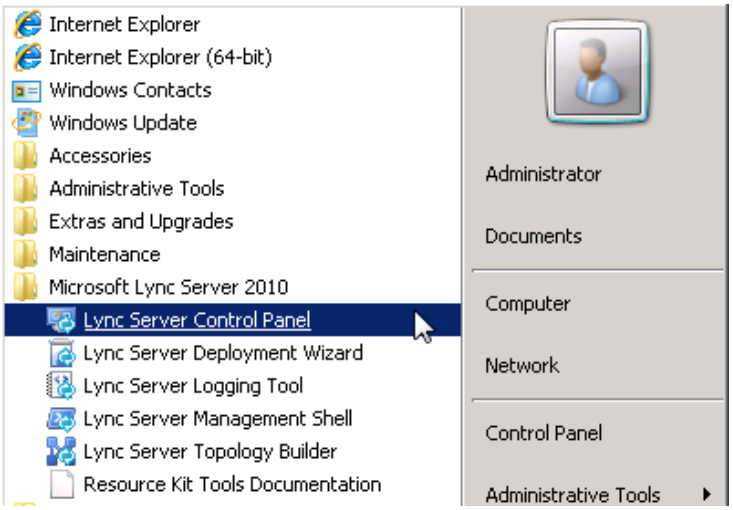
3.3 Configuring the Route on the Lync Server 2010

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).
2. Click **Start**.
3. Click **All Programs**, and select **Lync Server Control Panel**.

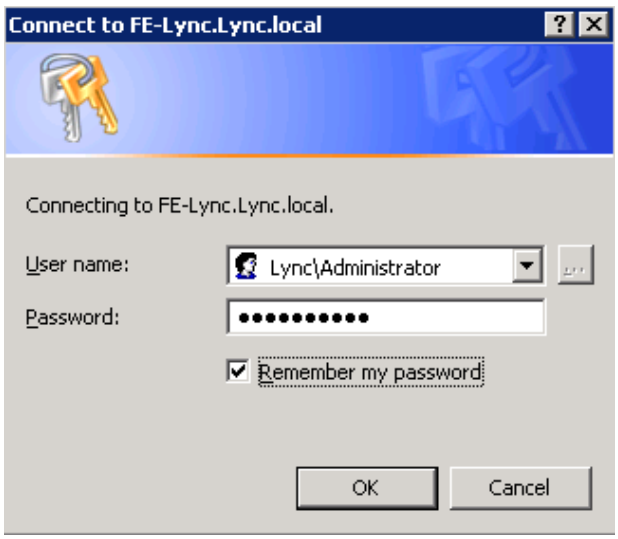
Figure 3-16: Opening the Lync Server Control Panel



The **Connect to FE-Lync.Lync.local** screen appears.

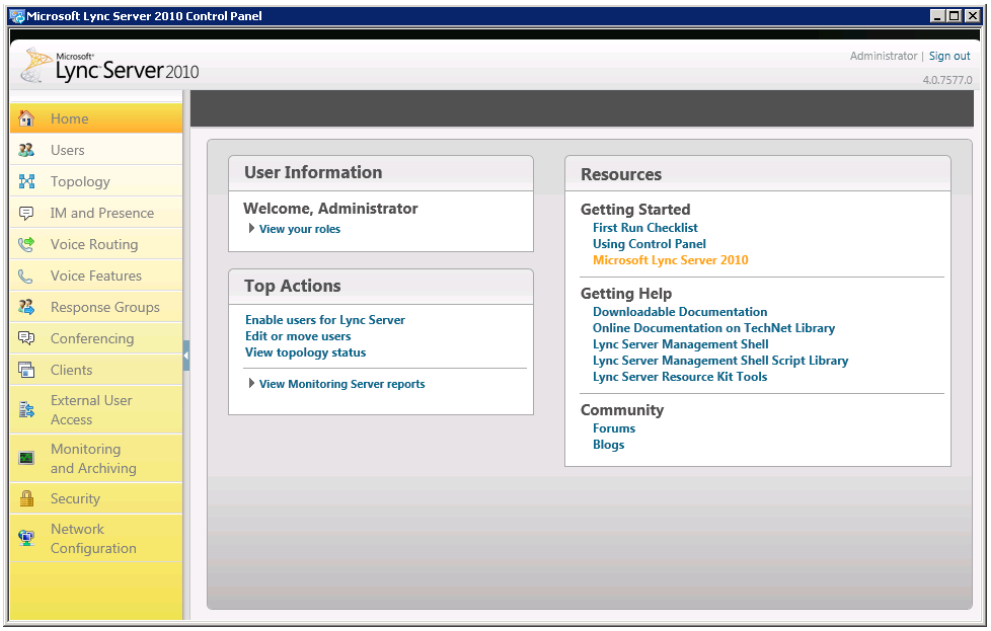
4. Enter your domain Username and Password and then click **OK**.

Figure 3-17: Lync Server Credentials



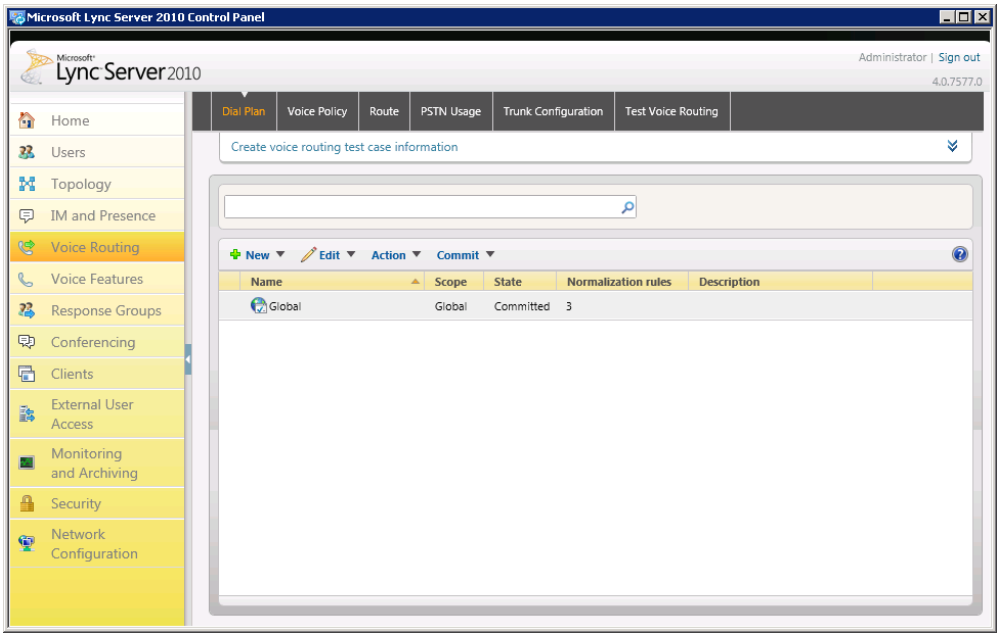
The CSCP Home page is displayed.

Figure 3-18: CSCP Home page



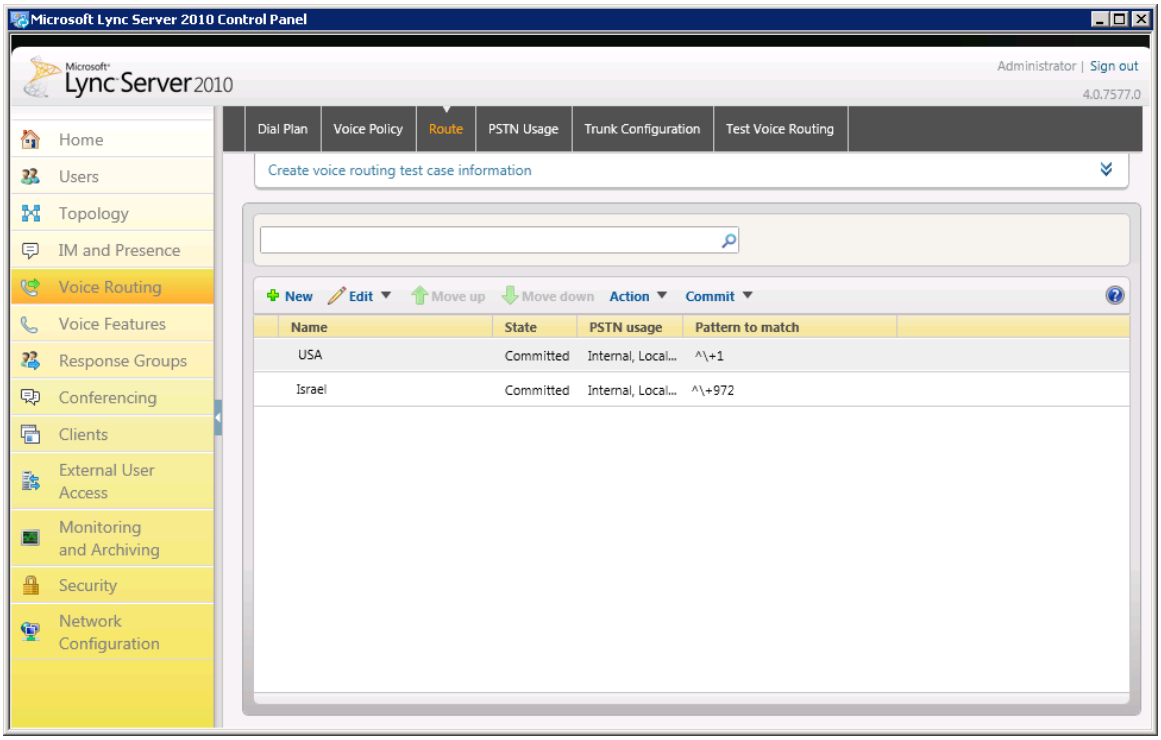
5. In the Navigation pane, select the **Voice Routing** menu option.

Figure 3-19: Voice Routing Option



6. In the Voice Routing menu at the top of the page, click the **Route** tab.

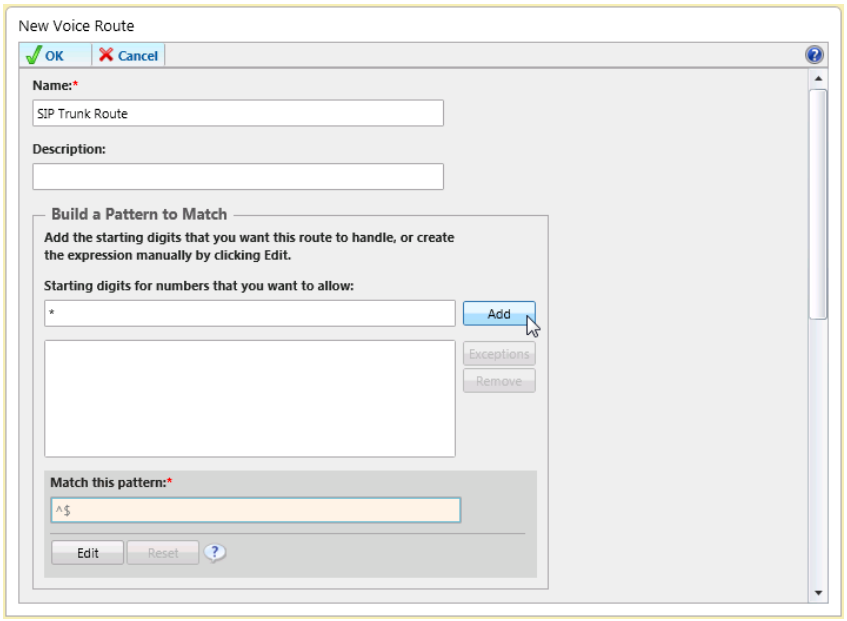
Figure 3-20: Route Option



7. In the content area toolbar, click 

; the following screen appears:

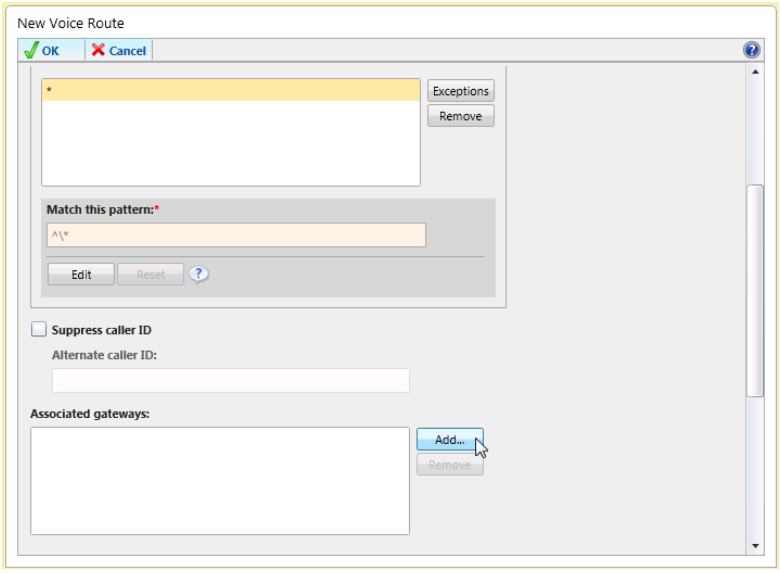
Figure 3-21: Adding New Voice Route



8. In the New Voice Route page, enter a Name for this route (i.e. SIP Trunk Route).
9. Under 'Build a Pattern to Match', add the starting digits you wish this route to handle. In this example, the pattern to match is '*', which means "to match all numbers".
10. Click **Add**.

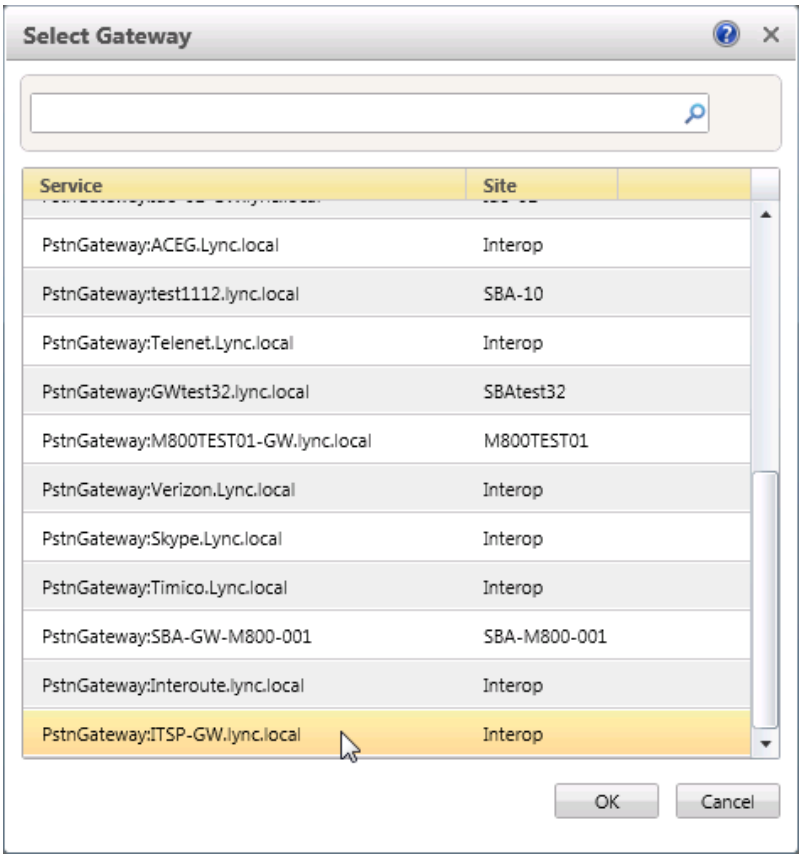
11. Associate the route with the E-SBC IP/PSTN gateway you created above by scrolling down to the Associated Gateways pane and click **Add**.

Figure 3-22: Adding New E-SBC Gateway



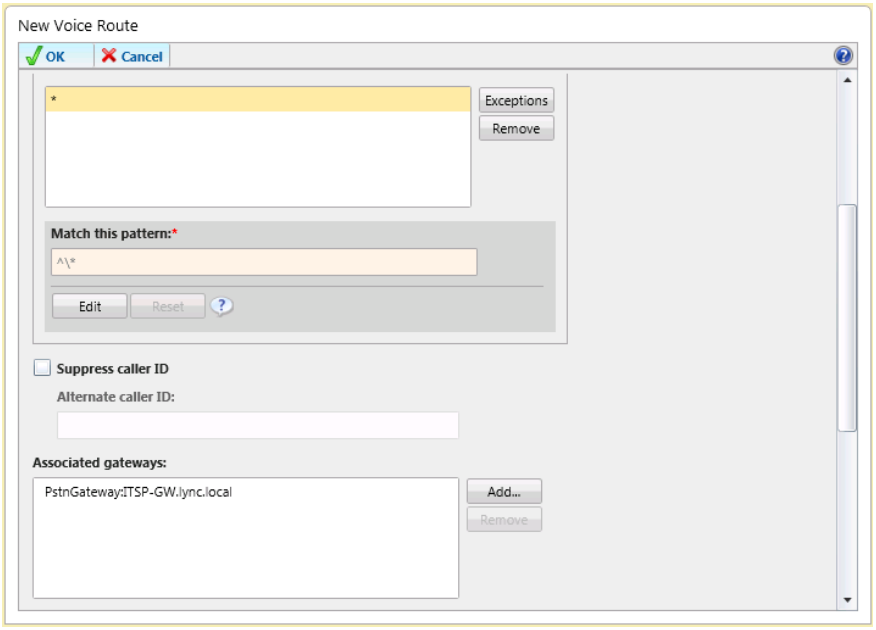
A list of all the deployed Gateways is displayed.

Figure 3-23: List of Deployed Gateways



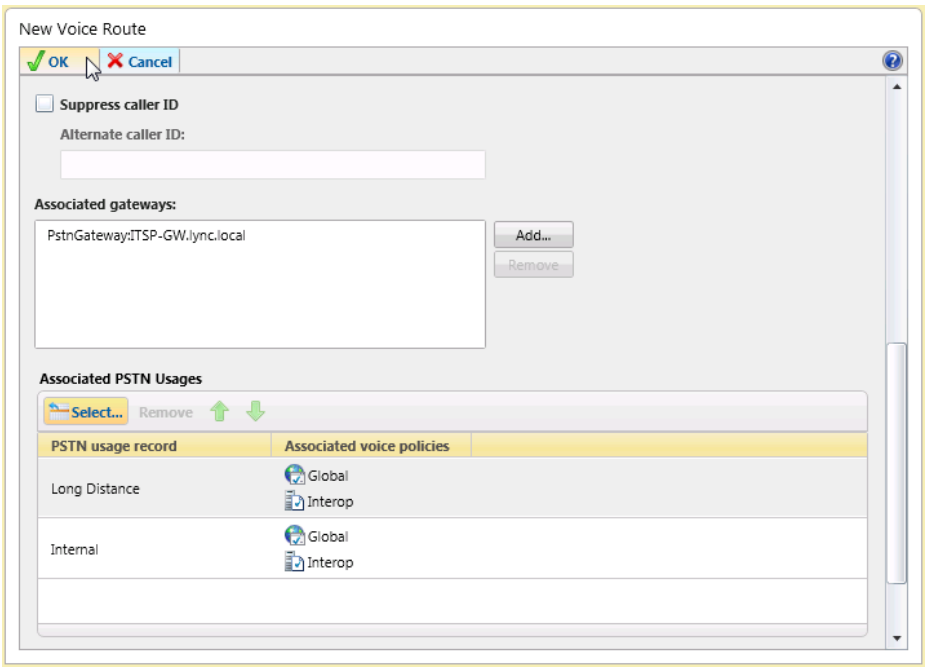
12. Select the E-SBC Gateway you created above and click **OK**.

Figure 3-24: Selected the E-SBC Gateway



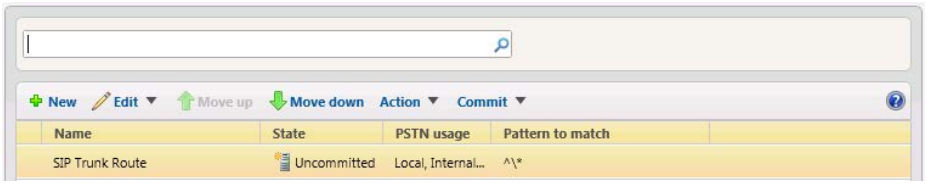
13. In the Associated PSTN Usages toolbar, click **Select** and add the associated PSTN Usage.

Figure 3-25: Associating PSTN Usage to E-SBC Gateway



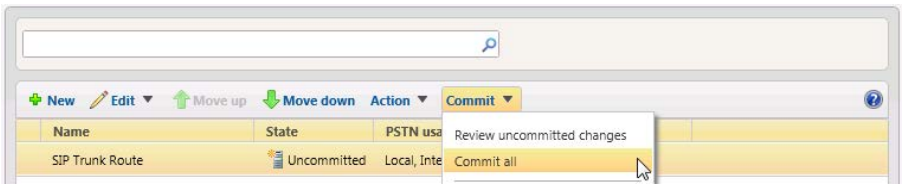
14. In the toolbar at the top of the New Voice Route pane, click **OK.** The New Voice Route (Uncommitted) is displayed.

Figure 3-26: Confirmation of New Voice Route



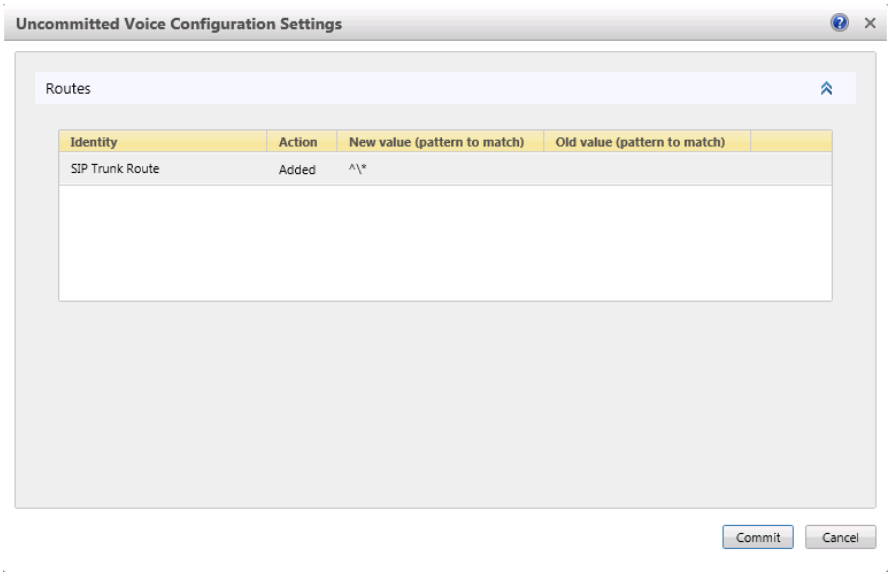
15. On the Content area Toolbar, from the Commit drop-down list, select **Commit all**.

Figure 3-27: Committing Voice Routes



16. In the Uncommitted Voice Configuration Settings page, click **Commit**.

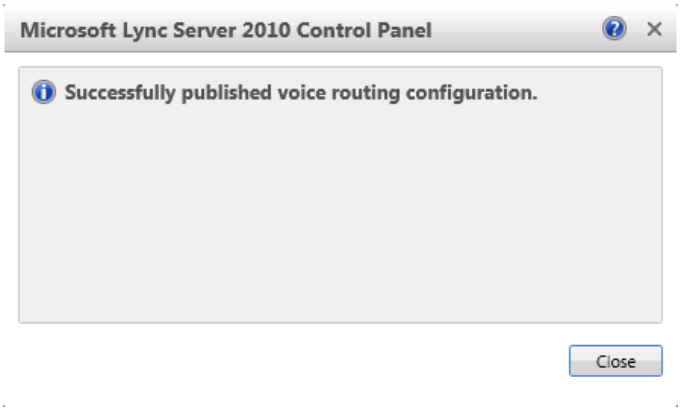
Figure 3-28: Uncommitted Voice Configuration Settings



A message is displayed, confirming a successful voice routing configuration.

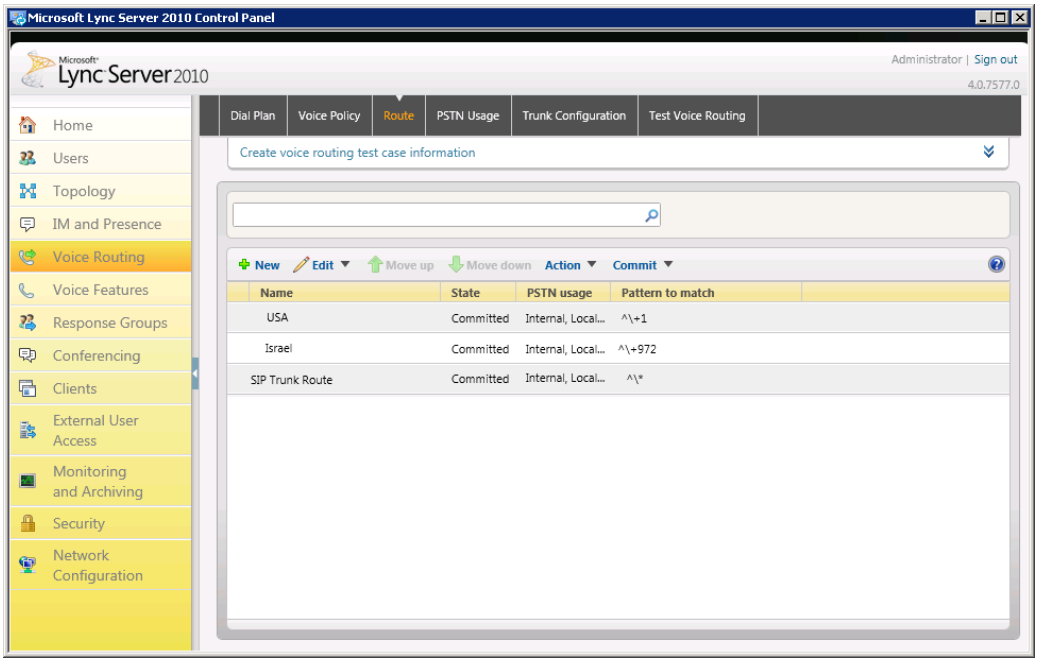
17. Click **Close**.

Figure 3-29: Voice Routing Configuration Confirmation



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

Figure 3-30: Voice Routing Screen Displaying Committed Routes



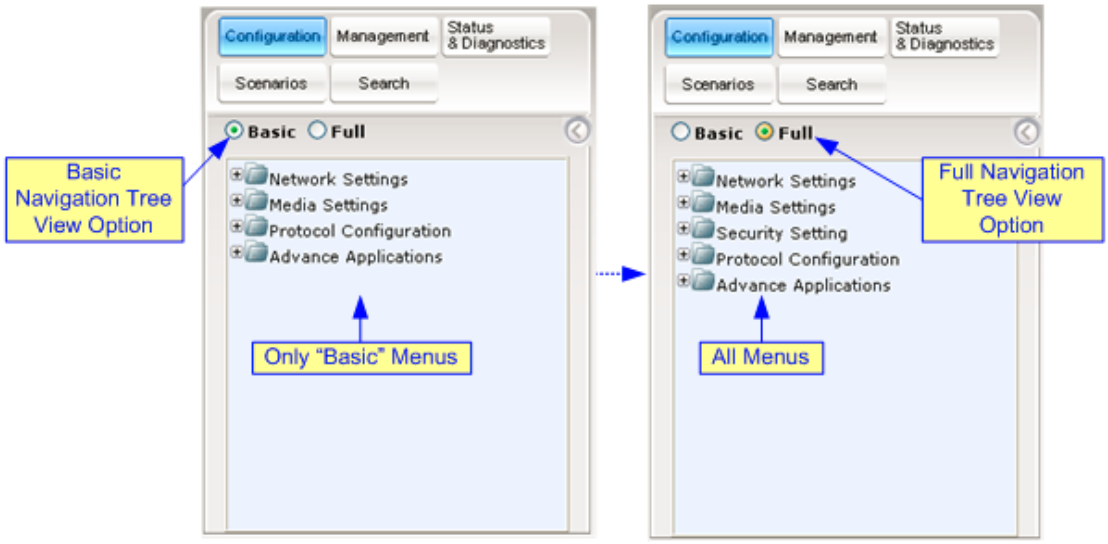
4 Configuring the E-SBC Device

This section describes the following steps for configuring the E-SBC device in the Spitfire SIP Trunking environment. The following describes the steps required to configure the E-SBC device:

- **Step 1:** Configuring IP Addresses. See Section 4.1 on page 30.
- **Step 2:** Configuring Port Forwarding. See Section 4.2 on page 33.
- **Step 3:** Enabling Application Mode. See Section 4.3 on page 35.
- **Step 4:** Configuring Secure Real-Time Transport Protocol (SRTP). See Section 4.4 on page 36.
- **Step 5:** Configuring IP Media. For more information, see Section 4.5 on page 37.
- **Step 6:** Configuring SIP General Parameters. For more information, see Section 4.6 on page 38.
- **Step 7:** Configuring DTMF & Dialing. See Section 4.7 on page 40.
- **Step 8:** Configuring Coders. See Section 4.8 on page 41.
- **Step 9:** Configuring Proxy & Registration. See Section 4.9 on page 42.
- **Step 10:** Configuring Proxy Sets Table. See Section 4.10 on page 43.
- **Step 11:** Configuring Coder Groups. See Section 4.11 on page 45.
- **Step 12:** Configuring IP Profile. See Section 4.12 on page 46. 4.13
- **Step 13:** Configuring IP Group Tables. See Section 4.13 on page 48.
- **Step 14:** Configuring Account Table. See Section 4.14 on page 50.
- **Step 15:** Configuring Routing. See Section 4.15 on page 51.
- **Step 16:** Configuring Manipulation Tables. See Section 4.16 on page 53.
- **Step 17:** Configuring Message Manipulations. See Section 4.17 2on page 55.
- **Step 18:** Configuring SIP TLS Connection. See Section 4.18 on page 58.
- **Step 19:** Resetting the Gateway. See Section 4.184.19 on page 64.

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



4.1 Step 1: Configuring IP Addresses

This step describes how to configure LAN IP addresses when the internal data-routing capabilities of the E-SBC device are used in order to connect to the Spitfire SIP Trunk. In this case, you must configure a separate WAN interface as described 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 below.

4.1.1 Configuring LAN IP Addresses

The following describes how to configure VoIP IP Settings and LAN Data-Routing IP Settings.

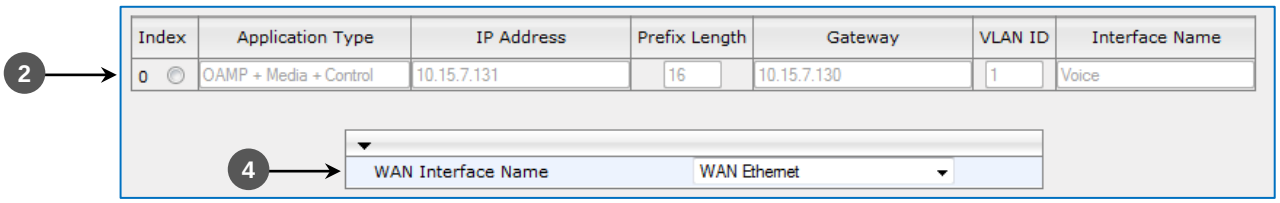
4.1.1.1 Configuring VoIP IP Settings

The section describes how to configure VoIP IP Settings.

➢ **To configure the VoIP IP settings:**

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

Figure 4-2: IP Settings



2. Click the **Index** option corresponding to the "OAMP + Media + Control" (i.e., VoIP and management interface) Application Type, and then click **Edit**.
3. Set the following parameters:
   - **IP-Address:** <Gateway IP-Address> (e.g., 10.15.7.131).
   - **Prefix Length:** The Subnet Mask in bits (e.g., 16 for 255.255.0.0).
   - **Gateway:** <Gateway Default Gateway> (e.g., 10.15.7.130). For Mediant 800 or Mediant 1000, this IP should be same as you set up in the LAN data-routing IP address. For Mediant 3000, it should be the corporate router IP.
4. From the 'WAN Interface Name' drop-down list, select **WAN Ethernet**. This is the WAN interface on which your VoIP traffic interfaces with the public network.

4.1.1.2 Configuring LAN Data-Routing IP Settings

The following describes how to configure LAN data-routing IP settings.

Notes: This step is only relevant for the Mediant 800 MSBG and the Mediant 1000 MSBG devices.

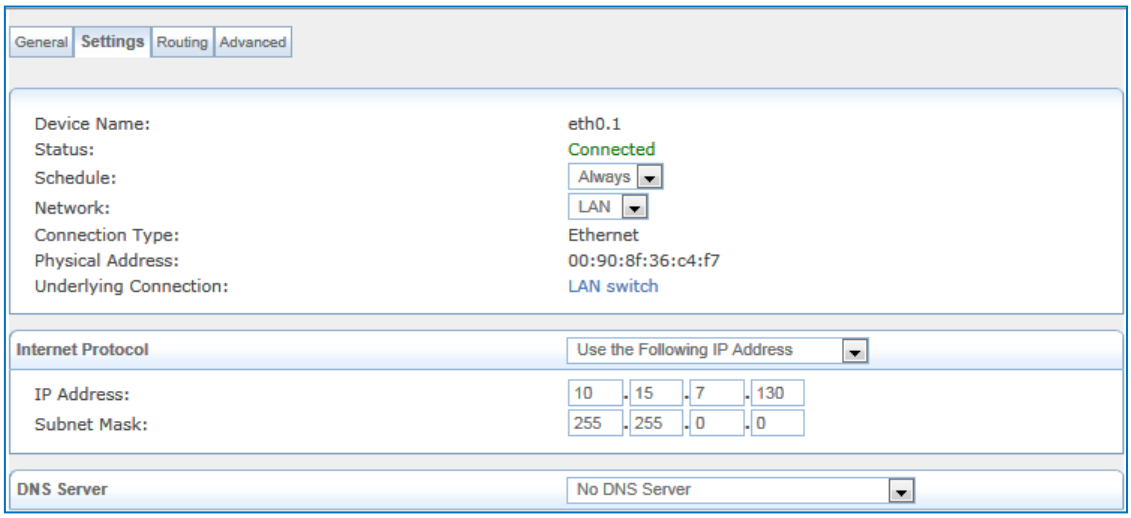
➢ **To define the MSBG device's LAN data-routing IP address:**

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

Figure 4-3: 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., 10.15.7.130) and subnet respectively, and then click **OK**.

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



4.1.2 Configuring WAN IP Addresses

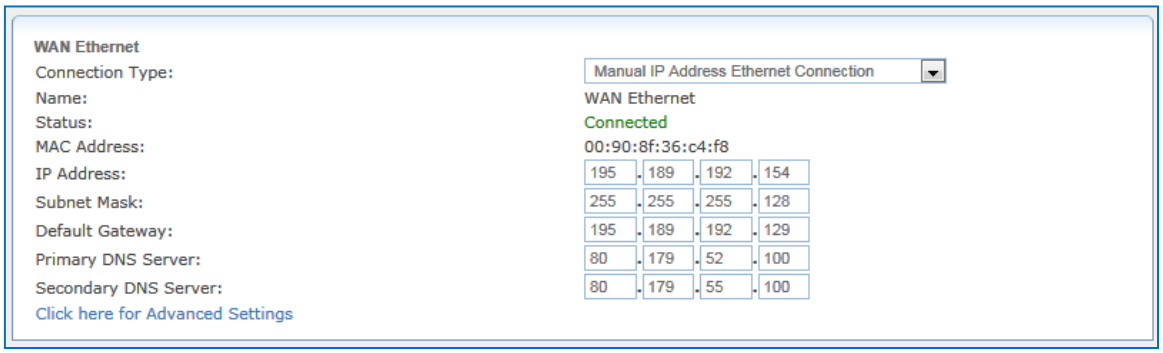
The following describes how to configure the MSBG device IP address used to connect to the WAN.

Notes: This step is only relevant for the Mediant 800 MSBG and the Mediant 1000 MSBG devices.

➢ **To configure the WAN IP address:**

1. Cable the MSBG device to the WAN network (i.e., ADSL or Cable modem), using the WAN port.
2. Open the Settings page (**Configuration** tab > **Data** menu > **WAN Access** > **Settings**).

Figure 4-5: WAN Settings



3. Set the following parameters:
   - **IP Address:** <WAN IP-Address> (e.g., 195.189.192.154).
   - **Subnet Mask:** <Subnet Mask> (e.g., 255.255.255.128).
   - **Default Gateway:** <WAN Default GW IP-Address> (e.g., 195.189.192.129).
   - **Primary DNS Server:** <First DATA DNS IP-Address> (e.g., 80.179.52.100).
   - **Secondary DNS Server:** <Second Data DNS IP-Address> (e.g., 80.179.55.100).

4.2 Step 2: Configuring Port Forwarding

This step describes how to configure the MSBG device's Port Forwarding.

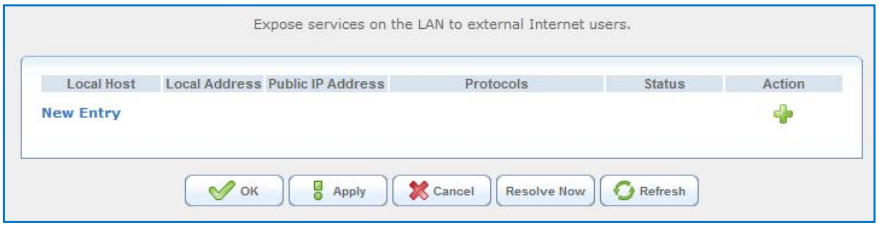
The Port Forwarding item enables you to define the applications that require special handling by the device. This allows you to select the application's protocol or ports (SIP and RTP) and the local IP address of the device (e.g., Gateway's IP: 10.15.7.131) that will be using the service.

Notes: This step is only relevant for Mediant 800 MSBG and Mediant 1000 MSBG devices.

➢ **To configure a port forwarding service:**

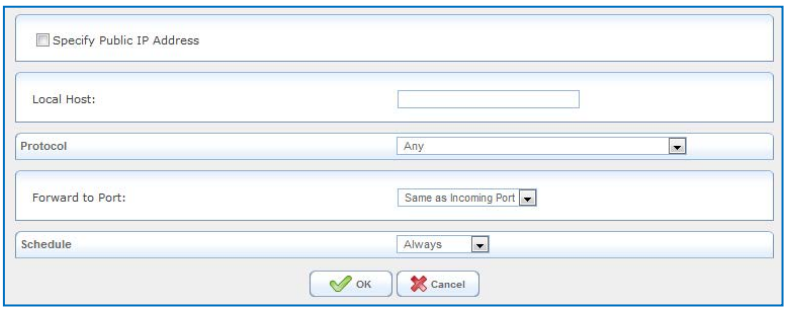
1. Open the Settings page (**Configuration** tab > **Data** menu > **Firewall and ACL** > **Port Forwarding**).

Figure 4-6: Configure Port Forwarding



2. Click the 'New Entry' link; the following page appears:

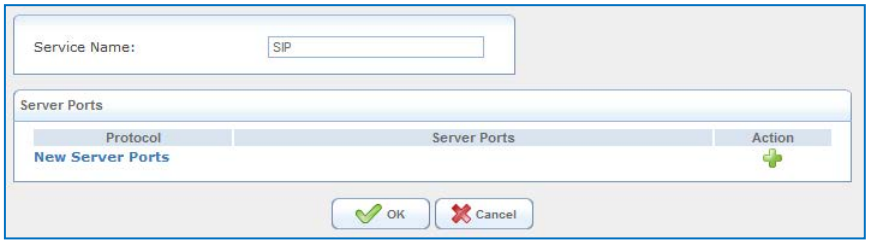
Figure 4-7: Adding Port Forwarding Rule



3. In the 'Local Host' field, enter the host name or IP address (e.g., 10.15.7.131).
4. From the 'Protocol' drop-down list, select or specify the type of protocol.
5. Add a new protocol using the 'User Defined' option, and then add a new Service, representing the protocol.

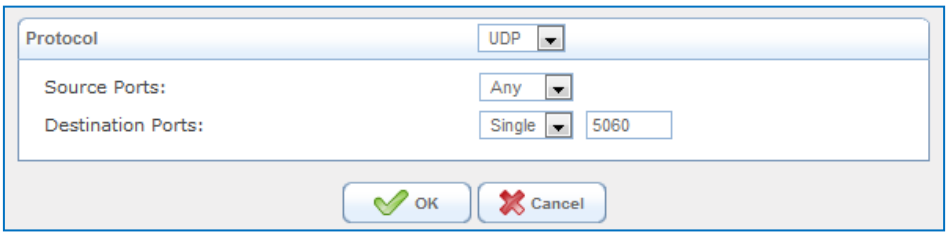
6. In the 'Service Name' name field, enter "SIP".
7. Click the **New Server Ports** link.

Figure 4-8: Adding a Service Protocol



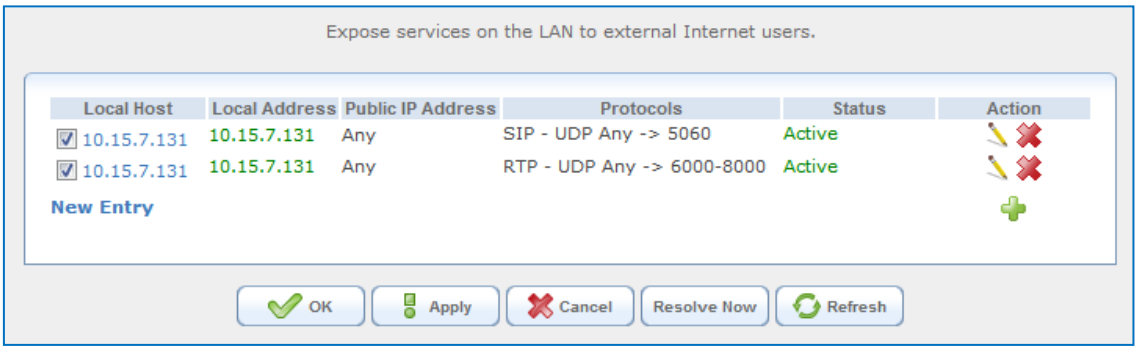
8. From the 'Protocol' drop-down list, select **UDP**.

Figure 4-9: Defining Service Server Ports



9. In the 'Destination Ports' field, enter the range (e.g., 5060 for SIP and 6000-8000 for RTP).
10. Click **OK**; the main Port Forwarding page displays a summary of the rules that you added:

Figure 4-10: Display Port Forwarding Rules



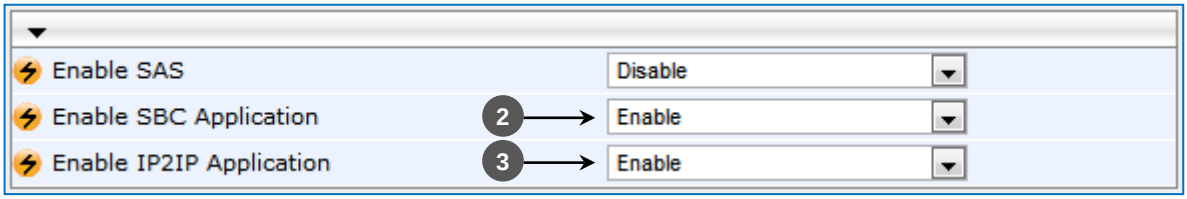
4.3 Step 3: Enabling Application Mode

The following describes how to enable the IP-to-IP and SBC application mode.

➢ **To enable the IP-to-IP application mode:**

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

Figure 4-6: Applications Enabling



2. From the 'Enable SBC Application' drop-down list, select **Enable**. (This application is enabled for the purpose of configuring the Message Manipulation Table on the SBC menu. This application can be disabled after setting the Message Manipulation Table).
3. From the 'Enable IP2IP Application' drop-down list, select **Enable**.

Notes:

- To enable the IP-to-IP capabilities on the AudioCodes gateway, your gateway must be loaded with the feature key that includes the **IP-to-IP** feature.
- The E-SBC device must be running SIP Version 6.2 or later.
- A reset with BURN to FLASH is required.

4.4 Step 4: Configuring Secure Real-Time Transport Protocol

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

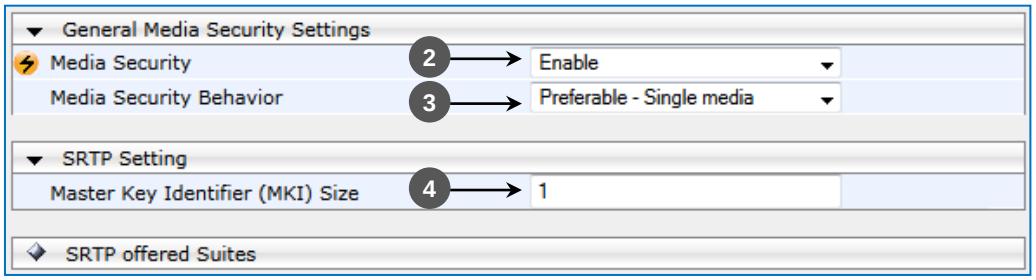
- **Required:** SRTP should be attempted, but do not use encryption if negotiation for SRTP is unsuccessful.
- **Optional:** Attempt to negotiate the use of SRTP to secure media packets. Use RTP if SRTP cannot be negotiated.
- **Not used:** Send media packets using RTP.

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 4-7: Media Security Page



2. From the 'Media Security' drop-down list, select **Enable**.
3. From the 'Media Security Behavior' drop-down list, select one of the following:
   - **Mandatory** - if the 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 configuration and reset the Gateway.

Notes: In order to set the Media Security Behavior to the IP Profile of the Mediation Server, see the IP Profile Settings (see Section 4.12 on page 46).

4.5 Step 5: Configuring IP Media

This step describes how to configure the number of media channels for the IP media. 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 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 800 E-SBC device is 30 (i.e., up to 15 IP-to-IP calls).

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).

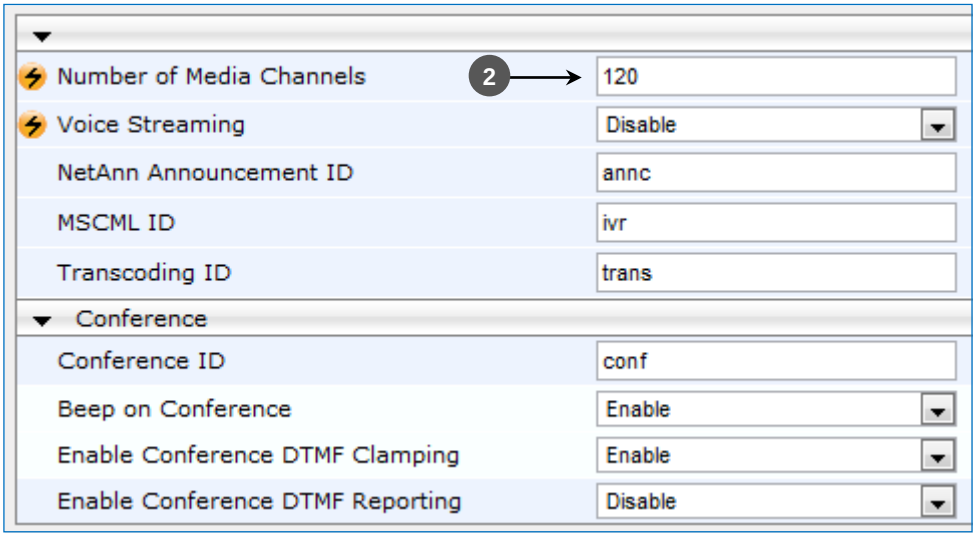
The maximum number of media channels available on the Mediant 3000 E-SBC device is 2016 (i.e., up to 1008 IP-to-IP calls).

In this configuration, 120 channels are configured.

➢ **To configure IP Media Settings:**

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

Figure 4-8: IP Media Settings



2. In the 'Number of Media Channels' field, enter "120".

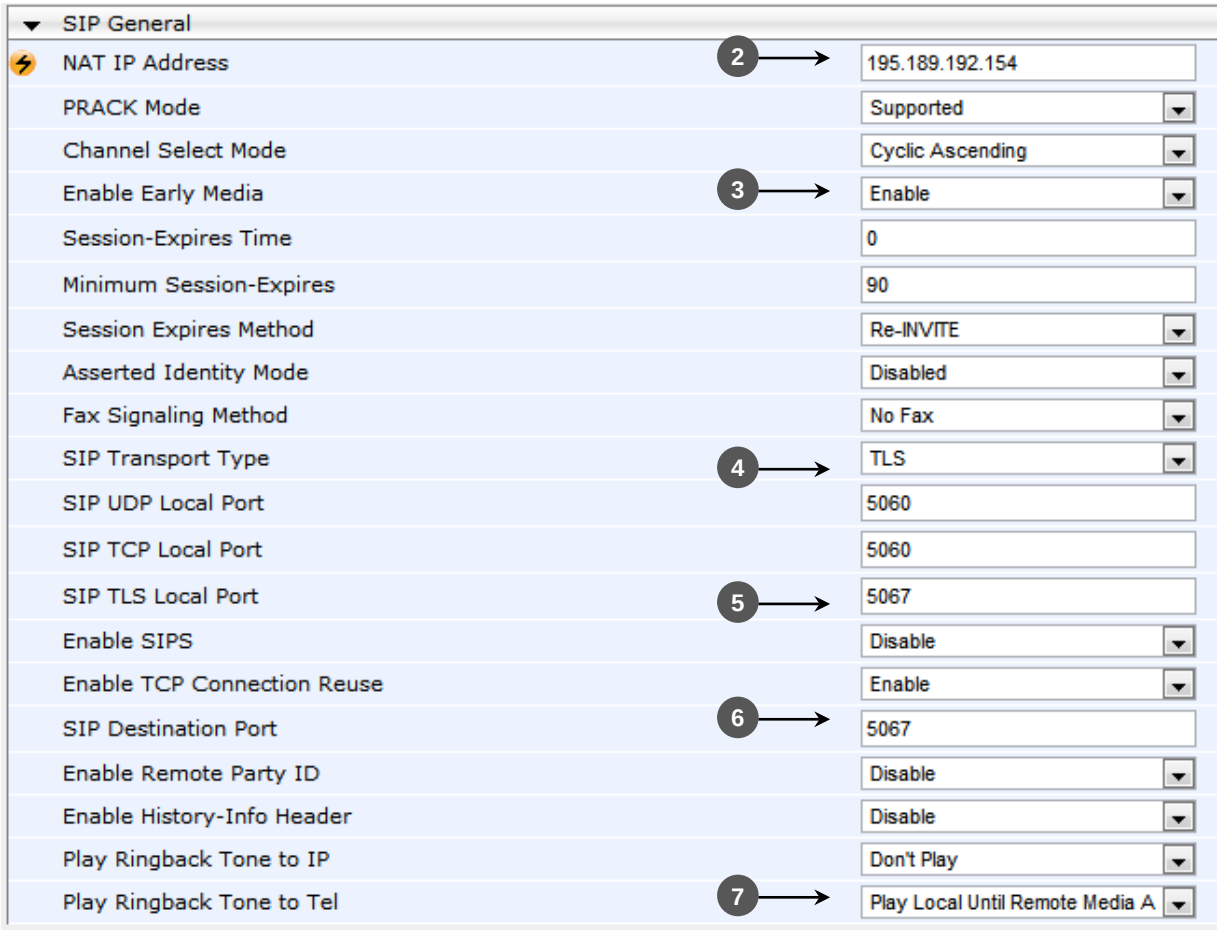
4.6 Step 6: Configuring SIP General Parameters

The following describes how to enable SIP General parameters.

➢ **To configure SIP General Parameters:**

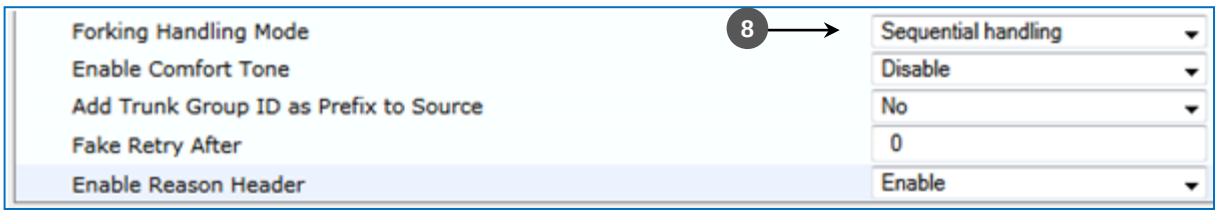
1. Open the Applications Enabling page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **General Parameters**).

Figure 4-9: General Parameters



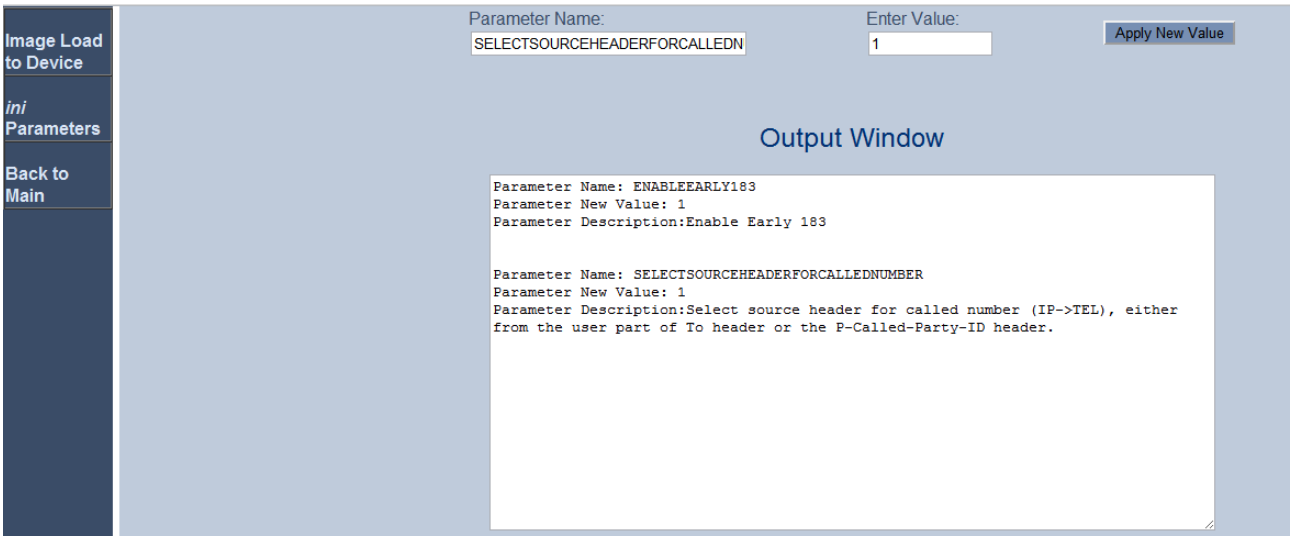
2. In the 'NAT IP Address' field, enter the Global (public) IP address of the E-SBC device.
3. From the 'Enable Early Media' drop-down list, select **Enable**.
4. From the 'SIP Transport Type' drop-down list, select **TLS**.
5. In the 'SIP TLS Local Port' field, enter "5067".
6. In the 'SIP Destination Port' field, enter "5067" (Lync Server listening port).
7. From the 'Play Ringback Tone to Tel' drop-down list, select **Play Local Until Remote Media Arrives.**

Figure 4-11: General Parameters (Cont.)



8. From the 'Forking Handling Mode' drop-down list, select **Sequential handling**.
9. 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://10.15.7.131/AdminPage).
10. On the left pane, click ***ini* Parameters**.

Figure 4-10: INI file Output Window



11. In the first 'Parameter Name' field, enter “ENABLEEARLY183”.
12. In the first 'Parameter New Value' field, enter "1".
13. In the second 'Parameter Name' field, enter “SELECTSOURCEHEADERFORCALLEDNUMBER”.
14. In the second ‘Parameter New Value’ field, enter "1".
15. Click **Apply New Value**.

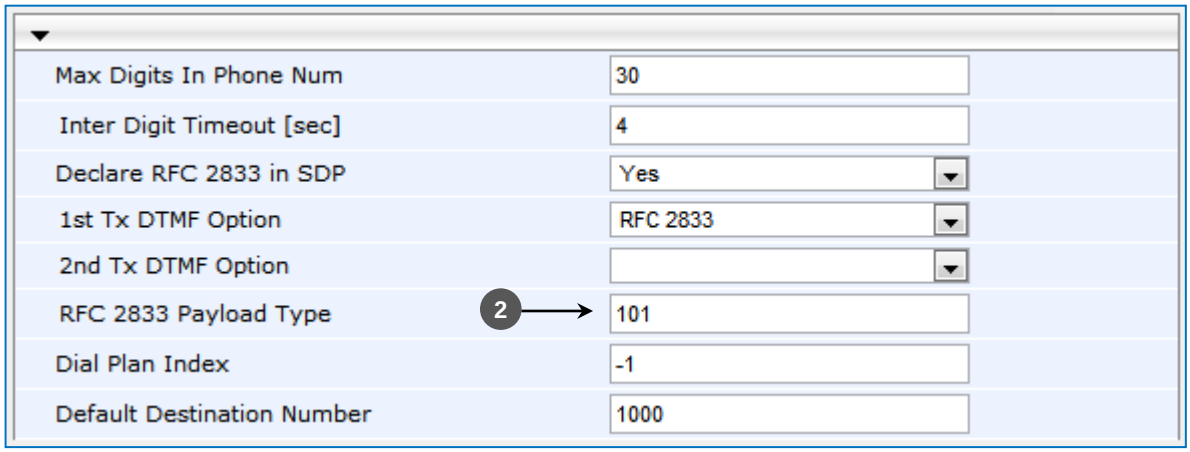
4.7 Step 7: Configuring DTMF and Dialing

The following describes how to configure the DTMF and Dialing settings.

➢ **To configure DTMF and Dialing:**

1. Open the DTMF and Dialing page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **DTMF and Supplementary** > **DTMF & Dialing**).

Figure 4-11: DTMF and Dialing



2. In the 'RFC 2833 Payload Type' field, enter "101".

4.8 Step 8: Configuring Coders

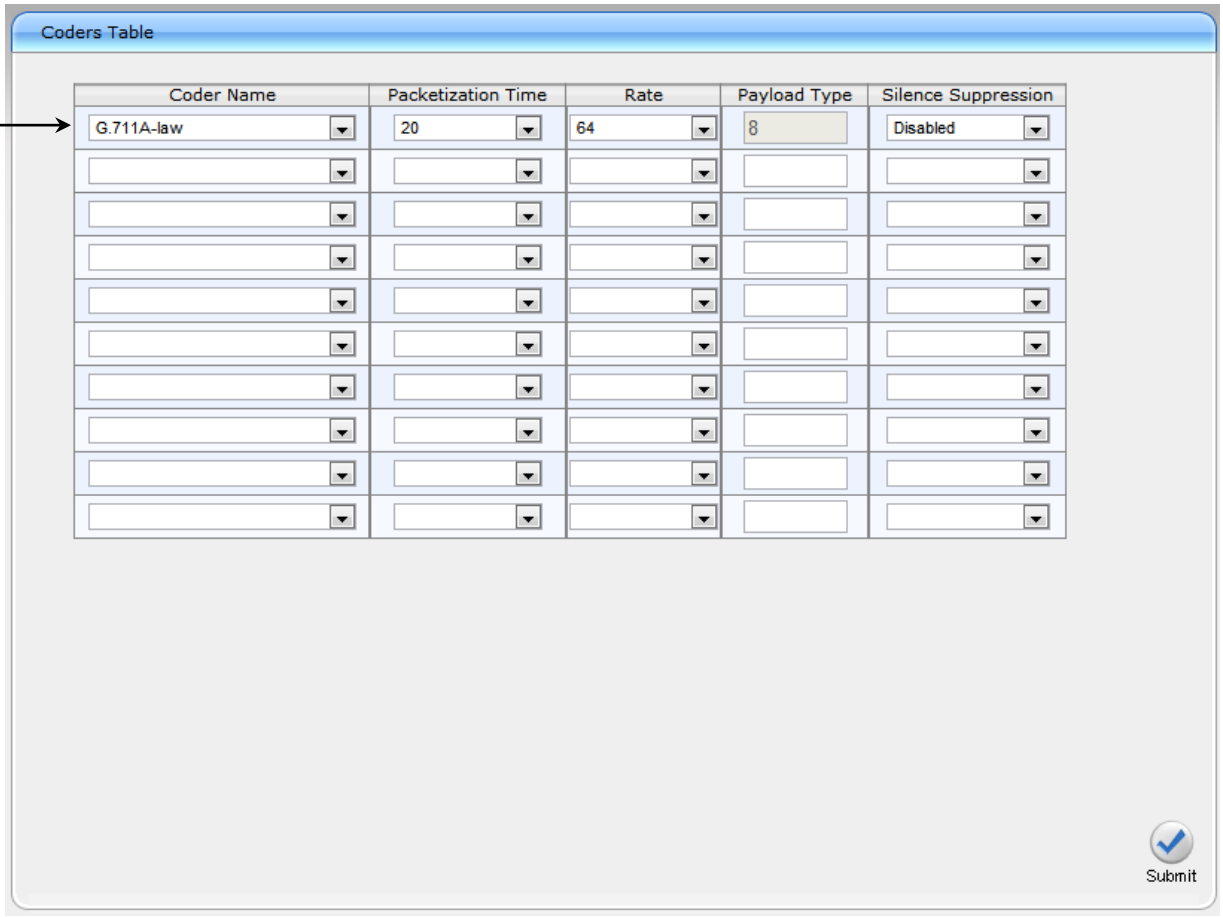
This step describes how to configure the SIP coders. This is the general coder table in this case scenario we are using coder group tables, see Section 4.11 on page 45.

The screen below show an example for the general coders' table configuration:

➢ **To configure coders:**

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

Figure 4-12: Coders



2. From the 'Coder Name' drop-down list, select **G.711A-law** and **G.711U-law**.
3. Click **Submit**.

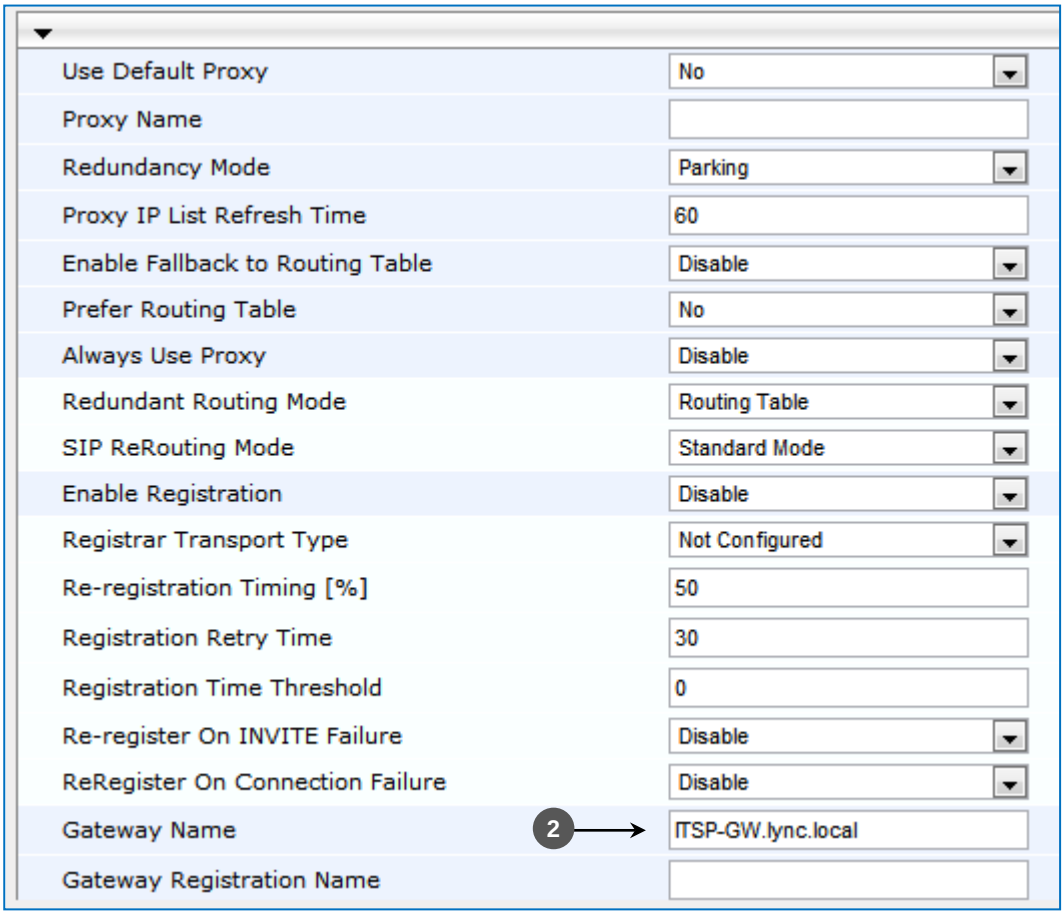
4.9 Step 9: Configuring Proxy and Registration

The following describes how to configure the SIP Proxy and Registration. This configuration includes setting a redundant route for the Microsoft Lync Proxy Set.

➢ **To configure Proxy and Registration:**

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

Figure 4-13: Proxy & Registration



2. In the 'Gateway Name' field, enter the Gateway FQDN Name (e.g., "ITSP-GW.lync.local").

Note: You configure this name in Section 4.18.3 on page 59).

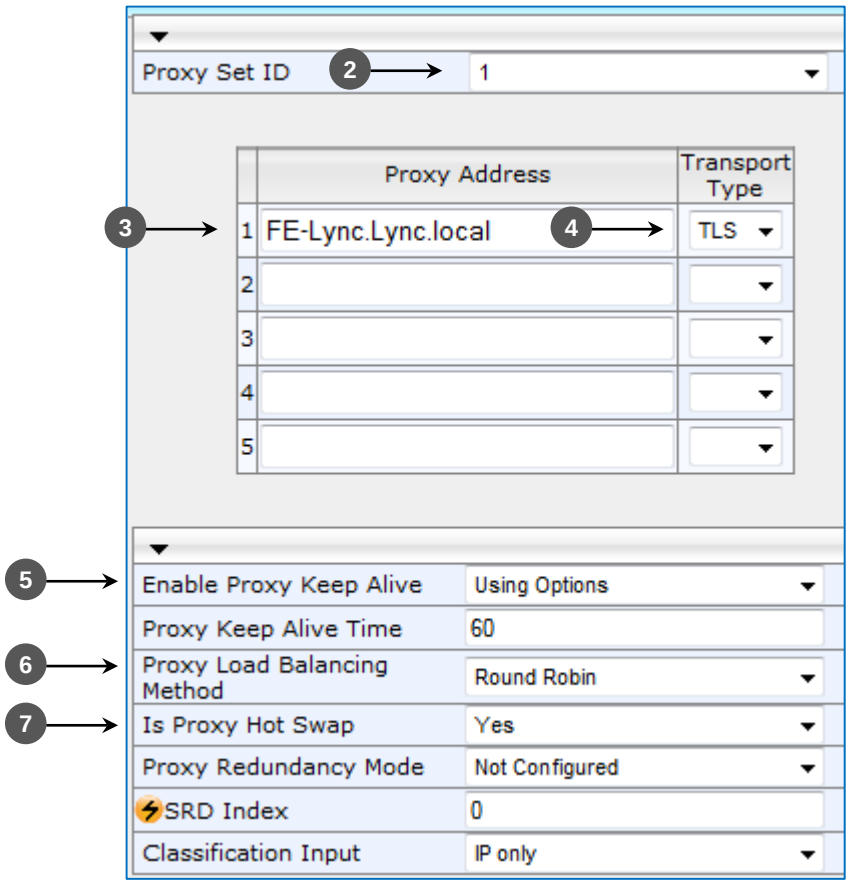
4.10 Step 10: Configuring Proxy Sets Tables

The following describes how to configure the proxy set tables. You need to configure two proxy sets - one for the Spitfire SIP trunk and the other for the Microsoft Lync server.

➢ **To configure Proxy Sets Table 1 for Microsoft Lync:**

1. Open the Proxy Sets Table page (**Configuration** tab > **VoIP** menu > **Control Network**> **Proxy Sets Table**).

Figure 4-14: Proxy Sets Table 1

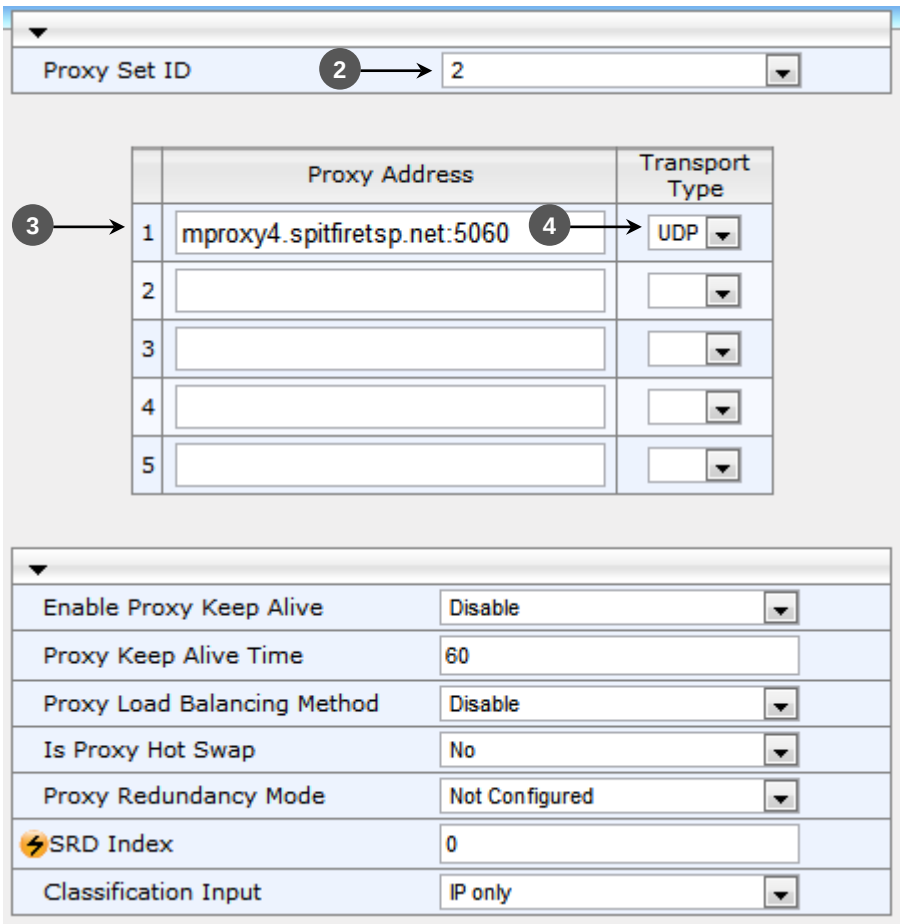


2. From the ' Proxy Set ID' drop-down list, select **1**.
3. Configure the Microsoft Lync Server SIP Trunking IP address or FQDN and Destination Port (e.g., "FE-Lync.Lync.local").
4. From the 'Transport Type' drop-down list, select **TLS**.
5. From the 'Enable Proxy Keep Alive' drop-down list, select **Using Options**.
6. From the 'Proxy Load Balancing Method' drop-down list, select **Round Robin**.
7. From the 'Is Proxy Hot Swap' drop-down list, select **Yes**.
8. Click **Submit**.

➢ **To configure Proxy Sets Table 2 for Spitfire SIP Trunk:**

1. Open the Proxy Sets Table page (**Configuration** tab > **VoIP** menu > **Control Network**> **Proxy Sets Table**).

Figure 4-15: Proxy Sets Table 2



2. From the 'Proxy Set ID' drop-down list, select **2**.
3. Configure the Spitfire IP address or FQDN and Destination Port (e.g., mproxy4.spitfiretsp.net:5060').
4. From the 'Transport Type' drop-down list, select **UDP**.
5. Click **Submit**.

4.11 Step 11: Configuring Coder Group

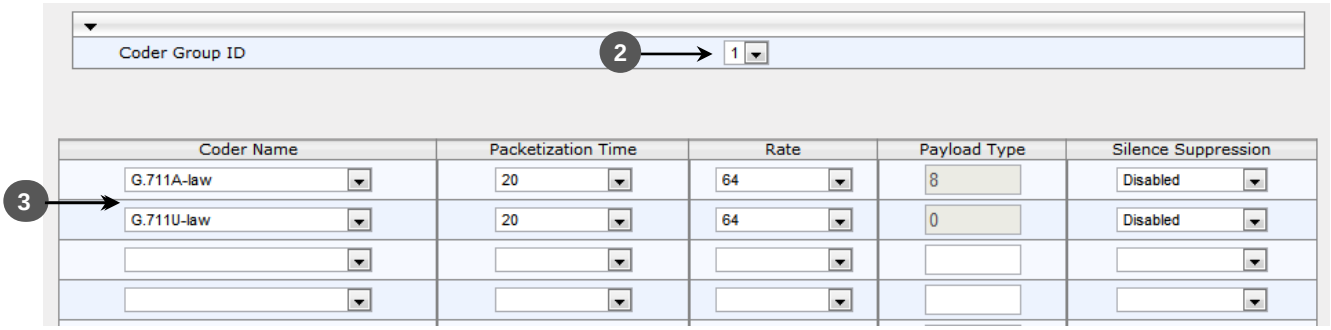
This step describes how to configure the Coder Groups. Microsoft Lync supports G.711 coders, while the network connection to Spitfire may restrict you to work with lower bandwidth coders, such as G.729.

The 'Coder Group Settings' allow you to define up to four different Coder Groups. These Coder Groups are then assigned to IP Profiles, where each IP profile is based on the respective supported coder (see Section 4.11 on page 45).

➢ **To configure Coders Group for Microsoft Lync connection:**

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

Figure 4-16: Coders Group Settings for Microsoft Lync Connection

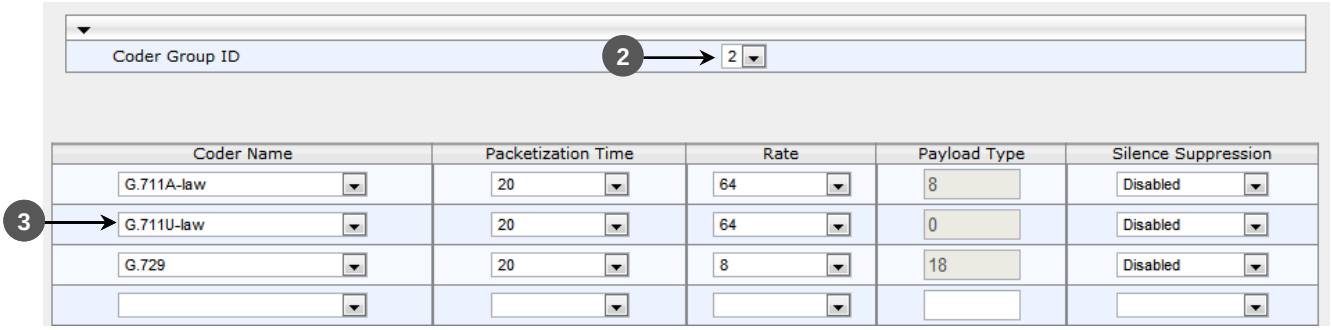


2. Select **Coder Group ID 1**.
3. Set Coder Name **G.711A-law** and **G.711U-law**.
4. Click **Submit**.

➢ **To configure Coders Group for IP Directions SIP Trunk connection:**

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

Figure 4-17: Coders Group Settings IP Directions for SIP Trunk Connection



2. Select **Coder Group ID 2**.
3. Set Coder Name **G.711A-law, G.711U-law** and **G.729.**
4. Click **Submit**.

4.12 Step 12: Configuring IP Profile

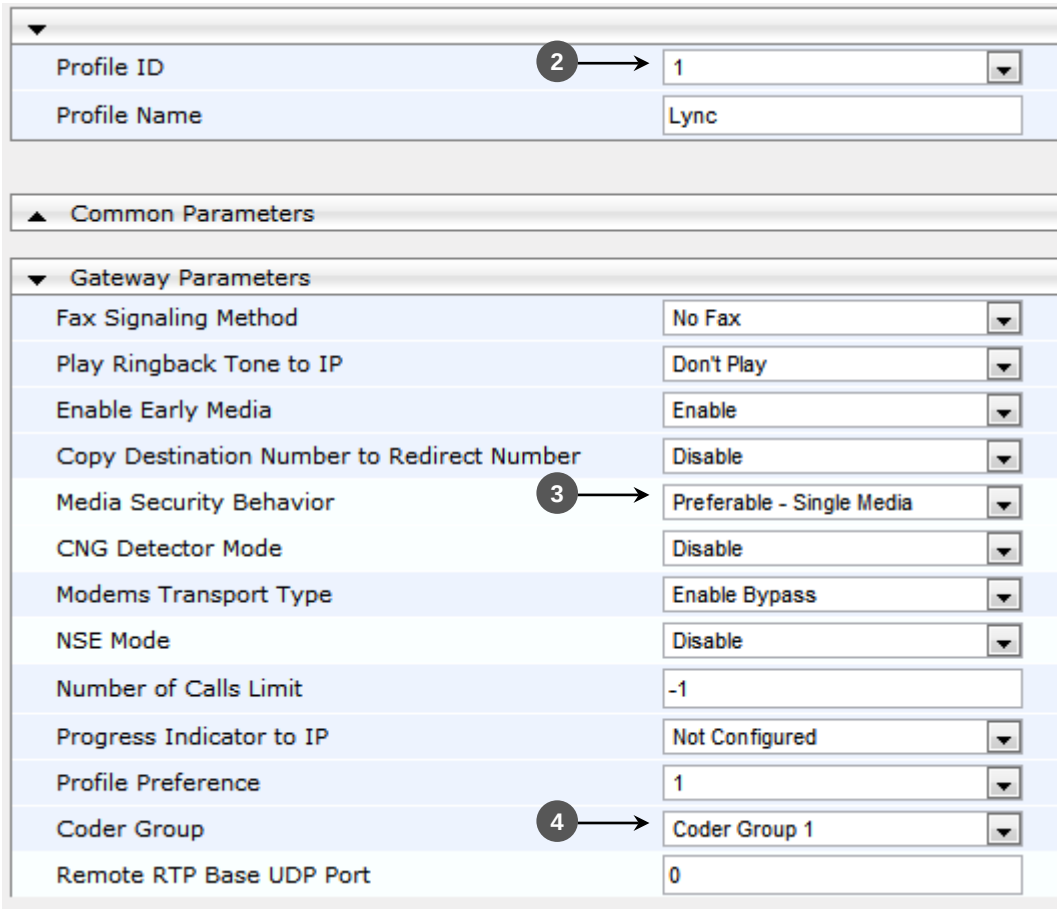
The following describes how to configure the IP Profile. In this configuration, the IP Profile is used to configure the SRTP/TLS mode and the Coder Group (see Section 4.8 on page 41).

You must configure Microsoft Lync to work in secure mode (SRTP/TLS); while, the Spitfire SIP trunk is configured in non-secure mode RTP/UDP.

➢ **To configure IP Profile for Microsoft Lync:**

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

Figure 4-18: IP Profile Settings for Microsoft Lync

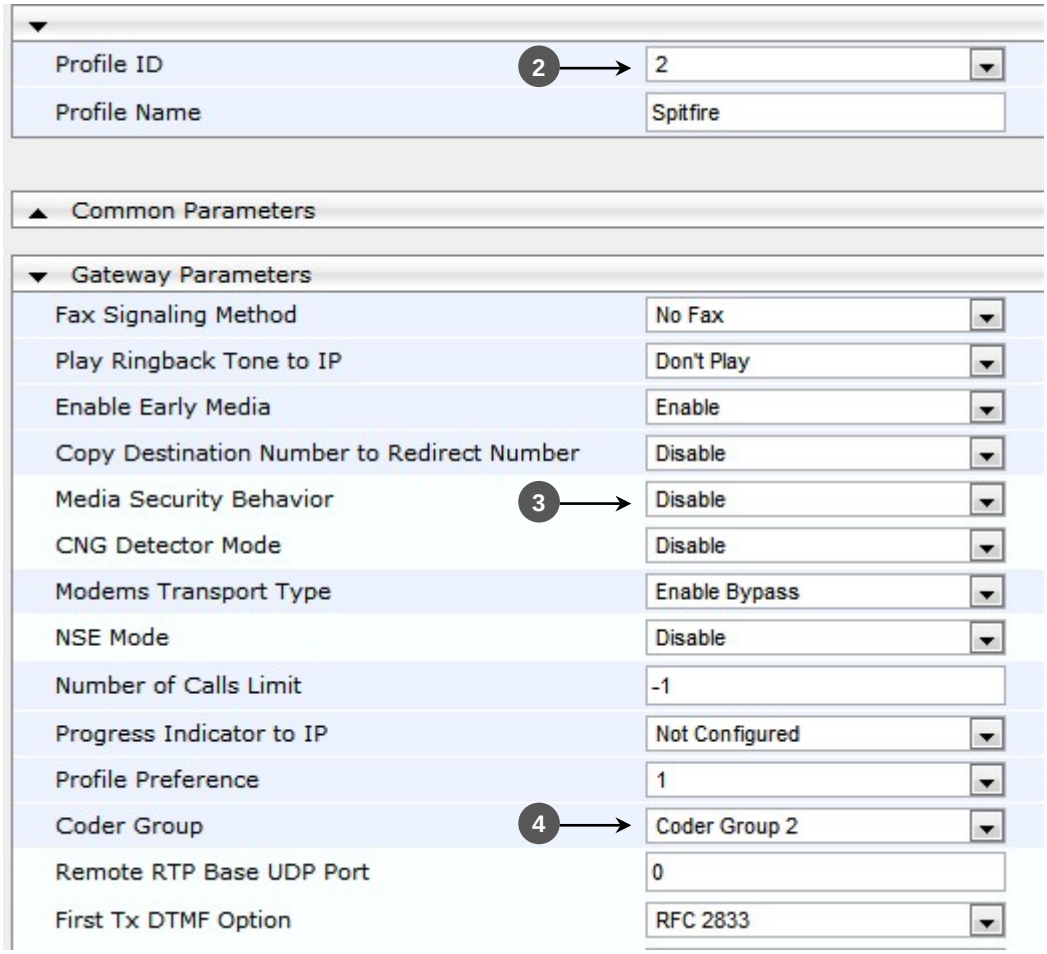


2. From the 'Profile ID' drop-down list, select **1**.
3. From the 'Media Security Behavior' drop-down list, select **Preferable – Single Media**.
4. From the 'Coder Group' drop-down list, select **Coder Group 1**.
5. Click **Submit**.

➢ To configure IP Profile for Spitfire SIP Trunk:

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

Figure 4-19: IP Profile Settings for Spitfire SIP Trunk



2. From the 'Profile ID' drop-down list, select **2**.
3. From the 'Media Security Behavior' drop-down list, select **Disable**.
4. Set Coder Group to **Coder Group 2**.
5. Click **Submit**.

4.13 Step 13: Configuring IP Group Tables

The following 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:

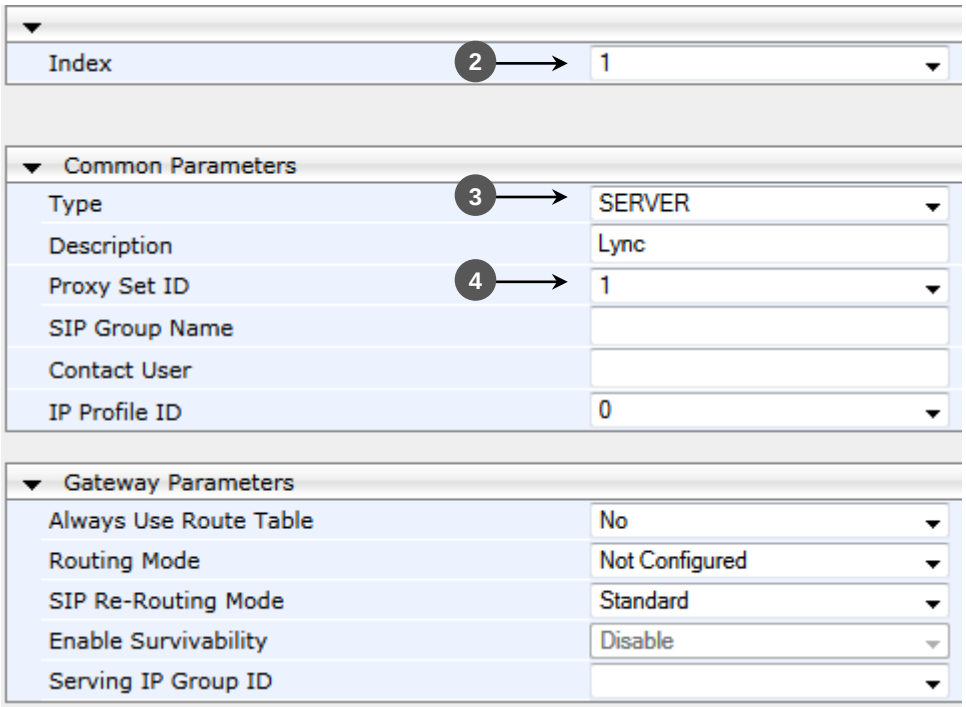
- Lync Server 2010 - Mediation Server
- Spitfire SIP Trunk

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

➢ **To configure IP Group Table 1:**

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

Figure 4-20: IP Group Table 1

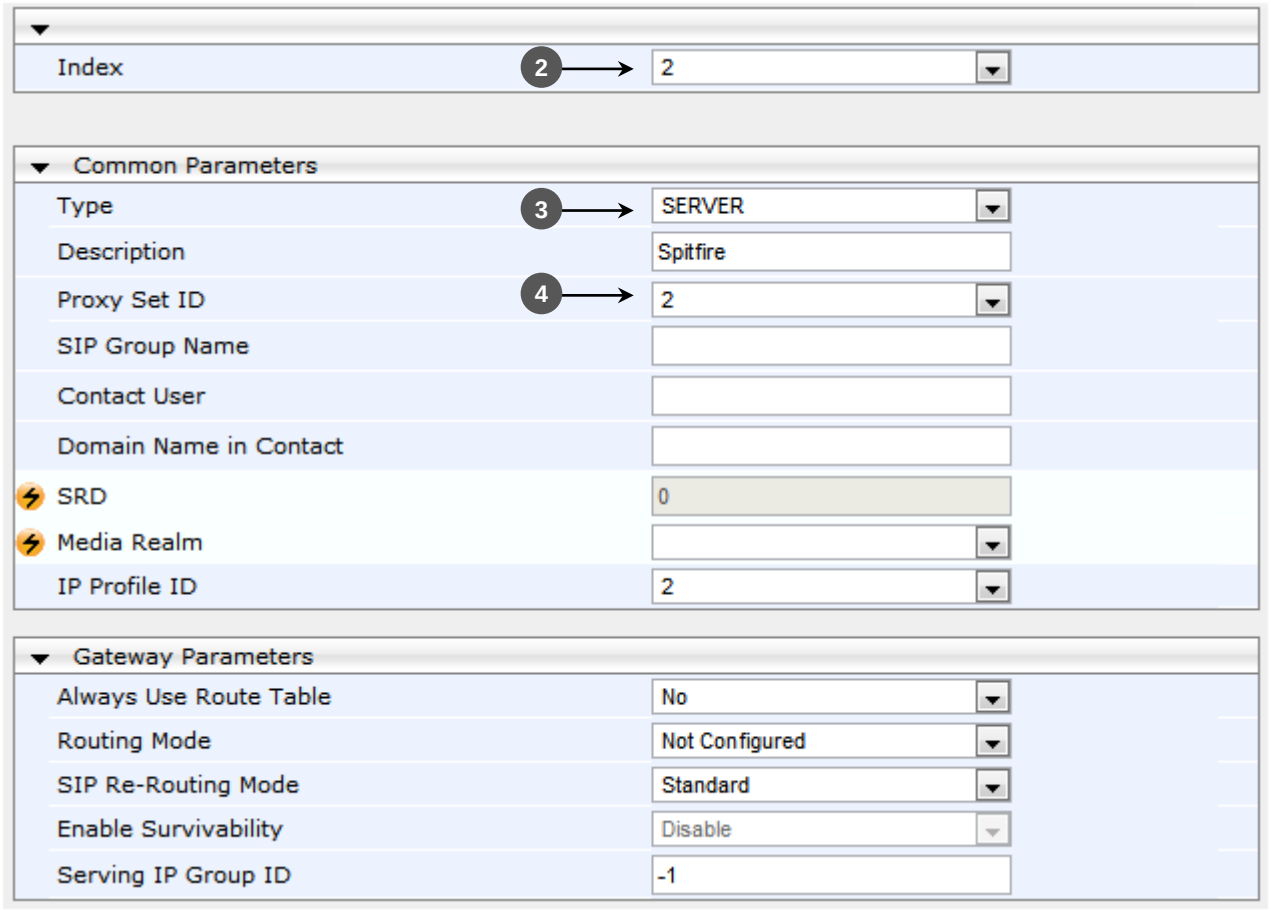


2. From the 'Index' drop-down list, select **1**.
3. From the 'Type' drop-down list, select **SERVER**.
4. From the 'Proxy Set ID' drop-down list, select **1**.
5. Click **Submit**.

➢ **To configure IP Group Table 2:**

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

Figure 4-21: IP Group Table 2



2. From the 'Index' drop-down list, select **2**.
3. From the 'Type' drop-down list, select **SERVER**.
4. From the 'Proxy Set ID' drop-down list, select **2**.
5. Click **Submit**.

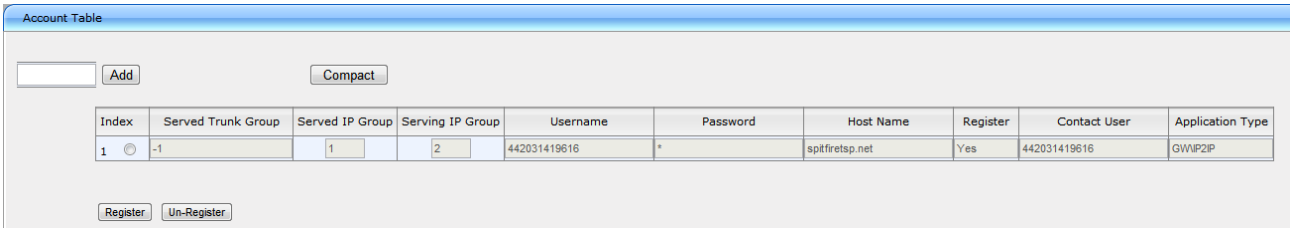
4.14 Step 14: Configuring Account Table

This step describes how to configure and register Spitfire client extensions on the SIP Trunk. The SIP trunk registers.

➢ **To configure Accounts:**

1. Open the 'Account Table' page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Account Table**).

Figure 4-22: Account Table



2. Enter an index table entry number, and then click **Add**.
3. Configure the account user entry according to the provided information. Fill the entry's table according to the above example
   - In the 'Served IP Group' field, enter "1" (i.e., Lync Server 2010).
   - In the 'Served IP Group' field, enter "2" (i.e., Spitfire SIP Trunk).
   - In the 'Username' field, enter your username.
   - In the 'Password' field, enter your password.
   - In the 'Host Name' field, enter "spitfire.net".
   - In the 'Register' field, enter "Yes".
   - In the 'Contact User' field, enter your username.
   - In the 'Application Type' field, enter "GWI\IP2IP".

Note: If there are more accounts, repeat steps 2 and 3 to add more user account entries.

4.15 Step 15: Configuring Routing

The following describes how to configure the IP-to-IP routing table.

The device IP-to-IP routing rules are configured in the 'IP to Trunk Group Routing' and 'Tel to IP Routing' tables. These tables provide enhanced IP-to-IP call routing capabilities for routing received SIP messages, such as INVITE messages to a destination IP address. The routing rule must match one of the following input characteristics:

- Source IP Group
- Source Phone Prefix and/or Source Host Prefix

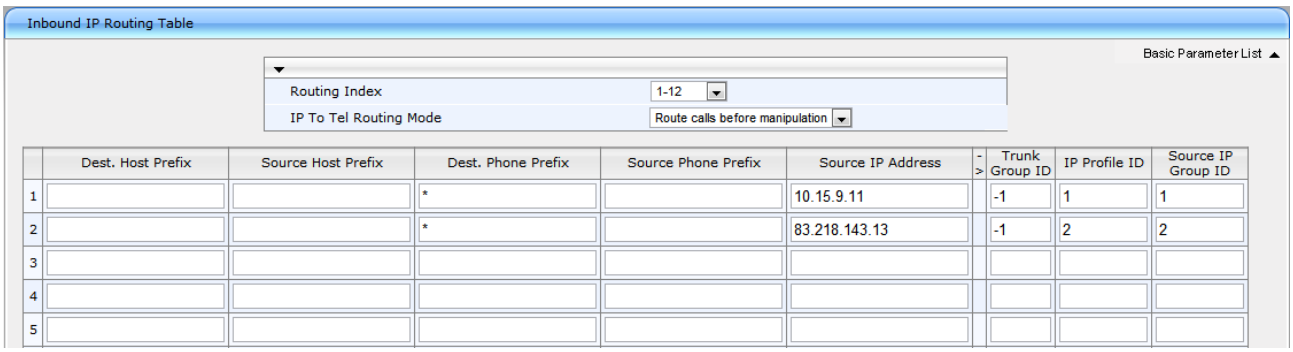
Note: It is crucial that you adhere to the following guidelines when configuring your IP-to-IP routing rules:

- Ensure that your routing rules are accurate and correctly defined.
- Ensure that your routing rules from **source IP Group** to **destination IP Group** are accurately defined to be eligible for the desired call routing outcome.
- Avoid (if possible) using the asterisk (*) symbol to indicate "any" for a specific parameter in your routing rules. This constitutes a weak routing rule. For strong routing rules, enter specific letter or numeric character values.

➢ **To configure the IP to Trunk Group Routing Table:**

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

Figure 4-23: IP to Trunk Group Routing Table

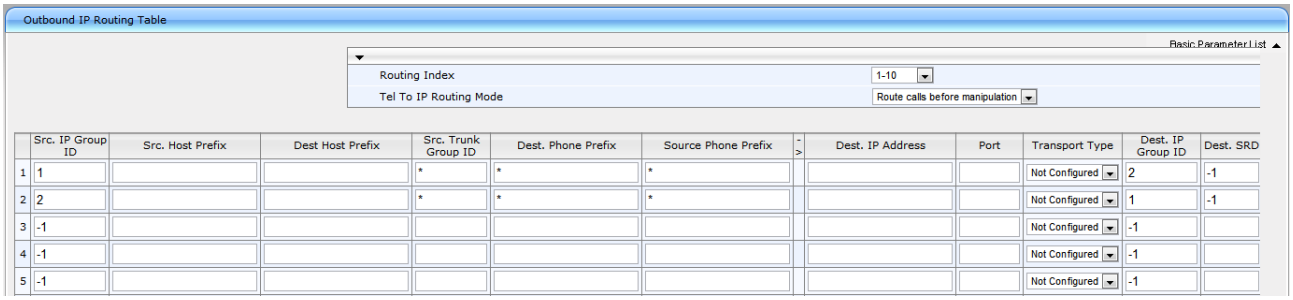


2. **Configure Row Index 1:** Calls arriving from 10.15.9.11 (i.e., Microsoft Lync server) are sent to the 'Trunk Group ID' **-1** (i.e., 'Tel to IP Routing Table') with 'IP Profile ID' = **1** and marked as 'Source IPGroup ID' = **1**.
3. **Configure Row Index 2:** Calls arriving from 83.218.143.13 (i.e., Spitfire Sip Trunk) are sent to the 'Trunk Group ID' **-1** (i.e., 'Tel to IP Routing Table') with 'IP Profile ID' = **2** and marked as 'Source IPGroup ID'=**2**.

➢ **To configure Tel to IP Routing Table:**

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

Figure 4-24: Tel to IP Routing Table



2. **Configure Row Index 1**: Calls from Source IPGroup ID **1** (i.e., from Microsoft Lync) are sent to 'Dest. IPGroup ID **2** (i.e., to Spitfire Sip trunk).
3. **Configure Row Index 2:** Calls from Source IPGroup ID **2** (i.e., from Spitfire Sip trunk) are sent to 'Dest. IPGroup ID **1** (i.e., to Microsoft Lync).

Note: The Routing configuration may change according to the local deployment topology.

4.16 Step 16: Configuring Manipulation Tables

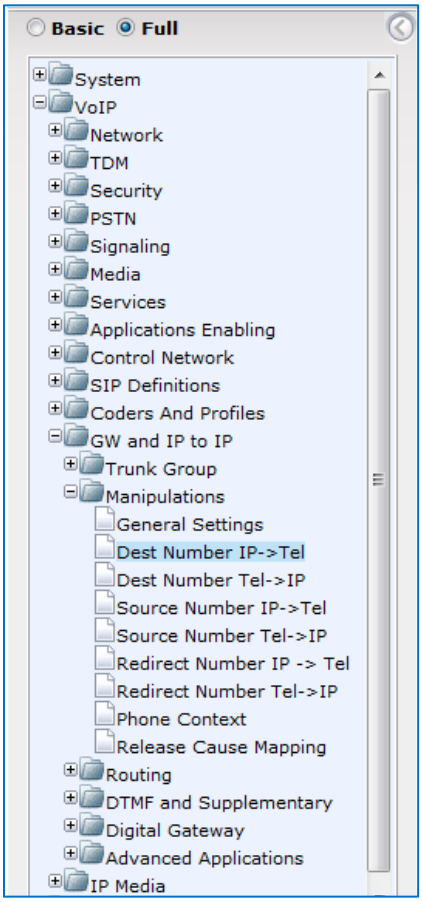
The following describes how to configure the manipulation tables. The Manipulation Tables sub-menu allows you to configure number of manipulations and mappings of NPI/TON to SIP messages.

Note: Adapt the manipulation table according to your environment dial plan.

➢ **To configure Manipulation Tables:**

1. Open the Manipulation Table page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Manipulations**).

Figure 4-25: Manipulation Tables

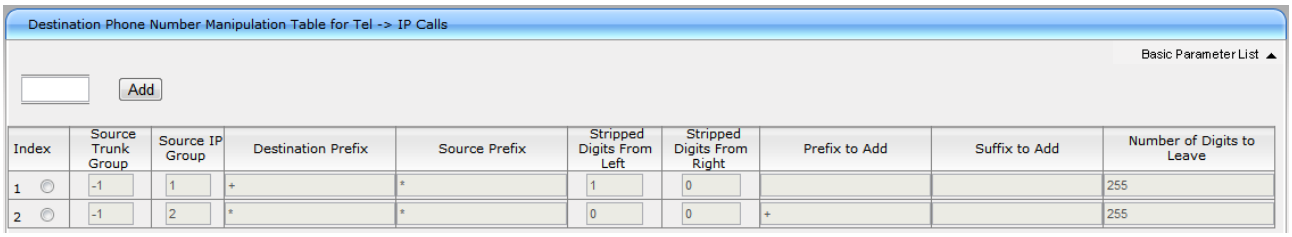


The following includes examples for number manipulation on destination and source numbers in the Tel-to-IP tables:

➢ **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** sub-menu > **Dest Number Tel** > **IP**).

Figure 4-29: Destination Phone Number Manipulation Table for Tel -> IP Calls

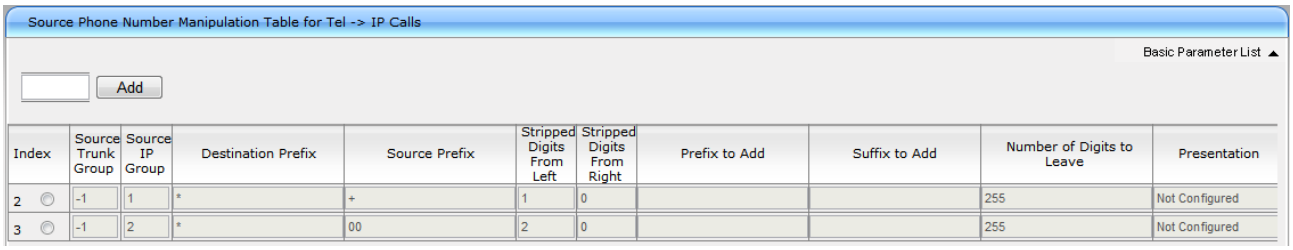


2. **Index #1** defines the destination number manipulation of calls from Lync Server.
   - For all calls received from Source IP Group 1 (i.e., from Lync Server) and the destination number prefix that begins with '+', remove the '+' from the number.
3. **Index #2** defines the destination number manipulation of calls from Spitfire SIP Trunk.
   - For all calls received from Source IP Group 2 (i.e., from Spitfire SIP Trunk) and the destination number prefix is '*' (i.e., any), add the '+' prefix to the number.

➢ **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** sub-menu > **Source Number Tel** > **IP**).

Figure 4-30: Source Phone Number Manipulation Table for Tel -> IP Calls Page



2. **Index #2** defines source number manipulation of calls from the Lync Server. All calls received from Source IP Group **1** (i.e., from Lync Server) and the source number prefix begins with '+', remove the '+' from the number.
3. **Index #3** defines source number manipulation of calls from the Spitfire SIP Trunk. All calls received from Source IP Group **2** (i.e., from Spitfire SIP Trunk) and the source number prefix begins with **00**, remove the '00' from the number.

4.17 Step 17: Configuring Message Manipulations

The Message Manipulations page allows you to define up to 200 SIP message manipulation rules. This manipulation includes insertion, removal, and/or modification of SIP headers. Multiple manipulation rules can be configured for the same SIP message. SIP message manipulation rules configured on this page will be assigned to an IP Group and determined whether they must be applied to inbound or outbound messages. This step describes the Message Manipulation for working with Spitfire SIP Trunk for the Call Transfer/Forward feature using Microsoft Lync.

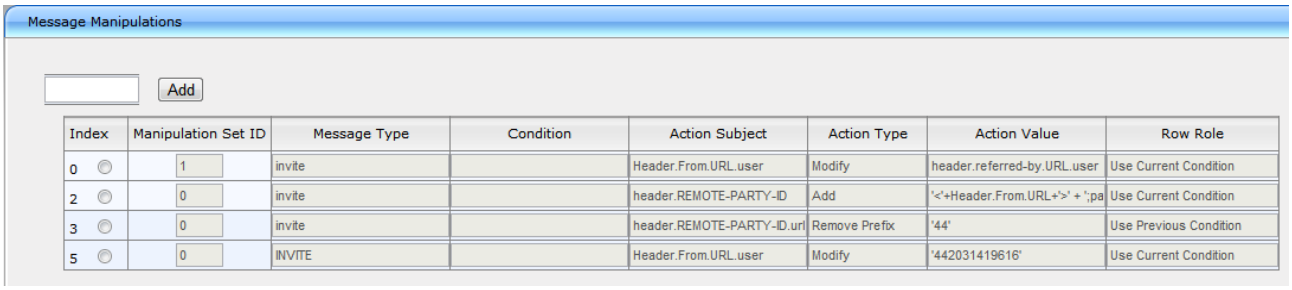
Two sets of manipulation are defined:

- **Set ID 1** is assigned to the gateway inbound manipulation set (Row index 1).
- **Set ID 0** is assigned to Spitfire IP Group (IP Group 2) as an Outbound Message Manipulation Set

➢ **To configure SIP message manipulations:**

1. Open the Message Manipulations page (**Configuration** tab > **VoIP** menu > **SBC** submenu > **Manipulations SBC** sub-menu > **Message**).

Figure 4-33: SIP Message Manipulation

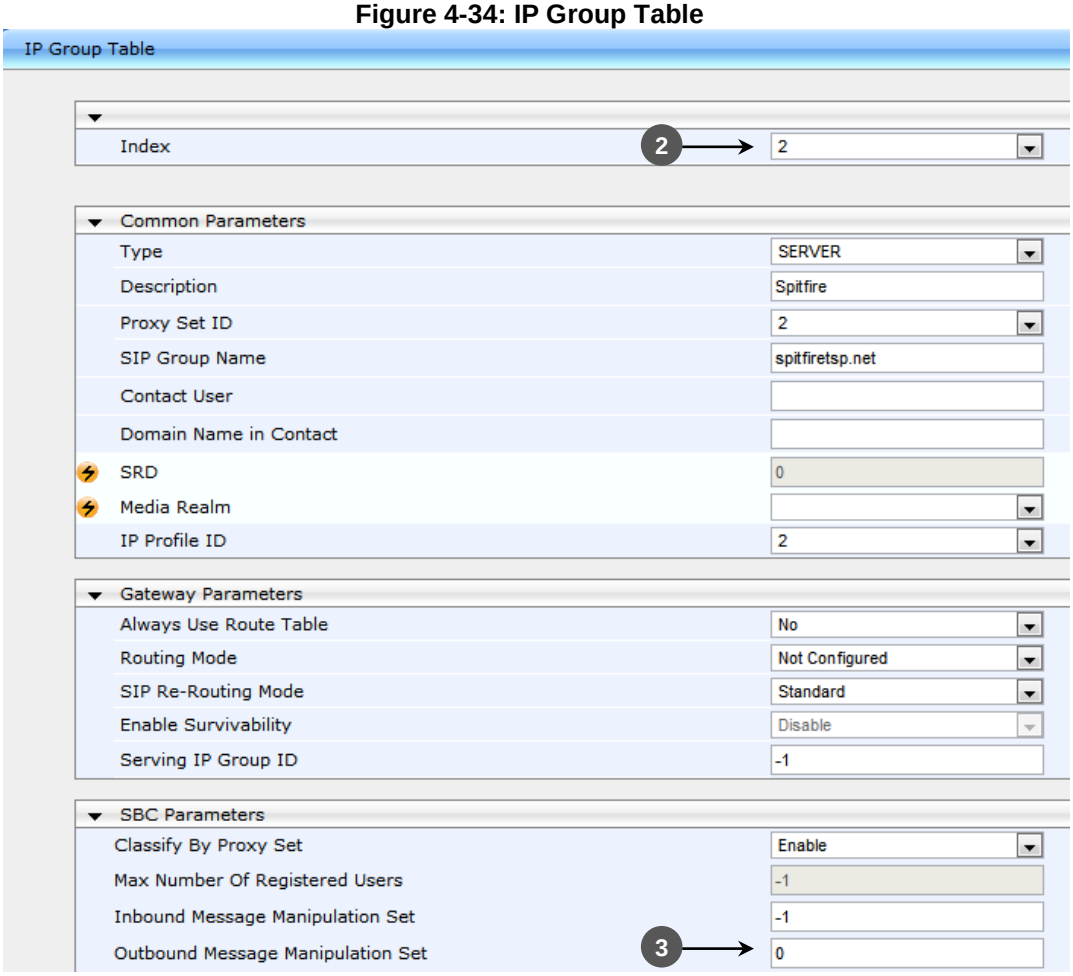


2. Configure the following manipulation rules:
   - **Row Index #0:** For any **INVITE** coming from the Lync Server, this manipulation row modifies the **user** part in the **From** header to the **user** part that appears in the **Referred-By** header (if exists).
   - **Row Index #2:** For any **INVITE** going to the Spitfire SIP Trunk, this manipulation row adds a **Remote-Party ID** Header with the **From URL** header and 'party=calling' syntax.
   - **Row Index #3:** In addition to the previous row (Index #2), this manipulation row removes the **user** part in the **Remote-Party ID** Prefix value of '**44**'.
   - **Row Index #5:** For any **INVITE** going to the Spitfire SIP Trunk, this manipulation row modifies the **user** part in the **From** header to the value of '**442031419616**'.

➢ **To assign a manipulation Set ID 1 to IP Group 2:**

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

Figure 4-34: IP Group Table

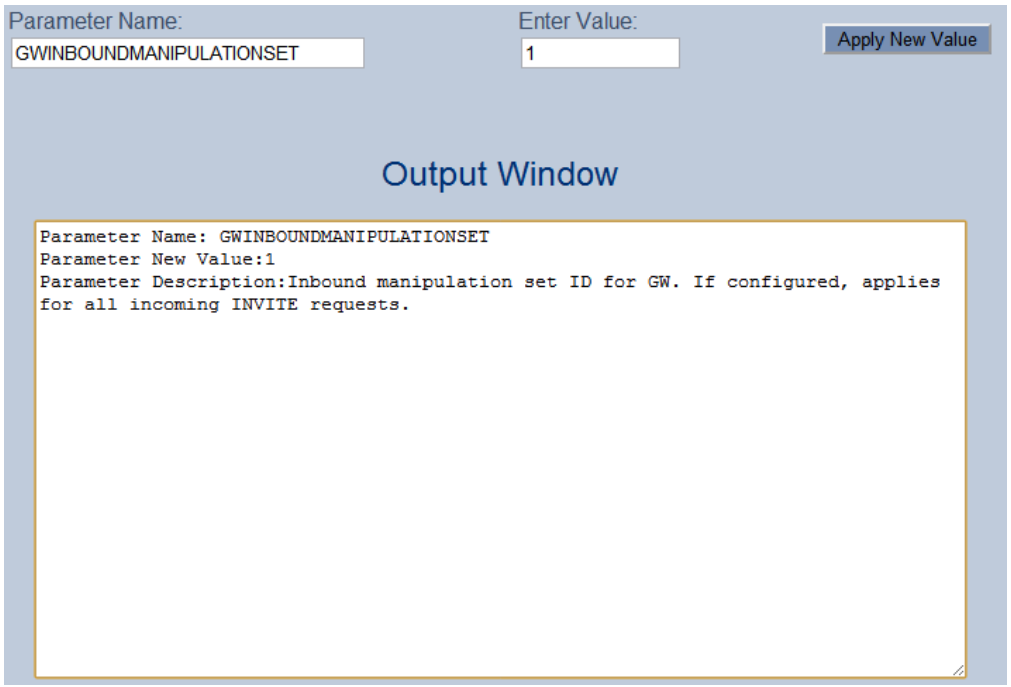


2. From the 'Index' drop-down list, select **2**.
3. In the 'Outbound Message Manipulation Set' field, enter "0".

➢ **To assign manipulation set ID 0 to gateway inbound manipulation set:**

1. 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://10.15.7.131/AdminPage).
2. On the left menu pane, click **ini Parameters**.
3. In the 'Parameter Name' field, enter "GWINBOUNDMANIPULATIONSET".
4. In the 'Enter Value' field, enter "0".

Figure 4-35: Output Window



5. Click **Apply New Value**.

4.18 Step 18: Configuring SIP TLS Connection

The following describes how to configure the AudioCodes gateways for implementing a TLS connection with the Microsoft Lync Mediation server. The steps described in this section are essential elements for the configuration of a secure SIP TLS connection.

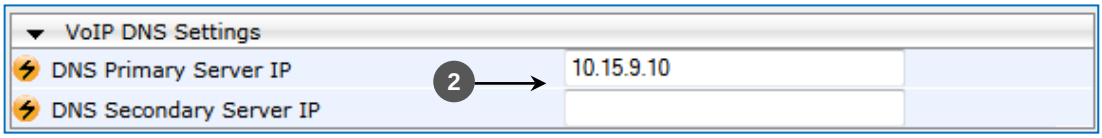
4.18.1 Step 18-1: Configuring VoIP DNS Settings

The following describes how to define the VoIP LAN DNS server, which is a necessary action when a FQDN is configured (as in this scenario configuration, see Section 4.9 on page 9).

➢ **To configure the VoIP DNS settings:**

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

Figure 4-26: VoIP DNS Settings



2. Set the following parameters:
   - **DNS Primary Server IP:** <Primary DNS IP-Address> (e.g., 10.15.9.10).
   - **DNS Secondary Server IP:** <Secondary DNS IP-Address>.

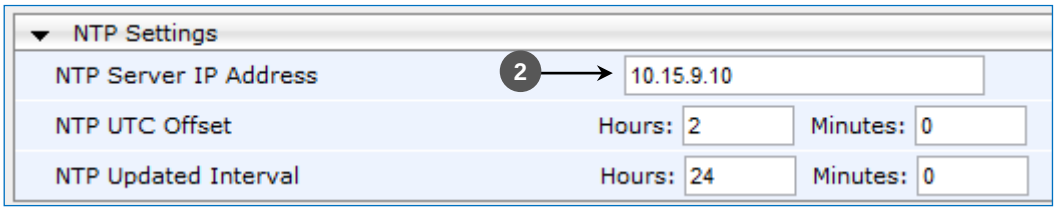
4.18.2 Step 18-2: Configuring NTP Server

The following describes how to configure the NTP Server IP address. It is recommended to implement an NTP server (third-party) so that the E-SBC device receives the accurate current date and time. This is necessary for validating remote parties' certificates.

➢ **To configure NTP Settings:**

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

Figure 4-27: NTP Settings



2. In the 'NTP Server IP Address' field, enter the NTP Server IP-Address (e.g., 10.15.9.10).

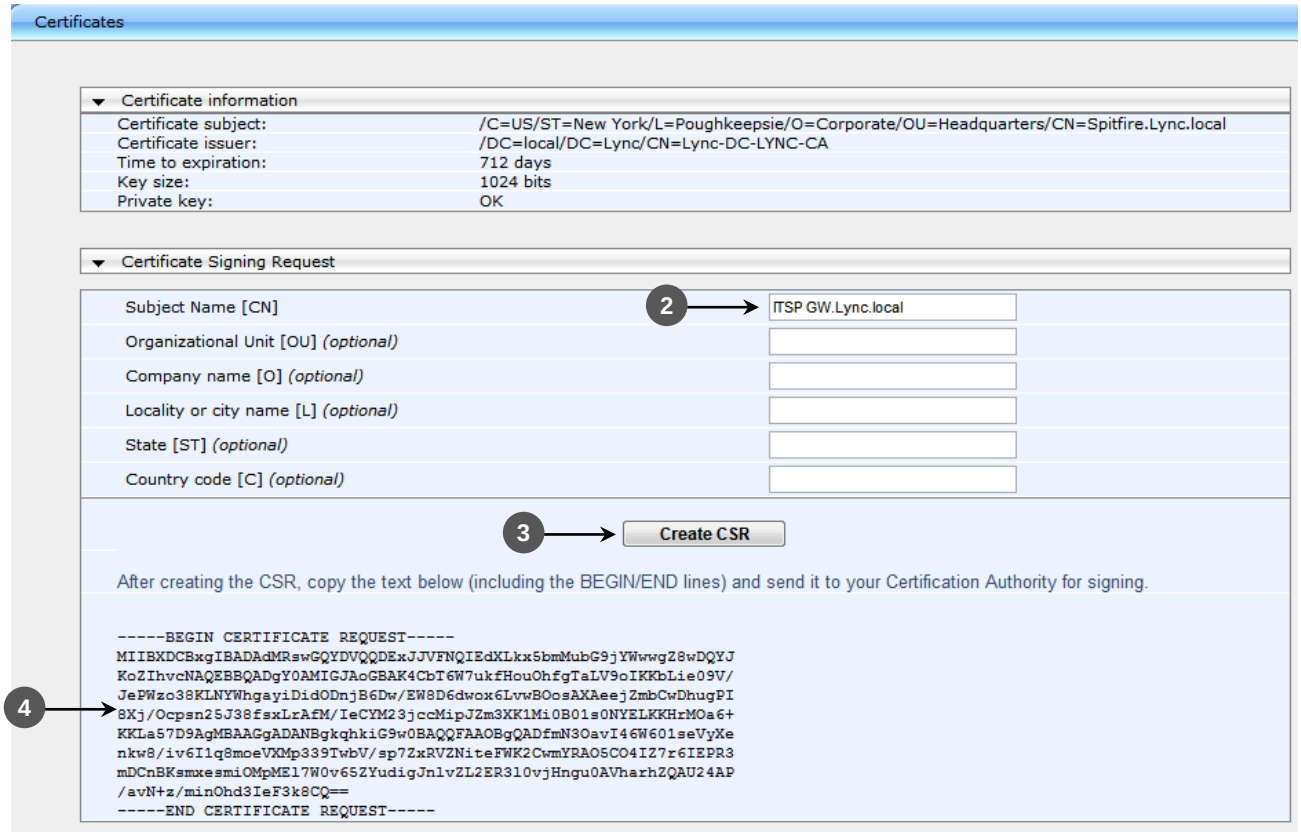
4.18.3 Step 18-3: Configuring a Certificate

The following 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 computer (i.e., the computer used to manage the E-SBC using the embedded Web interface.).

➢ **To configure a certificate:**

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

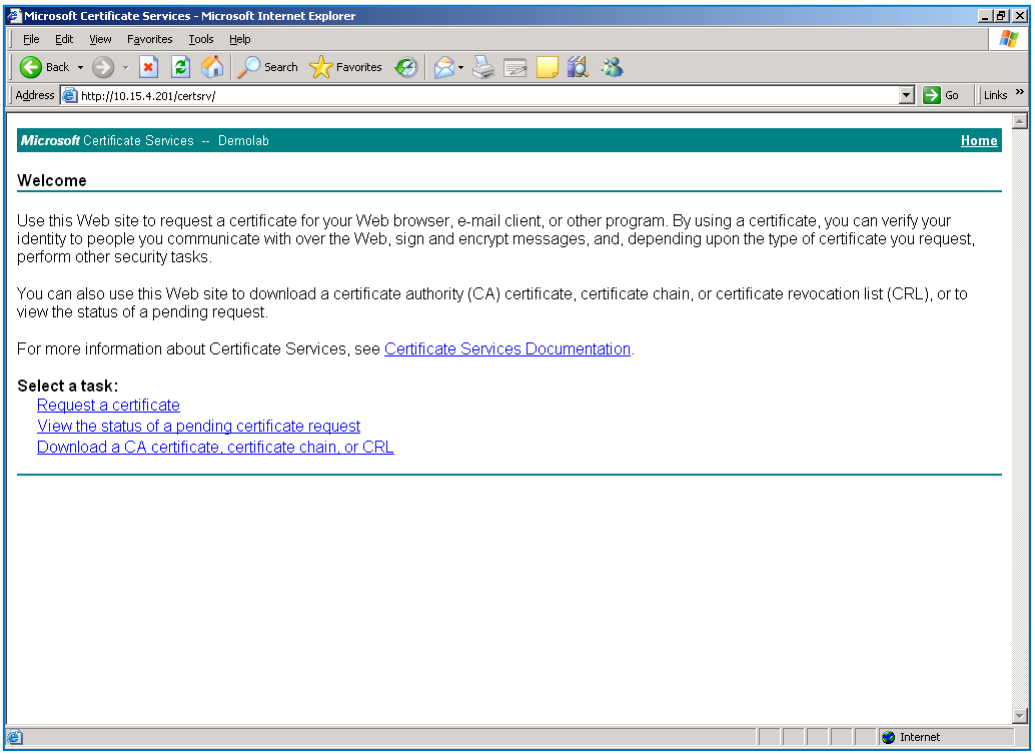
Figure 4-28: Certificates Page



2. In the 'Subject Name' field, enter the Media Gateway name (i.e., ITSP-GW.Lync.local)
3. Click **Generate CSR**; a Certificate request will be generated.
4. Copy the CSR (from the line "----BEGIN CERTIFICATE" to "END CERTIFICATE REQUEST----") to a .txt file (such as Notepad), and then save it to a folder on your computer as *certreq.txt*.

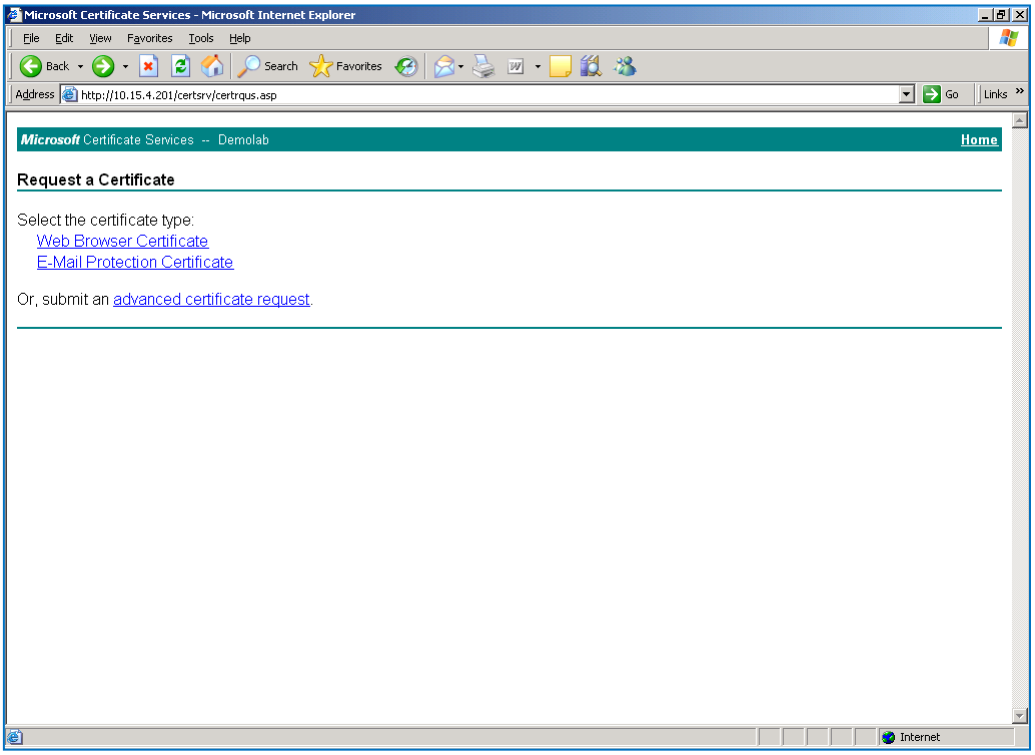
5. Navigate to the 'Server http://<Certificate Server>/CertSrv' certificate.

Figure 4-29: Microsoft Certificate Services Web Page



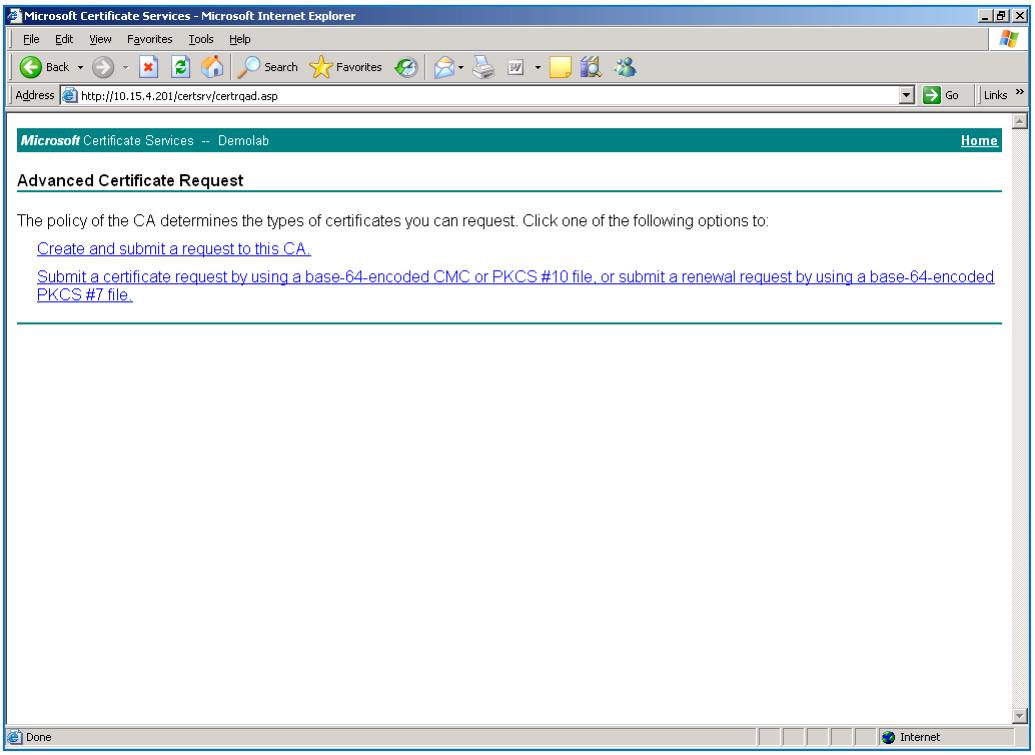
6. Click the **Request a Certificate** link; the following screen appears.

Figure 4-30: Request a Certificate Page



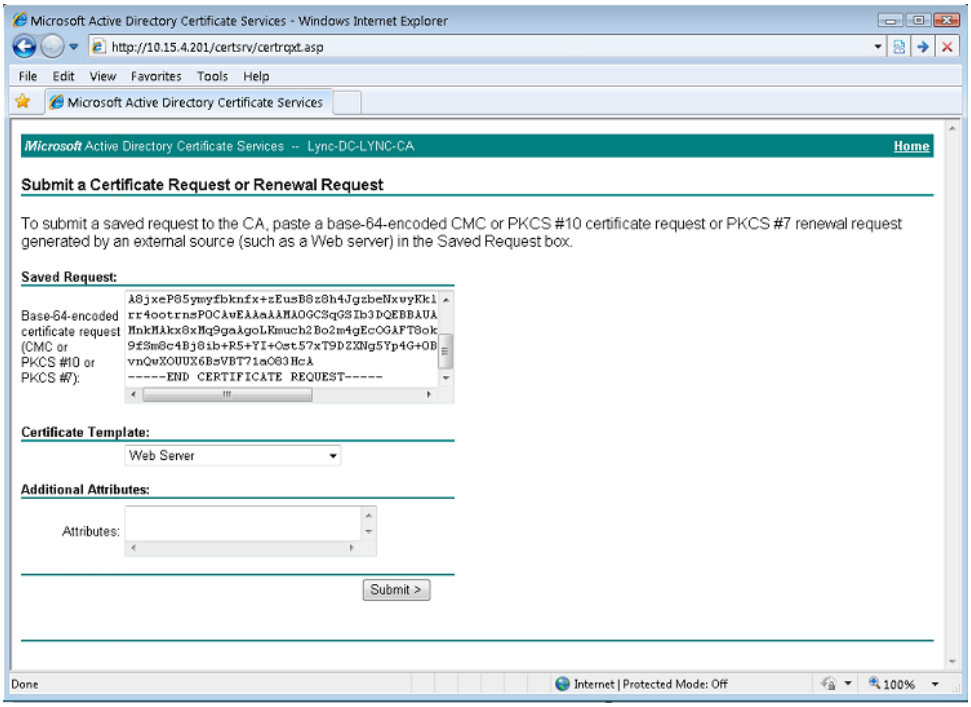
7. Click the **Advanced Certificate Request** link, and then click **Next**.

Figure 4-31: Advanced Certificate Request Page



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

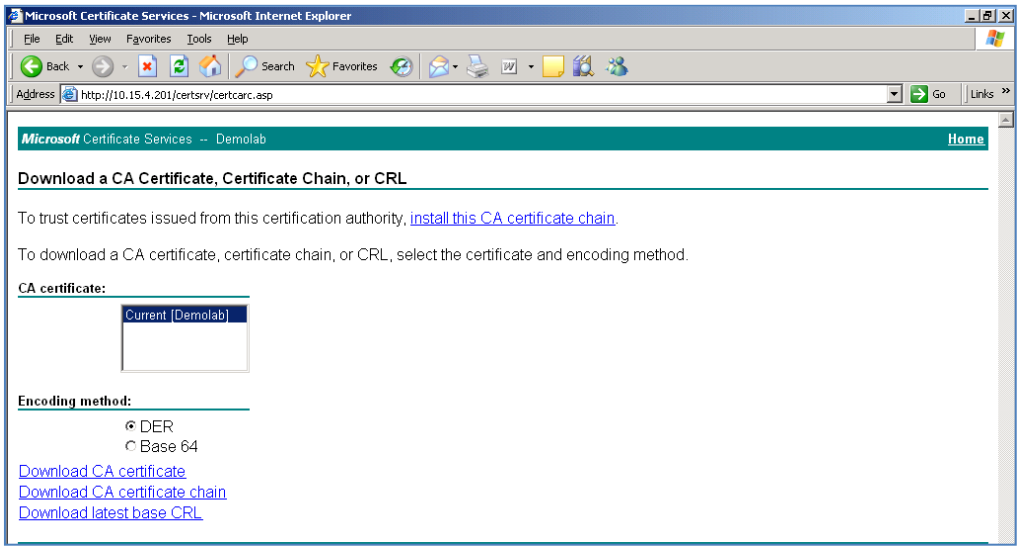
Figure 4-32: Submit a Certificate Request or Renewal Request Page



9. Open the *certreq.txt* file that you created and saved (see Step 4), and then copy its contents to the 'Base64 Encoded Certificate Request' text box.
10. From the 'Certificate Template' drop-down list, select **Web Server**.

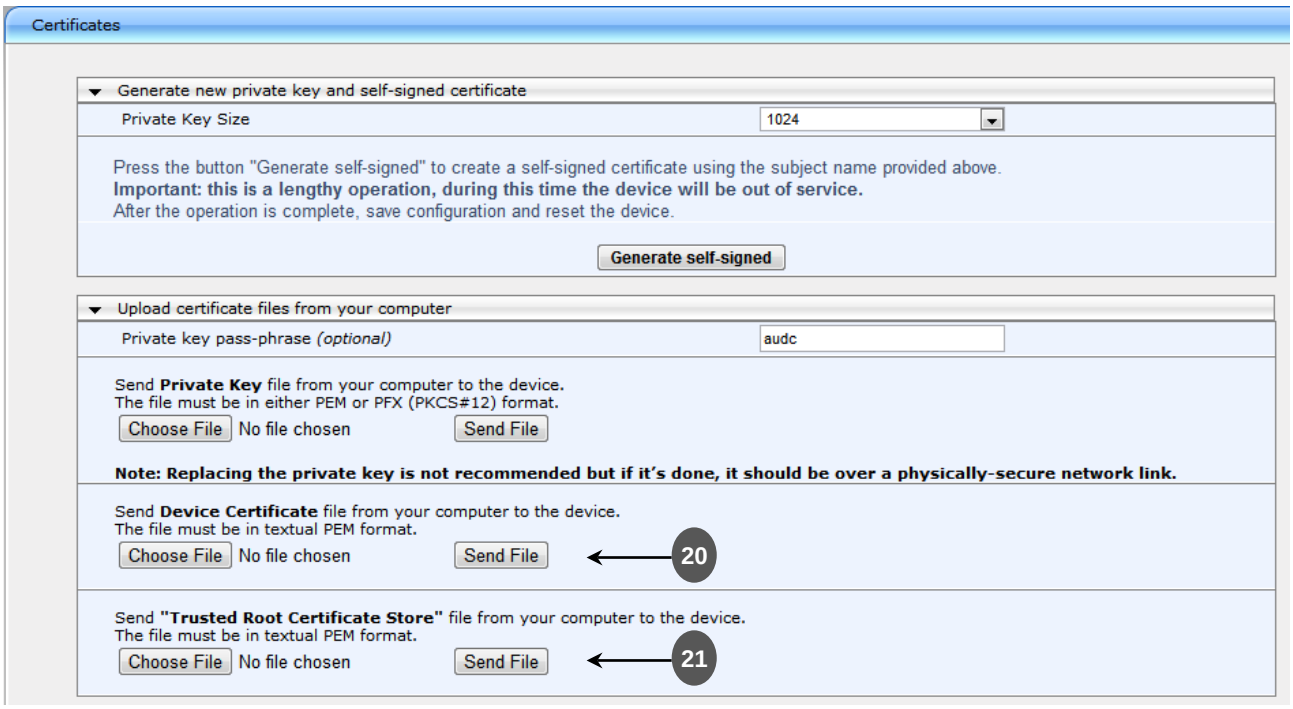
11. Click **Submit**; the following screen appears:

Figure 4-33: Download a CA Certificate, Certificate Chain, or CRL Page



12. Click the **Base 64** encoding option, and then click the **Download CA certificate** link.
13. Save the file as '*gateway.cer*' in a folder on your computer.
14. Navigate to the 'Server http://<Certificate Server>/CertSrv' certificate.
15. Click either one of the following links:
    - **Download a CA certificate**
    - **Download CA certificate chain**
    - **Download latest base CRL**
16. Under the Encoding method group, click the **Base 64** option.
17. Click the **Download CA certificate** link.
18. Save the file as 'certroot.cer' in a folder on your computer.
19. Navigate back (in the E-SBC device) to the Certificates page.

Figure 4-34: Certificates Page



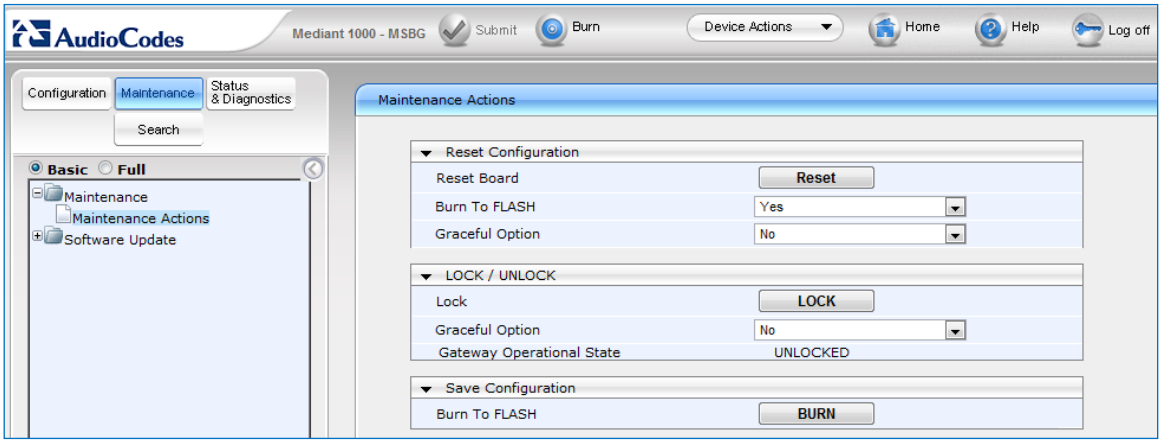
20. On the Certificates page, under the 'Send Device Certificate…' field, click **Choose File** and select the '*Gateway.cer'* certificate file that you saved on your local disk (see Step 13), and then click **Send File** to upload the certificate.
21. On the Certificates page, in the 'Trusted Root Certificate Store' field, click **Choose File** and select the '*Certroot.cer*' certificate file that you saved on your local disk (see Step 1818), and then click **Send File** to upload the certificate.
22. 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 **Maintenance** tab, and then in the Navigation tree, choose **Maintenance Actions**).

4.19 Step 19: Resetting the Gateway

After you have completed the gateway configuration as described in the steps above, you need to **save** ("burn") the configuration to the gateway's flash memory and then **reset** the gateway.

1. On the toolbar, from the 'Device Actions' drop-down list, choose **Reset**; the Maintenance Actions page appears.

Figure 4-35: Resetting the Gateway



2. Under the Reset Configuration group, click **Reset**. By default, the gateway burns the configuration to flash, before resetting the gateway.

A AudioCodes INI File

The following displays the E-SBC device ini file. This file reflects the configuration described in Section 4 on page 29.

```
;**************
;** Ini File **
;**************

;Board: Mediant 1000 - MSBG
;Serial Number: 3589366
;Slot Number: 1
;Software Version: 6.40A.022.009
;DSP Software Version: 620AE3 => 640.02
;Board IP Address: 10.15.7.131
;Board Subnet Mask: 255.255.0.0
;Board Default Gateway: 10.15.7.130
;Ram size: 512M   Flash size: 64M
;Num of DSP Cores: 13  Num DSP Channels: 51
;Profile: NONE
;Key features:;Board Type: Mediant 1000 - MSBG ;PSTN Protocols: ISDN
IUA=4 CAS ;Coders: G723 G729 GSM-FR G727 ILBC ;E1Trunks=4 ;T1Trunks=4
;IP Media: Conf VXML VoicePromptAnnounc(H248.9) ;Channel Type: RTP
DspCh=240 IPMediaDspCh=240 ;DSP Voice features: IpmDetector ;DATA
features: Routing FireWall&VPN WAN Advanced-Routing ;Security: IPSEC
MediaEncryption StrongEncryption EncryptControlProtocol ;Control
Protocols: MSFT MGCP MEGACO SIP SASurvivability SBC=120 ;Default
features:;Coders: G711 G726;

;--------- Mediant-1000 HW components ---------
;
; Slot # : Module type : # of ports : # of DSPs
;----------------------------------------------
;      1 : FALC56      :          2 :         3
;      2 : FXS         :          4 :         1
;      3 : Empty
;      4 : Empty
;      5 : Empty
;      6 : Empty
;----------------------------------------------


[SYSTEM Params]

DNSPriServerIP = 10.15.9.10
SyslogServerIP = 10.15.45.200
EnableSyslog = 1
NTPServerIP = 10.15.9.10
NTPServerUTCOffset = 7200
AllowWanHttp = 1
AllowWanHttps = 1
PM_VEDSPUtil = '1,43,48,15'
```

```
[BSP Params]

PCMLawSelect = 3
WanInterfaceName = 'GigabitEthernet 0/0'

[Analog Params]


[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]


[MEGACO Params]

EP_Num_0 = 0
EP_Num_1 = 1
EP_Num_2 = 1
EP_Num_3 = 0
EP_Num_4 = 0

[PSTN Params]


[SS7 Params]


[Voice Engine Params]

RFC2833TxPayloadType = 101
EnableAGC = 1
EnableDSPIPMDetectors = 1
ENABLEMEDIASECURITY = 1
SRTPTxPacketMKISize = 1
CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]

LogoWidth = '145'
HTTPSCipherString = 'RC4:EXP'

[SIP Params]

MEDIACHANNELS = 120
REGISTRATIONTIME = 600
SIPDESTINATIONPORT = 5067
PLAYRBTONE2TEL = 0
GWDEBUGLEVEL = 5
ENABLEEARLYMEDIA = 1
```

```
SIPGATEWAYNAME = 'ITSP-GW.Lync.Local'
STATICNATIP = 195.189.192.154
PROXYREDUNDANCYMODE = 1
ADDTON2RPI = 0
SIPTRANSPORTTYPE = 2
TLSLOCALSIPPORT = 5067
MEDIASECURITYBEHAVIOUR = 3
ENABLECONTACTRESTRICTION = 1
FORKINGHANDLINGMODE = 1
ENABLESBCAPPLICATION = 1
ENABLEIP2IPAPPLICATION = 1
SELECTSOURCEHEADERFORCALLEDNUMBER = 1
ENABLEEARLY183 = 1
GWINBOUNDMANIPULATIONSET = 1

[SCTP Params]


[VXML Params]


[IPsec Params]


[Audio Staging Params]


[SNMP Params]


[ PREFIX ]

FORMAT PREFIX_Index = PREFIX_DestinationPrefix, PREFIX_DestAddress,
PREFIX_SourcePrefix, PREFIX_ProfileId, PREFIX_MeteringCode,
PREFIX_DestPort, PREFIX_SrcIPGroupID, PREFIX_DestHostPrefix,
PREFIX_DestIPGroupID, PREFIX_SrcHostPrefix, PREFIX_TransportType,
PREFIX_SrcTrunkGroupID, PREFIX_DestSRD, PREFIX_CostGroup,
PREFIX_ForkingGroup;
PREFIX 0 = *, , *, 2, 255, 0, 1, , 2, , -1, -1, -1, , -1;
PREFIX 1 = *, , *, 1, 255, 0, 2, , 1, , -1, -1, -1, , -1;

[ \PREFIX ]


[ NumberMapTel2Ip ]

FORMAT NumberMapTel2Ip_Index = NumberMapTel2Ip_DestinationPrefix,
NumberMapTel2Ip_SourcePrefix, NumberMapTel2Ip_SourceAddress,
NumberMapTel2Ip_NumberType, NumberMapTel2Ip_NumberPlan,
NumberMapTel2Ip_RemoveFromLeft, NumberMapTel2Ip_RemoveFromRight,
NumberMapTel2Ip_LeaveFromRight, NumberMapTel2Ip_Prefix2Add,
NumberMapTel2Ip_Suffix2Add, NumberMapTel2Ip_IsPresentationRestricted,
NumberMapTel2Ip_SrcTrunkGroupID, NumberMapTel2Ip_SrcIPGroupID;
```

```
NumberMapTel2Ip 1 = +, *, *, 255, 255, 1, 0, 255, , , 255, -1, 1;
NumberMapTel2Ip 2 = *, *, *, 255, 255, 0, 0, 255, +, , 255, -1, 2;

[ \NumberMapTel2Ip ]


[ SourceNumberMapTel2Ip ]

FORMAT SourceNumberMapTel2Ip_Index =
SourceNumberMapTel2Ip_DestinationPrefix,
SourceNumberMapTel2Ip_SourcePrefix,
SourceNumberMapTel2Ip_SourceAddress, SourceNumberMapTel2Ip_NumberType,
SourceNumberMapTel2Ip_NumberPlan,
SourceNumberMapTel2Ip_RemoveFromLeft,
SourceNumberMapTel2Ip_RemoveFromRight,
SourceNumberMapTel2Ip_LeaveFromRight,
SourceNumberMapTel2Ip_Prefix2Add, SourceNumberMapTel2Ip_Suffix2Add,
SourceNumberMapTel2Ip_IsPresentationRestricted,
SourceNumberMapTel2Ip_SrcTrunkGroupID,
SourceNumberMapTel2Ip_SrcIPGroupID;
SourceNumberMapTel2Ip 2 = *, +, *, 255, 255, 1, 0, 255, , , 255, -1,
1;
SourceNumberMapTel2Ip 3 = *, 00, *, 255, 255, 2, 0, 255, , , 255, -1,
2;

[ \SourceNumberMapTel2Ip ]


[ PstnPrefix ]

FORMAT PstnPrefix_Index = PstnPrefix_DestPrefix,
PstnPrefix_TrunkGroupId, PstnPrefix_SourcePrefix,
PstnPrefix_SourceAddress, PstnPrefix_ProfileId,
PstnPrefix_SrcIPGroupID, PstnPrefix_DestHostPrefix,
PstnPrefix_SrcHostPrefix;
PstnPrefix 0 = *, -1, , 10.15.9.11, 1, 1, , ;
PstnPrefix 1 = *, -1, , 83.218.143.13, 2, 2, , ;

[ \PstnPrefix ]


[ ProxyIp ]

FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = FE-Lync.Lync.local:5067, 2, 1;
ProxyIp 1 = mproxy4.spitfiretsp.net:5060, 0, 2;

[ \ProxyIp ]


[ TxDtmfOption ]

FORMAT TxDtmfOption_Index = TxDtmfOption_Type;
```

```
TxDtmfOption 0 = 4;

[ \TxDtmfOption ]


[ IpProfile ]

FORMAT IpProfile_Index = IpProfile_ProfileName,
IpProfile_IpPreference, IpProfile_CodersGroupID, IpProfile_IsFaxUsed,
IpProfile_JitterBufMinDelay, IpProfile_JitterBufOptFactor,
IpProfile_IPDiffServ, IpProfile_SigIPDiffServ, IpProfile_SCE,
IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort,
IpProfile_CNGmode, IpProfile_VxxTransportType, IpProfile_NSEMode,
IpProfile_IsDTMFUsed, IpProfile_PlayRBTone2IP,
IpProfile_EnableEarlyMedia, IpProfile_ProgressIndicator2IP,
IpProfile_EnableEchoCanceller, IpProfile_CopyDest2RedirectNumber,
IpProfile_MediaSecurityBehaviour, IpProfile_CallLimit,
IpProfile_DisconnectOnBrokenConnection, IpProfile_FirstTxDtmfOption,
IpProfile_SecondTxDtmfOption, IpProfile_RxDTMFOption,
IpProfile_EnableHold, IpProfile_InputGain, IpProfile_VoiceVolume,
IpProfile_AddIEInSetup, IpProfile_SBCExtensionCodersGroupID,
IpProfile_MediaIPVersionPreference, IpProfile_TranscodingMode,
IpProfile_SBCAllowedCodersGroupID, IpProfile_SBCAllowedCodersMode,
IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior,
IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCAssertIdentity,
IpProfile_AMDSensitivityParameterSuit, IpProfile_AMDSensitivityLevel,
IpProfile_AMDMaxGreetingTime, IpProfile_AMDMaxPostSilenceGreetingTime,
IpProfile_SBCDiversionMode, IpProfile_SBCHistoryInfoMode,
IpProfile_EnableQSIGTunneling, IpProfile_SBCFaxCodersGroupID,
IpProfile_SBCFaxBehavior, IpProfile_SBCFaxOfferMode,
IpProfile_SBCFaxAnswerMode;
IpProfile 1 = Lync, 1, 1, 2, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0,
1, -1, 1, 0, 3, -1, 0, 4, -1, 1, 1, 0, 0, , -1, 0, 0, -1, 0, 0, 0, 0,
-1, 0, 8, 300, 400, -1, -1, 0, -1, 0, 0, 1;
IpProfile 2 = spitfire, 1, 2, 2, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0,
1, 1, -1, 1, 0, 2, -1, 0, 4, -1, 1, 1, 0, 0, , -1, 0, 0, -1, 0, 0, 0,
0, -1, 0, 8, 300, 400, -1, -1, 0, -1, 0, 0, 1;

[ \IpProfile ]


[ ProxySet ]

FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive,
ProxySet_ProxyKeepAliveTime, ProxySet_ProxyLoadBalancingMethod,
ProxySet_IsProxyHotSwap, ProxySet_SRD, ProxySet_ClassificationInput,
ProxySet_ProxyRedundancyMode;
ProxySet 0 = 0, 60, 0, 0, 0, 0, -1;
ProxySet 1 = 1, 60, 1, 1, 0, 0, -1;
ProxySet 2 = 0, 60, 0, 0, 0, 0, -1;

[ \ProxySet ]


[ IPGroup ]
```

```
FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Description,
IPGroup_ProxySetId, IPGroup_SIPGroupName, IPGroup_ContactUser,
IPGroup_EnableSurvivability, IPGroup_ServingIPGroup,
IPGroup_SipReRoutingMode, IPGroup_AlwaysUseRouteTable,
IPGroup_RoutingMode, IPGroup_SRD, IPGroup_MediaRealm,
IPGroup_ClassifyByProxySet, IPGroup_ProfileId,
IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet,
IPGroup_OutboundManSet, IPGroup_RegistrationMode,
IPGroup_AuthenticationMode, IPGroup_MethodList,
IPGroup_EnableSBCClientForking, IPGroup_ContactName;
IPGroup 1 = 0, Lync, 1, , , 0, -1, 0, 0, -1, 0, , 1, 1, -1, -1, -1, 0,
0, , 0, ;
IPGroup 2 = 0, Spitfire, 2, , , 0, -1, 0, 0, -1, 0, , 1, 2, -1, -1, 0,
0, 0, , 0, ;

[ \IPGroup ]


[ Account ]

FORMAT Account_Index = Account_ServedTrunkGroup,
Account_ServedIPGroup, Account_ServingIPGroup, Account_Username,
Account_Password, Account_HostName, Account_Register,
Account_ContactUser, Account_ApplicationType;
Account 1 = -1, 1, 2, 442031419616, *, spitfiretsp.net, 1,
442031419616, 0;

[ \Account ]


[ IPInboundManipulation ]

FORMAT IPInboundManipulation_Index =
IPInboundManipulation_IsAdditionalManipulation,
IPInboundManipulation_ManipulationPurpose,
IPInboundManipulation_SrcIPGroupID,
IPInboundManipulation_SrcUsernamePrefix,
IPInboundManipulation_SrcHost,
IPInboundManipulation_DestUsernamePrefix,
IPInboundManipulation_DestHost, IPInboundManipulation_RequestType,
IPInboundManipulation_ManipulatedURI,
IPInboundManipulation_RemoveFromLeft,
IPInboundManipulation_RemoveFromRight,
IPInboundManipulation_LeaveFromRight,
IPInboundManipulation_Prefix2Add, IPInboundManipulation_Suffix2Add;
IPInboundManipulation 2 = 0, 0, -1, *, *, *, *, 0, 0, 0, 0, 255, , ;

[ \IPInboundManipulation ]


[ CodersGroup0 ]

FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce;
CodersGroup0 0 = g711Alaw64k, 20, 0, -1, 0;
```

```
CodersGroup0 1 = g711Ulaw64k, 20, 0, -1, 0;

[ \CodersGroup0 ]


[ CodersGroup1 ]

FORMAT CodersGroup1_Index = CodersGroup1_Name, CodersGroup1_pTime,
CodersGroup1_rate, CodersGroup1_PayloadType, CodersGroup1_Sce;
CodersGroup1 0 = g711Alaw64k, 20, 0, -1, 0;
CodersGroup1 1 = g711Ulaw64k, 20, 0, -1, 0;

[ \CodersGroup1 ]


[ CodersGroup2 ]

FORMAT CodersGroup2_Index = CodersGroup2_Name, CodersGroup2_pTime,
CodersGroup2_rate, CodersGroup2_PayloadType, CodersGroup2_Sce;
CodersGroup2 0 = g711Alaw64k, 20, 0, -1, 0;
CodersGroup2 1 = g711Ulaw64k, 20, 0, -1, 0;
CodersGroup2 2 = g729, 20, 0, -1, 0;

[ \CodersGroup2 ]


[ MessageManipulations ]

FORMAT MessageManipulations_Index = MessageManipulations_ManSetID,
MessageManipulations_MessageType, MessageManipulations_Condition,
MessageManipulations_ActionSubject, MessageManipulations_ActionType,
MessageManipulations_ActionValue, MessageManipulations_RowRole;
MessageManipulations 0 = 1, invite, , Header.From.URL.user, 2,
header.referred-by.URL.user, 0;
MessageManipulations 2 = 0, invite, , header.REMOTE-PARTY-ID, 0,
"'<'+Header.From.URL+'>' + ';party=calling'", 0;
MessageManipulations 3 = 0, invite, , header.REMOTE-PARTY-ID.url.user,
6, '44', 1;
MessageManipulations 5 = 0, INVITE, , Header.From.URL.user, 2,
'442031419616', 0;

[ \MessageManipulations ]


[ RoutingRuleGroups ]

FORMAT RoutingRuleGroups_Index = RoutingRuleGroups_LCREnable,
RoutingRuleGroups_LCRAverageCallLength,
RoutingRuleGroups_LCRDefaultCost;
RoutingRuleGroups 0 = 0, 0, 1;

[ \RoutingRuleGroups ]
```

```
[ InterfaceTable ]

FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_VlanID, InterfaceTable_InterfaceName,
InterfaceTable_PrimaryDNSServerIPAddress,
InterfaceTable_SecondaryDNSServerIPAddress,
InterfaceTable_UnderlyingInterface;
InterfaceTable 0 = 6, 10, 10.15.7.131, 16, 10.15.7.130, 1, Voice,
10.15.9.10, 80.179.55.100, ;

[ \InterfaceTable ]


[ DspTemplates ]

;
;  *** TABLE DspTemplates ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts.
;

[ \DspTemplates ]
```

Reader's Notes

AudioCodes

Configuration Note

www.audiocodes.com