Configuration Note

Connecting Microsoft® Lync™ & Timico SIP Trunk using AudioCodes Mediant™ E-SBC Series

November 2011

Document #: LTRT-39290

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Notice

This document describes how to connect the Microsoft Lync 2010 with Timico SIP Trunk 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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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: Acronyms

Acronym	Meaning
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

This document describes how to setup the device to work with the Timico SIP Trunking and Microsoft Lync Communication platform.

This configuration note is intended for Installation Engineers or AudioCodes and Timico Partners who are installing and configuring the Timico SIP Trunking and Microsoft Lync 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 Gateway and E-SBC, enables 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 all-in-one multi-service access solution products 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 & Data services for business customers at wire speed, while maintaining SLA parameters for superior voice quality.

The Mediant 1000B Gateway and E-SBC enables connectivity and security between small and medium businesses (SMB) 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 Mediant 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 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'.

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_6.20A.022.003
Interface Type	SIP/IP
VoIP Protocol	SIP/UDP – to the Timico Sip Trunk SIP/TCP or TLS – to the Lync FE Server
Additional Notes	None

2.2 Timico SIP Trunking Version

Table 2-2: Timico Version

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

2.3 Microsoft Lync Version

Table 2-3: Microsoft Lync 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:

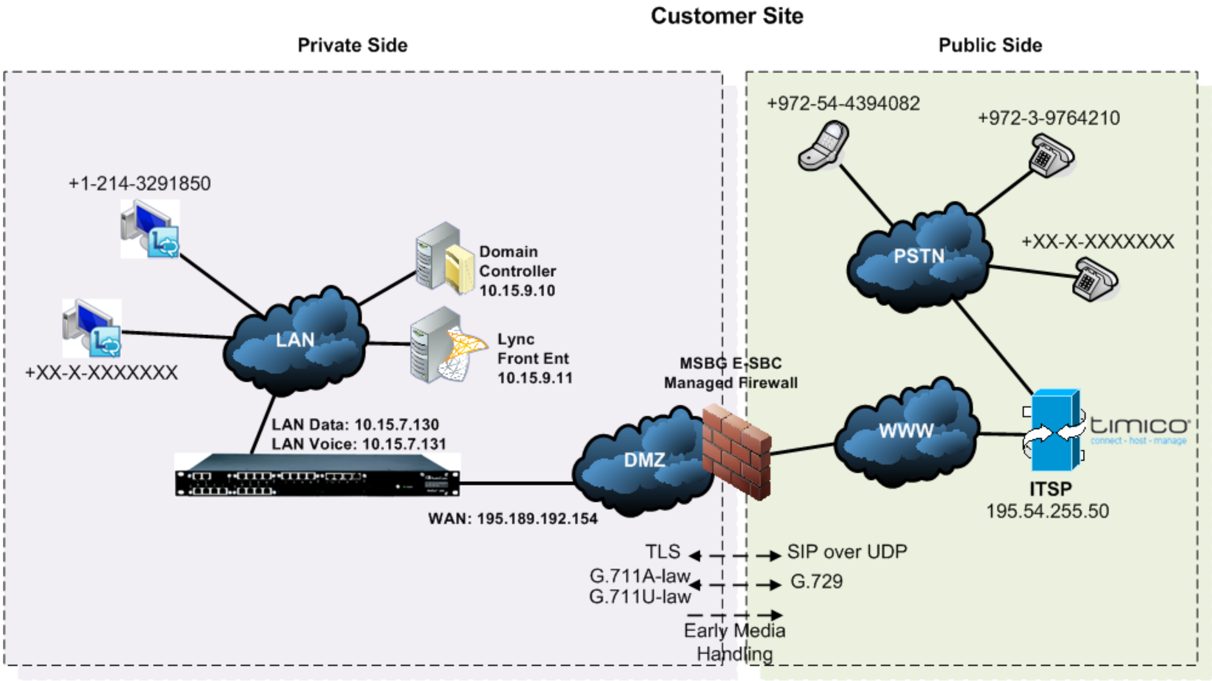
- An enterprise has a deployed Microsoft® Lync 2010 in its private network for enhanced communication within the company.
- The enterprise decides to offer its employees enterprise voice capabilities and to connect the company to the PSTN network using the Timico SIP Trunking service.
- AudioCodes Enterprise 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 Timico SIP trunk in the public network. Figure 2-1 below illustrates a typical topology of using the E-SBC device to connect the Microsoft® Lync Server 2010 LAN to the Timico SIP Trunking site.

The setup requirements are characterized as follows:

- While the Microsoft® Lync Server 2010 environment is located on the Enterprise's Local Area Network (LAN), the Timico SIP Trunks are located on the WAN.
- Since 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.
- Microsoft® Lync Server 2010 works with the TLS transport type, while the Timico SIP trunk works on the SIP over UDP transport type.
- Transcoding support: Microsoft® Lync Server 2010 supports G.711A-law and G.711U-law coders, while the Timico SIP Trunk also supports G.729 coder type.
- Support for early media handling.

Figure 2-1: Topology



Reader's Notes

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:

1. Configuring the E-SBC device as a ‘IP/PSTN Gateway’. See Section 3.1 on page 16.
2. Associating the ‘IP/PSTN Gateway’ with the Mediation Server. See Section 3.2 on page 20.
3. Configuring a ‘Route’ to utilize the SIP trunk connected to the E-SBC device. See Section 3.3 on page 26.

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

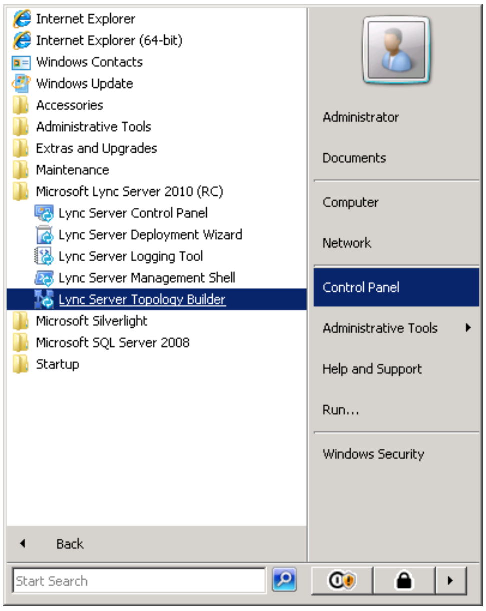
3.1 Configuring the E-SBC device as a 'IP/PSTN Gateway'

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 associating it with the Mediation Server:**

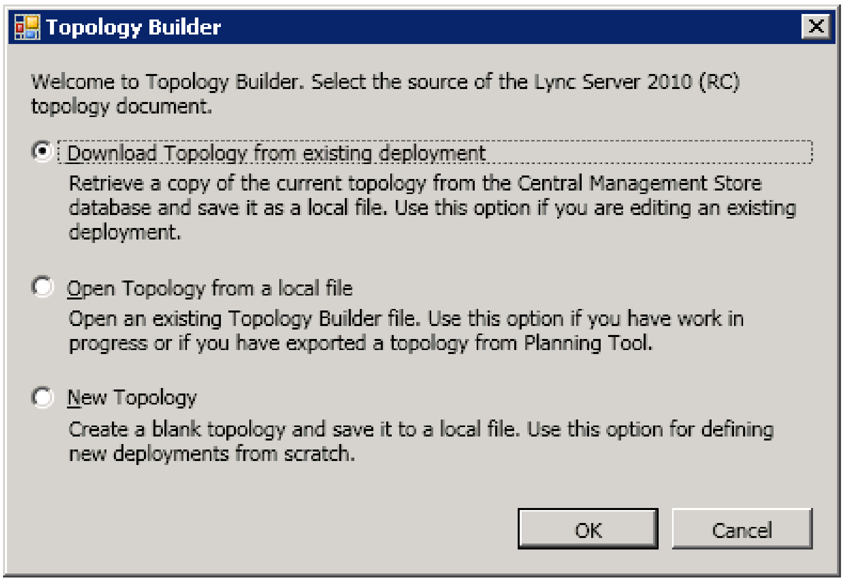
1. On the server where the Topology Builder is located, start the Lync Server 2010 **Topology Builder**: Click **Start**, select **All Programs**, then select **Lync Server Topology Builder**.

Figure 3-1: Opening the Lync Server Topology Builder



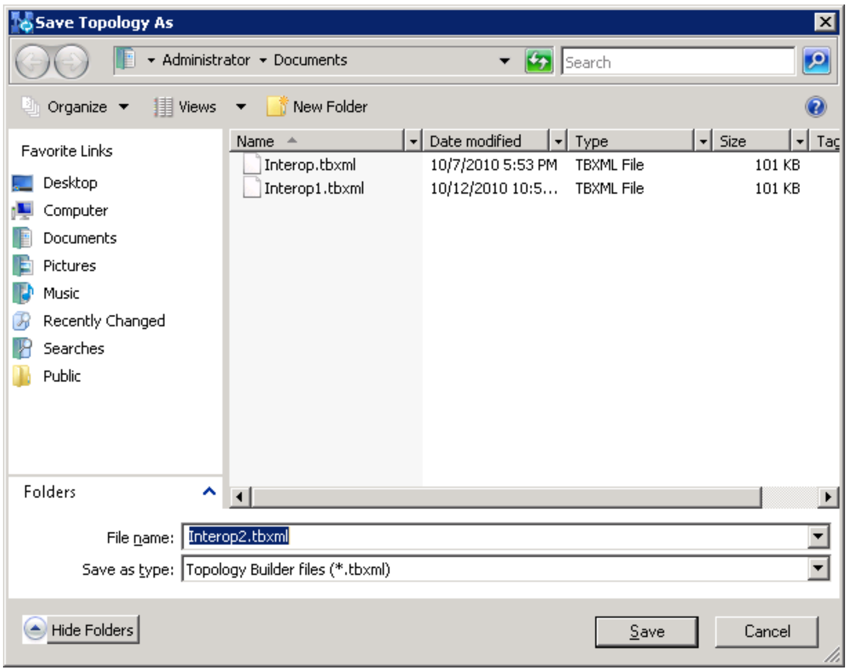
The following screen is displayed:

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.

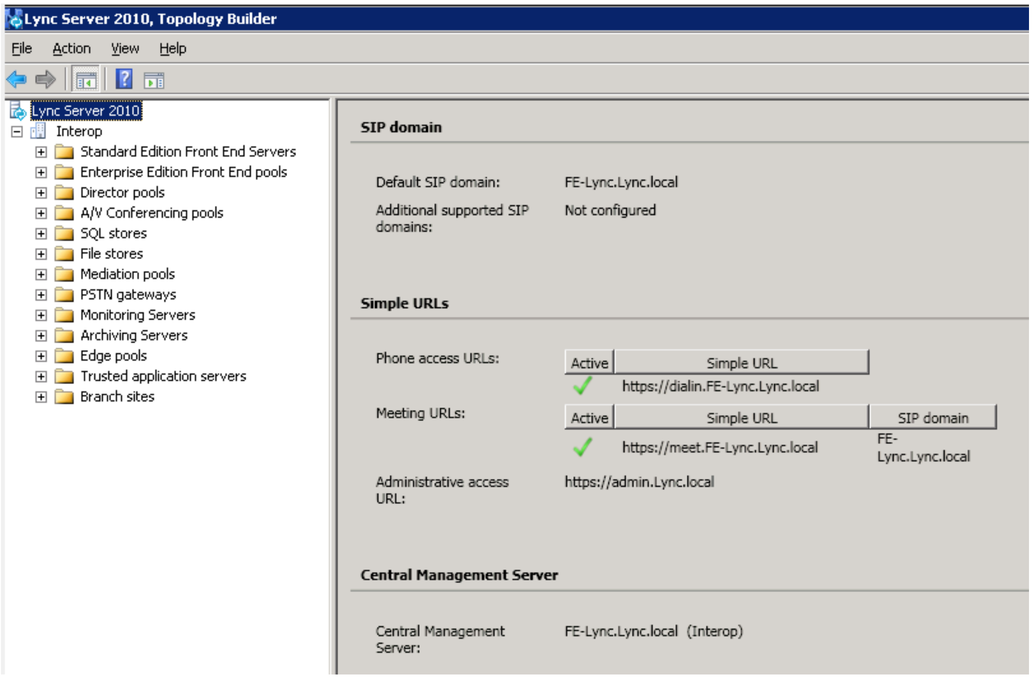
Figure 3-3: Save Topology



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

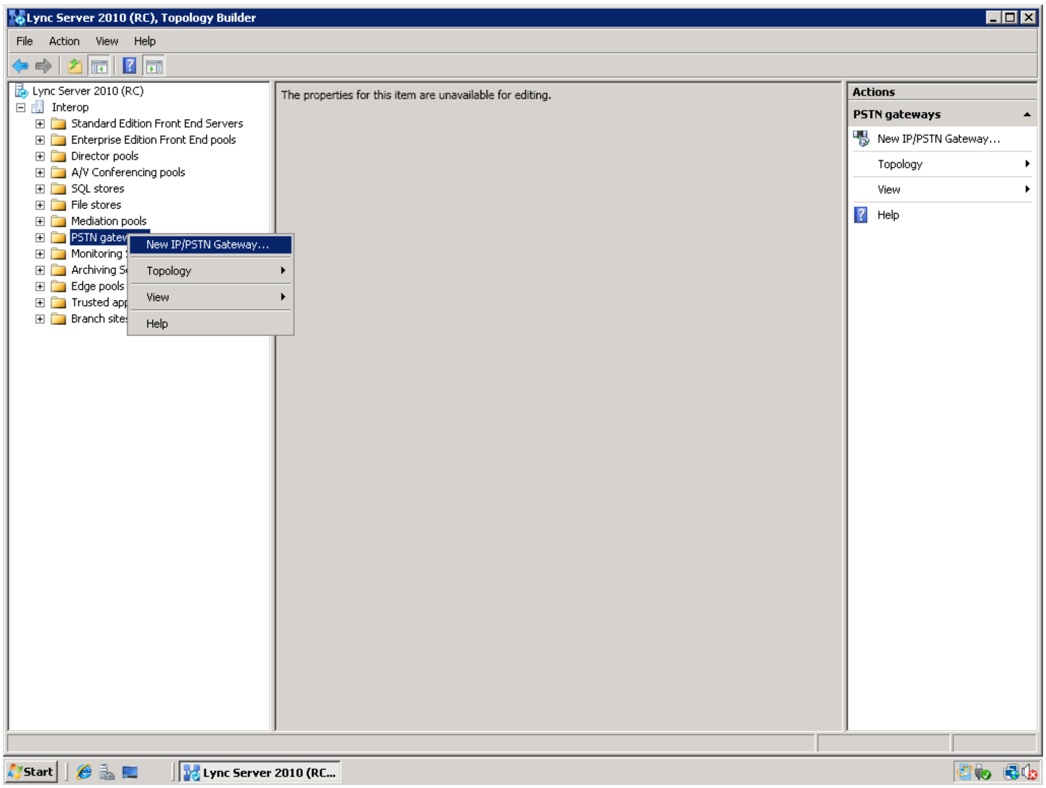
The Topology Builder screen with the topology downloaded is displayed.

Figure 3-4: Downloaded Topology



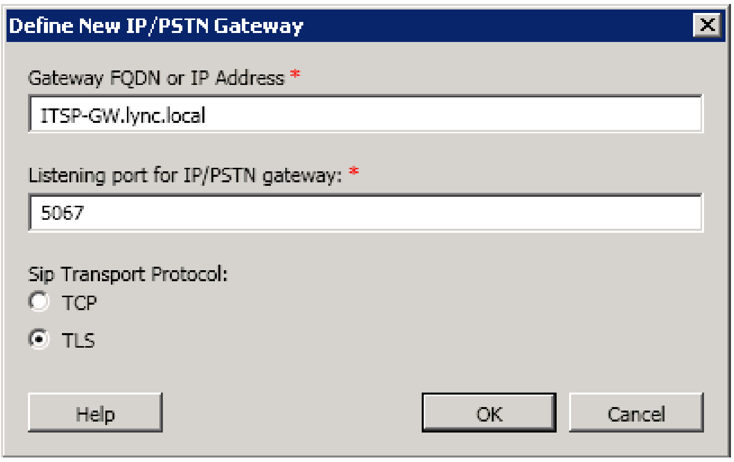
4. Expand the Site; right-click on the IP/PSTN Gateway and choose 'New IP/PSTN Gateway'.

Figure 3-5: New IP/PSTN Gateway



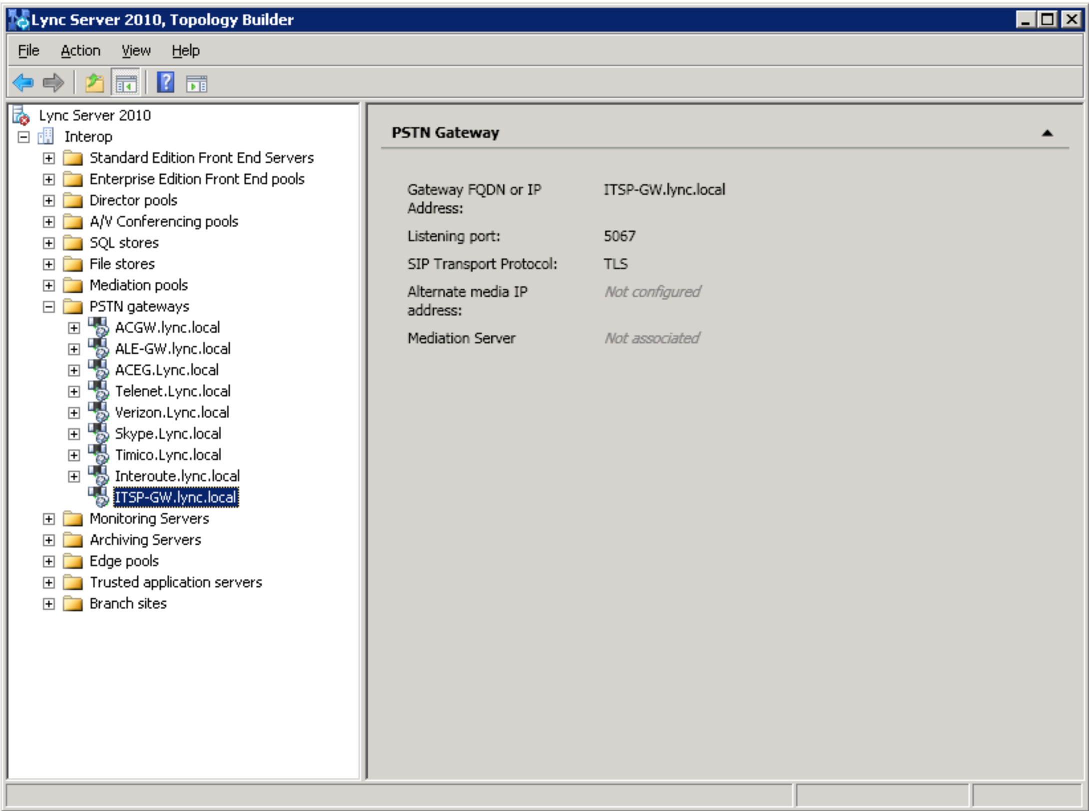
5. Enter the FQDN of the E-SBC (i.e. 'ITSP-GW.lync.local') and click **OK**.
   Note that the listening port for the Gateway is **5067** and the transport type is **TLS**.

Figure 3-6: Define New IP/PSTN Gateway



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

Figure 3-7: IP/PSTN Gateway



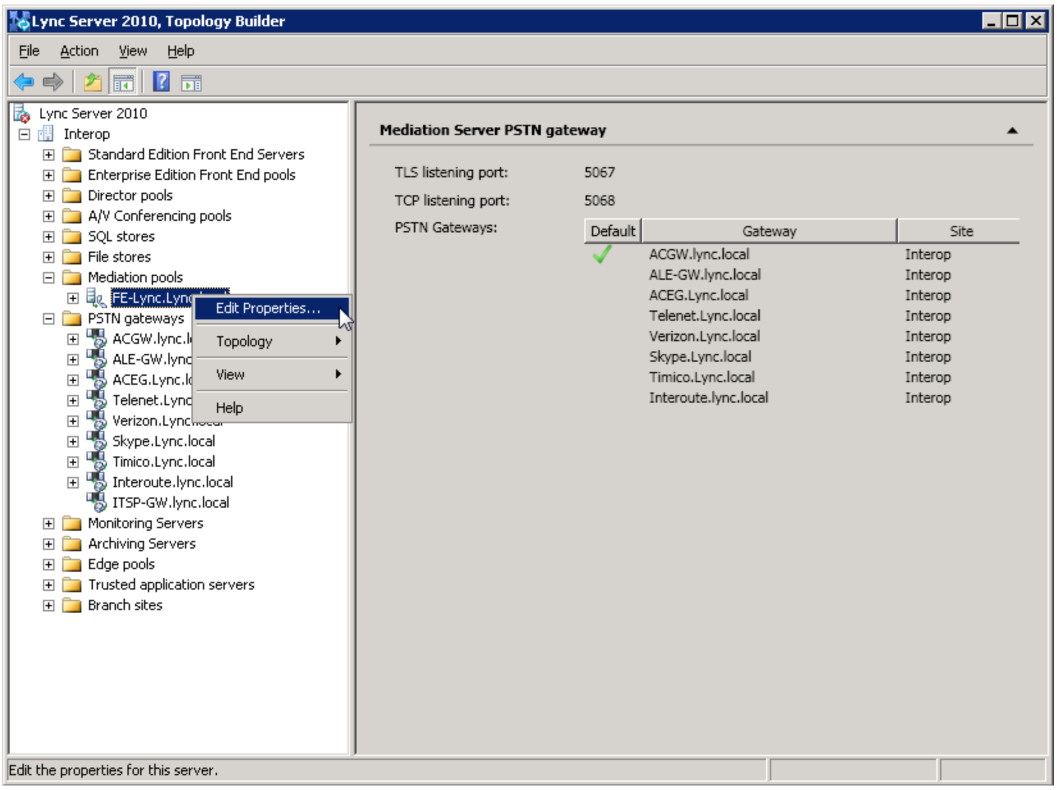
3.2 Associating the 'IP/PSTN Gateway' with the Mediation Server

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

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

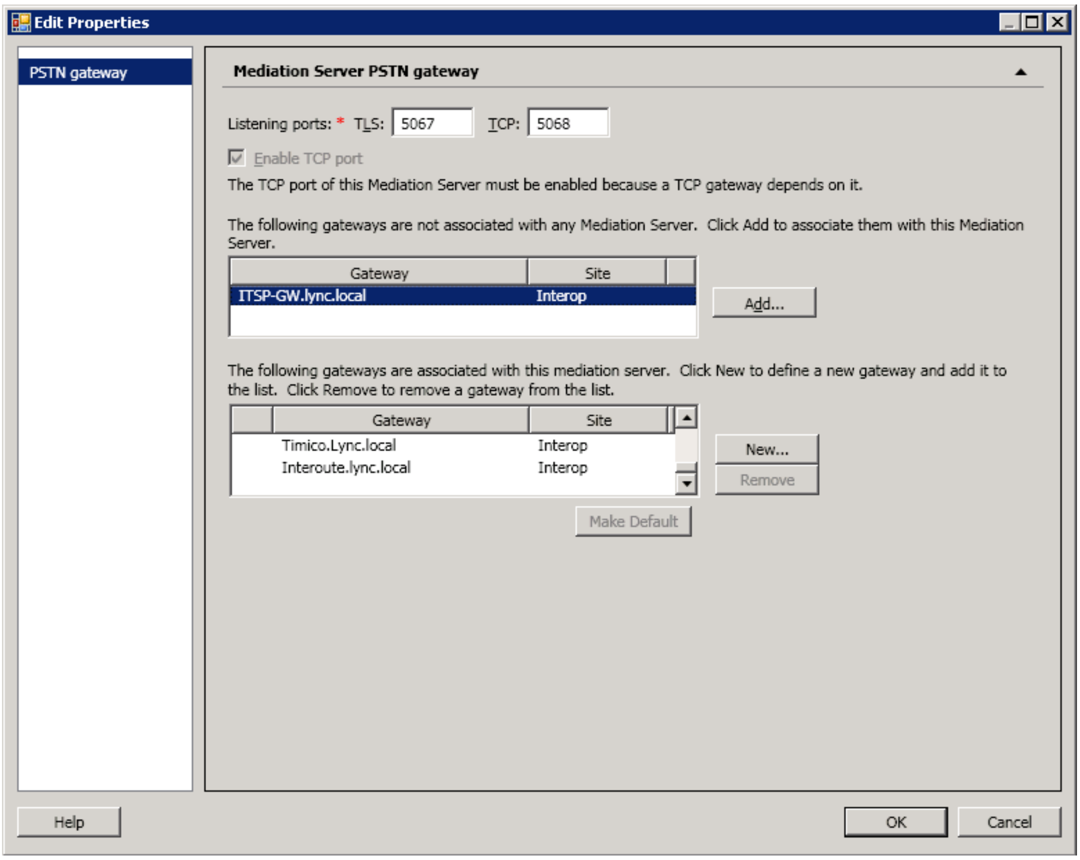
1. Right-click on the **Mediation server** that uses the E-SBC device (i.e. FE-Lync.Lync.local) and chooses **Edit Properties**.

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



The following screen is displayed:

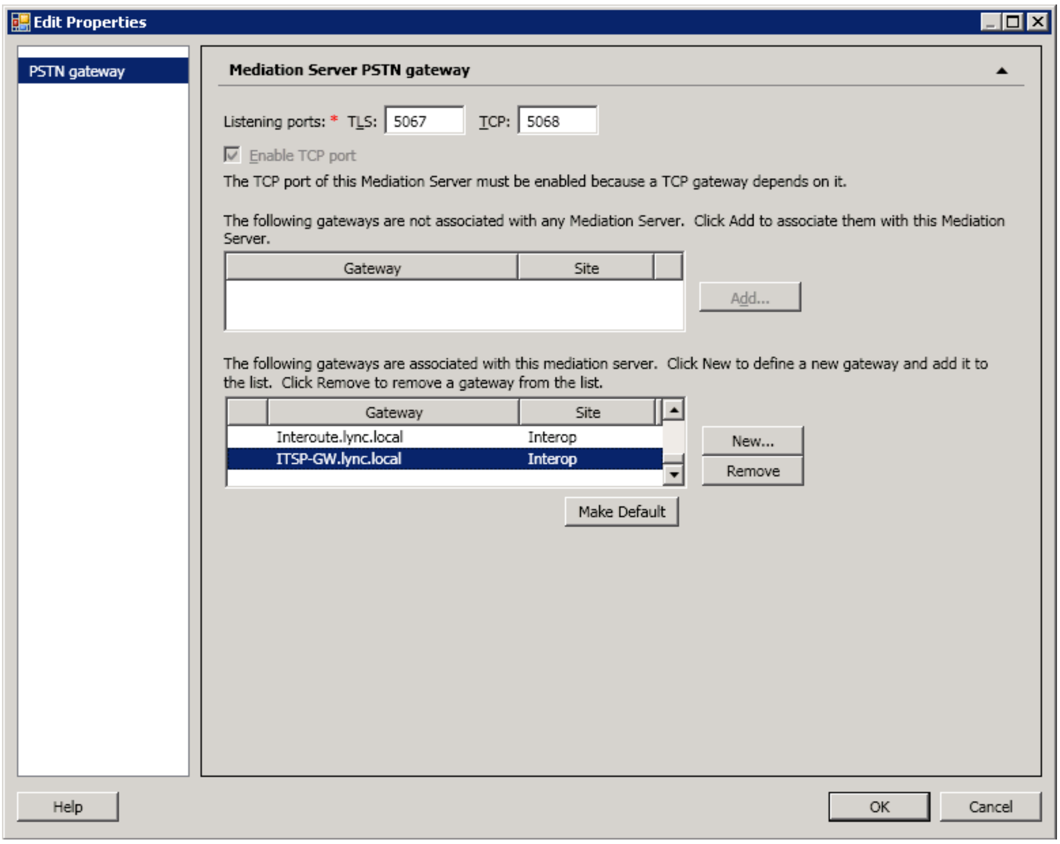
Figure 3-9: Before Associating IP/PSTN Gateway to Mediation Server



2. In the top-left corner, choose **PSTN gateway** and in the Mediation Server PSTN gateway pane, mark the E-SBC gateway (i.e. 'ITSP-GW.lync.local') and click **Add** to associate it with this Mediation Server.

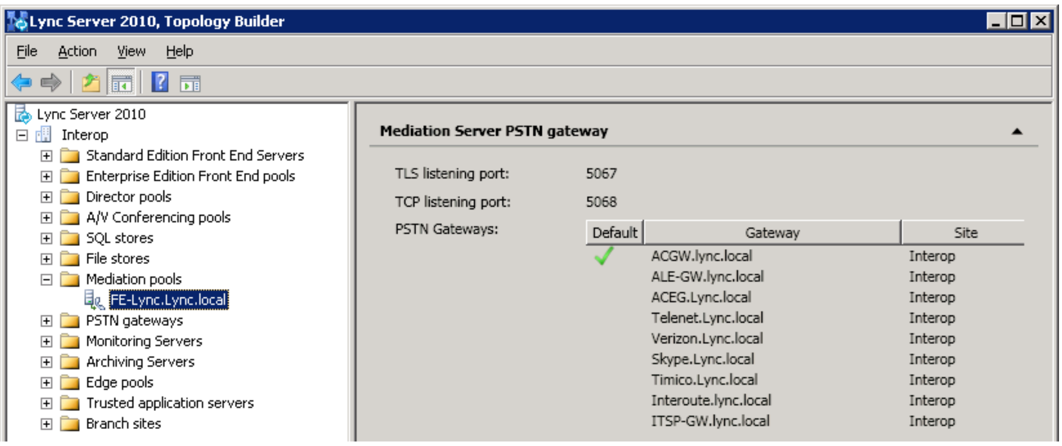
   Note that there are two sub-panes, one including a list of gateways not associated with the Mediation server and one including a list of gateways associated with the Mediation server.

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



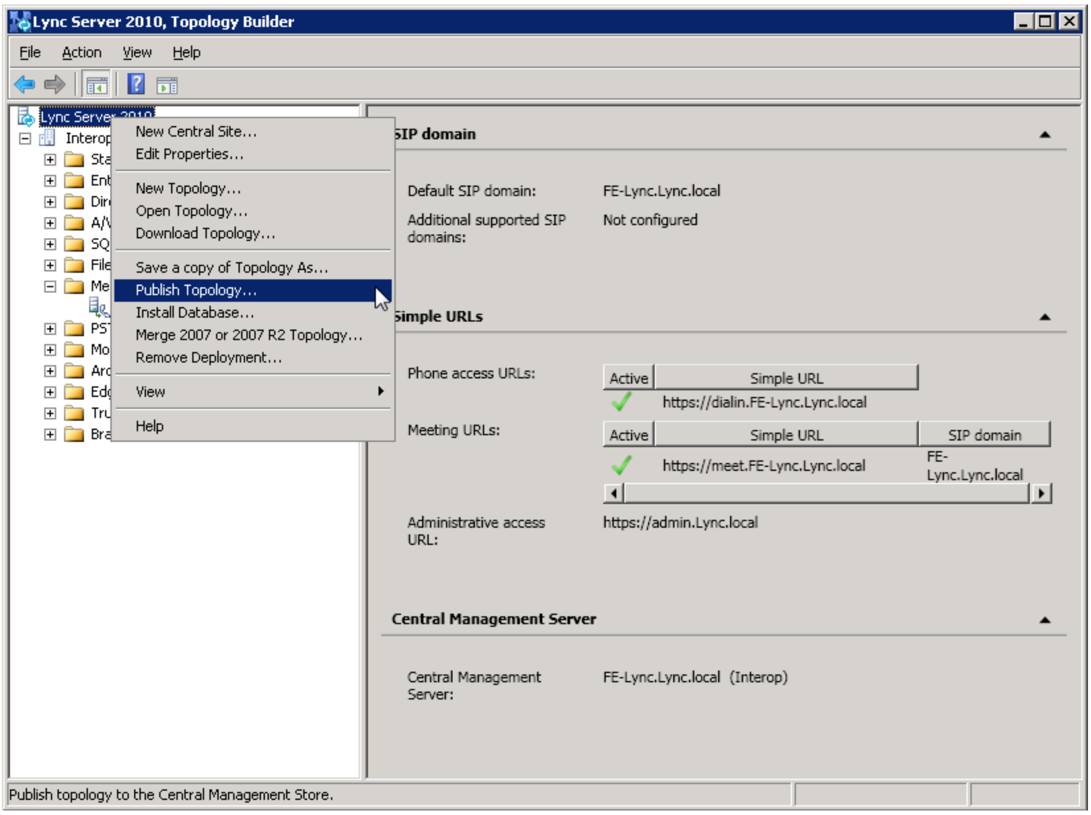
3. Click **OK**.

Figure 3-11: Media Server PSTN Gateway Association Properties



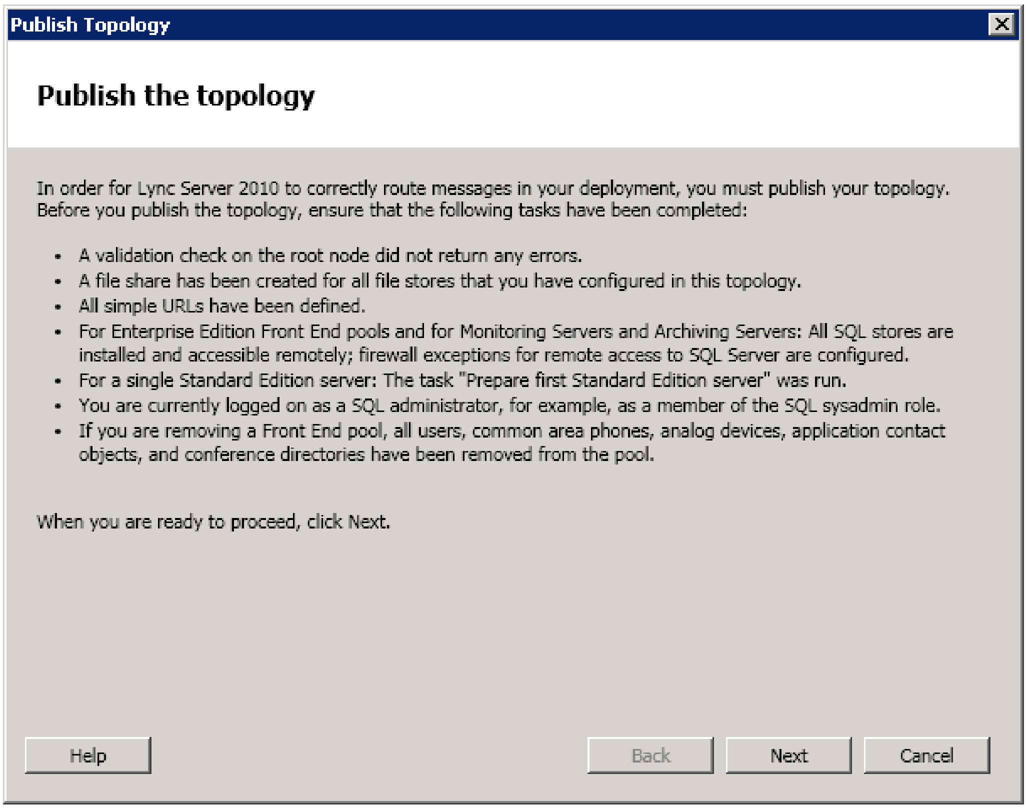
4. In the Lync Server main menu, choose **Action > Publish Topology**.

Figure 3-12: Publishing Topology



The Publish Topology screen is displayed.

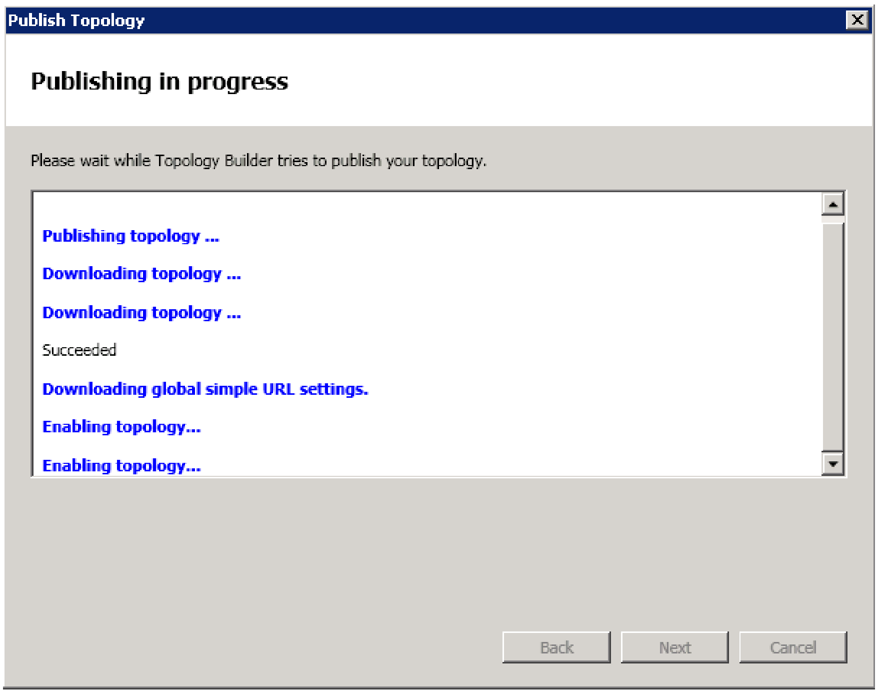
Figure 3-13: Publish Topology Confirmation



5. Click **Next**.

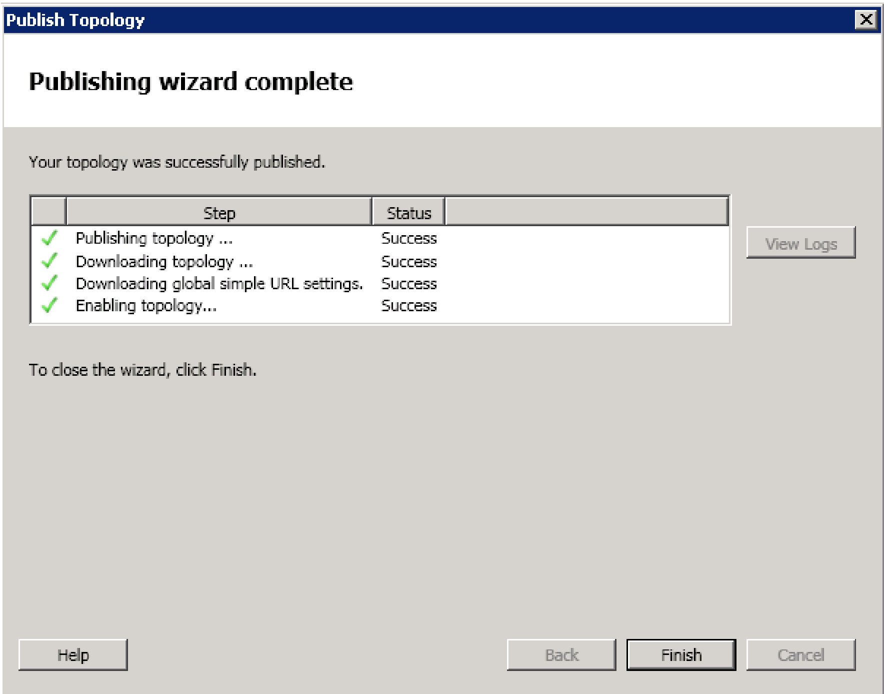
   The Topology Builder attempts to publish your topology.

Figure 3-14: Publish Topology Progress screen



Wait until the publish topology process has ended successfully.

Figure 3-15: Publish Topology Successfully Completed



6. Click **Finish**.

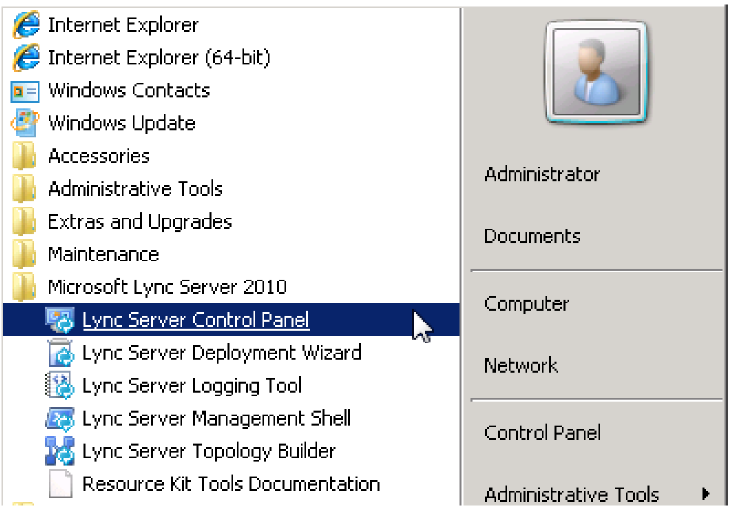
3.3 Configuring the 'Route' on the Lync Server 2010

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:**

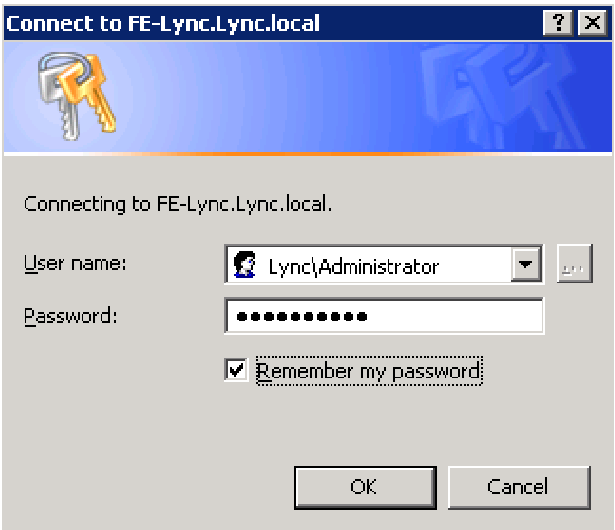
1. Open the Communication Server Control Panel (CSCP), click **Start**, select **All Programs**, and select **Lync Server Control Panel**.

Figure 3-16: Opening the Lync Server Control Panel



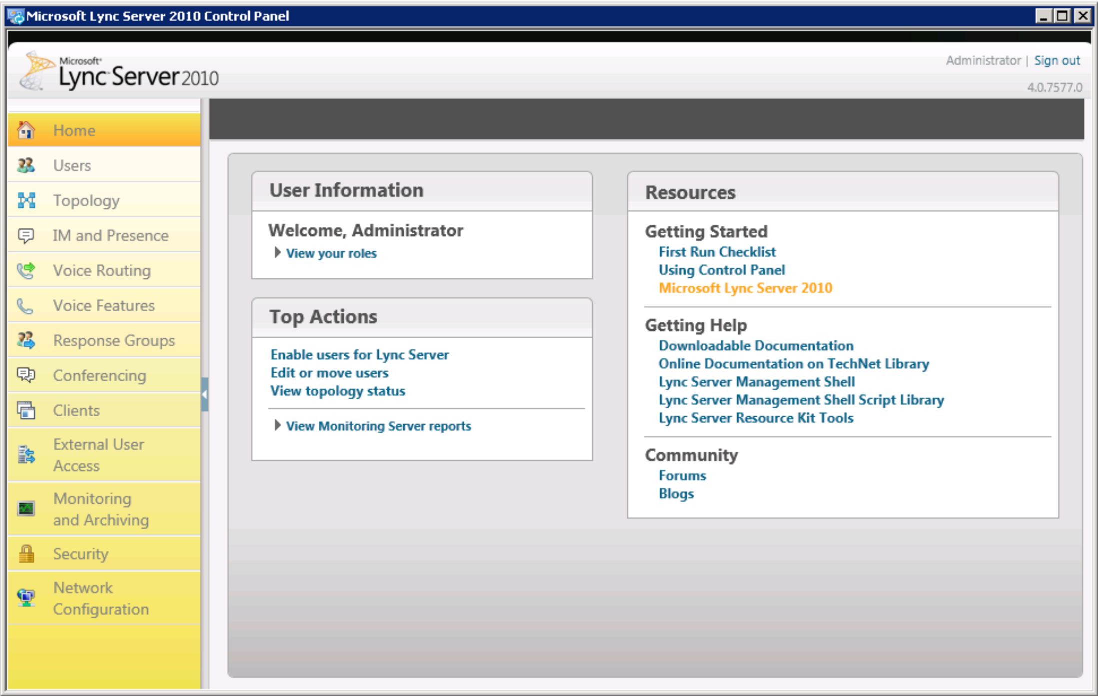
2. You are prompted for credentials; enter your domain username and password.

Figure 3-17: Lync Server Credentials



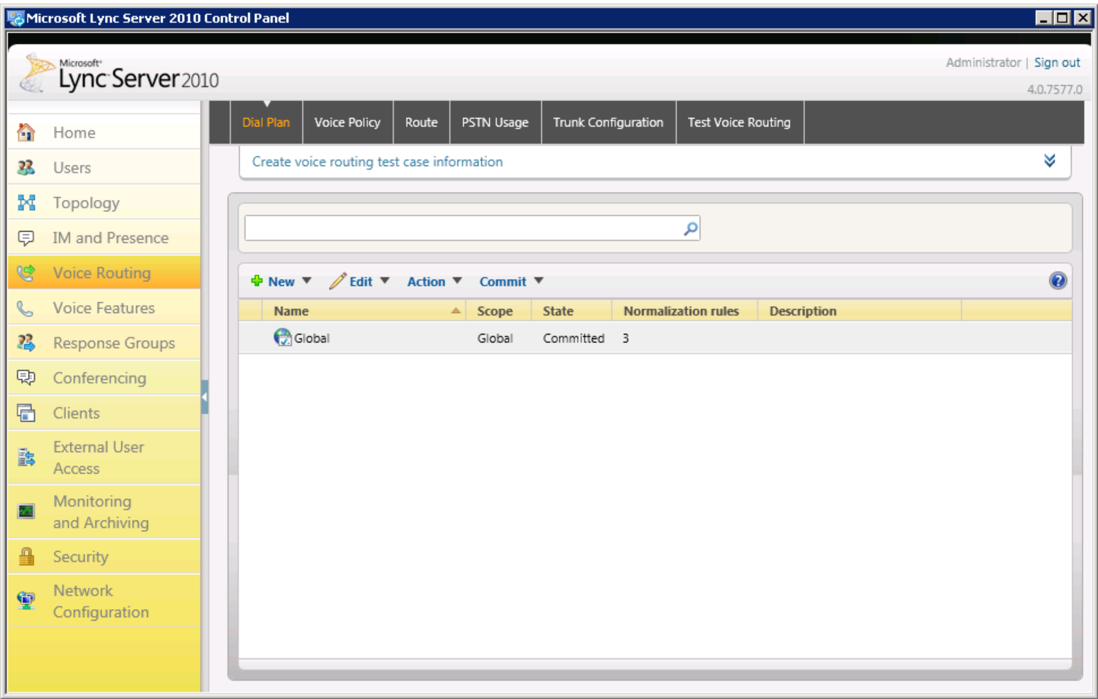
The CSCP Home page is displayed.

Figure 3-18: CSCP Home page



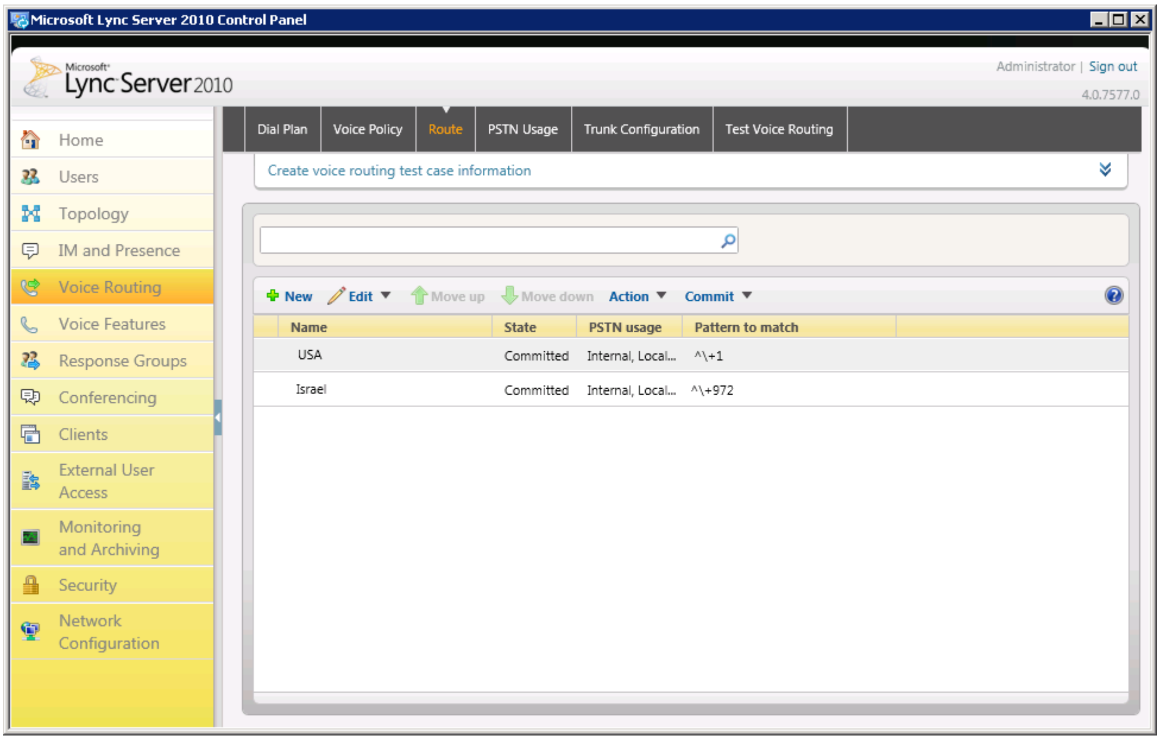
3. In the Navigation pane, select the **'Voice Routing'** option.

Figure 3-19: Voice Routing Option



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

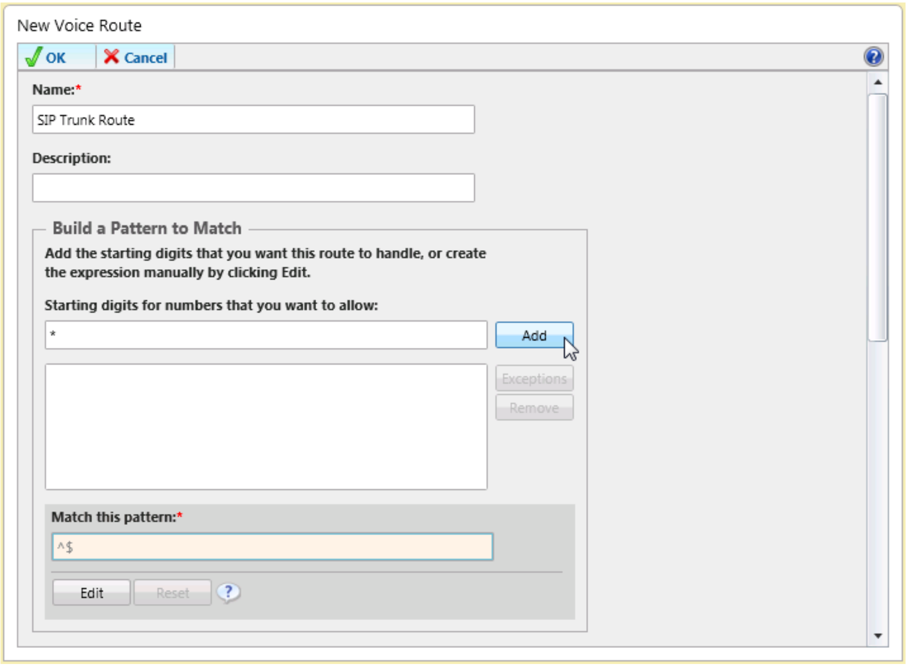
Figure 3-20: Route Option



5. In the content area toolbar, click 

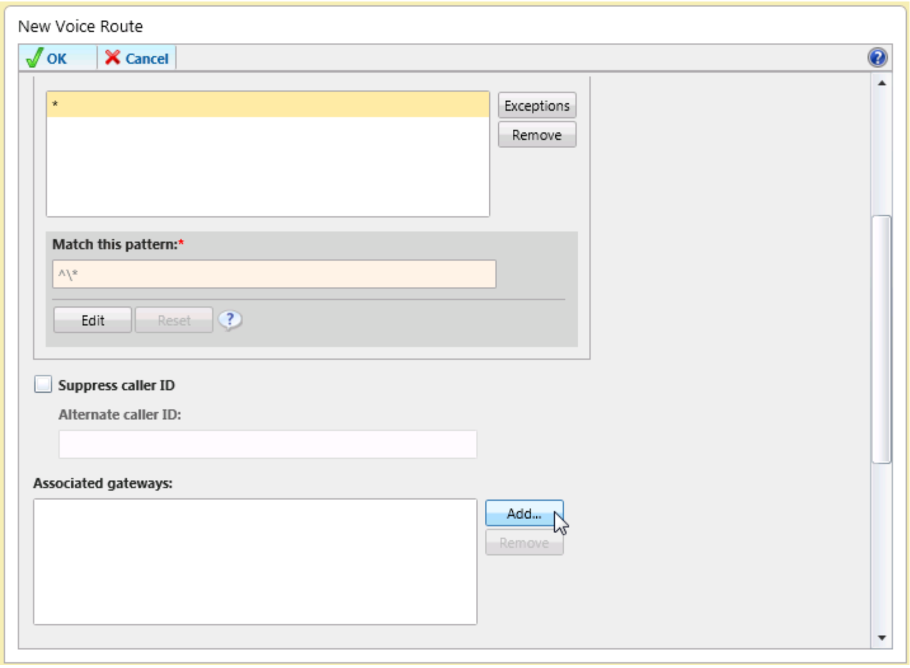
6. In the New Voice Route, fill in a Name for this route (i.e. SIP Trunk Route).
7. In the Build a Pattern to Match add the starting digits you wish this route to handle. In this example, the pattern to match is '*', which means "to match all numbers".
8. Click **Add**.

Figure 3-21: Adding New Voice Route



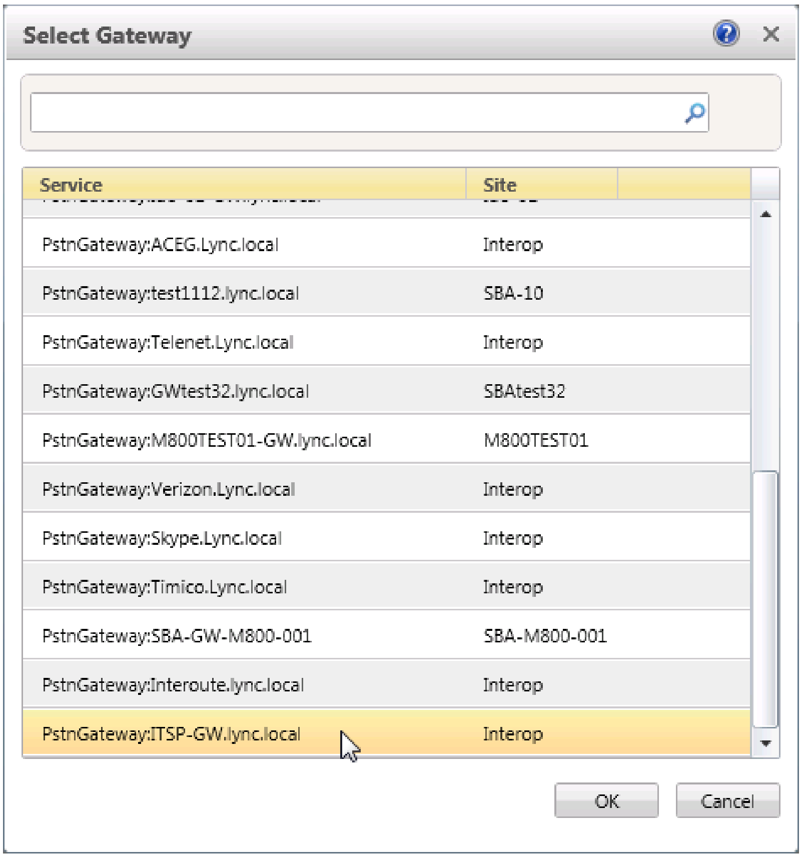
9. Associate the route with the E-SBC IP/PSTN gateway you created above; scroll down to the Associated Gateways pane and click **Add**.

Figure 3-22: Adding New E-SBC Gateway



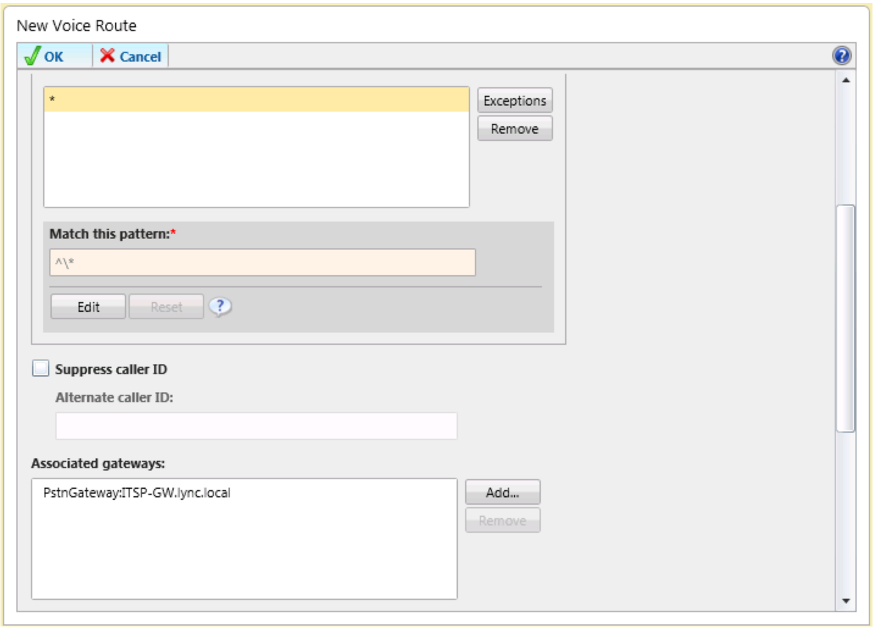
A list of all the deployed Gateways is displayed.

Figure 3-23: List of Deployed Gateways



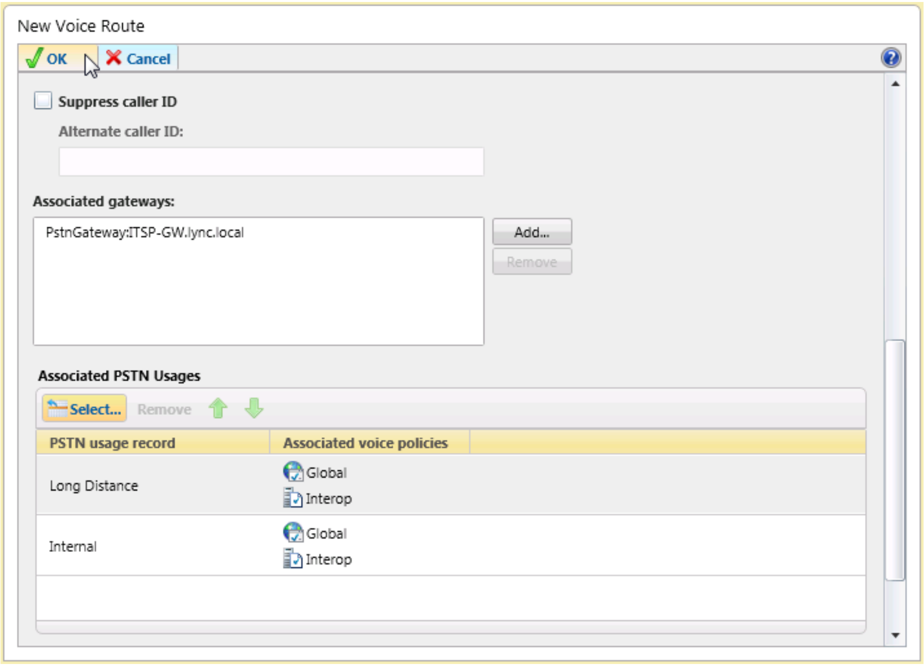
10. Select the E-SBC Gateway you created above and click **OK**.

Figure 3-24: Selected the E-SBC Gateway



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

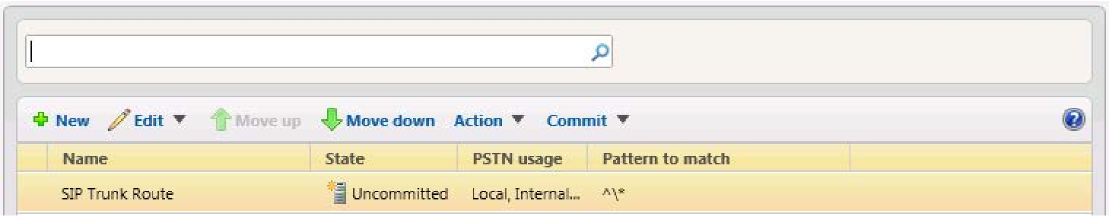
Figure 3-25: Associating PSTN Usage to E-SBC Gateway



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

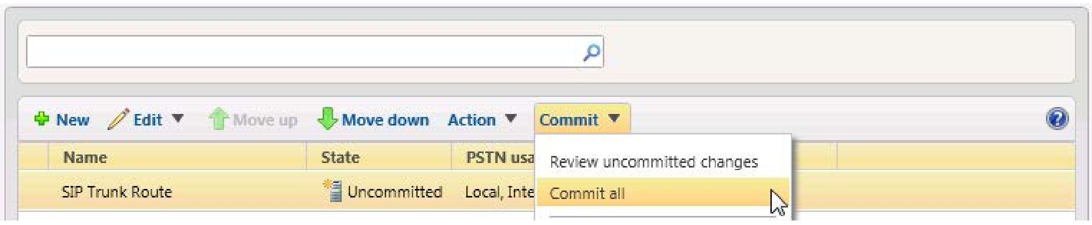
The New Voice Route (Uncommitted) is displayed.

Figure 3-26: Confirmation of New Voice Route



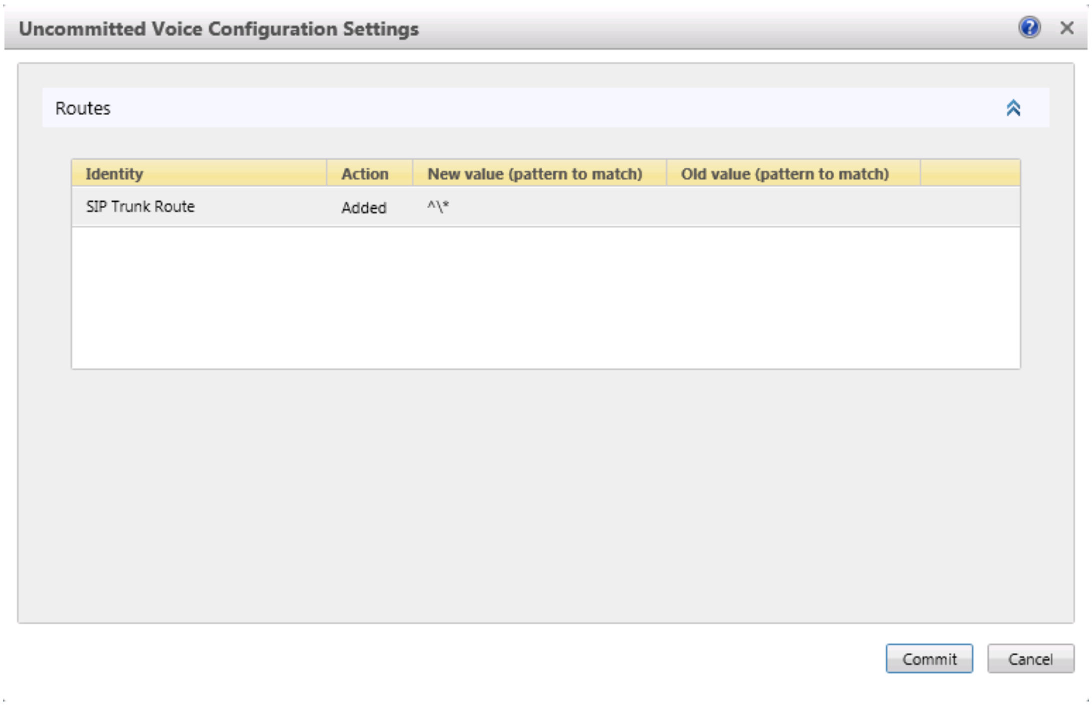
13. In the Content area Toolbar, click on the arrow adjacent to the **Commit** button; a drop-down menu is displayed; select the **Commit All** option.

Figure 3-27: Committing Voice Routes



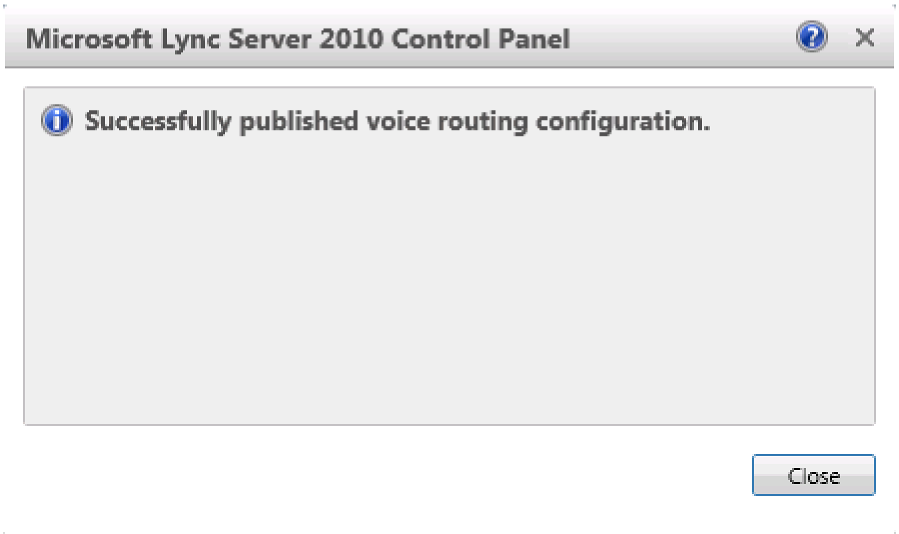
14. In the Uncommitted Voice Configuration Settings window, click **Commit**.

Figure 3-28: Uncommitted Voice Configuration Settings



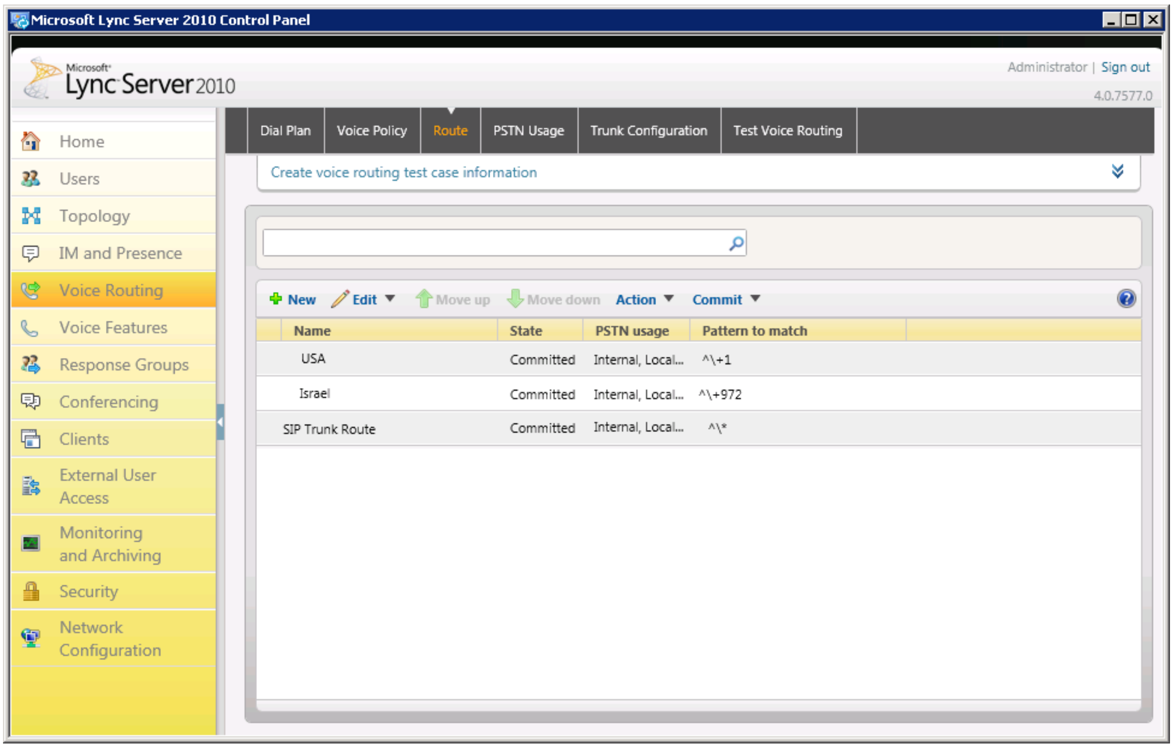
15. A message is displayed, confirming a successful voice routing configuration; in the **Microsoft Lync Server 2010 Control Panel** prompt, click **Close**.

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



Reader's Notes

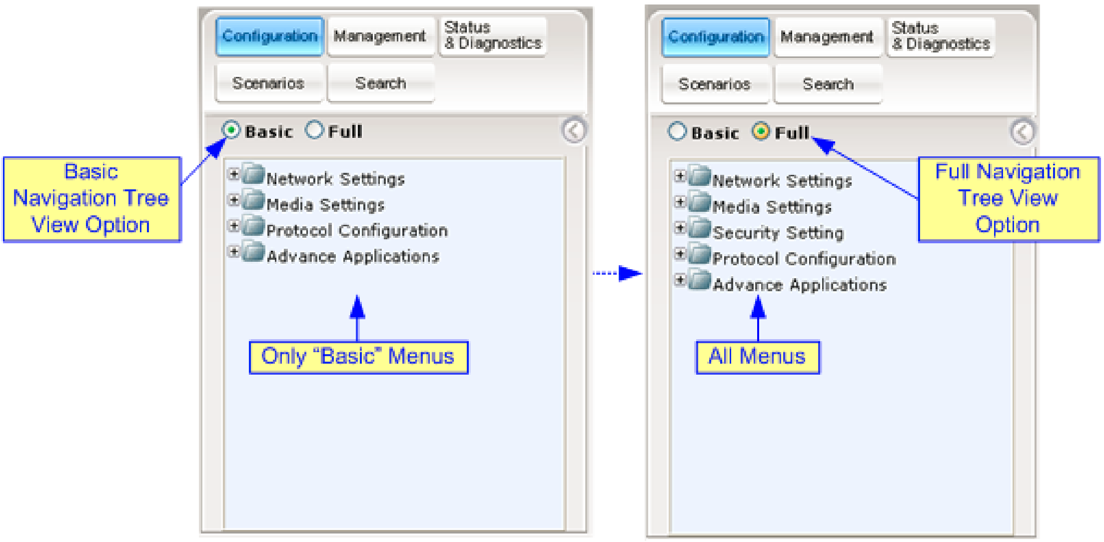
4 Configuring E-SBC Device

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

- **Step 1:** Configure IP Addresses. See Section 4.1 on page 37.
- **Step 2:** Configure Port Forwarding. See Section 4.2 on page 40.
- **Step 3:** Enable SIP IP2IP Application Mode. See Section 4.3 on page 42.
- **Step 4:** Configure Secure Real-Time Transport Protocol (SRTP). See Section 4.4 on page 43.
- **Step 5:** Configure IP Media. See Section 4.5 on page 44.
- **Step 6:** SIP General Parameters. See Section 4.6 on page 45.
- **Step 7:** DTMF & Dialing. See Section 4.7 on page 47.
- **Step 8:** Coders. See Section 4.8 on page 48.
- **Step 9:** Configure Proxy & Registration. See Section 4.9 on page 49.
- **Step 10:** Configure Proxy Sets table. See Section 4.10 on page 50.
- **Step 11:** Configure Coder Group. See Section 4.11 on page 52.
- **Step 12:** Configure IP Profile. See Section 4.12 on page 54.
- **Step 13:** Configure IP Group Tables. See Section 4.13 on page 56.
- **Step 14:** Routing. See Section. See Section 4.14 on page 58.
- **Step 15:** Manipulation. See Section 4.15 on page 60.
- **Step 16:** Define SIP TLS Connection. See Section 4.16 on page 62.
- **Step 17:** Resetting the Gateway. See Section 4.17 on page 69.

The procedures described in this section are performed using the E-SBC devices' Web-based management tool (i.e., embedded Web server). 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 Timico SIP Trunk. In this case, you must configure a separate WAN interface as described in this step.

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

This step describes how to configure the LAN addresses.

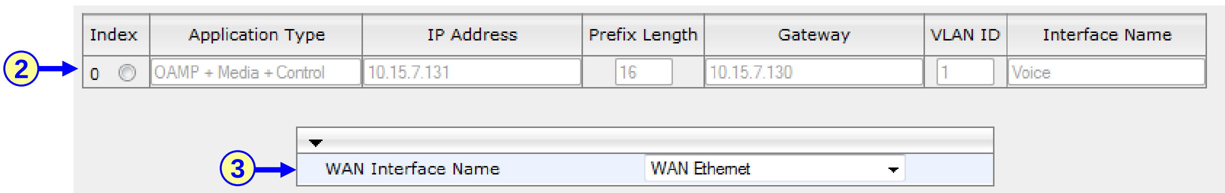
4.1.1.1 Configuring VoIP IP Settings

This 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. Select the 'Index' radio button corresponding to the Application Type "**OAMP + Media + Control** (i.e., VoIP and management interface), and then click **Edit**.

   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). in case of M800 or M1K this IP should be same as you setup in LAN data-routing IP address in case of M3K it should be the corporate router IP.
3. Set the **WAN Interface Name:** '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

This section 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.

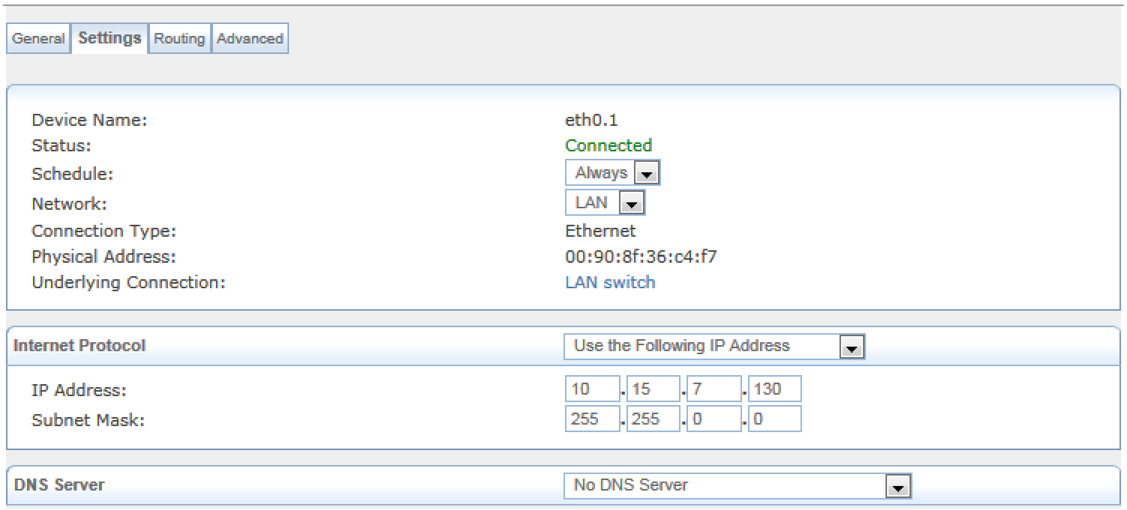
➢ To define the MSBG device's LAN data-routing IP address:

1. Access the MSBG device's Web interface with the IP address that you assigned to the VoIP and Management interface.
2. Access 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

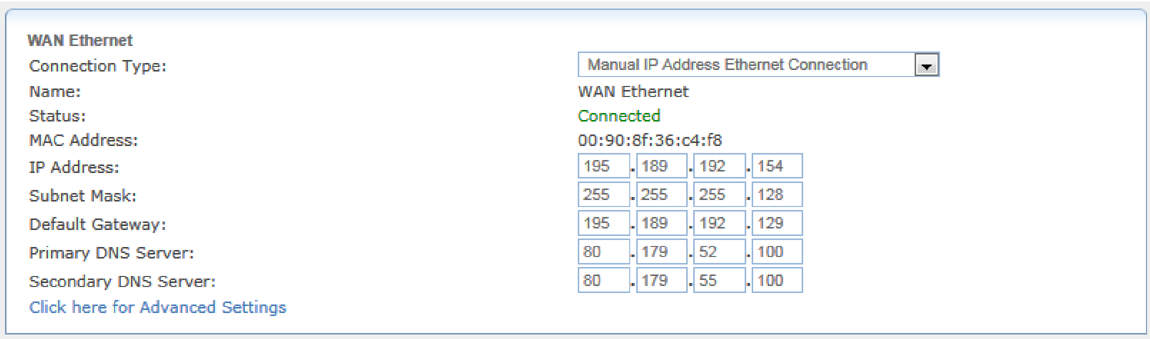
This step describes how to configure the MSBG device IP address used to connect to the WAN.

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 Port Forwarding.

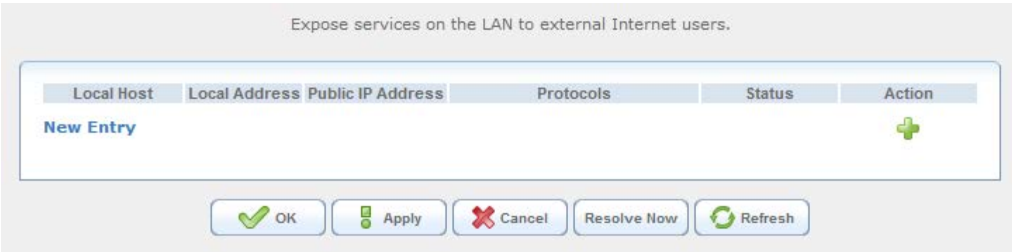
The Port Forwarding item enables you to define the applications that require special handling by the device. This allows you to select the application's protocol or ports (SIP and RTP) and the local IP address of the device (e.g., Gateway's IP: 10.15.7.131) that will be using the service.

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

➢ To configure a port forwarding service:

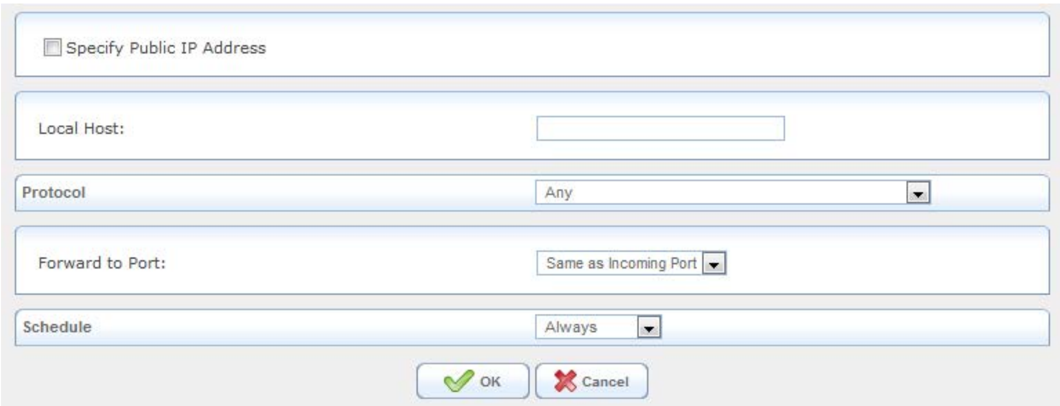
1. Open the 'Settings' page (**Configuration** tab > **Data** menu > **Firewall and ACL** > **Port Forwarding**).

Figure 4-6: Configure Port Forwarding



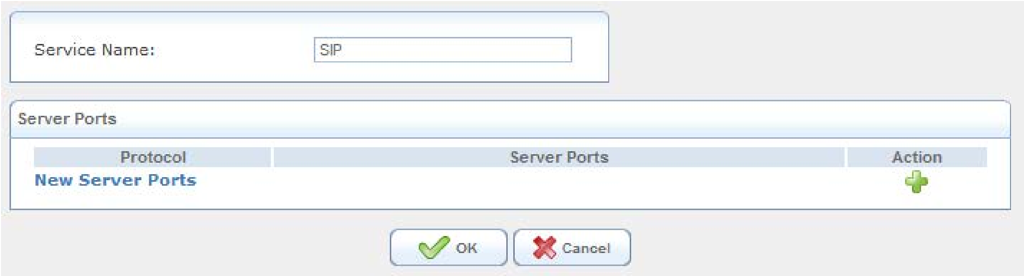
2. Click the 'New Entry' link; the following page appears:

Figure 4-7: Adding Port Forwarding Rule



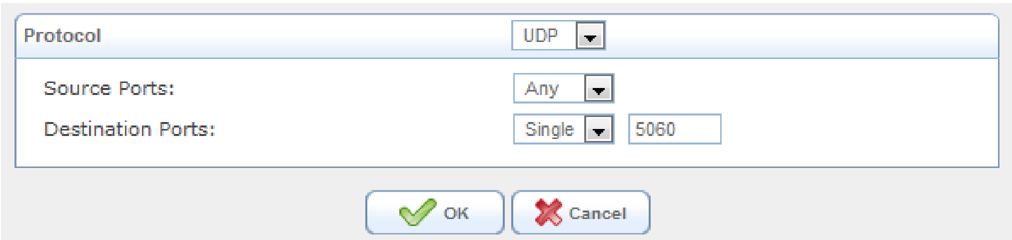
3. In the 'Local Host' field, enter the host name or IP address (e.g., 10.15.7.131).
4. From the 'Protocol' drop-down list, select or specify the type of protocol. Add a new one using the 'User Defined' option, and then add a new Service, representing the protocol.

Figure 4-8: Adding a Service Protocol



5. Write the new Service Name, (e.g., SIP , RTP)
6. Click the **New Server Ports** link.

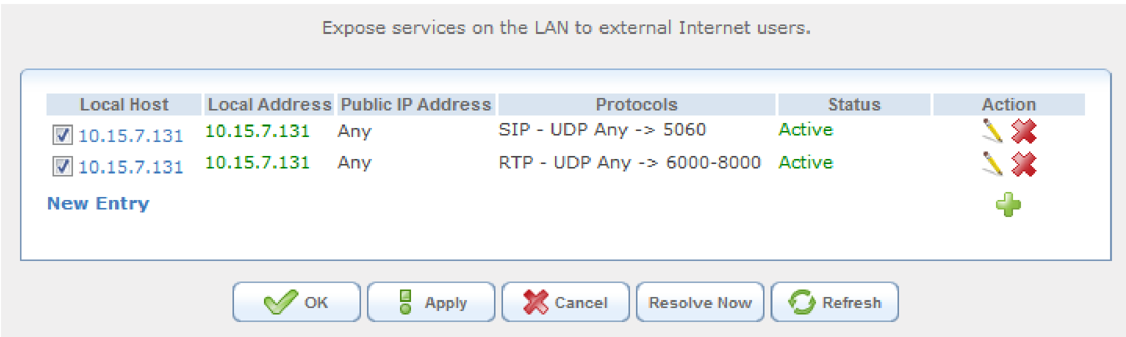
Figure 4-9: Defining Service Server Ports



7. Set the Protocol type (e.g., UDP).
8. Set the Destination Port Range (e.g., 5060 for SIP and 6000-8000 for RTP).
9. Click **OK** to save your changes.

   The main Port Forwarding page displays a summary of the rules that you added:

Figure 4-10: Display Port Forwarding Rules



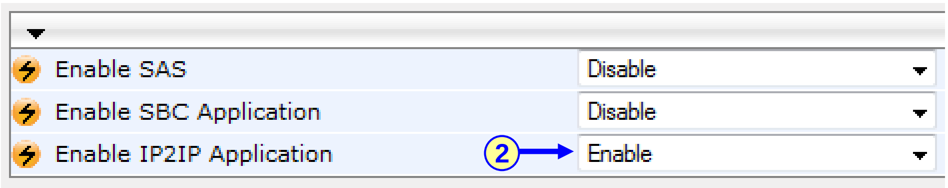
4.3 Step 3: Enabling Application Mode

This step describes how to enable the IP2IP application mode.

➢ To enable the application mode:

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

Figure 4-6: Applications Enabling



2. Enable **IP2IP Application**.

Notes:

1. To enable the IP2IP capabilities on the AudioCodes gateway, your gateway must be loaded with the feature key that includes the **IP2IP** feature.
2. The E-SBC device must be running SIP version 6.2 or later.
3. Reset with BURN to FLASH is required.

4.4 Step 4: Configuring 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:

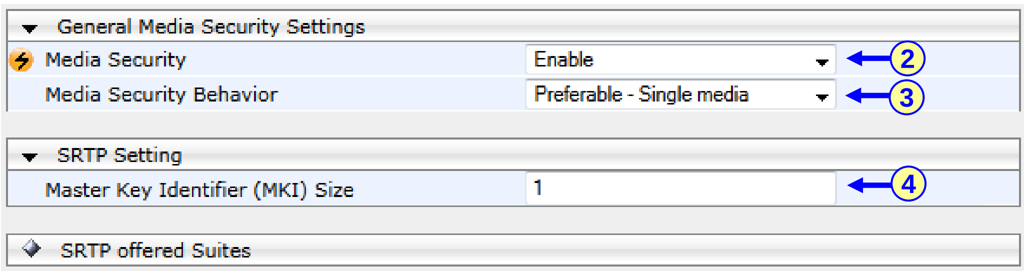
- **Required:** SRTP should be attempted, but do not use encryption if negotiation for SRTP is unsuccessful.
- **Optional:** Attempt to negotiate the use of SRTP to secure media packets. Use RTP if SRTP cannot be negotiated.
- **Not used:** Send media packets using RTP.

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 4-7: Media Security Page



2. Set the Media Security to **Enable**.
3. Set the Media Security Behavior:
   - 'Mandatory' if Mediation Server is configured to **SRTP Required**.
   - 'Preferable-Single media' if Mediation Server is configured to **SRTP Optional**.
4. Set the 'Master Key Identifier (MKI) Size' to **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 4.9 on page 49).

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).

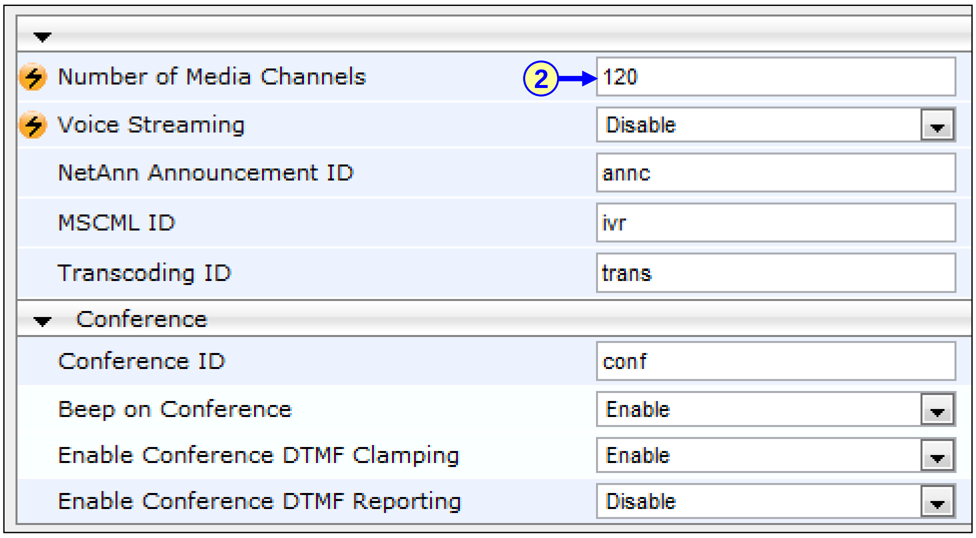
The maximum number of media channels available on the Mediant 3000 E-SBC device is 2016 (i.e., up to 1008 IP-to-IP calls).

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. Set 'Number of Media Channels' to **120**.

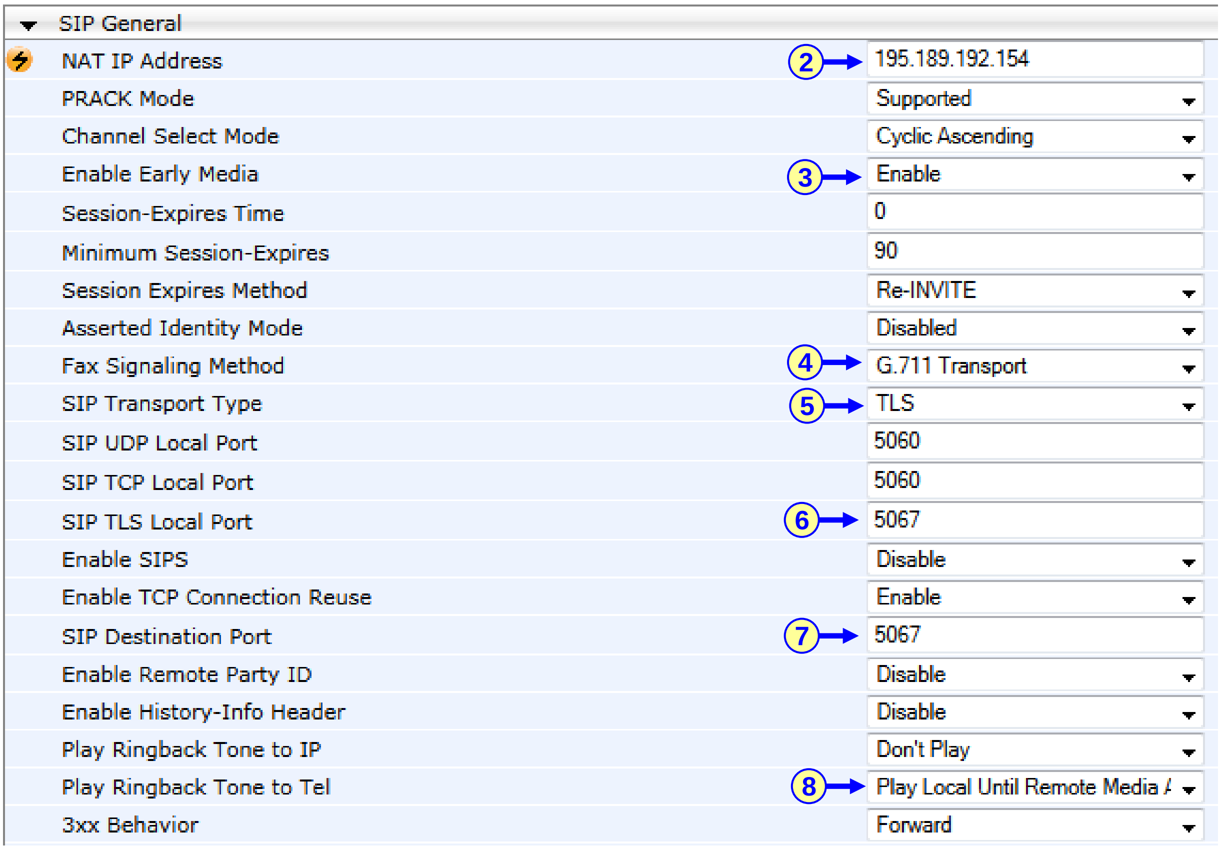
4.6 Step 6: Configuring SIP General Parameters

This step 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

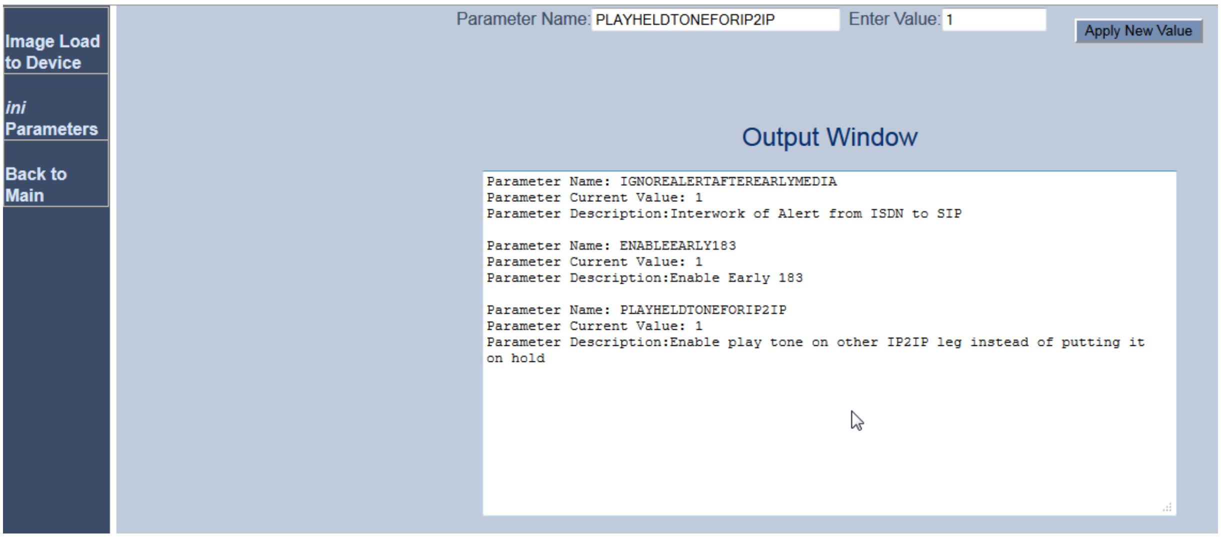


2. Set 'NAT IP Address', with the Global (public) IP address of the E-SBC device.
3. Set 'Enable Early Media' to **Enable**.
4. Set 'Fax Signaling Method' to **G.711 Transport**
5. Set 'SIP Transport Type' to **TLS**.
6. Set 'SIP TLS Local Port' to **5067** (Lync server port).
7. Set 'SIP Destination Port' to **5067** (Lync server port).
8. Set 'Play Ringback Tone' to Tel to **Play Local Until Remote Media Arrives**.

Figure 4-11: General Parameters (Cont.)

9. Set 'Forking Handling Mode' to **Sequential handling**.
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://10.15.7.131/AdminPage).
11. On the left pane, click ***ini*** Parameters.

Figure 4-10: INI file Output Window



12. In the 'Parameter Name' field, enter the following parameters:
    - **IGNOREALERTAFTEREARLYMEDIA**; In the 'Enter Value' field, enter **1**.
    - **ENABLEEARLY183**; In the Enter Value field, enter **1**.
    - **PLAYHELDTONEFORIP2IP**; In the Enter Value field, enter **1**
13. Click Apply New Value.

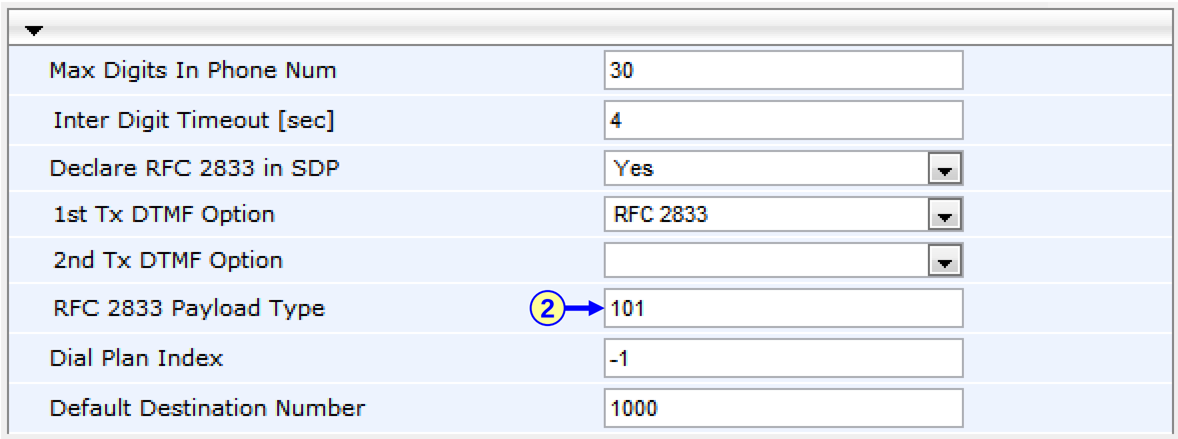
4.7 Step 7: Configuring DTMF and Dialing

This step describes how to configure the DTMF and Dialing settings.

➢ **To configure DTMF and Dialing:**

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

Figure 4-11: DTMF & Dialing



2. Set RFC 2833 Payload Type to **101**.

4.8 Step 8: Configuring Coders

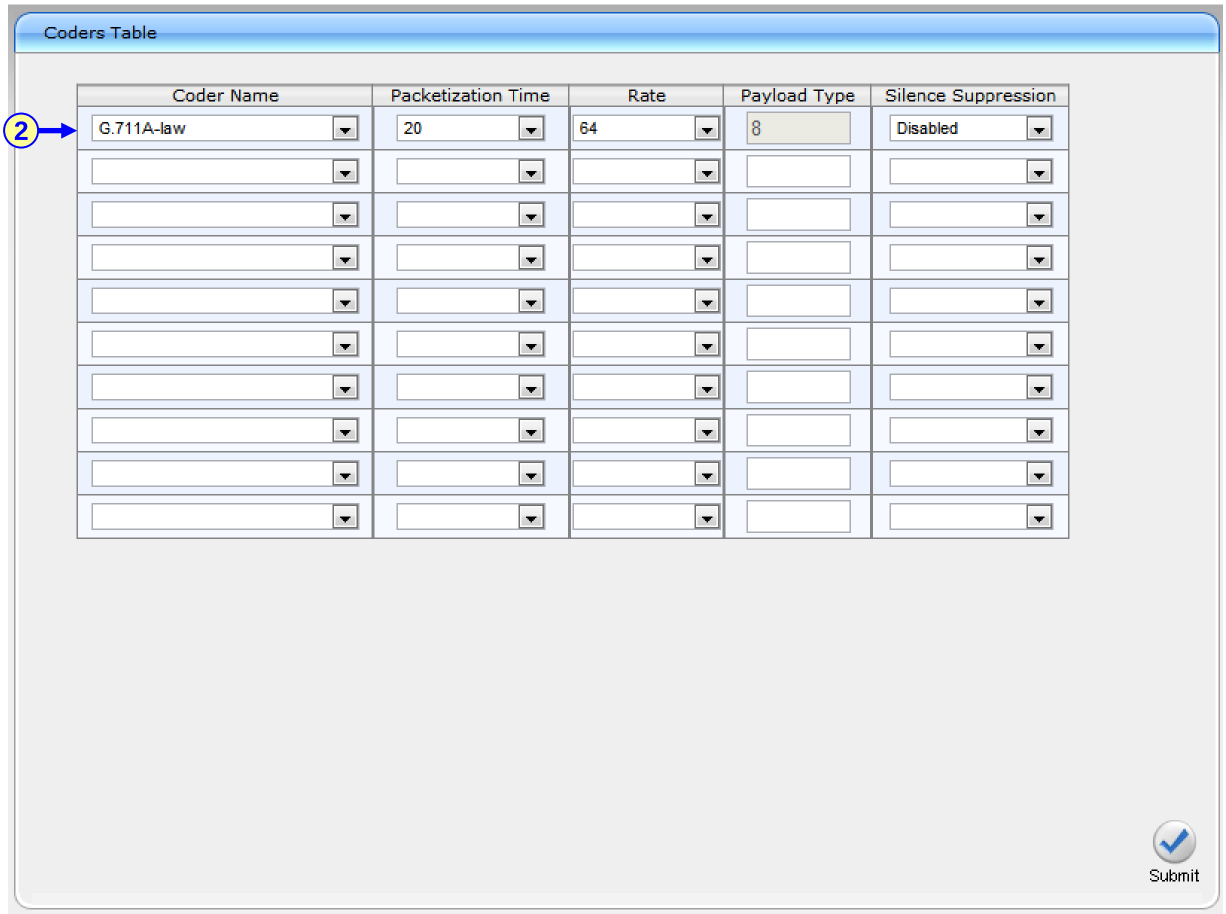
This step describes how to configure the SIP coders. This is the general coder table in this case scenario we are using coder group tables, see section 4.11.

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 the required coder.
3. Click **Submit**.

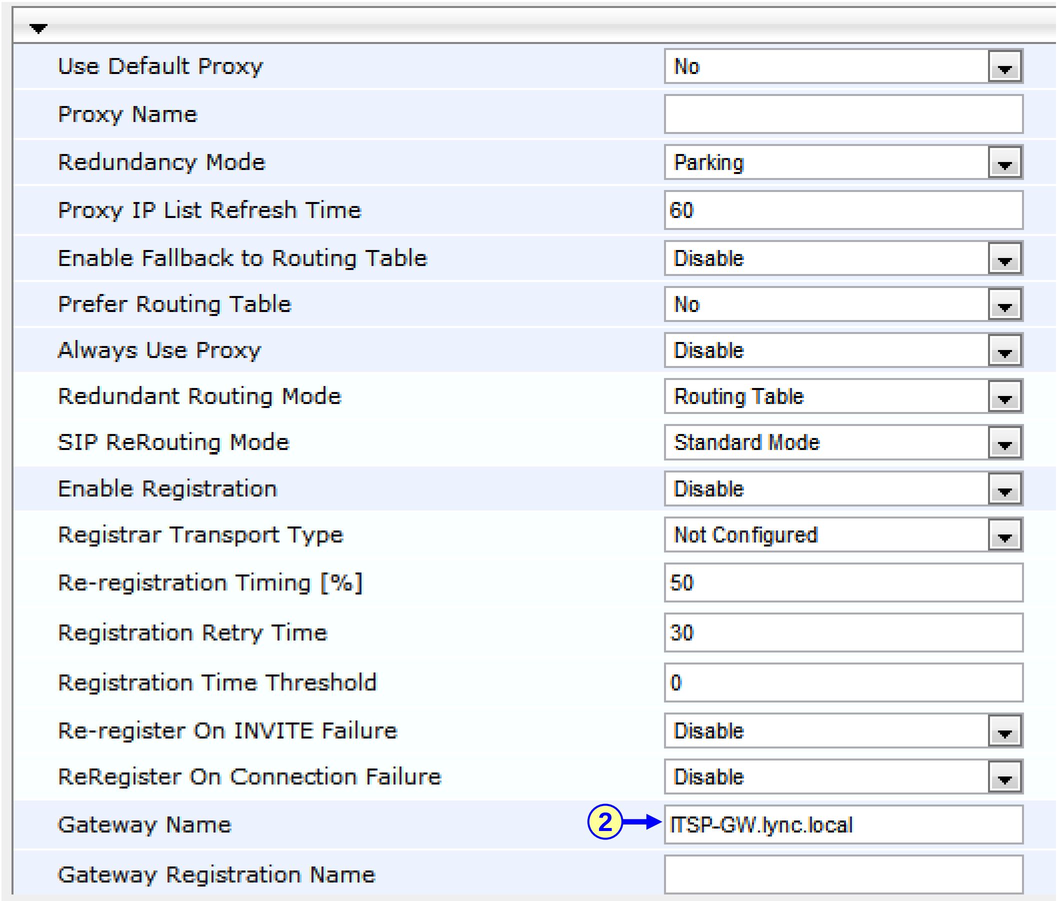
4.9 Step 9: Configuring Proxy and Registration

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

➢ **To configure Proxy & Registration:**

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

Figure 4-13: Proxy & Registration



2. Set 'Gateway Name' to **Gateway FQDN Name** (e.g., 'ITSP-GW.lync.local') (note that you configure this name in Section 4.16.3 on page 63).

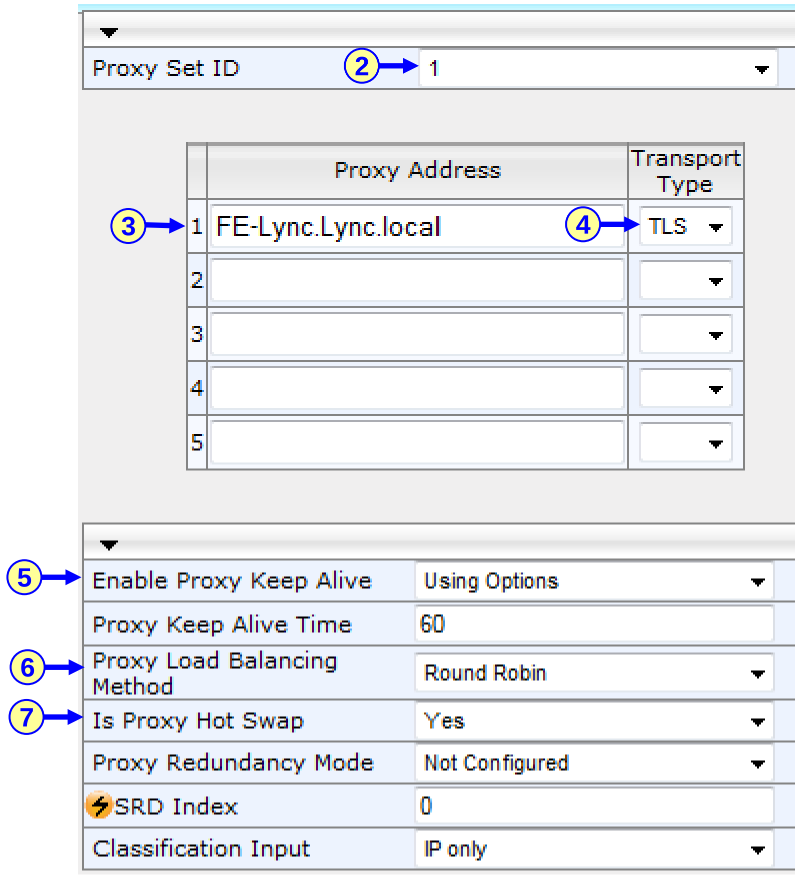
4.10 Step 10: Configuring Proxy Sets Tables

This step describes how to configure the proxy set tables. You need to configure two proxy sets, one for the Timico 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

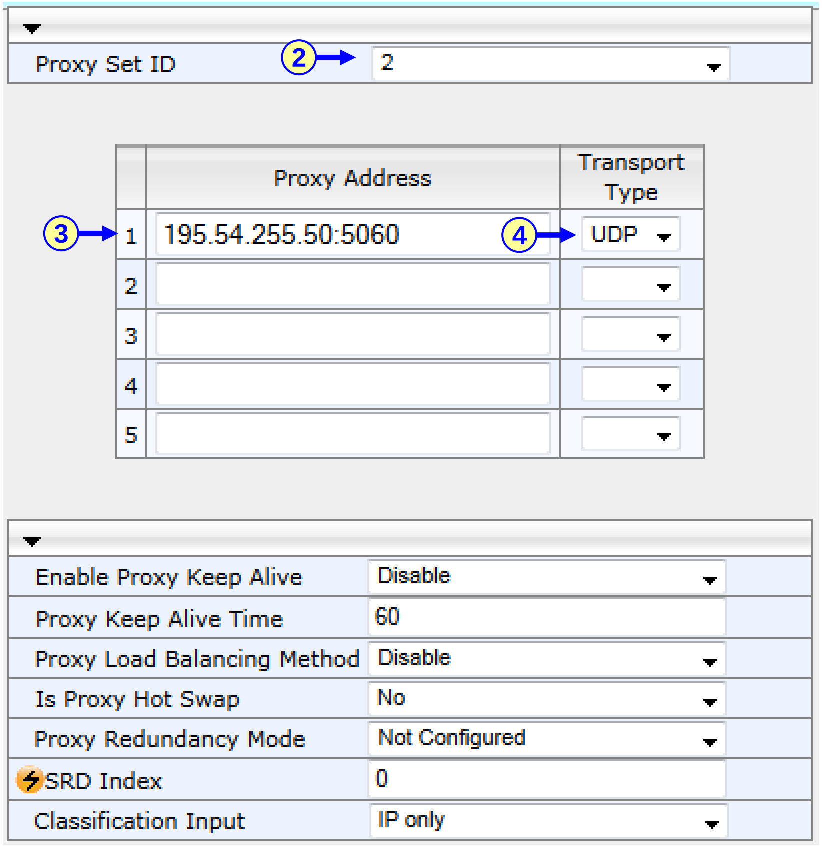


2. Set Proxy Set ID to **1**.
3. Configure Microsoft Lync Server SIP Trunking IP-Address or FQDN and Destination Port (e.g., FE-Lync.Lync.local).
4. Set 'Transport Type' to **TLS**.
5. Set 'Enable Proxy Keep Alive' to **Using Options**.
6. Set 'Proxy Load Balancing Method' to **Round Robin**.
7. Set 'Is Proxy Hot Swap' to **Yes**.

➢ To configure Proxy Sets Table 2 for Timico 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. Set 'Proxy Set ID' to **2**.
3. Configure Timico IP-Address or FQDN and Destination Port (e.g., 195.54.255.50:5060').
4. Set 'Transport Type' to **UDP**.

4.11 Step 11: Configuring Coder Group

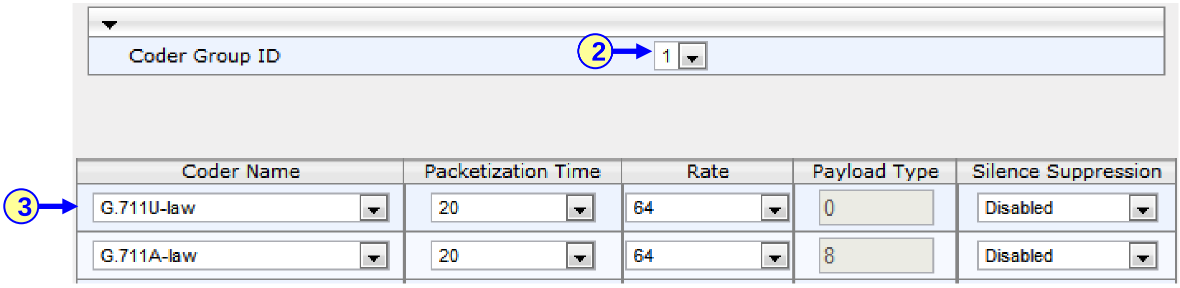
This step describes how to configure the Coder Groups. Microsoft Lync supports G.711 coders, while the network connection to Timico may restrict you to work with lower bandwidth coders, such as G.729.

The 'Coder Group Settings' allows 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 4.12 on page 54).

➢ 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

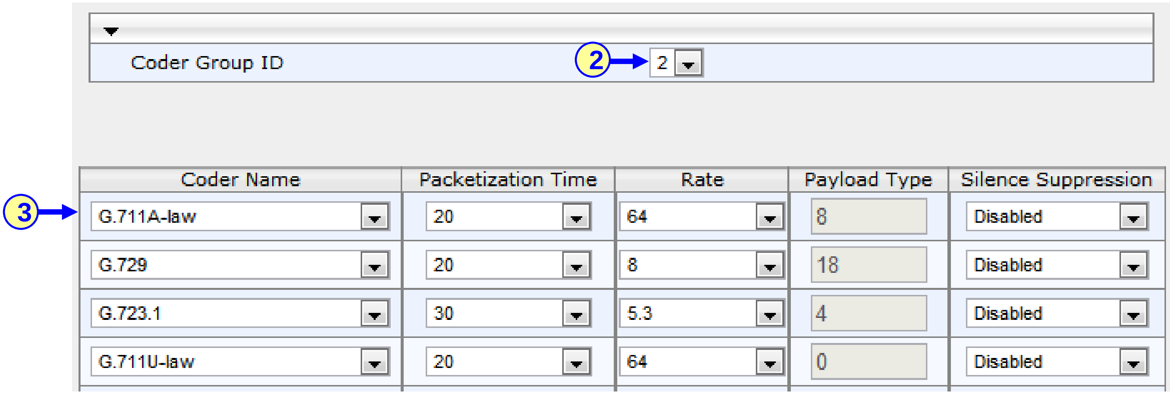


2. Select **Coder Group ID 1**.
3. Set Coder Name **G.711U-law** and **G.711A-law**.
4. Click **Submit**.

➢ To configure Coders Group for Timico 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



2. Select **Coder Group ID 2**.
3. Set Coder Name, **G.711A-law**, **G.729, G.723.1, and G.711U-law**.
4. Click **Submit**.

4.12 Step 12: Configuring IP Profile

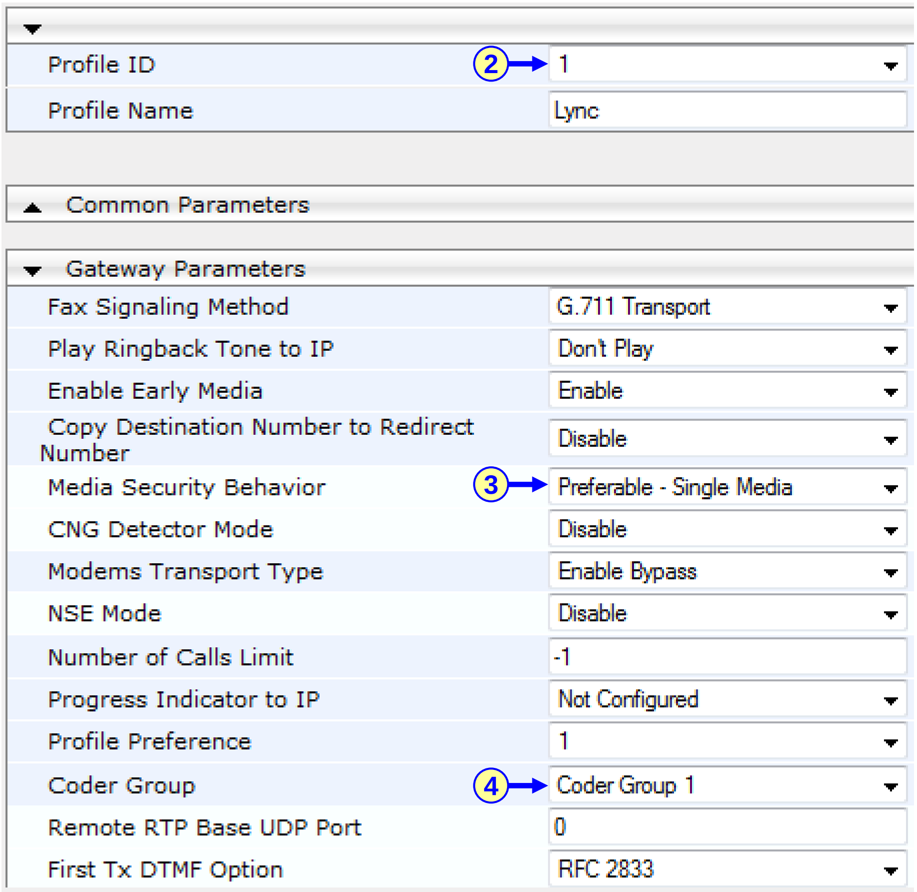
This step describes how to configure the IP Profile. In this configuration, the IP Profile is used to configure the SRTP/TLS mode and the Coder Group (see Section 4.11 on page 52).

You must configure Microsoft Lync to work in secure mode (SRTP/TLS); while, the Timico 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

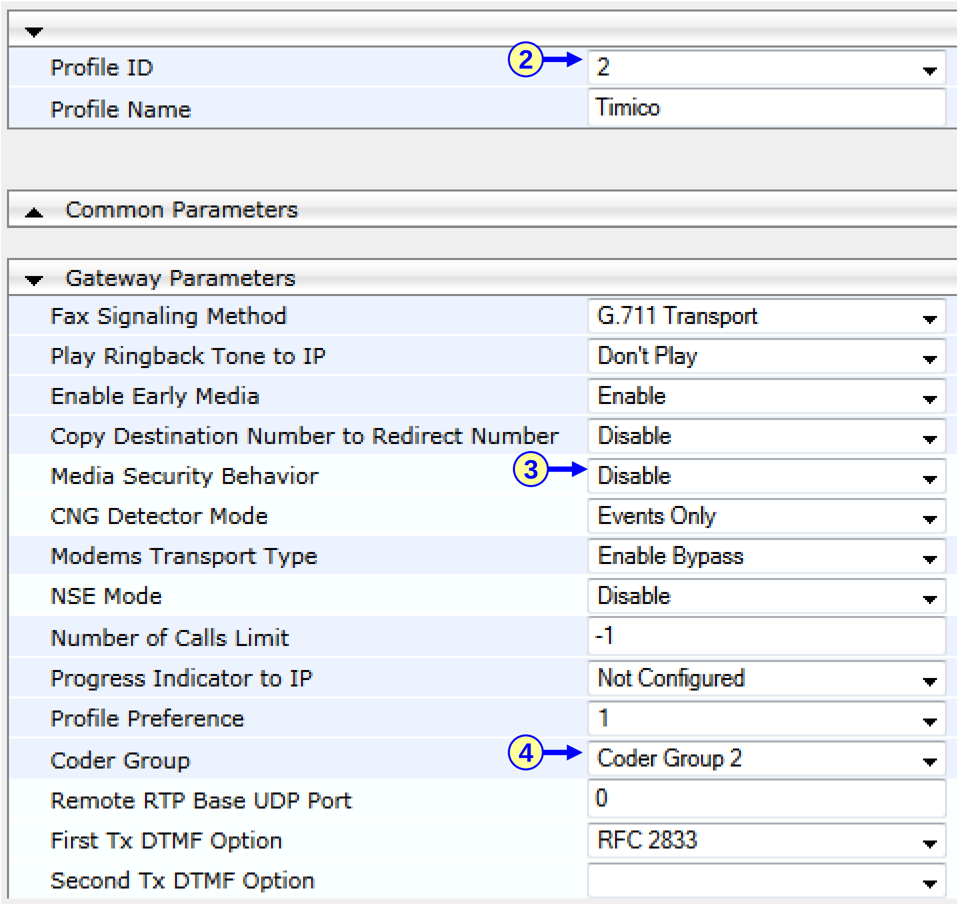


2. Select **Profile ID 1**.
3. Set Media Security Behavior to **Preferable – Single Media**.
4. Set Coder Group to **Coder Group 1**.
5. Click **Submit**.

➢ To configure IP Profile for Timico 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



2. Select Profile ID **2**.
3. Set Media Security Behavior to **Disable**.
4. Set Coder Group to **Coder Group 2**.
5. Click **Submit**.

4.13 Step 13: Configuring IP Group Tables

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:

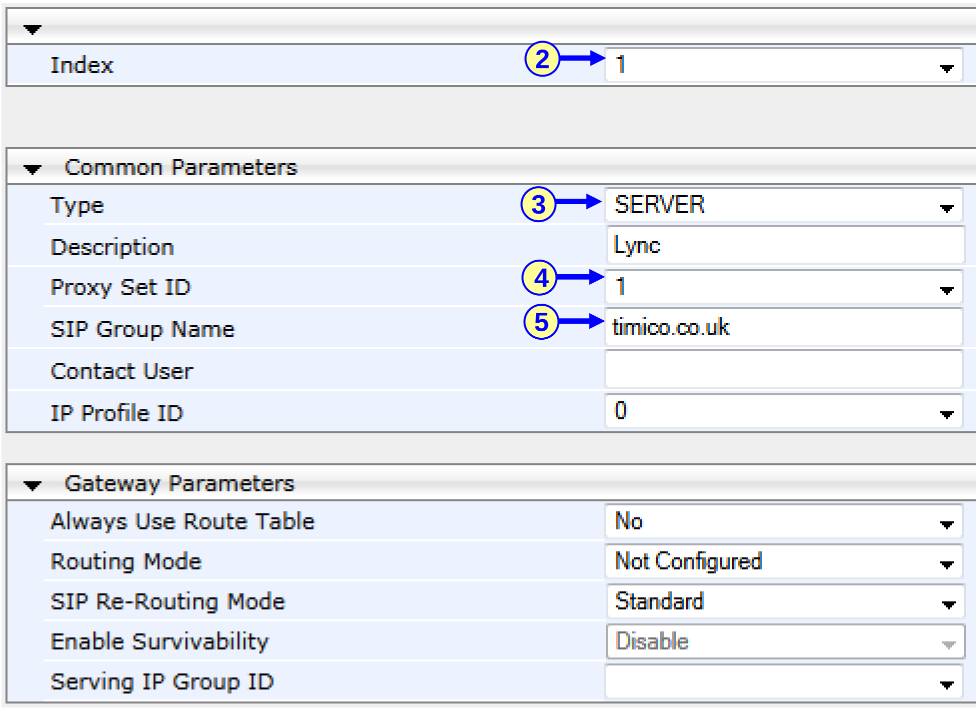
1. Lync Server 2010 – Mediation Server
2. Timico SIP Trunk

These IP groups are later used by the IP2IP 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

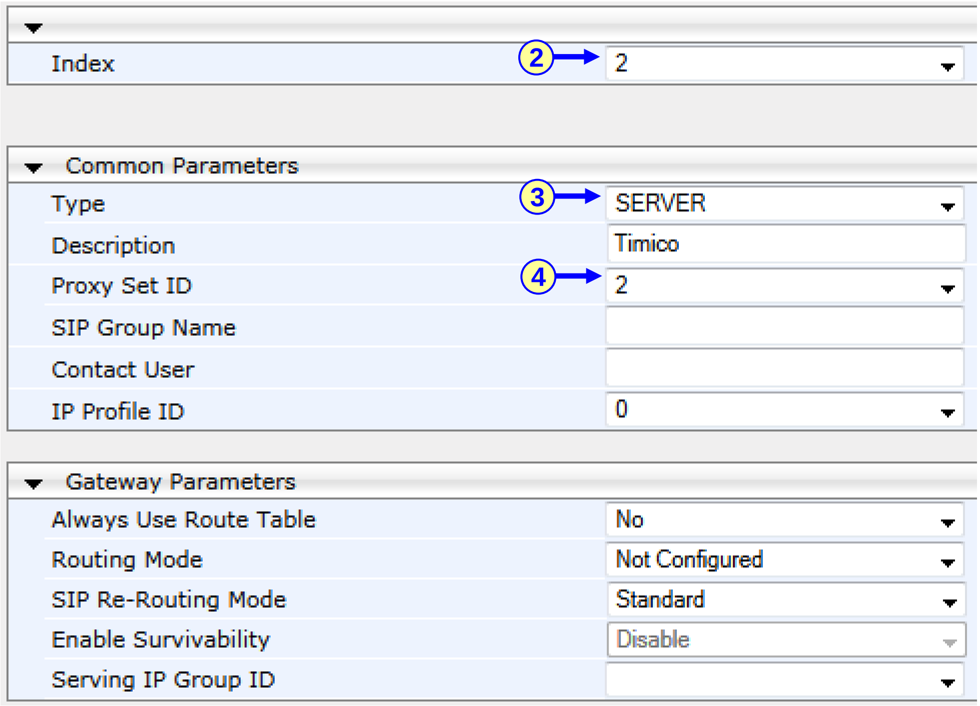


2. Set Index to **1**.
3. Set 'Type' to **SERVER**.
4. Set Proxy Set ID to **1**.
5. Set ‘SIP Group Name’ to **timico.co.uk**; this host name used in the INVITE messages to Timico IP Group.

➢ 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. Set Index to **2**.
3. Set 'Type' to **SERVER**.
4. Set 'Proxy Set ID' to **2**.

4.14 Step 14: Configuring Routing

This step 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. Those tables provides 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.

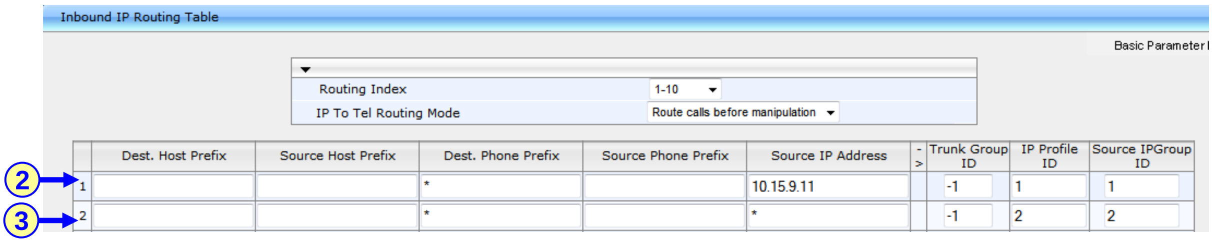
It is crucial that you adhere to the following guidelines when configuring your IP-to-IP routing rules:

- 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 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 4-22: IP to Trunk Group Routing Table

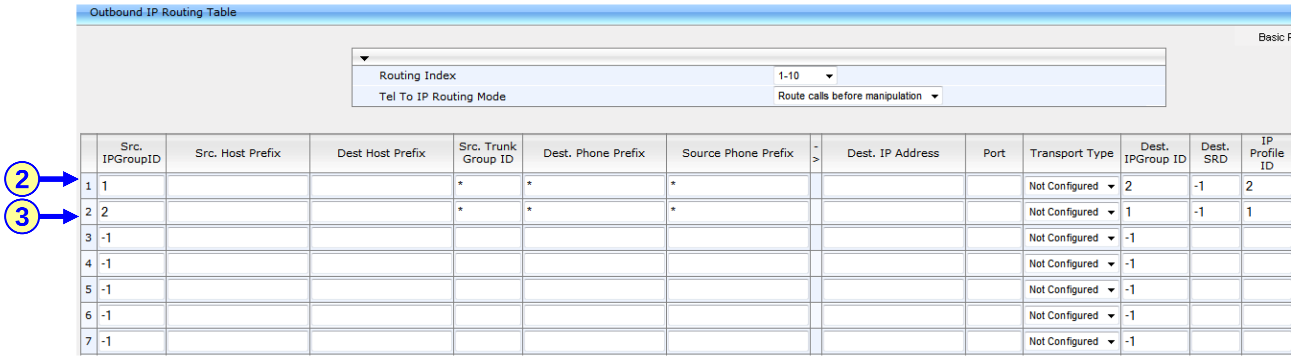


2. Calls arriving from the Microsoft Lync server are sent to the 'Tel to IP Routing Table' (-1) with 'IP Profile ID' = **1** and marked as 'Source IPGroup ID' = **1**.
3. Calls arriving from Timico are sent to the 'Tel to IP Routing Table' (-1) with 'IP Profile ID' = **2** and marked as 'Source IPGroup 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 4-23: Tel to IP Routing Table



	Src. IPGroupID	Src. Host Prefix	Dest Host Prefix	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Prefix	->	Dest. IP Address	Port	Transport Type	Dest. IPGroup ID	Dest. SRD	IP Profile ID
1	1			*	*	*				Not Configured	2	-1	2
2	2			*	*	*				Not Configured	1	-1	1
3	-1									Not Configured	-1		
4	-1									Not Configured	-1		
5	-1									Not Configured	-1		
6	-1									Not Configured	-1		
7	-1									Not Configured	-1		

2. Calls from Source IPGroup ID **1** (e.g., from Microsoft Lync) will be send to 'Dest. IPGroup ID **2** (e.g., To Timico).
3. Calls from Source IPGroup ID **2** (e.g., from Timico) will be send to 'Dest. IPGroup ID **1** (e.g., To Lync).

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

4.15 Step 15: Configuring Manipulation

This step describes how to configure the manipulation tables.

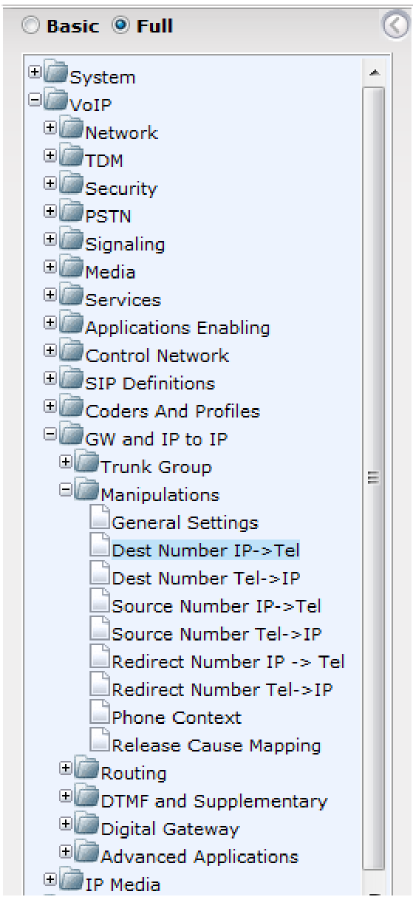
The Manipulation Tables submenu allows you to configure number manipulation and mapping of NPI/TON to SIP messages.

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

➢ To configure Manipulation Tables:

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

Figure 4-24: 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** submenu > **Dest Number Tel** > **IP**).

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

Destination Phone Number Manipulation Table for Tel -> IP Calls

Basic Parameter List

3 Add

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 Digits to Leave
1	-1	2	+	*	0	0			255
2	-1	2	1	*	0	0	+		255

- **Index #1** defines destination number manipulation of calls from Timico Sip Trunk. All calls received from Source IP Group 2 (i.e., from Timico SIP Trunk) and the destination number prefix begins with '+', do not perform any changes to the number.
- **Index #2** defines destination number manipulation of calls from Timico Sip Trunk. All calls received from Source IP Group 2 (i.e., from Timico SIP Trunk) and the destination number prefix begins with '1', 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** submenu > **Source Number Tel** > **IP**).

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

Source Phone Number Manipulation Table for Tel -> IP Calls

Basic Parameter List

3 Add

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 Digits to Leave	Presentation
1	-1	2	*	+	0	0			255	Not Configured
2	-1	2	*	1	0	0	+		255	Not Configured

- **Index #1** defines Source number manipulation of calls from Timico SIP Trunk. All calls received from Source IP Group **2** (i.e., from Timico SIP Trunk) and the Source number prefix begins with '+', do not perform any changes to the number.
- **Index #2** defines Source number manipulation of calls from Timico SIP Trunk. All calls received from Source IP Group **2** (i.e., from Timico SIP Trunk) and the Source number prefix begins with **1**, Add a ‘+’ as a prefix to the number.

4.16 Step 16: Configuring SIP TLS Connection

This step describes how to configure 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.

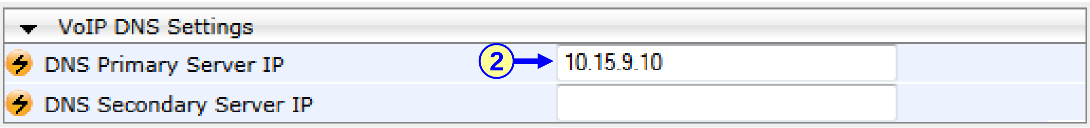
4.16.1 Step 16-1: Configuring VoIP DNS Settings

This step 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 4-25: 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>.

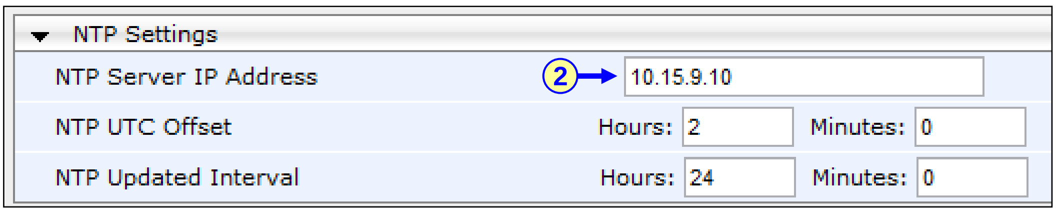
4.16.2 Step 16-2: Configuring NTP Server

This step describes how to configure the NTP Server IP address. It is recommended to implement an N TP 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 4-26: NTP Settings



2. Set the **NTP Server IP Address** to <NTP Server IP-Address> (e.g., 10.15.9.10).

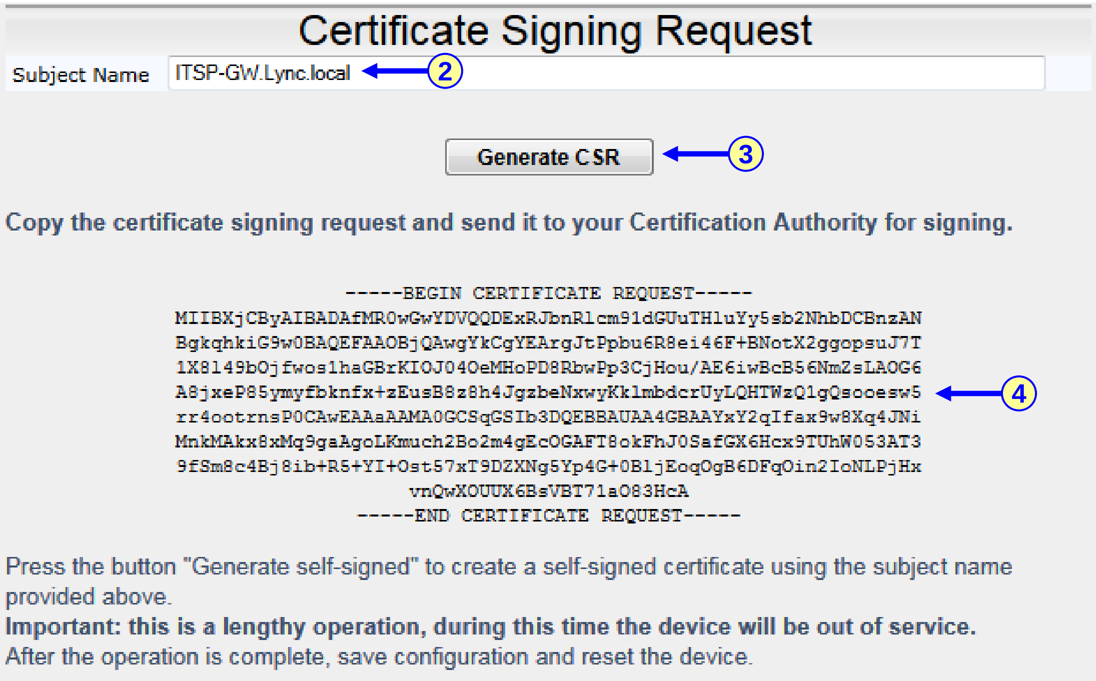
4.16.3 Step 16-3: Configuring a Certificate

This step 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**).

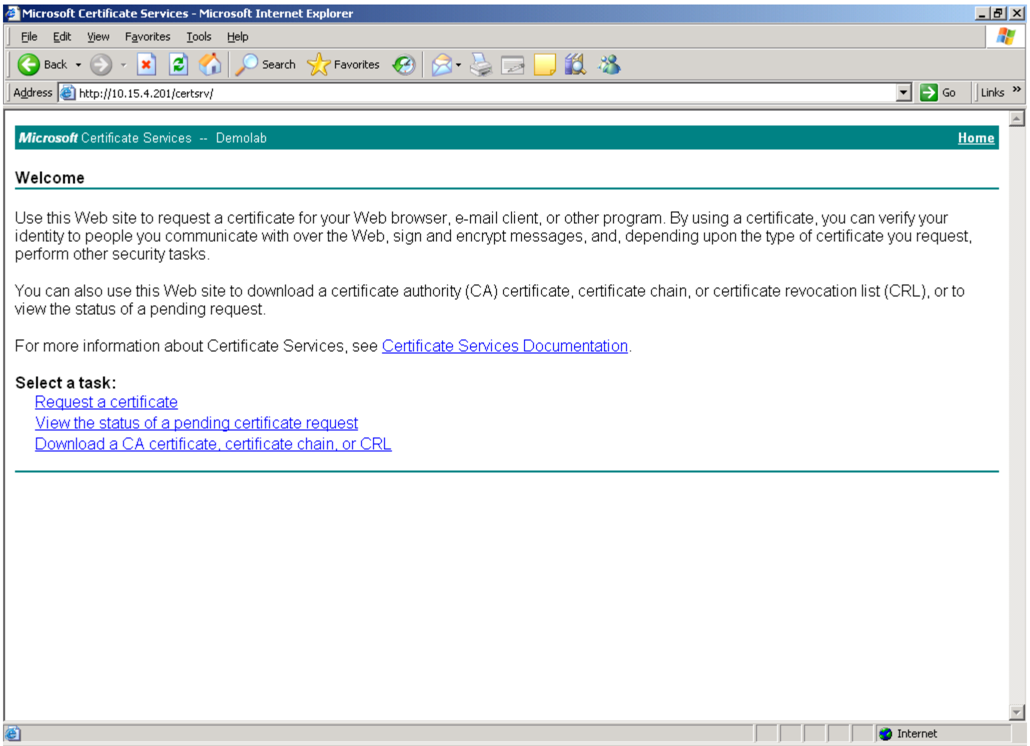
Figure 4-27: Certificates Page



2. In the 'Subject Name' field, enter the Media Gateway name (i.e., **ITSP-GW.Lync.local)**
3. Click on **Generate CSR**; a Certificate request will be generated.
4. 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*.

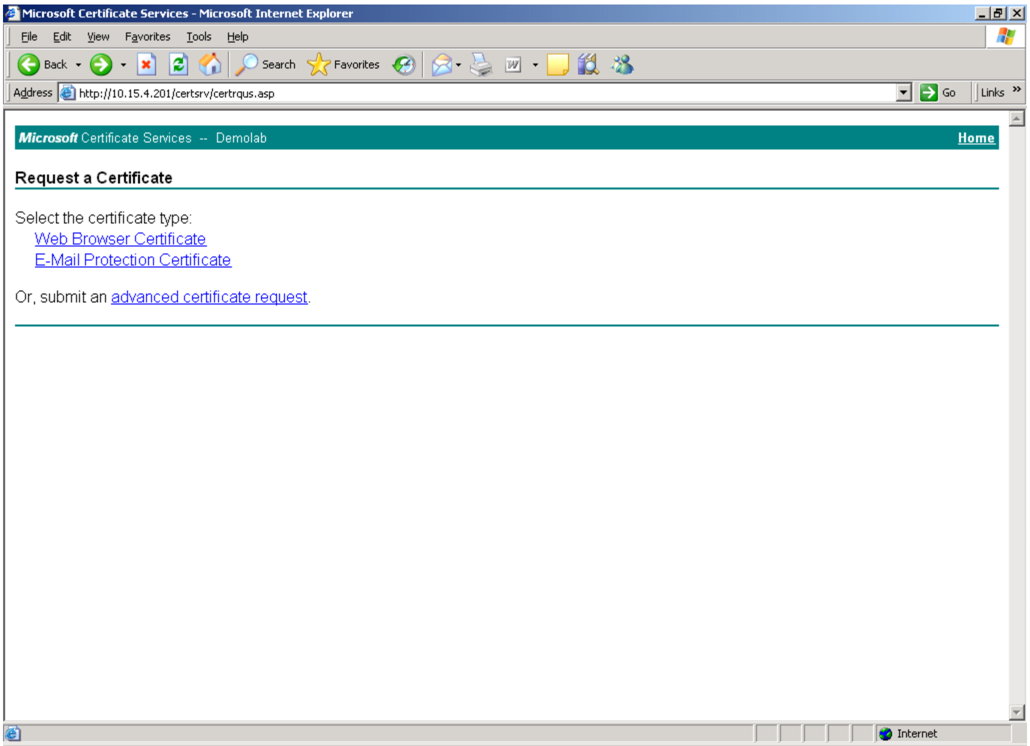
5. Navigate to the certificate 'Server http://<Certificate Server>/CertSrv'.

Figure 4-28: Microsoft Certificate Services Web Page



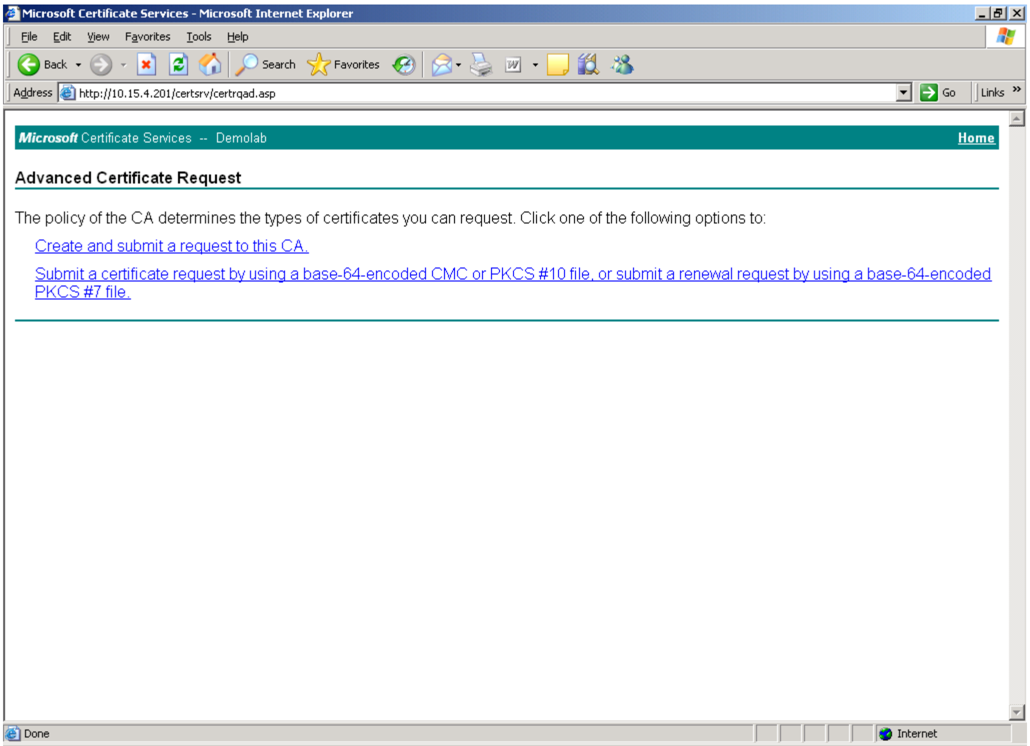
6. Click the link **Request a Certificate**.

Figure 4-29: Request a Certificate Page



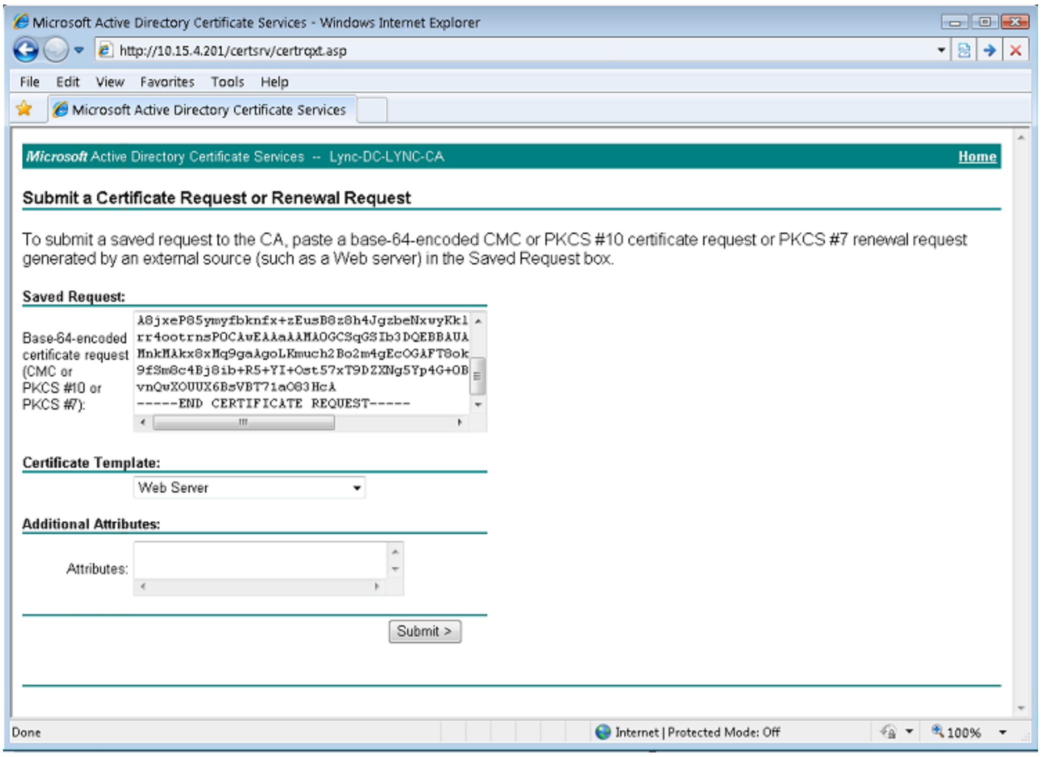
7. Click the link **Advanced Certificate Request**, and then click **Next**.

Figure 4-30: Advanced Certificate Request Page



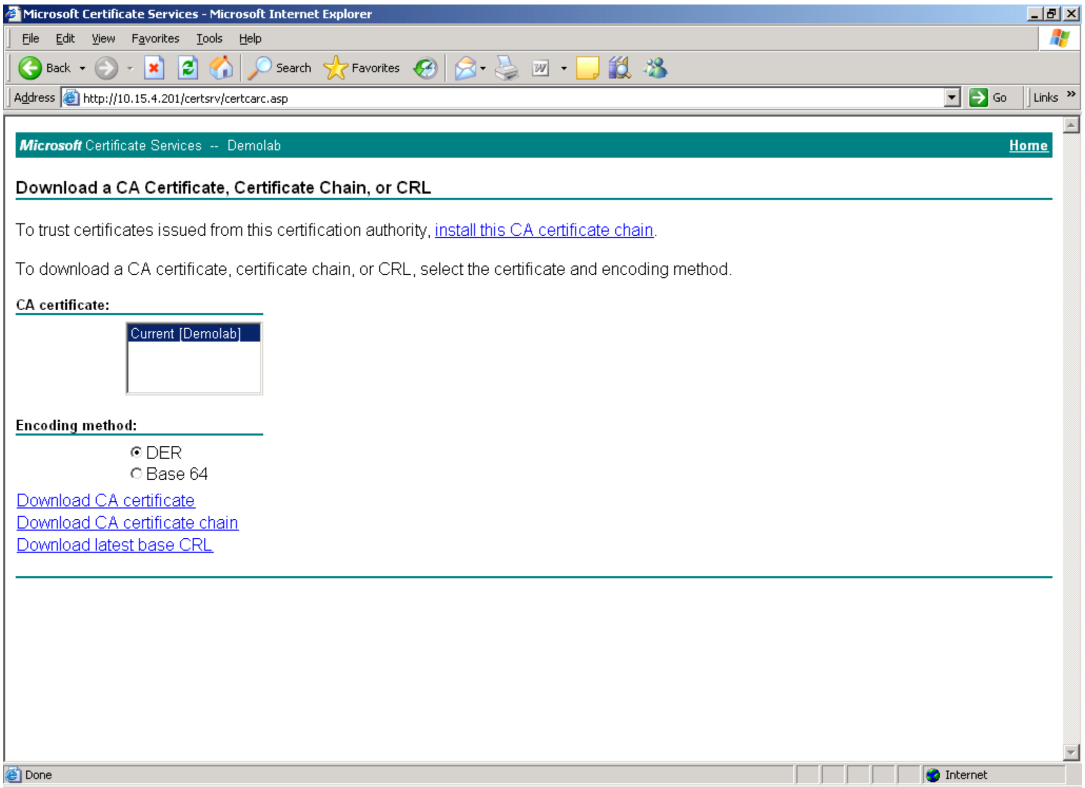
8. Click the link **Submit a Certificate request by using base64 encoded...**, and then click **Next**.

Figure 4-31: 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. Select 'Web Server' from the **Certificate Template** drop-down box.
11. Click **Submit**.
12. Choose the 'Base 64' encoding option, and then click the link **Download CA certificate**.
13. Save the file as '*gateway.cer'* in a folder on your PC.
14. Navigate to the certificate Server http://<Certificate Server>/CertSrv.
15. Click the link **Download a CA Certificate**, **Certificate Chain or CRL**.

Figure 4-32: Download a CA Certificate, Certificate Chain, or CRL Page



16. Under the Encoding method group, perform the following:
17. Select the 'Base 64' encoding method option.
18. Click the link Download CA certificate.
19. Save the file as '*certroot.cer'* in a folder on your PC.
20. Navigate back (in the E-SBC device) to the 'Certificates' page.

Figure 4-33: Certificates Page

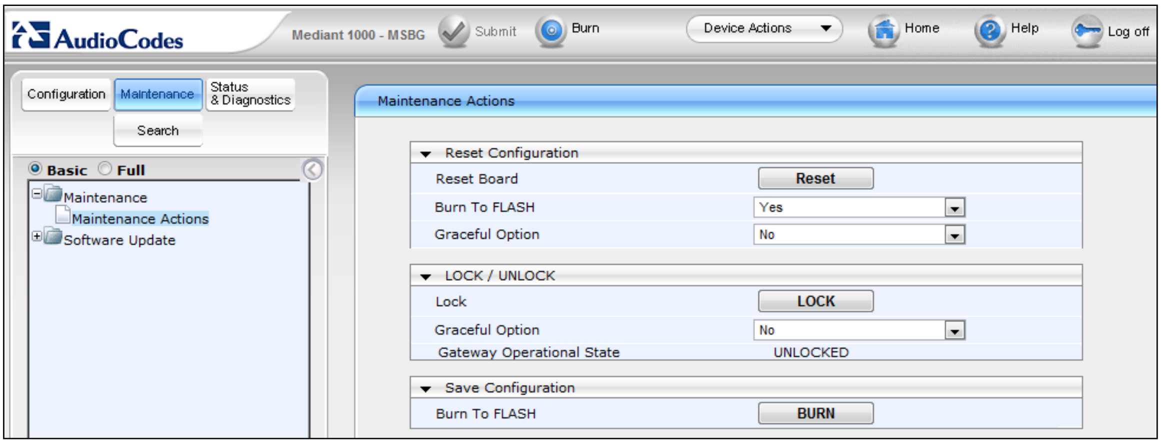
21. In the ‘Certificates’ page, in the ‘Server Certificate’ field, click **Browse** and select the ‘*Gateway.cer’* certificate file that you saved on your local disk (see Step 13), and then click **Send File** to upload the certificate.
22. In the ‘Certificates’ page, in the 'Trusted Root Certificate Store' field, click **Browse** and select the '*Certroot.cer*' certificate file that you saved on your local disk (see Step 19), and then click **Send File** to upload the certificate.
23. 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**).

4.17 Step 17: Resetting the Gateway

After you have completed the gateway configuration as described in the steps above, burn the configuration to the gateway's flash memory and reset the gateway.

- Click the **Reset** button to burn the configuration to flash and reset the gateway (ensure that the 'Burn to FLASH' field is set to **Yes**).

Figure 4-34: Reset the Gateway



Note: Reset with BURN to FLASH is required.

Reader's Notes

A AudioCodes INI File

This step shows the E-SBC device INI file. This file reflects the configuration described in Section 4 on page 35.

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

;Board: Mediant 1000 - MSBG
;Serial Number: 3589366
;Slot Number: 1
;Software Version: 6.20A.022.003
;DSP Software Version: 620AE3 => 620.08
;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 PCI
DspCh=240 IPMediaDspCh=240 ;DSP Voice features: EC128mSec
AdditionTimeslotSummation FastSlowPlayback BargeIn PatternDetector
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
```

```
TelnetServerEnable = 1
PM_VEDSPUtil = '1,64,72,15'

[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
DIGITMAPPING = ''

[PSTN Params]


[SS7 Params]


[Voice Engine Params]

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

[WEB Params]

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

[SIP Params]
```

```
MEDIACHANNELS = 120
SIPDESTINATIONPORT = 5067
PLAYRBTONE2TEL = 3
CHANNELSELECTMODE = 1
GWDEBUGLEVEL = 5
ENABLEEARLYMEDIA = 1
SIPGATEWAYNAME = 'ITSP-GW.Lync.local'
STATICNATIP = 195.189.192.154
PRACKMODE = 1
ISFAXUSED = 2
SIPTRANSPORTTYPE = 2
TCPLOCALSIPPORT = 5068
TLSLOCALSIPPORT = 5067
MEDIASECURITYBEHAVIOUR = 3
IGNOREALERTAFTEREARLYMEDIA = 1
FORKINGHANDLINGMODE = 1
ENABLEIP2IPAPPLICATION = 1
ENABLEEARLY183 = 1
PLAYHELDTONEFORIP2IP = 1
 [SCTP Params]


[VXML Params]


[IPsec Params]


[Audio Staging Params]


[SNMP Params]


;
;  *** TABLE InterfaceTable ***
;
;

[ InterfaceTable ]
FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_VlanID, InterfaceTable_InterfaceName;
InterfaceTable 0 = 6, 10, 10.15.7.131, 16, 10.15.7.130, 1, Voice;

[ \InterfaceTable ]

;
```

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

;
;  *** TABLE PREFIX ***
;
;

[ 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 0 = *, , *, 2, 255, 0, 1, , 2, , -1, -1, -1;
PREFIX 1 = *, , *, 1, 255, 0, 2, , 1, , -1, -1, -1;

[ \PREFIX ]

;
;  *** TABLE NumberMapIp2Tel ***
;
;

[ NumberMapIp2Tel ]
FORMAT NumberMapIp2Tel_Index = NumberMapIp2Tel_DestinationPrefix,
NumberMapIp2Tel_SourcePrefix, NumberMapIp2Tel_SourceAddress,
NumberMapIp2Tel_NumberType, NumberMapIp2Tel_NumberPlan,
NumberMapIp2Tel_RemoveFromLeft, NumberMapIp2Tel_RemoveFromRight,
NumberMapIp2Tel_LeaveFromRight, NumberMapIp2Tel_Prefix2Add,
NumberMapIp2Tel_Suffix2Add, NumberMapIp2Tel_IsPresentationRestricted,
NumberMapIp2Tel_SrcTrunkGroupID, NumberMapIp2Tel_SrcIPGroupID;
NumberMapIp2Tel 1 = +, *, *, 255, 255, 0, 0, 255, , , 255, -1, -1;
NumberMapIp2Tel 2 = *, *, *, 255, 255, 0, 0, 255, +, , 255, -1, -1;

[ \NumberMapIp2Tel ]

;
;  *** TABLE NumberMapTel2Ip ***
;
;

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

[ \NumberMapTel2Ip ]

;
;  *** TABLE PstnPrefix ***
;
;

[ 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, , *, 2, 2, , ;

[ \PstnPrefix ]

;

;
;  *** TABLE ProxyIp ***
;
;

[ ProxyIp ]
; ** NOTE: Changes were made to active configuration.
; **       The data below is different from current values.
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = FE-Lync.Lync.local, 2, 1;
ProxyIp 1 = 195.54.255.50:5060, 0, 2;

[ \ProxyIp ]

;
;  *** TABLE TxDtmfOption ***
;
;

[ TxDtmfOption ]
FORMAT TxDtmfOption_Index = TxDtmfOption_Type;
TxDtmfOption 0 = 4;

[ \TxDtmfOption ]
```

```
;
;  *** TABLE IpProfile ***
;
;

[ 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 1 = Lync, 1, 1, 2, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 1,
-1, 1, 0, 3, -1, 1, 4, -1, 1, 1, 0, 0, , -1, 0, 0, -1, 0, 0, 0, 0, -1,
0, 8, 300, 400, -1, -1;
IpProfile 2 = Timico, 1, 2, 2, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0,
1, -1, 1, 0, 2, -1, 1, 4, -1, 1, 1, 0, 0, , -1, 0, 0, -1, 0, 0, 0, 0, -
1, 0, 8, 300, 400, -1, -1;

[ \IpProfile ]

;
;  *** TABLE ProxySet ***
;
;

[ 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 ]
```

```
;
;  *** TABLE IPGroup ***
;
;

[ 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_ContactName;
IPGroup 1 = 0, Lync, 1, timico.co.uk, , 0, -1, 0, 0, -1, 0, , 1, 0, -1,
-1, -1, ;
IPGroup 2 = 0, Timico, 2, , , 0, -1, 0, 0, -1, 0, , 1, 2, -1, -1, -1, ;
[ \IPGroup ]

;
;  *** TABLE CodersGroup0 ***
;
;

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

[ \CodersGroup0 ]

;
;  *** TABLE CodersGroup1 ***
;
;

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

[ \CodersGroup1 ]

;
;  *** TABLE CodersGroup2 ***
;
;
```

```
[ CodersGroup2 ]
FORMAT CodersGroup2_Index = CodersGroup2_Name, CodersGroup2_pTime,
CodersGroup2_rate, CodersGroup2_PayloadType, CodersGroup2_Sce;
CodersGroup2 0 = g711Alaw64k, 20, 0, -1, 0;
CodersGroup2 1 = g729, 20, 0, -1, 0;
CodersGroup2 2 = g7231, 30, 0, -1, 0;
CodersGroup2 3 = g711Alaw64k, 20, 0, -1, 0;

[ \CodersGroup2 ]
```

Configuration Note