Enterprise Session Border Controllers (E-SBC)

AudioCodes™ Mediant™ Series

Interoperability Lab

Configuration Note

Combined Fax and Microsoft® Lync™ Server 2013 and Swisscom VoIP Gate using Mediant E-SBC

Microsoft Partner
Gold Communications

Lync

swisscom

AudioCodes

Version 6.6

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Document # LTRT-39351

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Notice

This document describes how to connect the Microsoft Lync Server 2013 and Swisscom VoIP Gate using AudioCodes Mediant E-SBC product series, which includes the Mediant 800 Gateway & E-SBC, Mediant 1000B Gateway & E-SBC, Mediant 3000 Gateway & E-SBC, Mediant 2600 E-SBC, and Mediant 4000 E-SBC.

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1 Introduction

This Configuration Note describes how to set up AudioCodes Enterprise Session Border Controller (hereafter, referred to as *E-SBC*) for interworking between Swisscom's VoIP Gate SIP Trunk and Microsoft's Lync Server 2013 environment combined with analog or fax devices connected on one or more AudioCodes MediaPack 11x analog gateways.

1.1 Intended Audience

The document is intended for engineers, or AudioCodes and Swisscom Partners who are responsible for installing and configuring Swisscom VoIP Gate and Microsoft's Lync Server 2013 for enabling VoIP calls using AudioCodes E-SBC.

1.2 About AudioCodes E-SBC Product Series

AudioCodes' family of E-SBC devices enables reliable connectivity and security between the Enterprise's and the service provider's VoIP networks.

The E-SBC provides perimeter defense as a way of protecting Enterprises 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.

Designed as a cost-effective appliance, the E-SBC is based on field-proven VoIP and network services with a native host processor, allowing the creation of purpose-built multiservice appliances, providing smooth connectivity to cloud services, with integrated quality of service, SLA monitoring, security and manageability. The native implementation of SBC provides a host of additional capabilities that are not possible with standalone SBC appliances such as VoIP mediation, PSTN access survivability, and third-party value-added services applications. This enables Enterprises to utilize the advantages of converged networks and eliminate the need for standalone appliances.

AudioCodes E-SBC is available as an integrated solution running on top of its field-proven Mediant Media Gateway and Multi-Service Business Router platforms, or as a software-only solution for deployment with third-party hardware.

1.3 About AudioCodes MediaPack 11x Product Series

The MediaPack series analog media gateways are cost-effective, cutting edge technology products. These stand-alone analog VoIP devices provide superior voice technology for connecting legacy telephones, fax machines and Private Branch Exchange (PBX) systems to IP-based telephony networks, as well as for integration with new IP-based PBX architectures. These devices are designed and tested to be fully interoperable with leading softswitches and SIP servers.

The device enables users to make local or international telephone and / or fax calls over the Internet between distributed company offices, using their existing telephones and fax.

The devices applicable in this configuration note provide analog ports for direct connection to phones, fax machines, and modems (FXS). Depending on model, the device can support up to 24 simultaneous VoIP calls. The device is also equipped with a 10/100Base-TX Ethernet port for connection to the IP network. The device provides LEDs for indicating operating status of the various interfaces.

The device is a compact unit that can be easily mounted on a desktop, wall, or in a 19-inch rack.

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2 Component Information

2.1 AudioCodes E-SBC Version

Table 2-1: AudioCodes E-SBC Version

SBC Vendor	AudioCodes
Models	▪ Mediant 800 Gateway & E-SBC ▪ Mediant 1000B Gateway & E-SBC ▪ Mediant 3000 Gateway & E-SBC ▪ Mediant 2600 E-SBC ▪ Mediant 4000 E-SBC
Software Version	SIP_6.60A.257.004
Protocol	▪ SIP/TCP (to the Swisscom VoIP Gate SIP Trunk) ▪ SIP/TCP or TLS (to the Lync FE Server)
Additional Notes	None

2.2 AudioCodes MediaPack Version

Table 2-2: AudioCodes E-SBC Version

SBC Vendor	AudioCodes
Models	▪ MediaPack 112 Gateway ▪ MediaPack 114 Gateway ▪ MediaPack 118 Gateway ▪ MediaPack 124D Gateway
Software Version	SIP_ 6.60A.245
Protocol	▪ SIP/TCP (to the AudioCodes E-SBC)
Additional Notes	None

2.3 SIP Trunking Version

Table 2-3: Swisscom Version

Vendor/Service Provider	Swisscom
SSW Model/Service	VoIP Gate
Software Version	
Protocol	SIP
Additional Notes	None

2.4 Microsoft Lync Server 2013 Version

Table 2-4: Microsoft Lync Server 2013 Version

Vendor	Microsoft
Model	Microsoft Lync
Software Version	Lync 2013 – CU 3 or higher
Protocol	SIP
Additional Notes	None

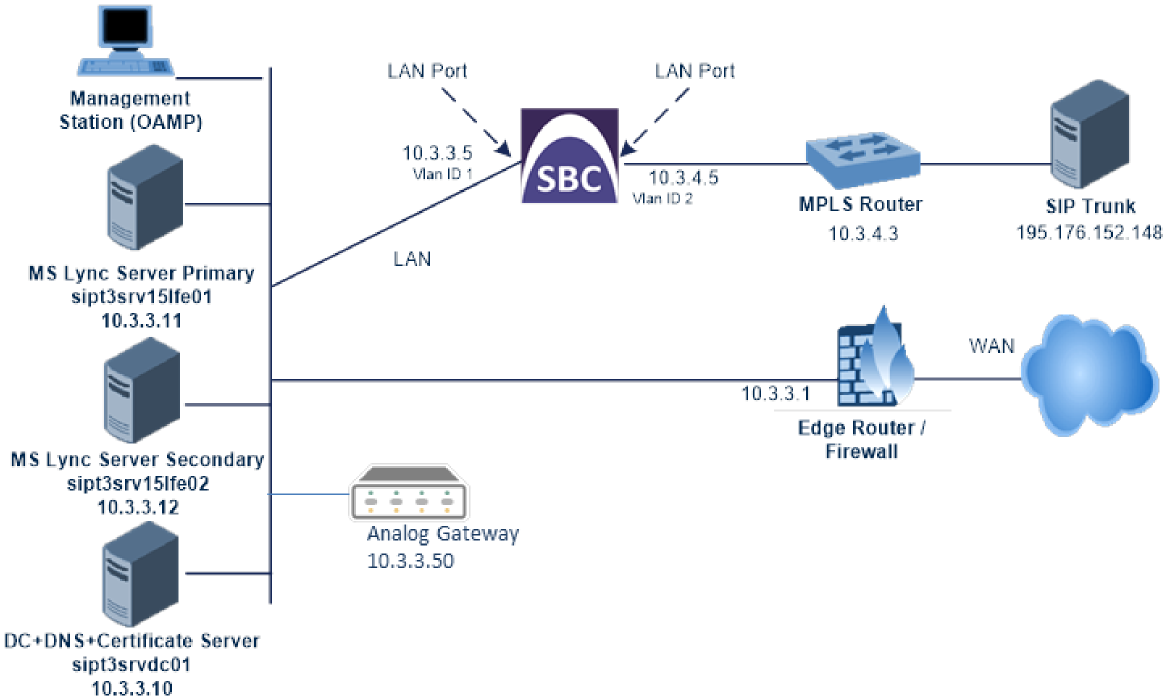
2.5 Interoperability Test Topology

The interoperability testing between AudioCodes E-SBC and Swisscom VoIP Gate with Lync 2013 was done using the following topology setup:

- Enterprise deployed with Microsoft Lync Server 2013 in its private network for enhanced communication within the Enterprise.
- Enterprise requires the use of Fax or analog devices which are not integrated into the Microsoft Lync Server topology.
- Enterprise wishes to offer its employees enterprise-voice capabilities and to connect the Enterprise to the PSTN network using Swisscom's VoIP Gate SIP Trunking service.
- A single instance of Swisscom VoIP Gate is used to support communication with both Lync and Fax.
- AudioCodes E-SBC is implemented to interconnect between the Enterprise LAN and the SIP Trunk.
    - **Session:** Real-time voice session using the IP-based Session Initiation Protocol (SIP).
    - **Border:** IP-to-IP network border between Lync Server 2013 network in the Enterprise LAN and Swisscom VoIP Gate located in the public network.
- The proposed configuration uses dynamic routing from the E-SBC towards the MediaPack analog gateways.
    - The MediaPack registers all analog endpoints on the E-SBC
    - The E-SBC uses the SIP registration information to create a dynamic database containing the relevant phone numbers and IP-addresses (AOR)
    - Calls received from Lync Server 2013 will be verified against this AOR database, and routed to the relevant MediaPack Analog Gateway if a match has been found. If no match is found, the call is relayed towards the Swisscom VoIP Gate.
    - Calls received from the Swisscom VoIP Gate follow the same logic, if a match is found in the AOR the call is forwarded to the relevant MediaPack Analog Gateway, if no match is found, the call continues towards the Lync Server 2013.

The figure below illustrates this interoperability test topology:

Figure 2-1: Interoperability Test Topology between E-SBC and Microsoft Lync with Swisscom VoIP Gate



2.5.1 Environment Setup

The interoperability test topology includes the following environment setup:

Table 2-5: Environment Setup

Area	Setup
Network	▪ Microsoft Lync Server 2013 environment is located on the Enterprise's LAN ▪ The AudioCodes MediaPack is located on the Enterprise’s LAN ▪ Swisscom VoIP Gate is located on the WAN
Signaling Transcoding	▪ Microsoft Lync Server 2013 operates with SIP-over-TCP or TLS transport type ▪ Swisscom VoIP Gate operates with SIP-over-TCP transport type
Codecs Transcoding	▪ Microsoft Lync Server 2013 supports G.711A-law ▪ Swisscom VoIP Gate supports G.711A-law, G711A-law_VBD coders. and T.38 coder
Media Transcoding	▪ Microsoft Lync Server 2013 operates with SRTP or RTP media type ▪ Swisscom VoIP Gate operates with RTP media type

Note: This interoperability test did not include TLS or SRTP between the Lync Server 2013 and the AudioCodes E-SBC. If applicable, refer to Step 8 and Step 9 in our generic *LTRT-54004 Mediant E-SBC* SIP *Trunking for Microsoft Lync 2013 Configuration Note* for additional instructions, in case TLS and SRTP are required for your environment.

2.5.2 Known Limitations

There were no limitations observed in the interoperability tests done for the AudioCodes E-SBC interworking between Microsoft Lync Server 2013 or AudioCodes MediaPack and Swisscom VoIP Gate.

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3 Configuring Lync Server 2013

This chapter describes how to configure Microsoft Lync Server 2013 to operate with AudioCodes E-SBC.

> **Note:** Dial plans, voice policies, and PSTN usages are also necessary for Enterprise voice deployment; however, they are beyond the scope of this document.

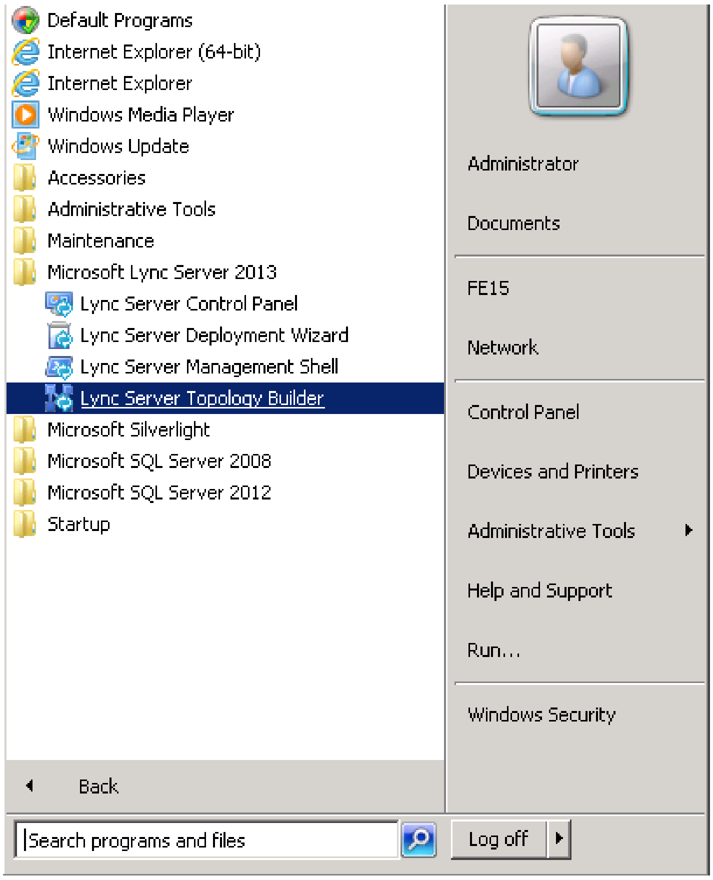
3.1 Configuring the E-SBC as an IP / PSTN Gateway

The procedure below describes how to configure the E-SBC as an IP / PSTN Gateway.

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

1. On the server where the Topology Builder is installed, start the Lync Server 2013 Topology Builder (Windows **Start** menu > **All Programs** > **Lync Server Topology Builder**), as shown below:

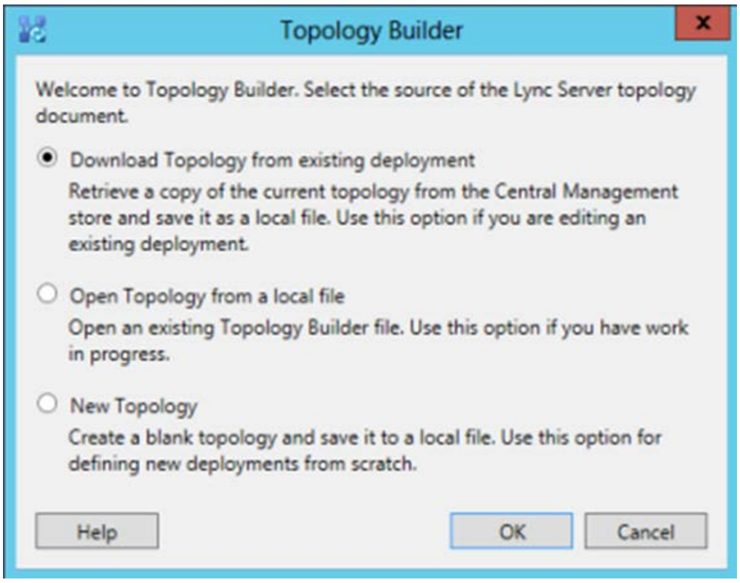
Figure 3-1: Starting the Lync Server Topology Builder



> **Note:** When using Lync on Windows Server 2012, use the modern UI equivalent to start the Lync Server Topology Builder.

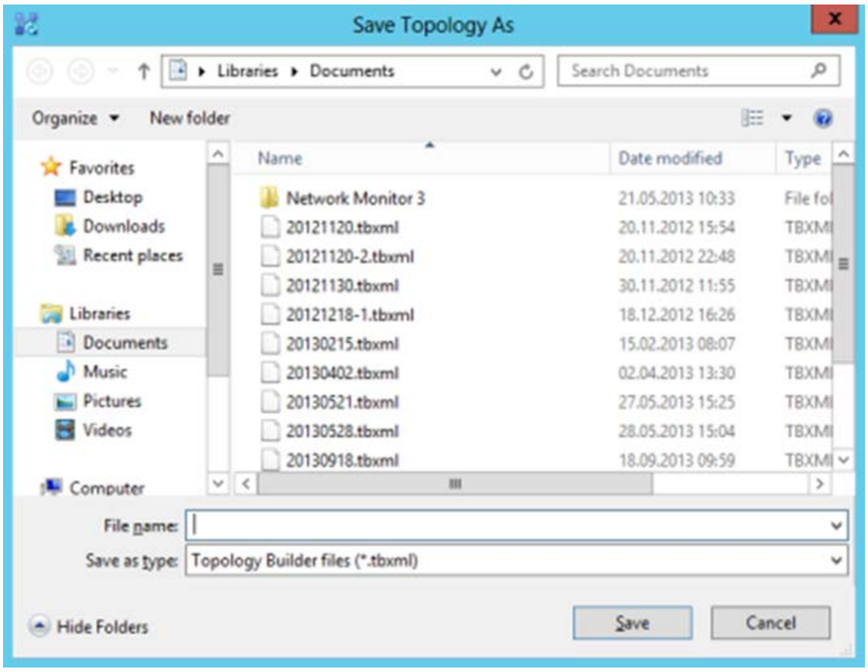
The following is displayed:

Figure 3-2: Topology Builder Dialog Box



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

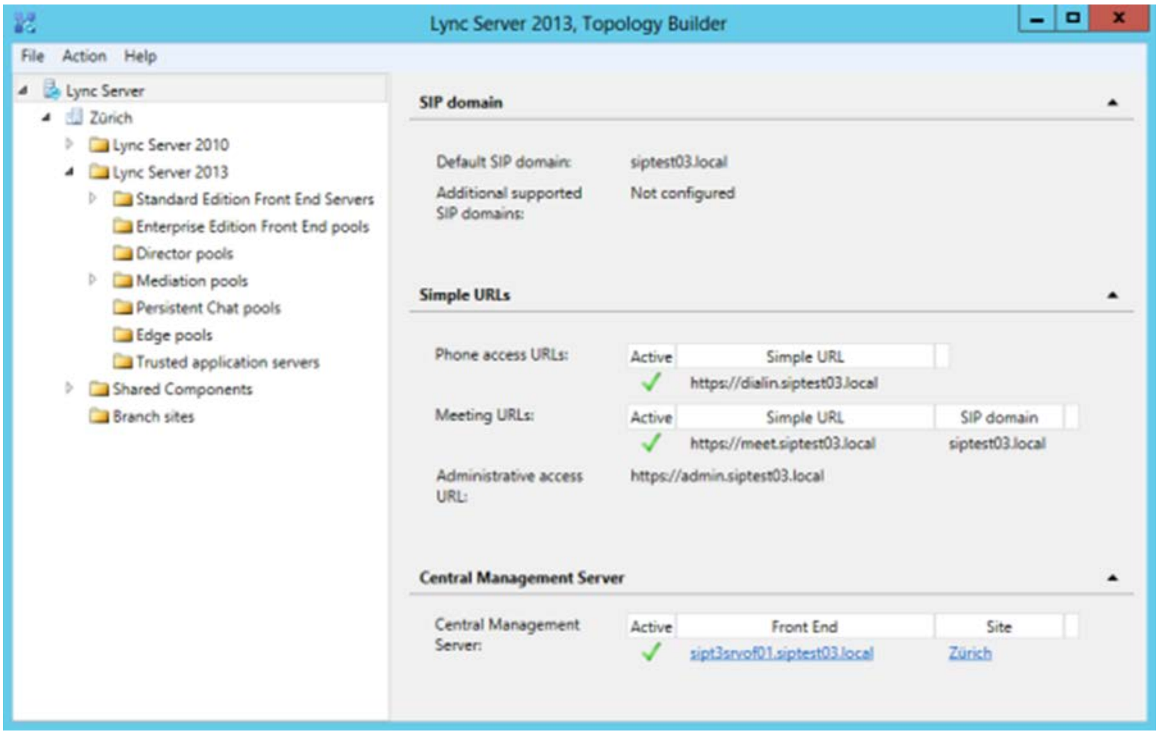
Figure 3-3: Save Topology Dialog Box



3. Enter a name for the Topology file, and then click **Save**. This step enables you to roll back from any changes you make during the installation.

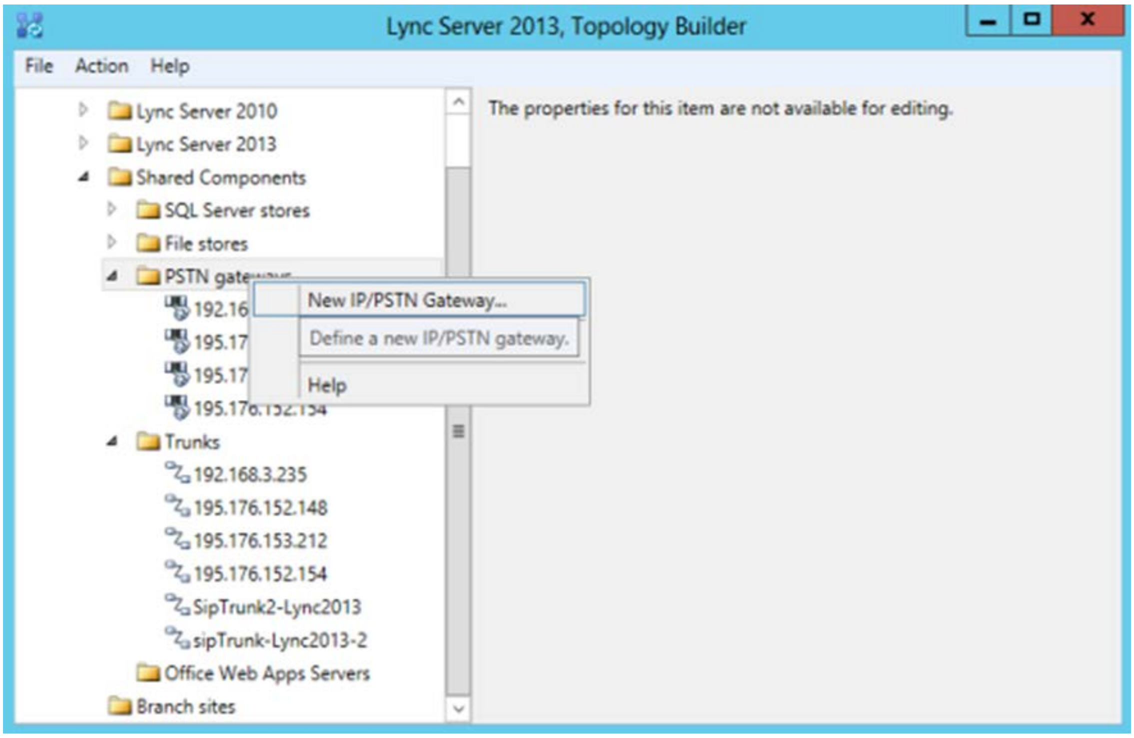
   The Topology Builder screen with the downloaded Topology is displayed:

Figure 3-4: Downloaded Topology



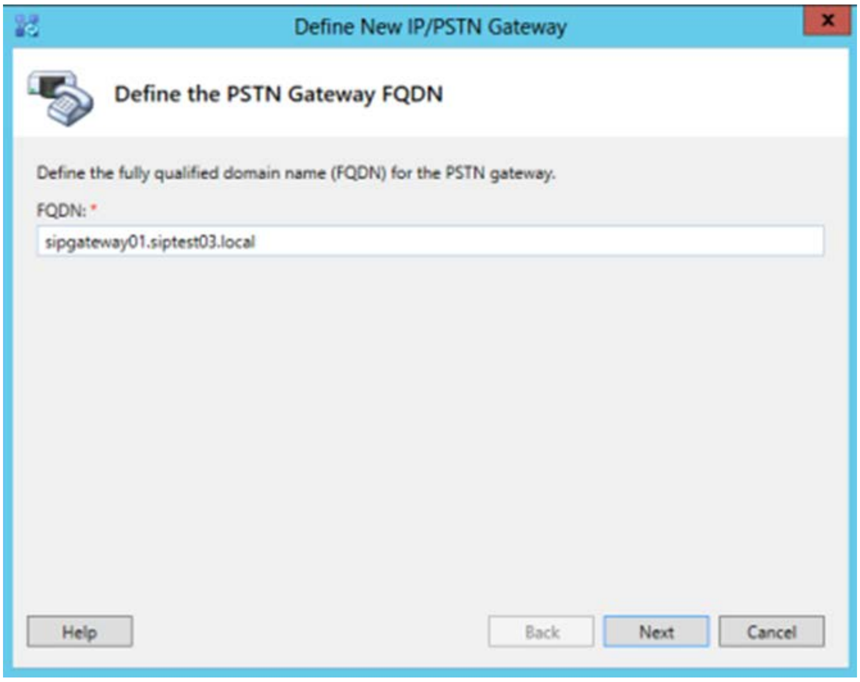
4. Under the **Shared Components** node, right-click the **PSTN gateways** node, and then from the shortcut menu, choose **New IP/PSTN Gateway**, as shown below:

Figure 3-5: Choosing New IP/PSTN Gateway



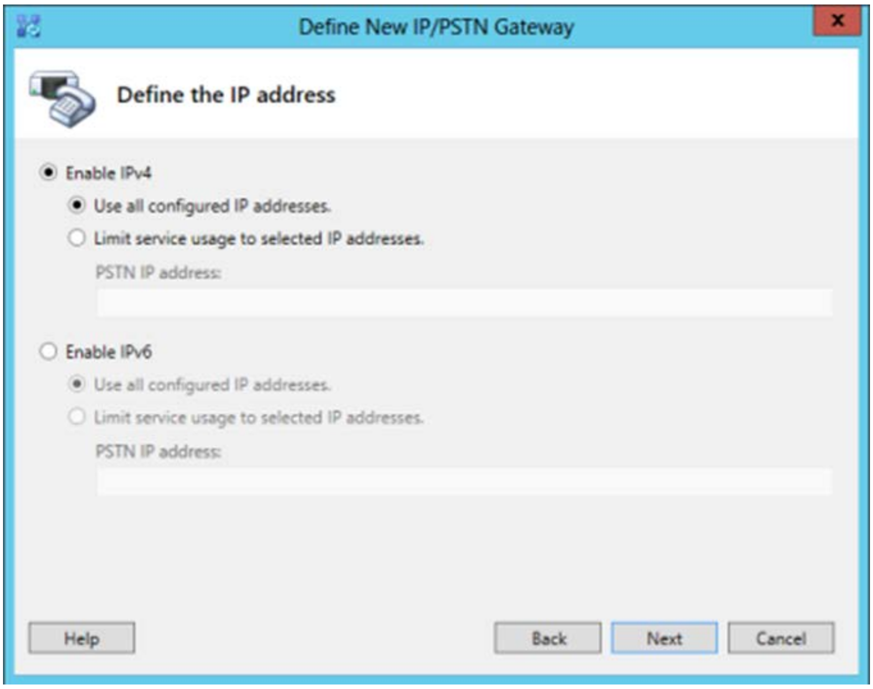
The following is displayed:

Figure 3-6: Define the PSTN Gateway FQDN



5. Enter the Fully Qualified Domain Name (FQDN) of the E-SBC (e.g., **sipgateway01.siptest03.local**). Update this FQDN in the relevant DNS record, and then click **Next**; the following is displayed:

Figure 3-7: Define the IP Address



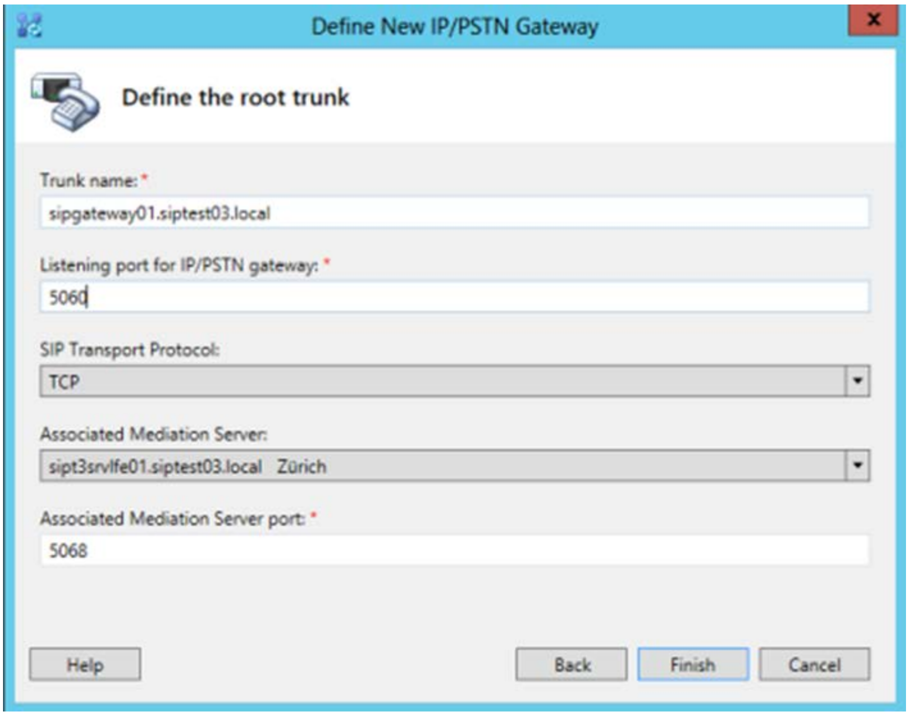
6. Define the listening mode (IPv4 or IPv6) of the IP address of your new PSTN gateway, and then click **Next**.

7. Define a *root trunk* for the PSTN gateway. A trunk is a logical connection between the Mediation Server and a gateway uniquely identified by the following combination: Mediation Server FQDN, Mediation Server listening port (TLS or TCP), gateway IP and FQDN, and gateway listening port.

Notes:

- When defining a PSTN gateway in Topology Builder, you must define a root trunk to successfully add the PSTN gateway to your topology.
- The root trunk cannot be removed until the associated PSTN gateway is removed.

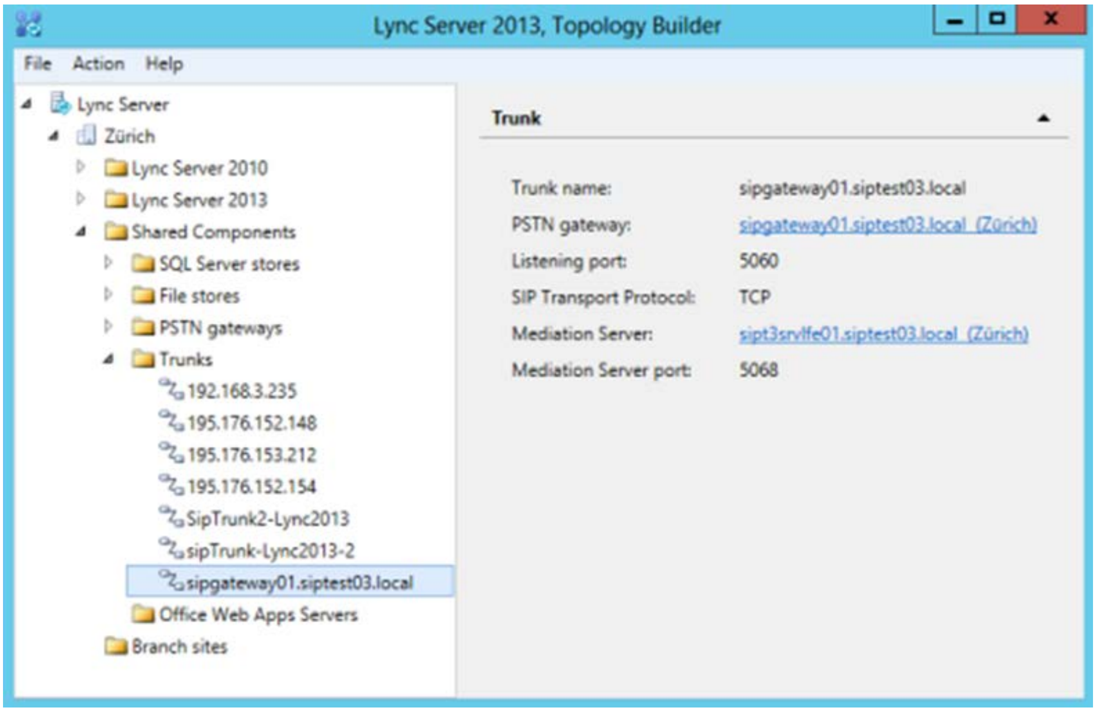
Figure 3-8: Define the Root Trunk



a. In the 'Listening Port for IP/PSTN Gateway' field, enter the listening port that the E-SBC will use for SIP messages from the Mediation Server that will be associated with the root trunk of the PSTN gateway (e.g., **5060**).
b. In the 'SIP Transport Protocol' field, select the transport type (e.g., **TCP**) that the trunk uses.
c. In the 'Associated Mediation Server' field, select the Mediation Server pool to associate with the root trunk of this PSTN gateway.
d. In the 'Associated Mediation Server Port' field, enter the listening port that the Mediation Server will use for SIP messages from the SBC (e.g., **5068**).
e. Click **Finish**.

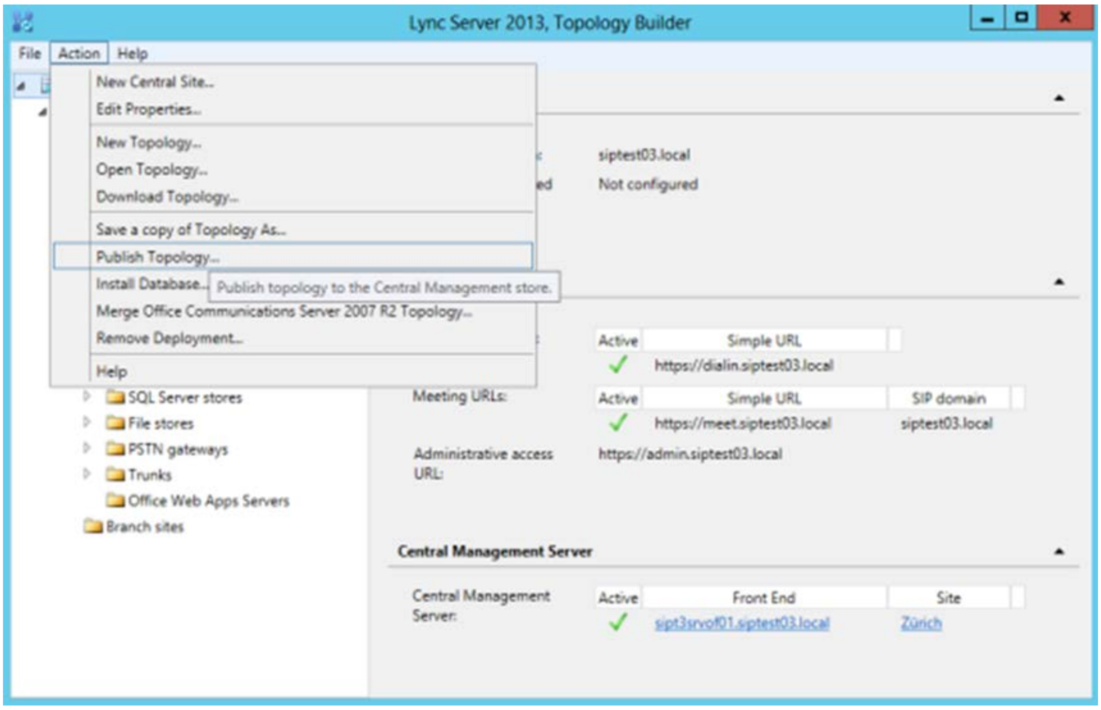
The E-SBC is added as a PSTN gateway, and a trunk is created as shown below:

Figure 3-9: E-SBC added as IP/PSTN Gateway and Trunk Created



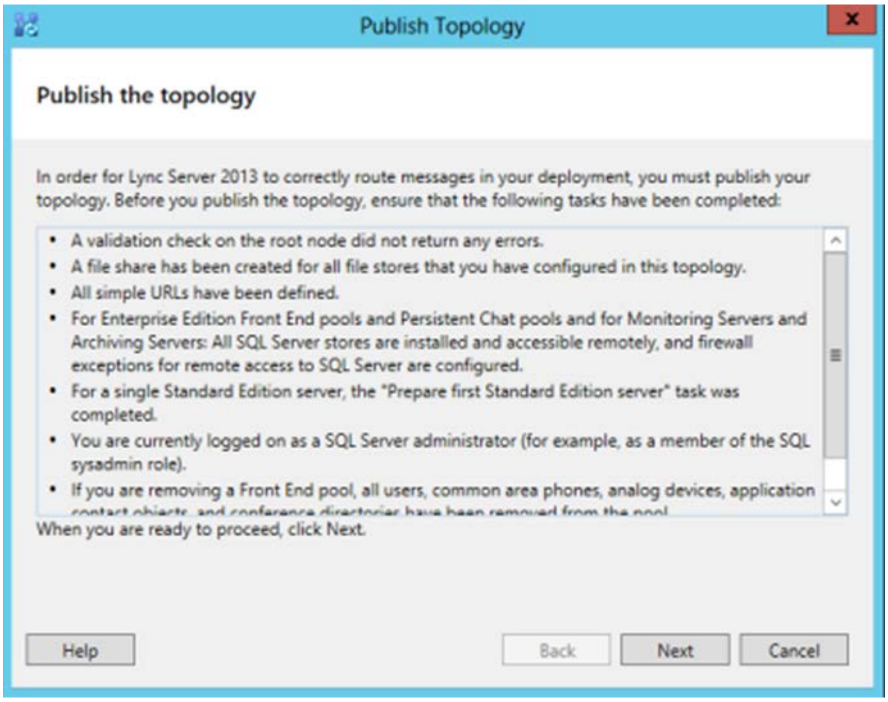
8. Publish the Topology: In the main tree, select the root node **Lync Server**, and then from the **Action** menu, choose **Publish Topology**, as shown below:

Figure 3-10: Choosing Publish Topology



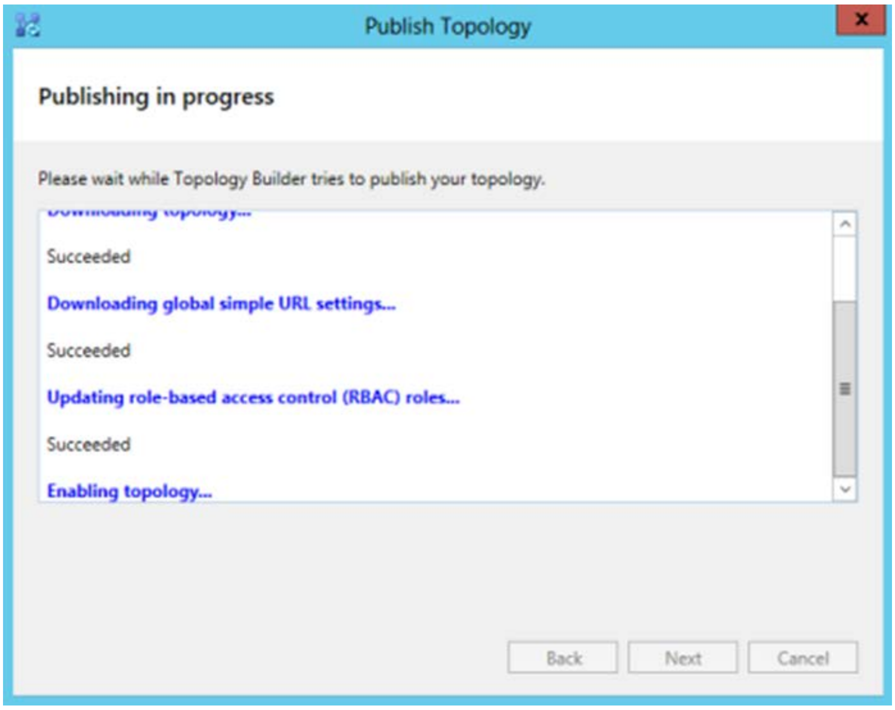
The following is displayed:

Figure 3-11: Publish the Topology



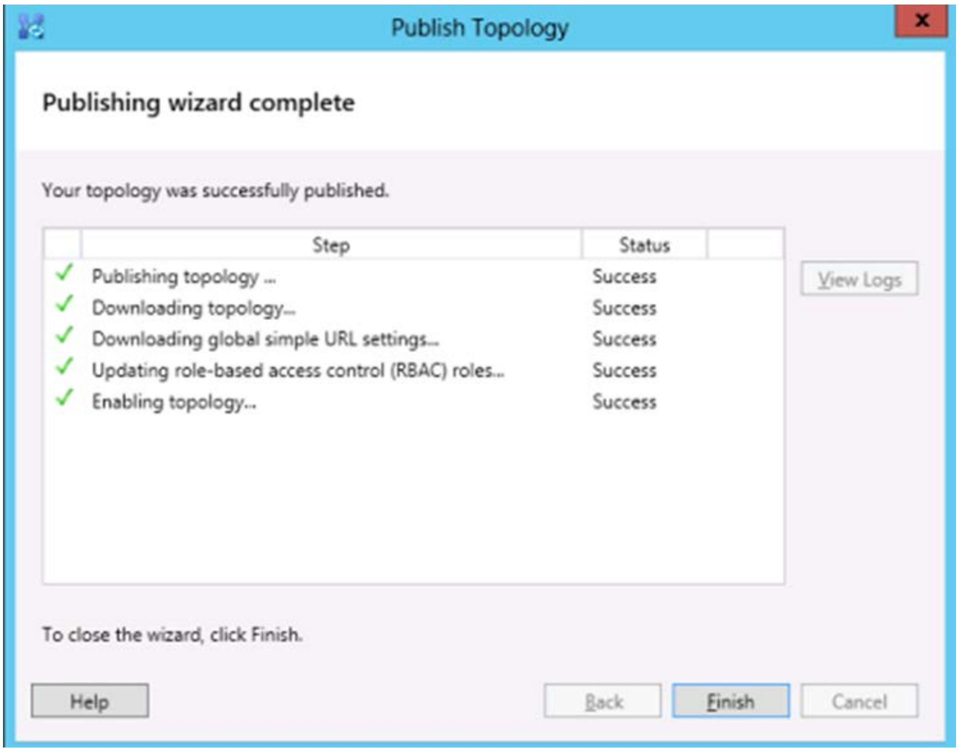
9. Click **Next**; the Topology Builder starts to publish your topology, as shown below:

Figure 3-12: Publishing in Progress



10. Wait until the publishing topology process completes successfully, as shown below:

Figure 3-13: Publishing Wizard Complete



11. Click **Finish**.

3.2 Configuring the "Route" on Lync Server 2013

The procedure below describes how to configure a "Route" on the Lync Server 2013 and to associate it with the E-SBC PSTN gateway.

➢ **To configure the "route" on Lync Server 2013:**

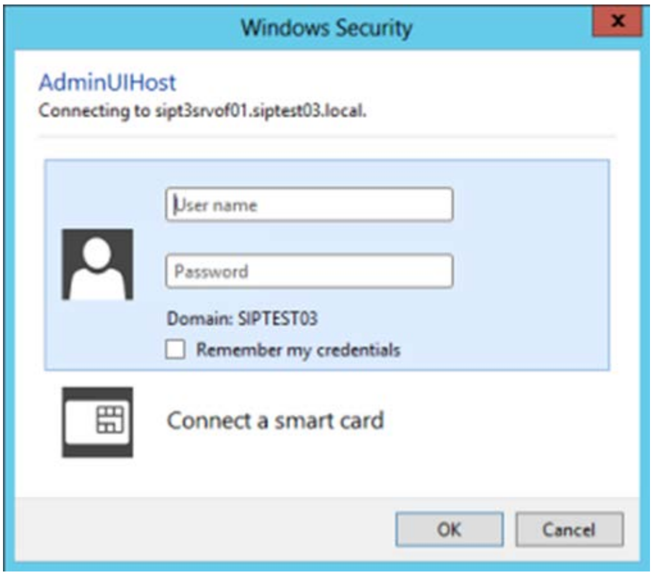
1. Start the Microsoft Lync Server 2013 Control Panel (**Start** > **All Programs** > **Microsoft Lync Server 2013** > **Lync Server Control Panel**), as shown below:

Figure 3-14: Opening the Lync Server Control Panel

Note: When using Lync on Windows Server 2012, use the modern UI equivalent to start the Lync Server Control Panel.

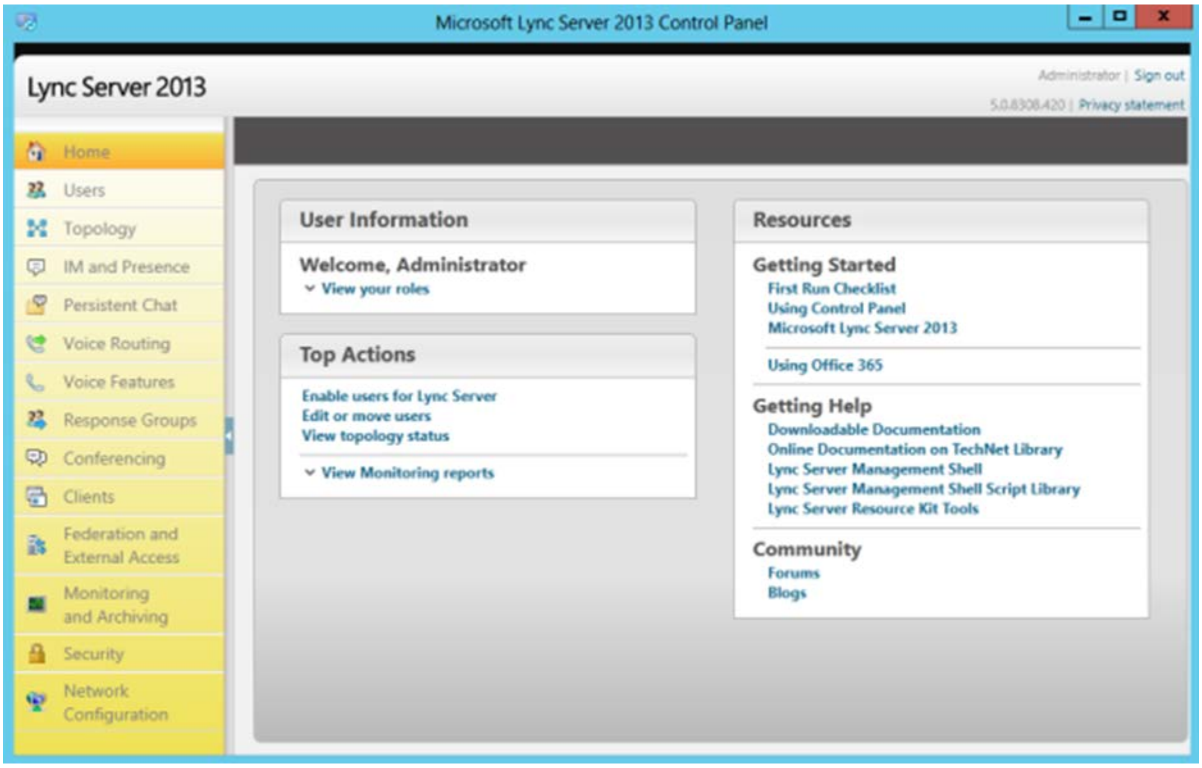
You are prompted to enter your login credentials:

Figure 3-15: Lync Server Credentials



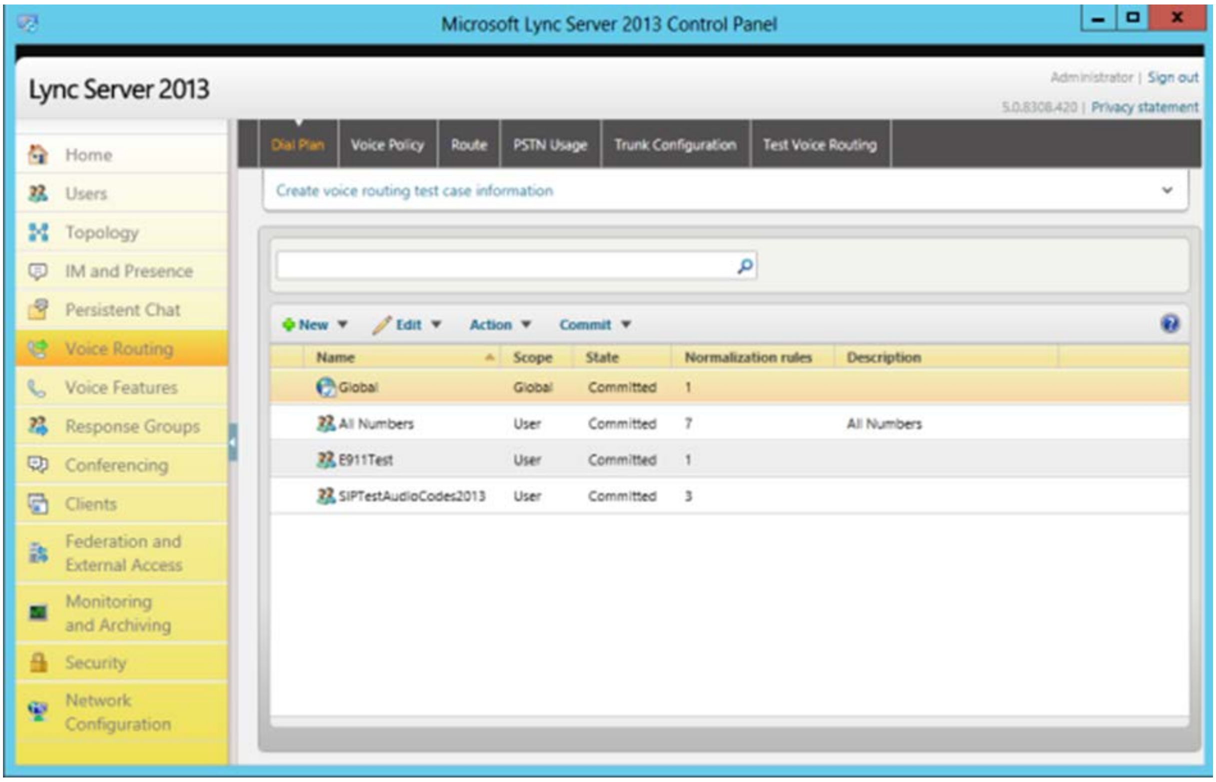
2. Enter your domain username and password, and then click **OK**; the Microsoft Lync Server 2013 Control Panel is displayed:

Figure 3-16: Microsoft Lync Server 2013 Control Panel



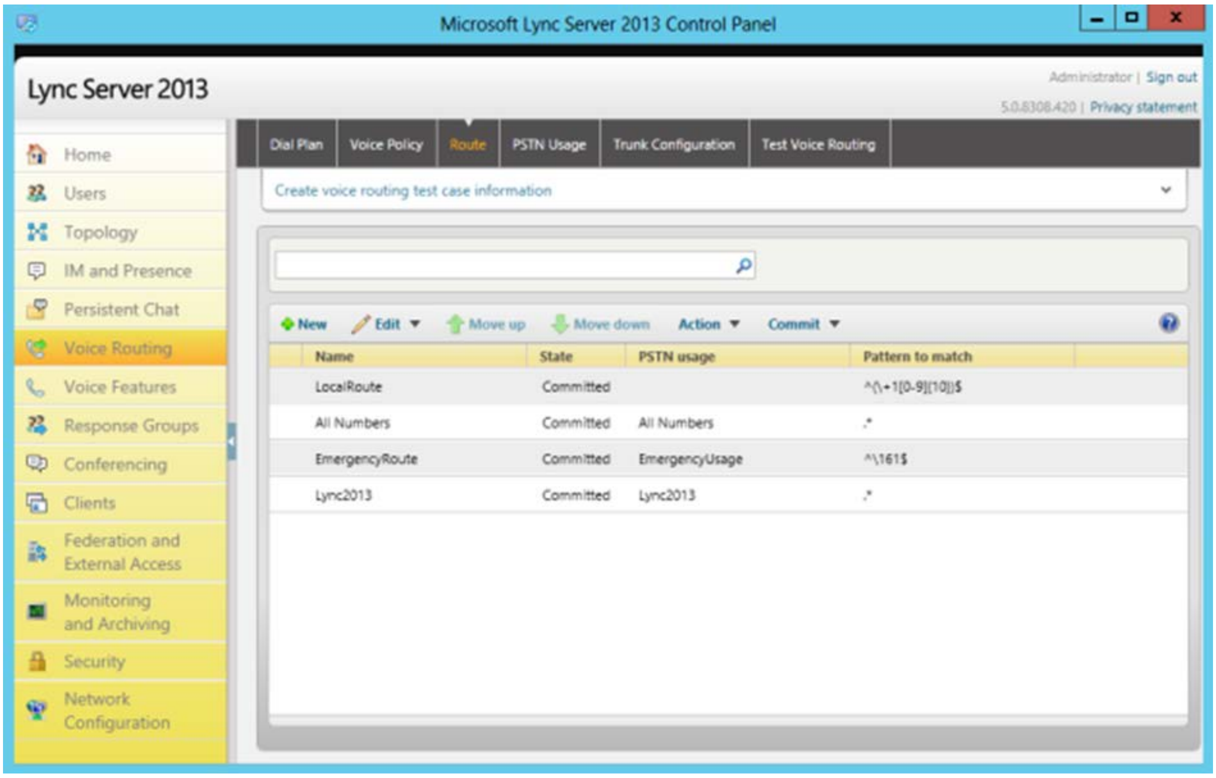
3. In the left navigation pane, select **Voice Routing**.

Figure 3-17: Voice Routing Page



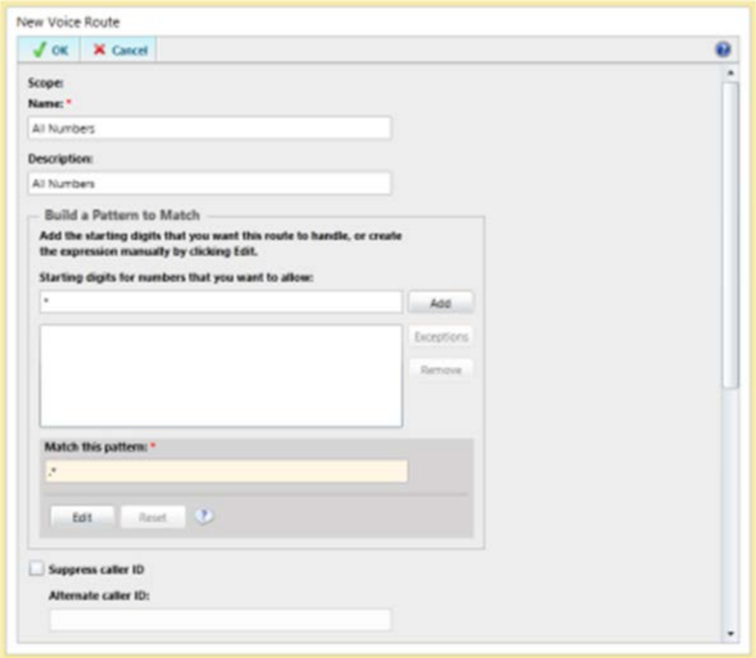
4. In the Voice Routing page, select the **Route** tab.

Figure 3-18: Route Tab



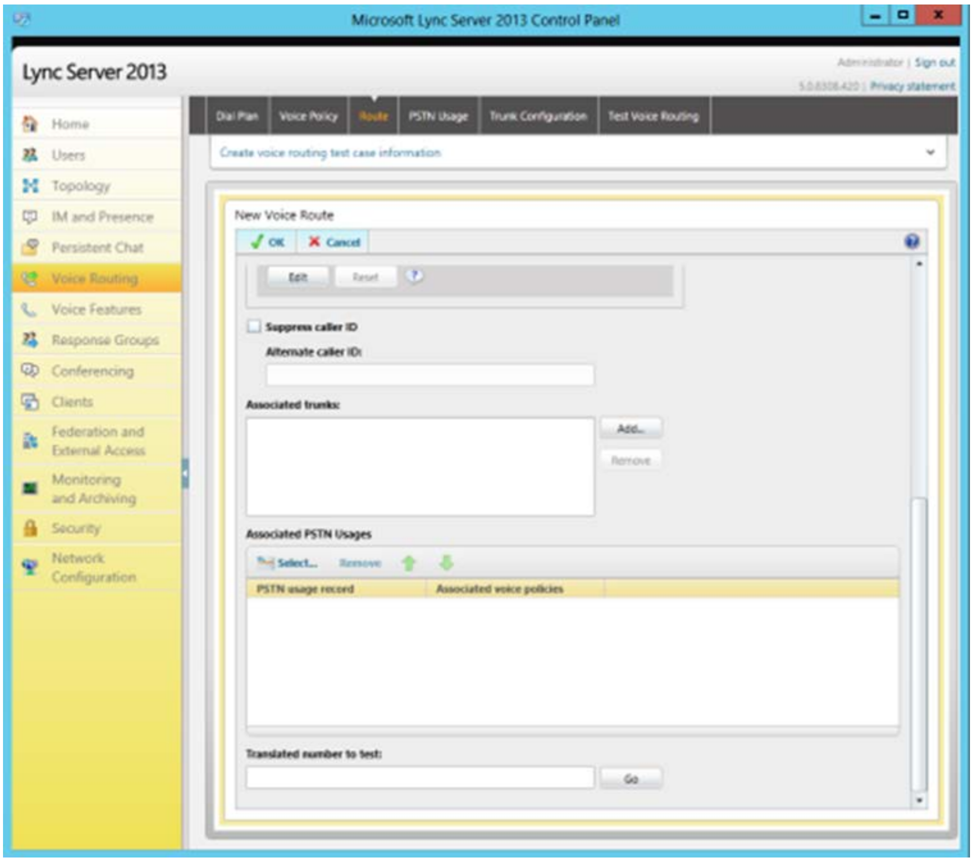
5. Click **New**; the New Voice Route page appears:

Figure 3-19: Adding New Voice Route



6. In the 'Name' field, enter a name for this route (e.g., **SIP Trunk Route**).
7. In the 'Starting digits for numbers that you want to allow' field, enter the starting digits you want this route to handle (e.g., * to match all numbers), and then click **Add**.

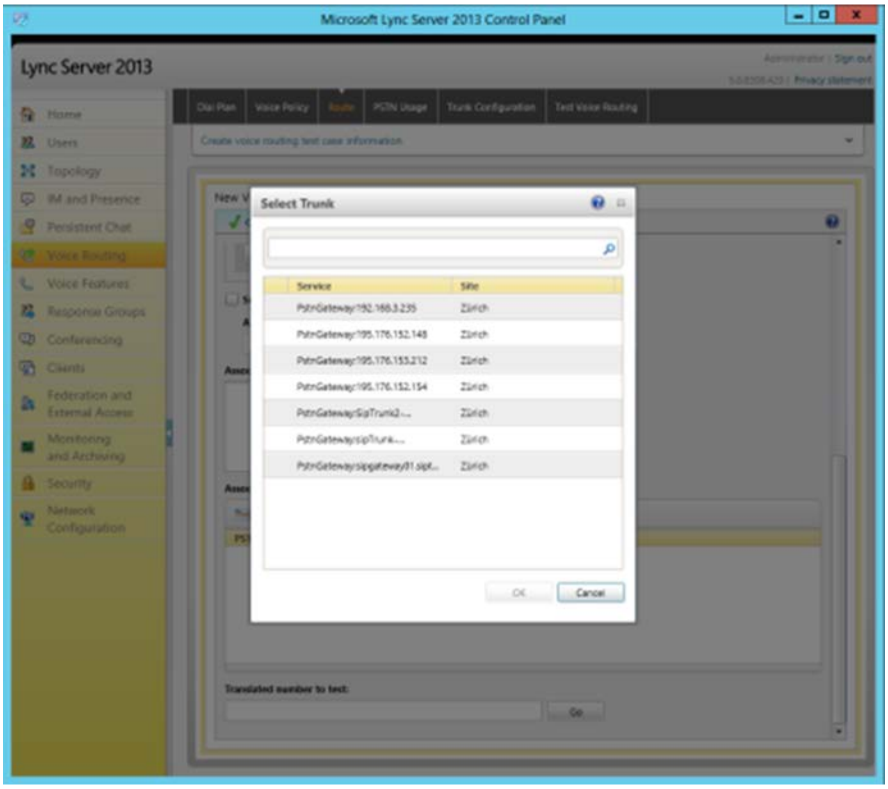
Figure 3-20: Adding New Trunk



8. Associate the route with the E-SBC Trunk that you created:

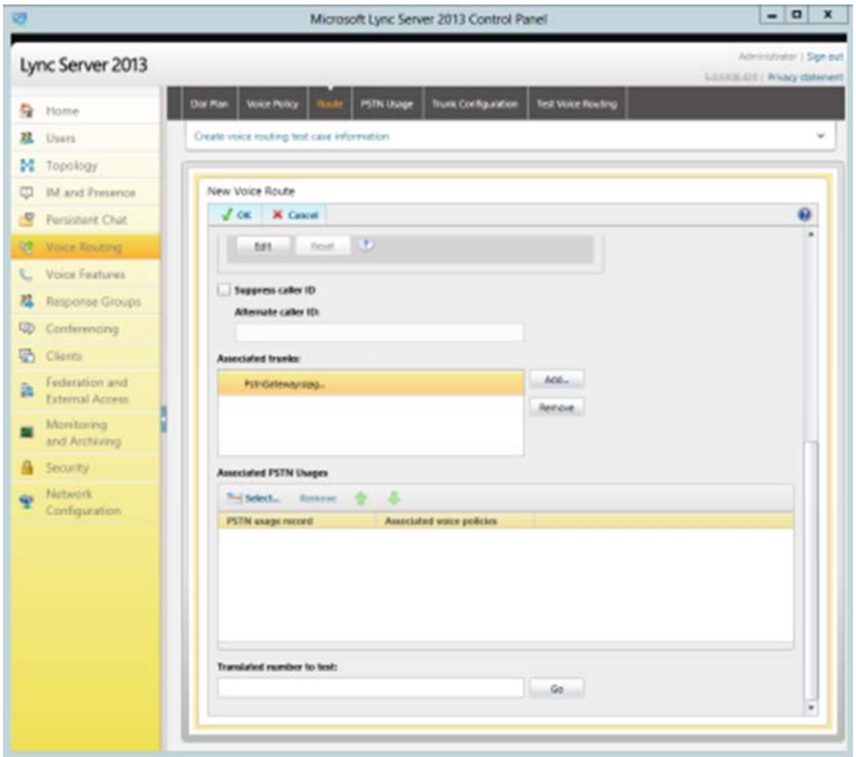
   a. Under the 'Associated Trunks' group, click **Add**; a list of all the deployed gateways is displayed:

Figure 3-21: List of Deployed Trunks



   b. Select the E-SBC Trunk you created, and then click **OK**; the trunk is added to the 'Associated Trunks' group list:

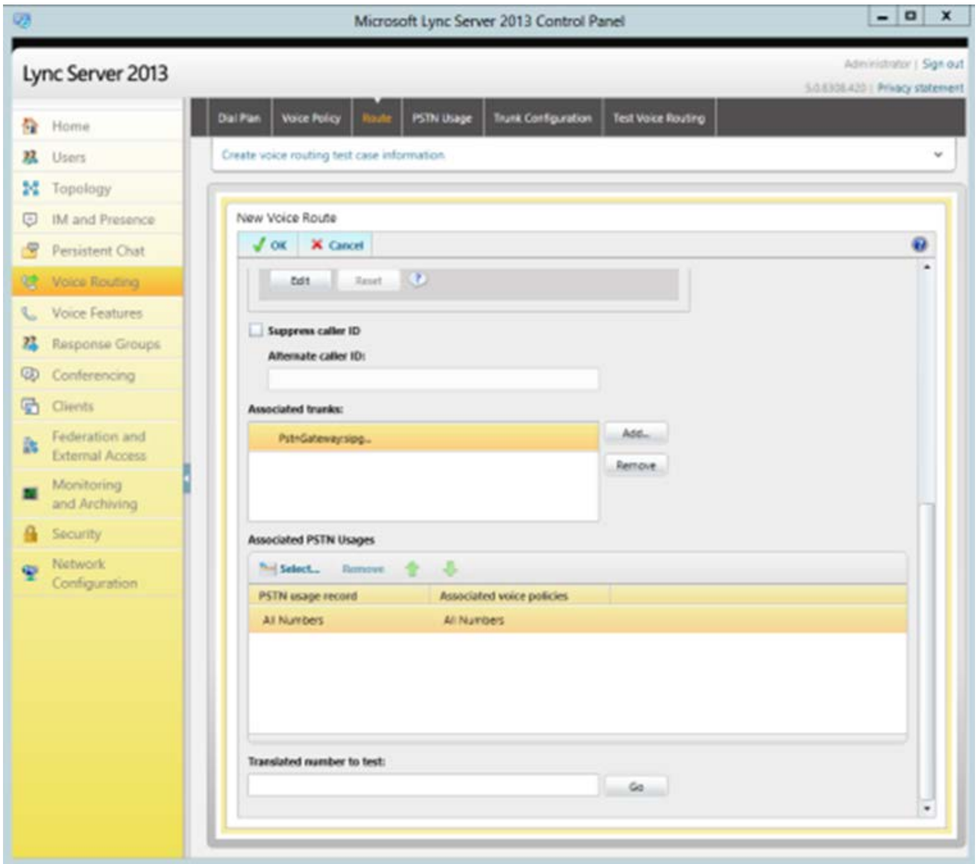
Figure 3-22: Selected E-SBC Trunk



9. Associate a PSTN Usage to this route:

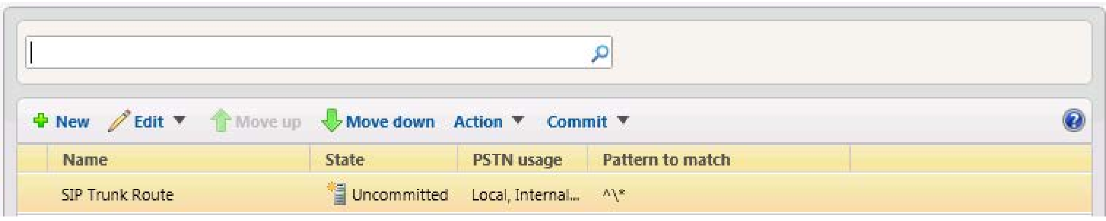
   a. Under the 'Associated PSTN Usages' group, click **Select** and then add the associated PSTN Usage.

Figure 3-23: Associating PSTN Usage to Route



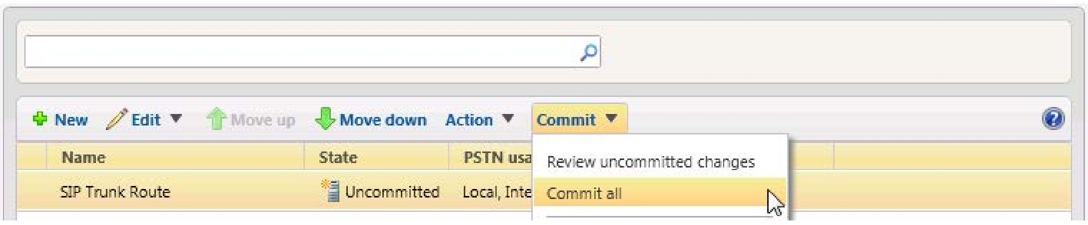
10. Click **OK** (located on the top of the New Voice Route page); the New Voice Route (Uncommitted) is displayed:

Figure 3-24: Confirmation of New Voice Route



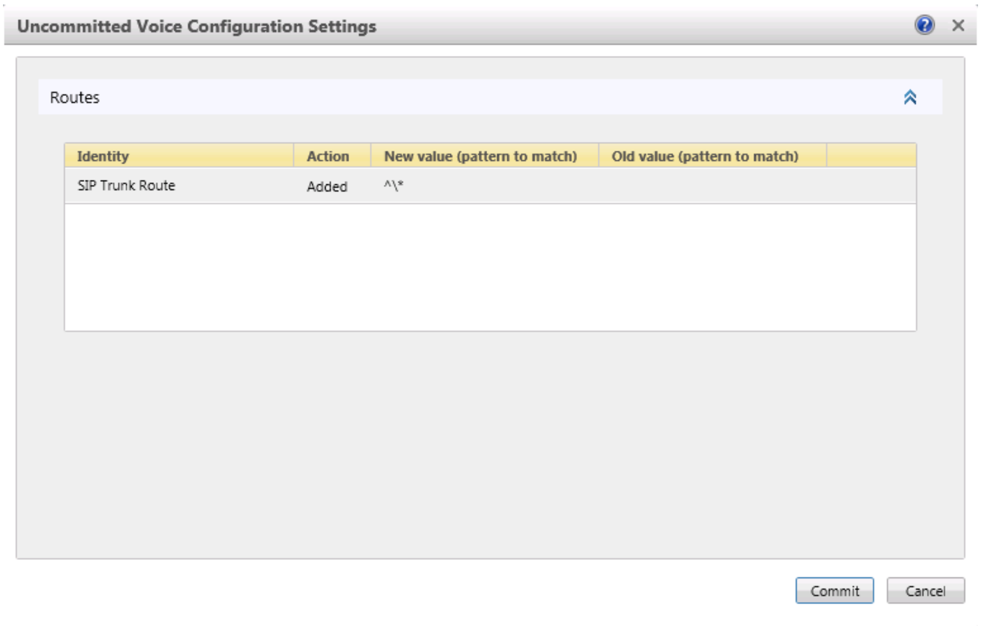
11. From the **Commit** drop-down list, choose **Commit all**, as shown below:

Figure 3-25: Committing Voice Routes



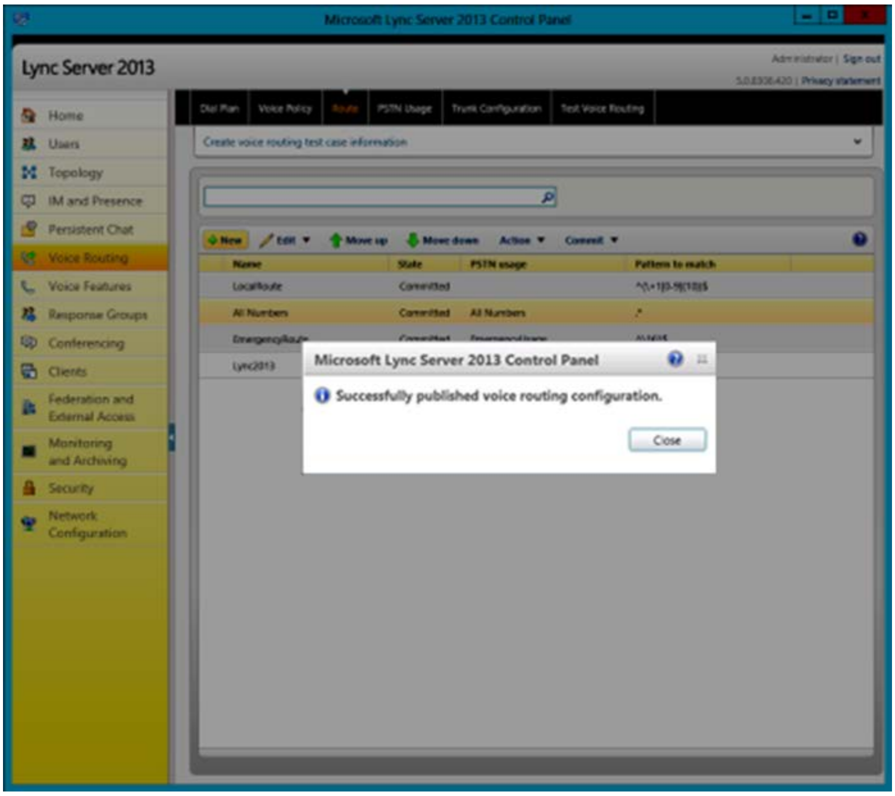
The Uncommitted Voice Configuration Settings page appears:

Figure 3-26: Uncommitted Voice Configuration Settings



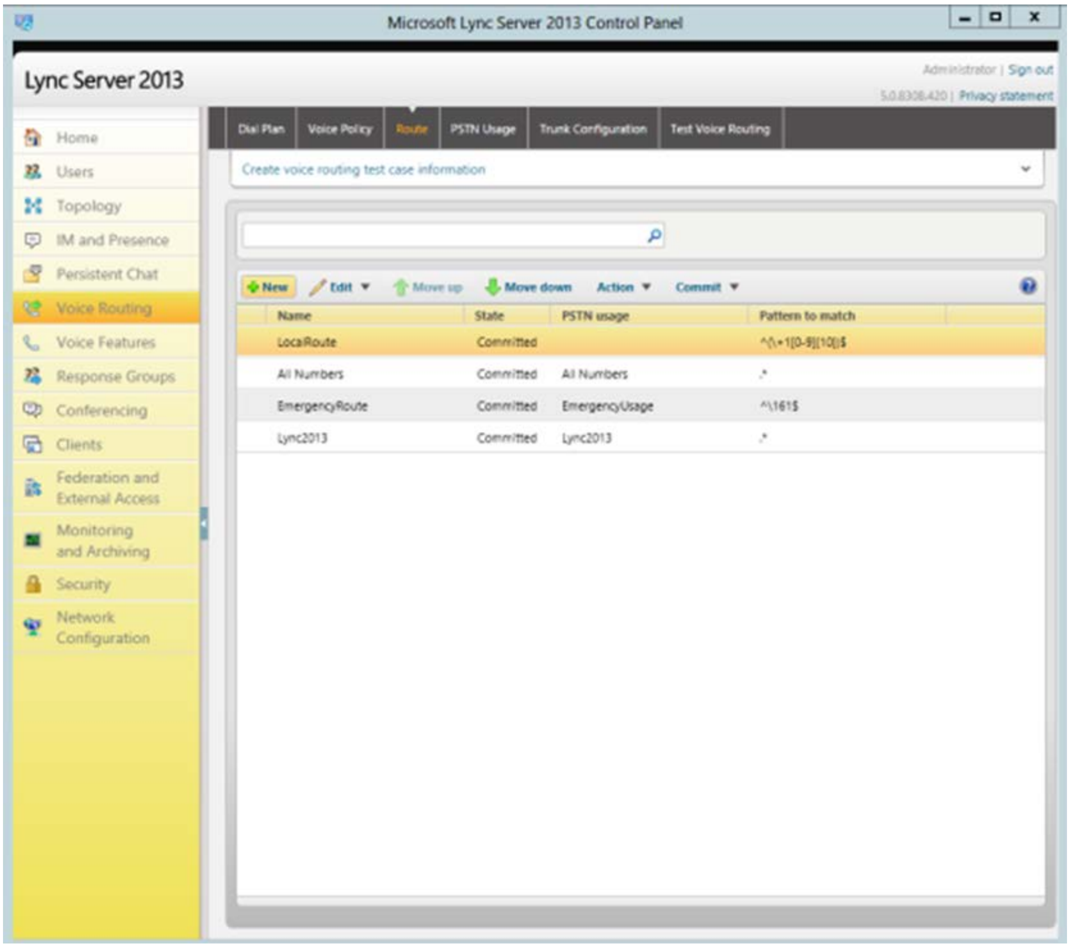
12. Click **Commit**; a message is displayed confirming a successful voice routing configuration, as shown below:

Figure 3-27: Confirmation of Successful Voice Routing Configuration



13. Click **Close**; the new committed Route is displayed in the Voice Routing page, as shown below:

Figure 3-28: Voice Routing Screen Displaying Committed Routes

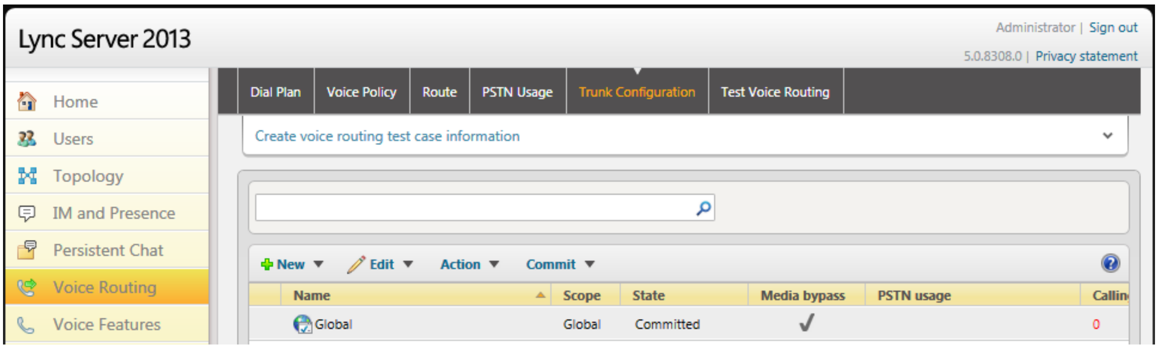


14. For correct interworking with the Swisscom VoIP Gate, continue with the following steps:

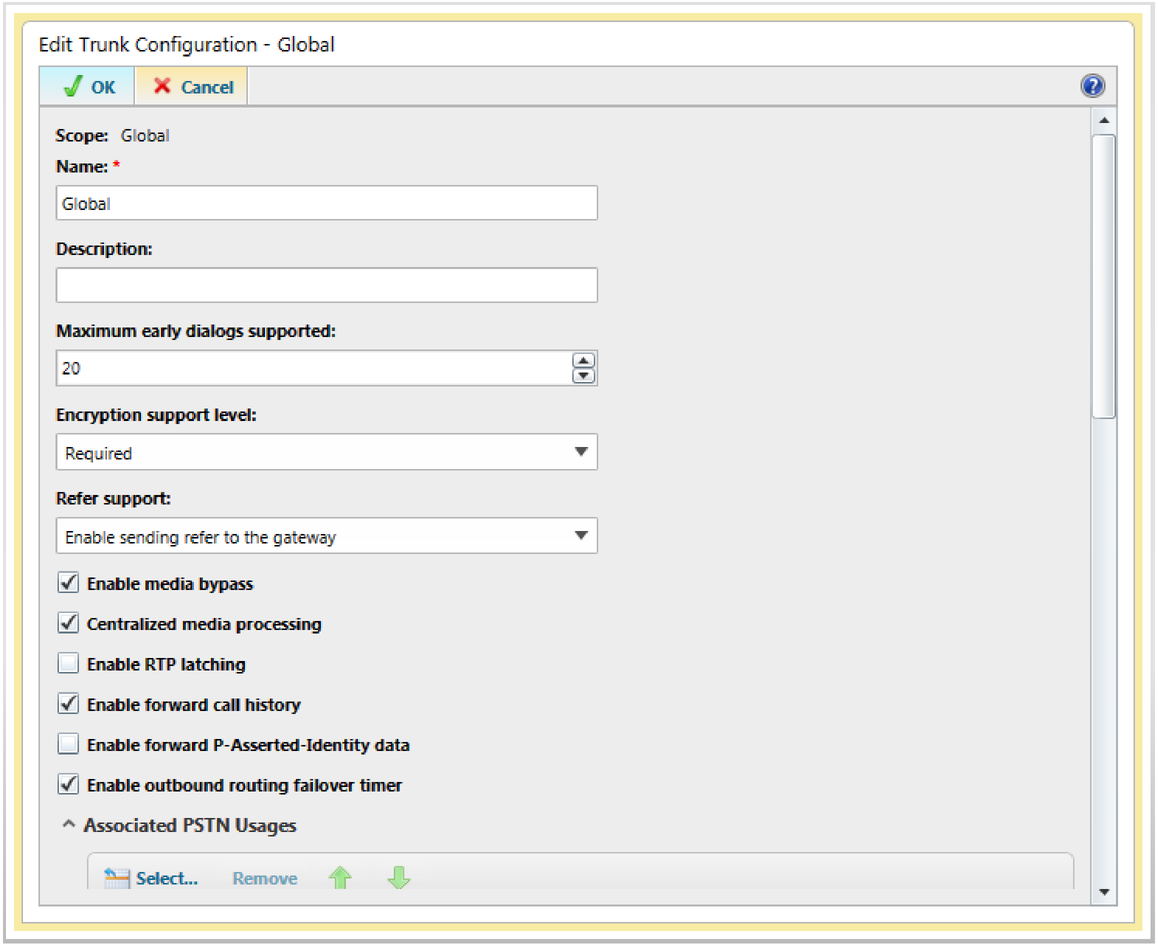
> **Note:** The SIP History-Info header provides a method to verify the identity (ID) of the call forwarder (i.e., the Lync user number). This ID is optional in Lync, but shall remain disabled for use with Swisscom VoIP Gate.

a. In the Voice Routing page, select the **Trunk Configuration** tab. Note that you can add and modify trunk configuration by site or by pool.

Figure 3-29: Voice Routing Screen – Trunk Configuration Tab



b. Click **Edit**; the Edit Trunk Configuration page appears:



c. **De-**select the **Enable forward call history** check box, and then click **OK**.

d. Verify de-activation of RTCP and Session timers. Since these parameters are not visible on the graphical interface, the Lync server Management Shel must be used.

e. Use the following command on the Lync server Management Shell after reconfiguration to verify correct values.

- Get-CsTrunkConfiguration

```
Identity                                  :
sipgateway01.siptest03.local
OutboundTranslationRulesList              : {}
SipResponseCodeTranslationRulesList       : {}
OutboundCallingNumberTranslationRulesList : {}
PstnUsages                                : {}
Description                               :
ConcentratedTopology                      : True
EnableBypass                              : False
EnableMobileTrunkSupport                  : False
EnableReferSupport                        : False
EnableSessionTimer                        : False
EnableSignalBoost                         : False
MaxEarlyDialogs                           : 20
RemovePlusFromUri                         : False
RTCPActiveCalls                           : False
RTCPCallsOnHold                           : False
SRTPMode                                  : Not Supported
EnablePIDFLOSupport                       : False
EnableRTPLatching                         : False
EnableOnlineVoice                         : False
ForwardCallHistory                        : False
Enable3pccRefer                           : False
ForwardPAI                                : False
EnableFastFailoverTimer                   : True
EnableLocationRestriction                 : False
```

Note: Testing has been conducted with the following changes to the default trunk settings:

- EnableBypass, EnableReferSupport,RTCPActiveCalls and RTCPCallsonHold must be set to false
- SRTPMode is set to not supported
- EnableSessionTimer and ForwardCallHistory must be remain on their default value (False)

f. Repeat Steps 11 through 13 to commit your settings.

4 Configuring AudioCodes E-SBC

This chapter provides step-by-step procedures on how to configure AudioCodes E-SBC for interworking between Microsoft Lync Server 2013 and the Swisscom VoIP Gate. These configuration procedures are based on the interoperability test topology described in Section 2.5 on page 11, and includes the following main areas:

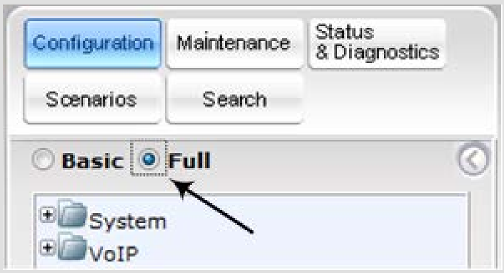
- E-SBC WAN interface - Swisscom VoIP Gate environment
- E-SBC LAN interface - Lync Server 2013 environment including MediaPack Analog Gateways

This configuration is done using the E-SBC's embedded Web server (hereafter, referred to as *Web interface*).

Notes:

- For implementing Microsoft Lync and Swisscom SIP Trunk based on the configuration described in this section, AudioCodes E-SBC must be installed with a Software License Key that includes the following software features:
  - √ **Microsoft**
  - √ **SBC**
  - √ **Security**
  - √ **DSP**
  - √ **RTP**
  - √ **SIP**
  - √ **FEU**

  For more information about the Software License Key, contact your AudioCodes sales representative.
- 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, refer to the *Recommended Security Guidelines* document.
- Before you begin configuring the E-SBC, ensure that the E-SBC's Web interface Navigation tree is in Full-menu display mode. To do this, select the **Full** option, as shown below:



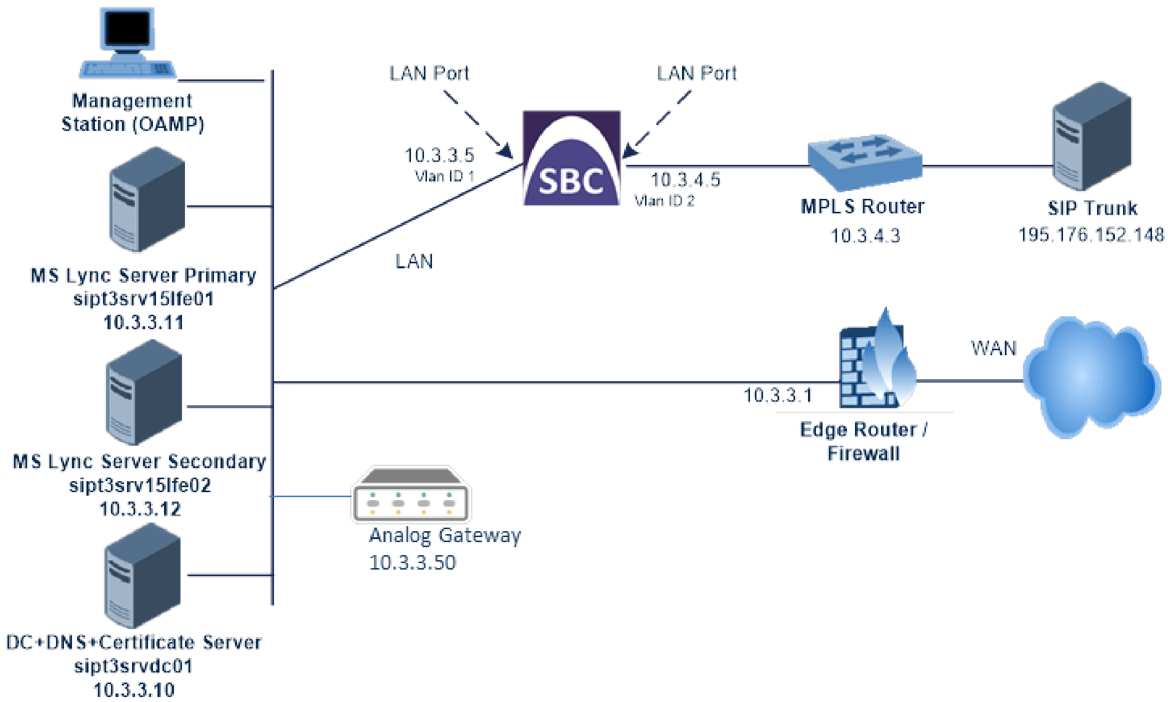
Note that when the E-SBC is reset, the Navigation tree reverts to Basic-menu display.

4.1 Step 1: IP Network Interfaces Configuration

This step describes how to configure the E-SBC's IP network interfaces. There are several ways to deploy the E-SBC; however, this interoperability test topology employs the following deployment method:

- E-SBC interfaces with the following IP entities:
  - Lync servers, located on the LAN
  - MediaPack analog Gateways, located on the LAN
  - Swisscom VoIP Gate, located on the WAN
- E-SBC connects to the WAN through a DMZ network
- Physical connection: The type of physical connection to the LAN depends on the method used to connect to the Enterprise's network. In the interoperability test topology, E-SBC connects to the LAN and WAN using dedicated LAN ports (i.e., two ports and two network cables are used).
- E-SBC also uses two logical network interfaces:
  - LAN (VLAN ID 1)
  - WAN (VLAN ID 2)

Figure 4-1: Network Interfaces in Interoperability Test Topology



4.1.1 Step 1a: Configure Network Interfaces

This step describes how to configure the IP network interfaces for each of the following interfaces:

- LAN VoIP (assigned the name "KUNDE")
- WAN VoIP (assigned the name "SWISSCOM")

➢ **To configure the IP network interfaces:**

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **Network** > **IP Interfaces Table**).
2. Modify the existing LAN network interface:
   a. Select the 'Index' radio button of the **OAMP + Media + Control** table row, and then click **Edit**.
   b. Configure the interface LAN side as follows:

Parameter	Value
IP Address	**10.3.3.5** (IP address of E-SBC)
Prefix Length	**24** (subnet mask in bits for 255.255.255.0)
Gateway	**10.3.3.1** (LAN side default GW)
VLAN ID	**1**
Interface Name	**KUNDE** (arbitrary descriptive name)
Primary DNS Server IP Address	**10.3.3.10** (LAN side DNS)
Underlying Interface	**GROUP_1** (Ethernet port group)

3. Add a network interface for the WAN side:
   a. Enter **1**, and then click **Add Index**.
   b. Configure the interface as follows:

Parameter	Value
Application Type	**Media + Control**
IP Address	**10.3.4.5** (WAN IP address)
Prefix Length	**24** (for 255.255.255.0)
Gateway	**10.3.4.3** (WAN router's IP address)
VLAN ID	**2**
Interface Name	**SWISSCOM**
Primary DNS Server IP Address	**0.0.0.0**
Underlying Interface	**GROUP_2**

4. Click **Apply**, and then **Done**.

The configured IP network interfaces are shown below:

Figure 4-2: Configured Network Interfaces in IP Interfaces Table

IP Interfaces Table

Add Index   Done

Index	Application Type	Interface Mode	IP Address	Prefix Length	Gateway	VLAN ID	Interface Name	Primary DNS Server IP Address	Secondary DNS Server IP Address	Underlying Interface
1	OAMP + Media + Control	IPv4 Manual	10.3.3.5	24	10.3.3.1	1	KUNDE	10.3.3.10	0.0.0.0	GROUP_1
2	Media + Control	IPv4 Manual	10.3.4.5	24	10.3.4.3	2	SWISS	0.0.0.0	0.0.0.0	GROUP_2

4.1.2 Step 1b: Configure the Native VLAN ID

This step describes how to configure the Native VLAN ID for the LAN and WAN interfaces.

➢ **To configure the Native VLAN ID for each network interface:**

1. Open the Physical Ports Settings page (**Configuration** tab> **VoIP** menu > **Network** > **Physical Ports Settings**).
2. For the **GROUP_1** member ports, set the 'Native Vlan' field to **1**. This VLAN was assigned to network interface "KUNDE".
3. For the **GROUP_2** member ports, set the 'Native Vlan' field to **2**. This VLAN was assigned to network interface "SWISSCOM".

Figure 4-3: Configured Port Native VLAN

Index	Port	Mode	Native Vlan	Speed&Duplex	Description	Group Member	Group Status
1	GE_0_1	Enable	1	Auto Negotiation	User Port #0	GROUP_1	Active
2	GE_0_2	Enable	1	Auto Negotiation	User Port #1	GROUP_1	Redundant
3	GE_7_1	Enable	2	Auto Negotiation	User Port #2	GROUP_2	Active
4	GE_7_2	Enable	2	Auto Negotiation	User Port #3	GROUP_2	Redundant

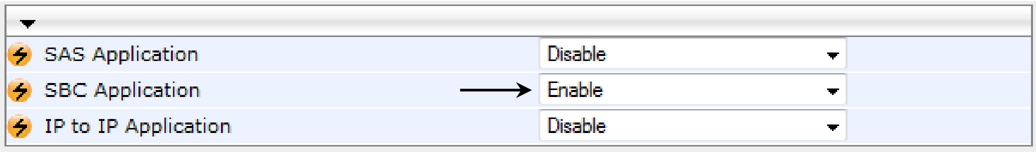
4.2 Step 2: Enable the SBC Application

This step describes how to enable the SBC application.

➢ **To enable the SBC application:**

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

Figure 4-4: Enabling SBC Application



2. From the 'SBC Application' drop-down list, select **Enable**.
3. Click **Submit**.
4. Reset the E-SBC with a burn to flash for this setting to take effect (see Section 4.16 on page 72).

4.3 Step 3: Signaling Routing Domains Configuration

This step describes how to configure Signaling Routing Domains (SRD). The SRD represents a logical VoIP network. Each logical or physical connection requires an SRD, for example, since this configuration note is based on the E-SBC interfaces having physically separated LAN and WAN Ethernet ports, a different SRD is required for each one.

The SRD is composed of the following:

- Media Realm: defines a UDP port range for RTP/SRTP (media) traffic on a specific logical IP network interface of the E-SBC.
- SIP Interface: defines a listening port and type (UDP, TCP, or TLS) for SIP signaling traffic on a specific logical IP network interface of the E-SBC.

4.3.1 Step 3a: Configure Media Realms

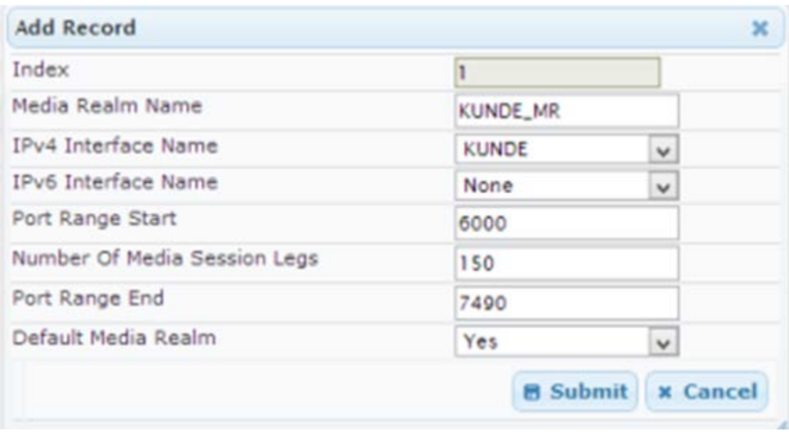
This step describes how to configure Media Realms. The simplest configuration is to create two Media Realms - one for internal (LAN) traffic and one for external (WAN) traffic.

➢ **To configure Media Realms:**

1. Open the Media Realm Table page (**Configuration** tab > **VoIP** menu > **Media** > **Media Realm Table**).
2. Configure a Media Realm for LAN traffic:

Parameter	Value
Index	**1**
Media Realm Name	**KUNDE_MR** (descriptive name)
IPv4 Interface Name	**KUNDE**
Port Range Start	**6000** (represents lowest UDP port number used for media on LAN)
Number of Media Session Legs	**150** (media sessions assigned with port range)

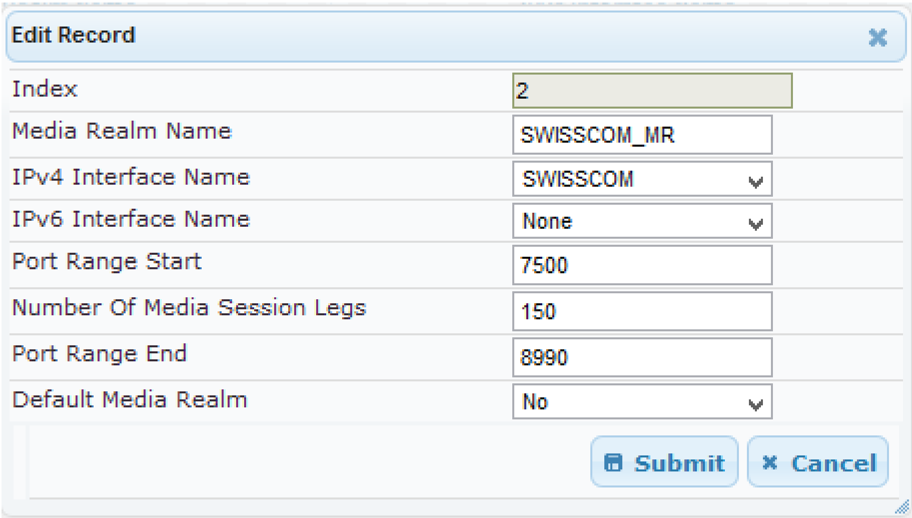
Figure 4-5: Configuring Media Realm for LAN



3. Configure a Media Realm for WAN traffic:

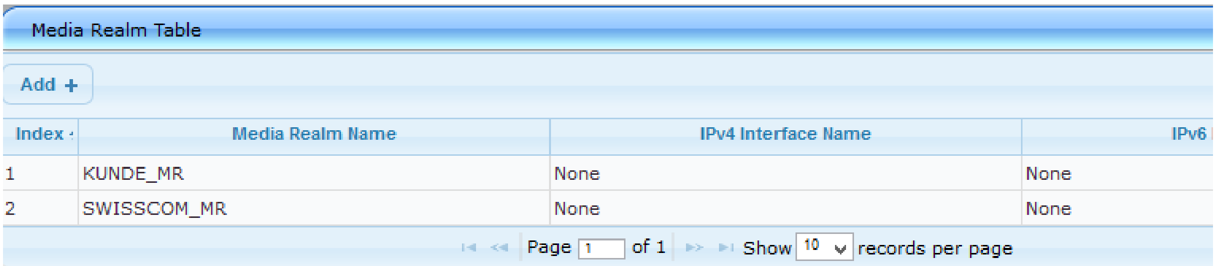
Parameter	Value
Index	**2**
Media Realm Name	**SWISSCOM_MR** (arbitrary name)
IPv4 Interface Name	**SWISSCOM**
Port Range Start	**7500** (represents lowest UDP port number used for media on WAN)
Number of Media Session Legs	**150** (media sessions assigned with port range)

Figure 4-6: Configuring Media Realm for WAN



The configured Media Realms are shown in the figure below:

Figure 4-7: Configured Media Realms in Media Realm Table



4.3.2 Step 3b: Configure SRDs

This step describes how to configure the SRDs.

➢ **To configure SRDs:**

1. Open the SRD Settings page (**Configuration** tab > **VoIP** menu > **Control Network** > **SRD Table**).
2. Configure an SRD for the E-SBC's internal interface (toward Lync Server 2013):

Parameter	Value
SRD Index	**1**
SRD Name	**KUNDE_SRD** (descriptive name for SRD)
Media Realm	**KUNDE_MR** (associates SRD with Media Realm)

Figure 4-8: Configuring LAN SRD

3. Configure an SRD for the E-SBC's external interface (toward the Swisscom VoIP Gate):

Parameter	Value
SRD Index	**2**
SRD Name	**SWISSCOM_SRD**
Media Realm	**SWISSCOM_MR**

Figure 4-9: Configuring WAN SRD

4.3.3 Step 3c: Configure SIP Signaling Interfaces

This step describes how to configure SIP Interfaces. For the interoperability test topology, an internal and external SIP Interface must be configured for the E-SBC.

➢ **To configure SIP Interfaces:**

1. Open the SIP Interface Table page (**Configuration** tab > **VoIP** menu > **Control Network** > **SIP Interface Table**).
2. Configure a SIP interface for the LAN:

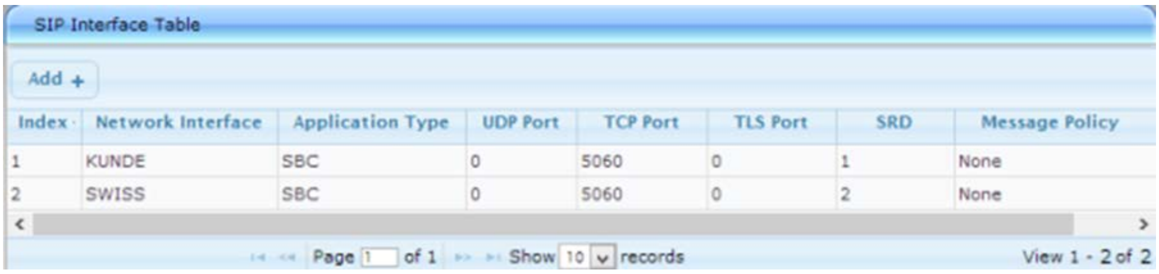
Parameter	Value
Index	**1**
Network Interface	**KUNDE**
Application Type	**SBC**
UDP and TLS Port	**0**
TCP Port	**5060**
SRD	**1**

3. Configure a SIP interface for the WAN:

Parameter	Value
Index	**2**
Network Interface	**SWISSCOM**
Application Type	**SBC**
UDP and TLS Port	**0**
TCP Port	**5060**
SRD	**2**

The configured SIP Interfaces are shown in the figure below:

Figure 4-10: Configured SIP Interfaces in SIP Interface Table



SIP Interface Table

Add +

Index	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	SRD	Message Policy
1	KUNDE	SBC	0	5060	0	1	None
2	SWISS	SBC	0	5060	0	2	None

Page 1 of 1 Show 10 records View 1 - 2 of 2

4.4 Step 4: Configure Proxy Sets

This step describes how to configure Proxy Sets. The Proxy Set defines the destination address (IP address or FQDN) of the IP entity server. Proxy Sets can also be used to configure load balancing between multiple servers.

For the interoperability test topology, two Proxy Sets need to be configured for the following IP entities:

- Microsoft Lync Server 2013
- Swisscom VoIP Gate

These Proxy Sets will later be associated with IP Groups.

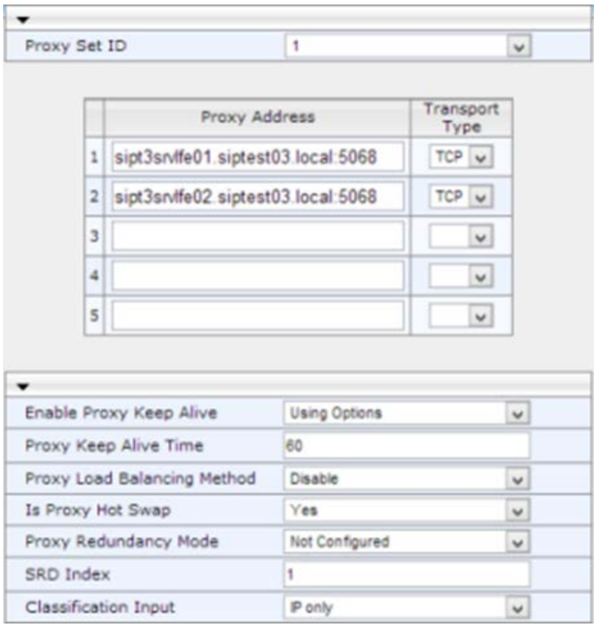
Note: The IP address of the VoIP Gate SIP proxy will be delivered by Swisscom.

➢ **To configure Proxy Sets:**

1. Open the Proxy Sets Table page (**Configuration** tab > **VoIP** menu > **Control Network** > **Proxy Sets Table**).
2. Configure a Proxy Set for Lync Server 2013:

Parameter	Value
Proxy Set ID	**1**
Proxy Address	**sipt3srvlfe01.siptest03.local:5068** (Lync Server 2013 IP address / FQDN and destination port)
Transport Type	**TCP**
Enable Proxy Keep Alive	**Using Options**
Proxy Load Balancing Method	**Disable**
Is Proxy Hot Swap	**Yes**
SRD Index	**1**

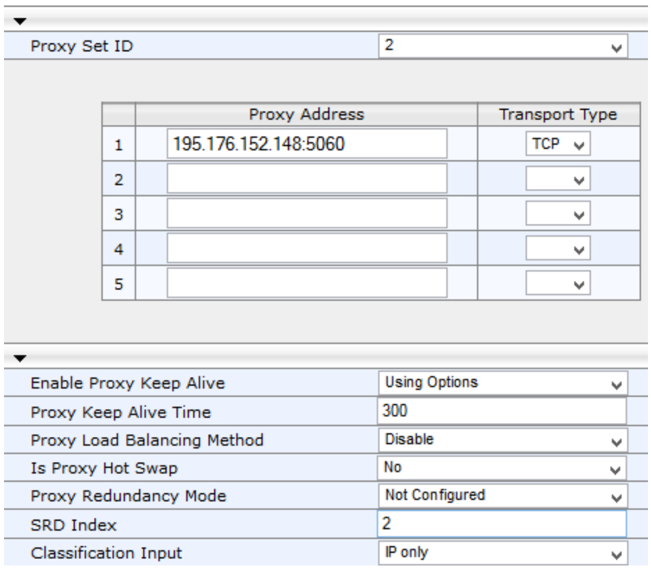
Figure 4-11: Configuring Proxy Set for Microsoft Lync Server 2013



3. Configure a Proxy Set for the Swisscom VoIP Gate:

Parameter	Value
Proxy Set ID	**2**
Proxy Address	**nn.nn.nn.nn:5060** (Swisscom IP address or FQDN including destination port)
Transport Type	**TCP**
Enable Proxy Keep Alive	**Using Options**
Is Proxy Hot Swap	**No**
SRD Index	**2** (enables classification by Proxy Set for SRD of IP Group belonging to Swisscom VoIP Gate)

Figure 4-12: Configuring Proxy Set for Swisscom VoIP Gate



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4.5 Step 5: Configure IP Groups

This step describes how to configure IP Groups. The IP Group represents an IP entity on the network with which the E-SBC communicates. This can be a server (e.g., IP PBX or ITSP) or it can be a group of users (e.g., LAN IP phones). For servers, the IP Group is typically used to define the server's IP address by associating it with a Proxy Set. A typical deployment consists of multiple IP Groups associated with the same SRD. For example, you can have two LAN IP PBXs sharing the same SRD, and two ITSPs / SIP Trunks sharing the same SRD. Once IP Groups are configured, they are used to configure IP-to-IP routing rules for denoting source and destination of the call.

In this interoperability test topology, IP Groups must be configured for the following IP entities:

- Lync Server 2013 (Mediation Server) located on LAN
- Swisscom VoIP Gate located on WAN
- MediaPack Analog Gateway located on LAN

➢ **To configure IP Groups:**

1. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **Control Network** > **IP Group Table**).
2. Configure an IP Group for the Lync Server 2013 Mediation Server:

Parameter	Value
Index	**1**
Type	**Server**
Description	**Lync** (arbitrary descriptive name)
Proxy Set ID	**1**
SIP Group Name	
SRD	**1**
Media Realm Name	**KUNDE_MR**
IP Profile ID	**1**
Outbound MessageManipulationSet	**1**

3. Configure an IP Group for the Swisscom VoIP Gate:

Parameter	Value
Index	**2**
Type	**Server**
Description	**Swisscom** (arbitrary descriptive name)
Proxy Set ID	**2**
SIP Group Name	**nn.nn.nn.nn**
SRD	**2**
Media Realm Name	**SWISSCOM_MR**
IP Profile ID	**2**
Outbound MessageManipulationSet	**2**

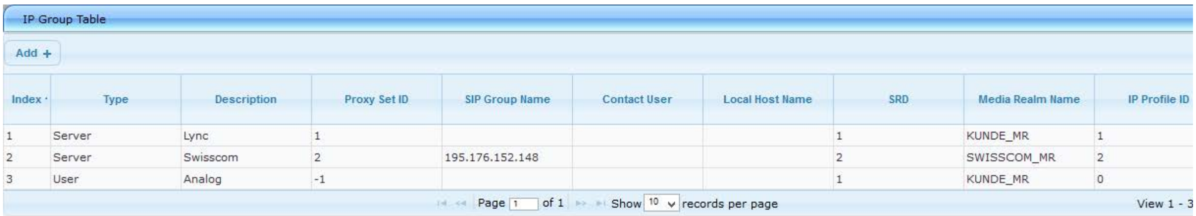
Note: The SIP Group Name of the VoIP Gate IP Group is identical to the IP address of the SIP proxy which will be delivered by Swisscom.

4. The Configure an IP Group for the MediaPack Analog Gateways:

Parameter	Value
Index	**3**
Type	**User** (the MediaPack gateway will dynamically register to the e-SBC)
Description	**Analog** (arbitrary descriptive name)
Proxy Set ID	**N/A**
SIP Group Name	
SRD	**1**
Media Realm Name	**KUNDE_MR**
IP Profile ID	**0**
Classify By Proxy Set	**Disable**

The configured IP Groups are shown in the figure below:

Figure 4-13: Configured IP Groups in IP Group Table



IP Group Table

Add +

Index	Type	Description	Proxy Set ID	SIP Group Name	Contact User	Local Host Name	SRD	Media Realm Name	IP Profile ID
1	Server	Lync	1				1	KUNDE_MR	1
2	Server	Swisscom	2	195.176.152.148			2	SWISSCOM_MR	2
3	User	Analog	-1				1	KUNDE_MR	0

Page 1 of 1 Show 10 records per page View 1 - 3

IP Group 3 will require classification rules for proper operation. The classification is based on the hostname used in the SIP signaling received from the MediaPack as well as the source username prefix applicable to all analog devices connected to the MediaPack.

The configuration of the classification is explained in Step 6.

4.6 Step 6: Configure the Condition Table

This step describes how to configure the Condition Table. The Condition Table allows advanced verifications of incoming SIP Signaling which can be used for various purposes. During this interoperability test a condition is required to assist in classification of incoming SIP signaling from the MediaPack Analog Gateways into the correct IPGroup 3. Refer to the next section for the use of this condition in a Classification rule.

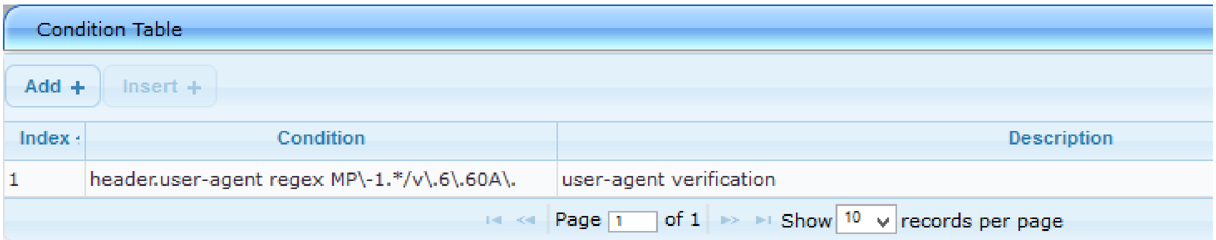
➢ **To configure the Message Condition Table:**

1. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **Condition Table**).
2. Configure a Condition for the MediaPack analog Gateway:

Parameter	Value
Index	**1**
Condition	**header.user-agent regex MP\-1.*/v\.6\.60A\.**
Description	**user-agent verification**

The configured Condition Table is shown in the figure below:

Figure 4-14: Configured rules in Condition Table



4.7 Step 7: Configure the Classification Table

This step describes how to configure the Classification Table. The Classification Table is used to assign incoming SIP signaling to an IP Group. In cases where incoming SIP signaling is always using the IP addresses specified in the proxy set attached to the IP Group, the classification can be done on proxy set level, and does not require use of this classification table. Such classification mechanism is used for IP Groups 1 and 2, for signaling from the Lync Server 2013 and Swisscom VoIP Gate respectively.

The MediaPack connects to an IP group of type user through dynamic registration which means that the E-SBC cannot classify its SIP signaling based through a proxy set. Instead, a classification rule uses the SIP hostname as a first step in the classification. Since E-SBC and MediaPack share the same SIP hostname in the Lync Topology, the previously created message condition is added for proper recognition of MediaPack communications.

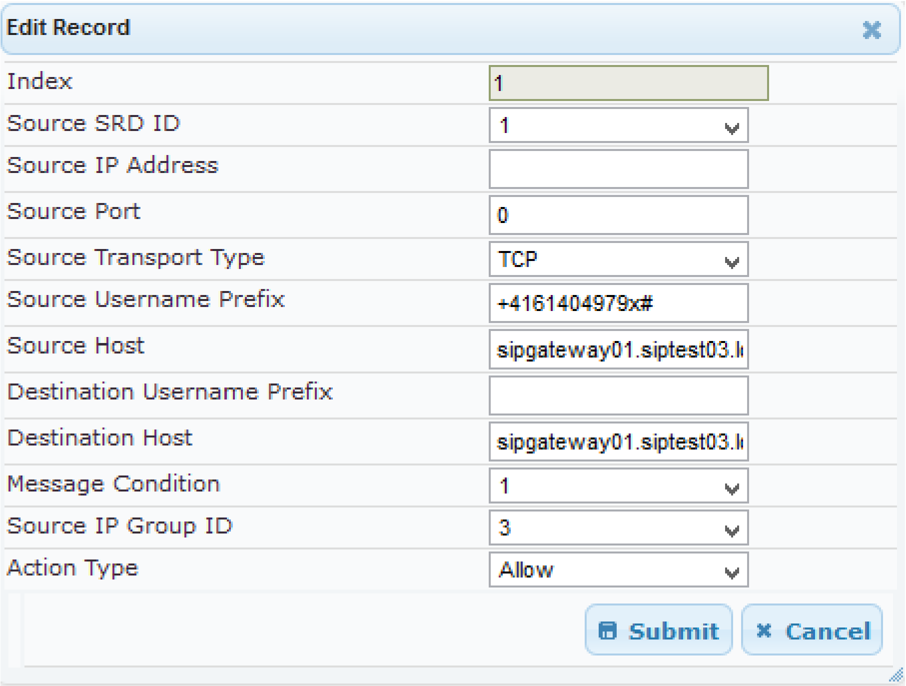
➢ **To configure the Classification Table:**

1. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **Classification Table**).
2. Configure an Classification rule for the MediaPack analog Gateway:

Parameter	Value
Index	**1**
Source SRD ID	**1**
Source Transport Type	**TCP**
Source Username Prefix	**+4161404979x#** (a prefix common to all analog devices connected to the MediaPack)
Source Host	**sipgateway01.siptest03.local** (the FQDN name used by the E-SBC in the Lync Topology is also used by the MediaPack in its SIP signaling)
Destination Host	**sipgateway01.siptest03.local**
Message Condition	**1**
Source IP Group ID	**3**

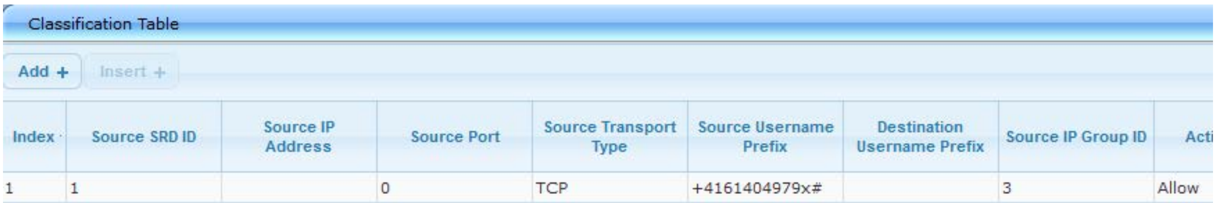
The screen for entering a Classification is shown in the figure below:

Figure 4-15: Classification Table entry



The configured Classification is shown in the figure below:

Figure 4-16: Configured rules in Classification Table



Index	Source SRD ID	Source IP Address	Source Port	Source Transport Type	Source Username Prefix	Destination Username Prefix	Source IP Group ID	Acti
1	1		0	TCP	+4161404979x#		3	Allow

4.8 Step 8: Configure IP Profiles

This step describes how to configure IP Profiles. The IP Profile defines a set of call capabilities relating to signaling (e.g., SIP message terminations such as REFER) and media (e.g., coder and transcoding method).

In this interoperability test topology, IP Profiles need to be configured for the following IP entities:

- Microsoft Lync Server 2013 - to operate in non-secure mode using RTP/SRTP transparency and TCP
- Swisscom VoIP Gate - to operate in non-secure mode using RTP/SRTP transparency and TCP

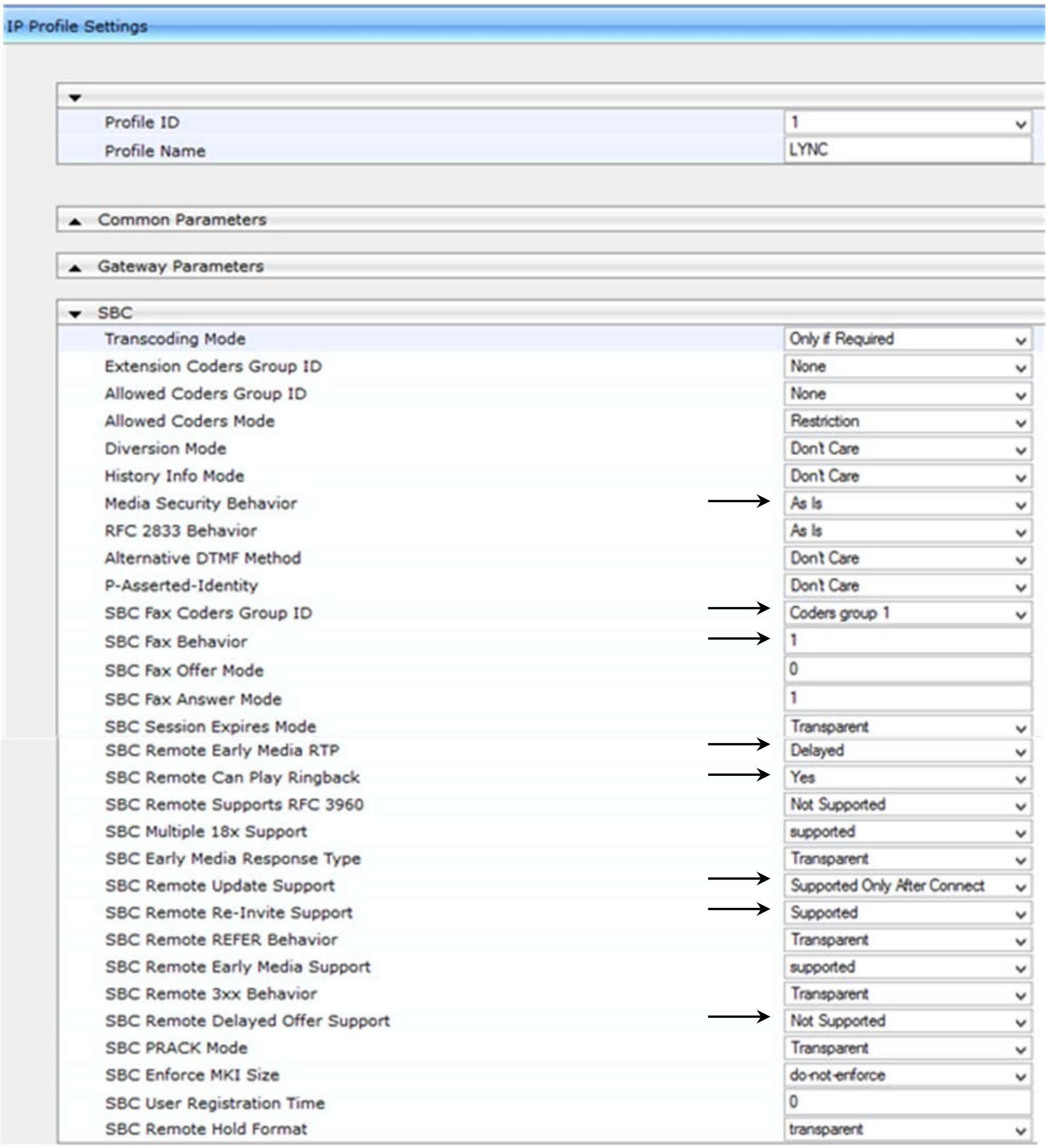
Note: The IP Profiles were assigned to these entities (i.e., IP Groups) in the previous step (see Section 4.5 on page 45).

To configure IP Profiles:

1. Open the IP Profile Settings page (**Configuration** tab > **VoIP** > **Coders and Profiles** > **IP Profile Settings**).
2. Configure an IP Profile for Lync Server 2013:

Parameter	Value
Profile ID	**1**
Media Security Behavior	**As Is**
SBC Fax Coders Group ID	**Coders Group 1** **(**this must point to a non-existing coders group. The mechanism is used to remove T.38 from any communications towards Lync 2013)
SBC Fax Behavior	**1**
SBC Remote Early Media RTP	**Delayed** (required, as Lync Server 2013 does not send RTP immediately to remote side when it sends a SIP 18x response)
SBC Remote Can Play Ringback	**yes**
SBC Remote Update Support	**Supported Only After Connect**
SBC Remote Early Media Support	**Supported**
SBC Remote Delayed Offer Support	**Not Supported**

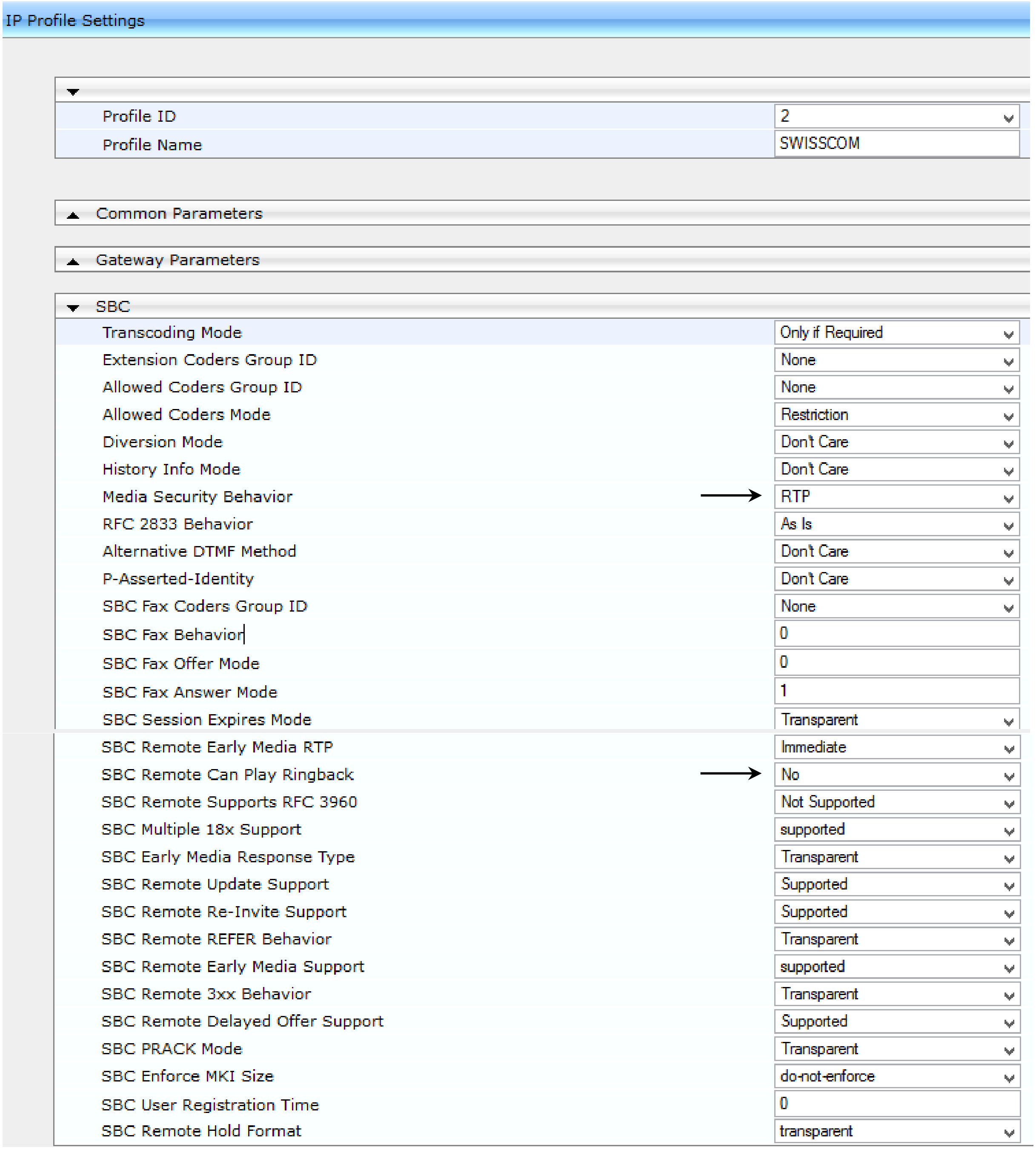
Figure 4-17: Configuring IP Profile for Lync Server 2013



3. Configure an IP Profile for the Swisscom VoIP Gate:

Parameter	Value
Profile ID	**2**
Media Security Behavior	**RTP**
SBC Remote Can Play Ringback	**No**

Figure 4-18: Configuring IP Profile for Swisscom VoIP Gate



4.9 Step 9: Configure the NTP Server Address

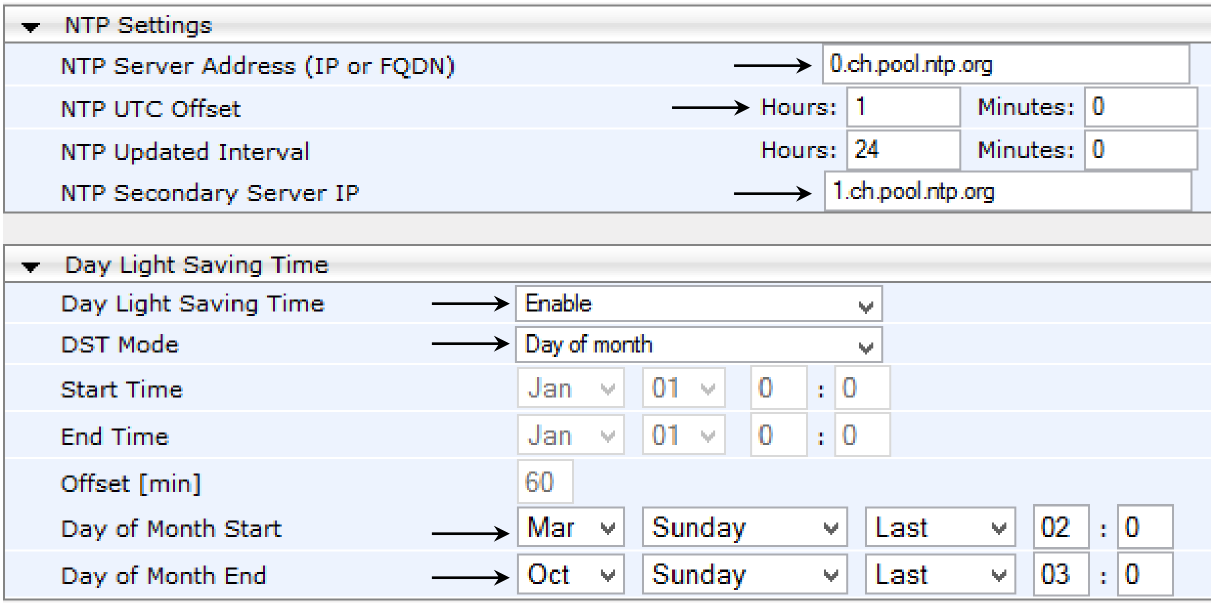
This step describes how to configure the NTP server's IP address. It is recommended to implement an NTP server (Microsoft NTP server or a third-party server) to ensure that the E-SBC receives the accurate and current date and time. This is an optional step in the current configuration, but would be a mandatory step for validating certificates of remote parties when TLS is used.

➢ **To configure the NTP server address:**

1. Open the Application Settings page (**Configuration** tab > **System** menu > **Application Settings**).
2. Configure the following parameters:

Parameter	Value
NTP Server Address (IP or FQDN)	**0.ch.pool.ntp.org**
NTP UTC Offset	**1** (Hours)
NTP Secondary Server IP	**1.ch.pool.ntp.org**
Day Light Saving Time	**Enable**
DST Mode	**Day of month**
Day of Month Start	**Mar Sunday Last 02 00**
Day of Month End	**Oct Sunday Last 03 00**

Figure 4-19: Configuring NTP Settings and Day Light Saving Time



3. Click **Submit**.

4.10 Step 10: Configure SRTP

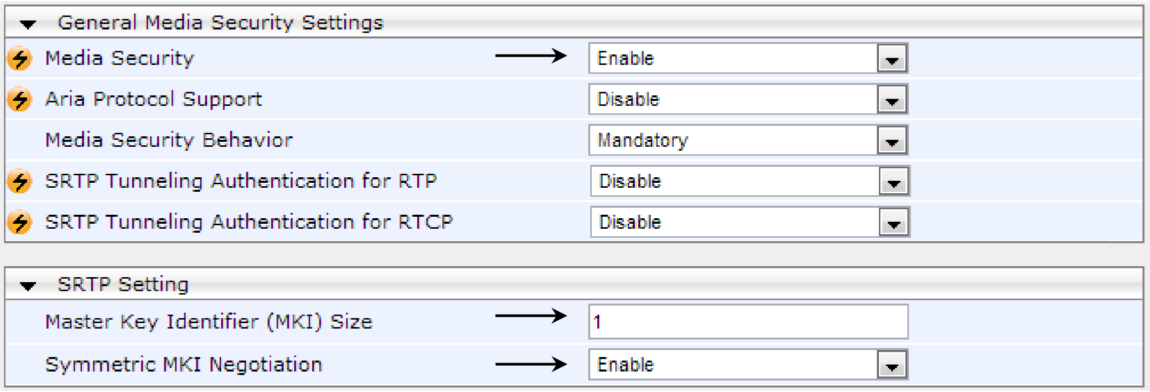
This step describes how to configure media security. If you configure the Microsoft Mediation Server to use SRTP, you need to configure the E-SBC to operate in the same manner. Note that SRTP was enabled for Lync Server 2013 when you configured an IP Profile for Lync Server 2013 (see Section 4.8 on page 50).

➢ **To configure media security:**

1. Open the Media Security page (**Configuration** tab > **Media** menu > **Media Security**).
2. Configure the parameters as follows:

Parameter	Value
Media Security	**Enable**
Master Key Identifier (MKI) Size	**1**
Symmetric MKI Negotiation	**Enable**

Figure 4-20: Configuring SRTP



3. Click **Submit**.
4. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.16 on page 72).

4.11 Step 11: Configure Maximum IP Media Channels

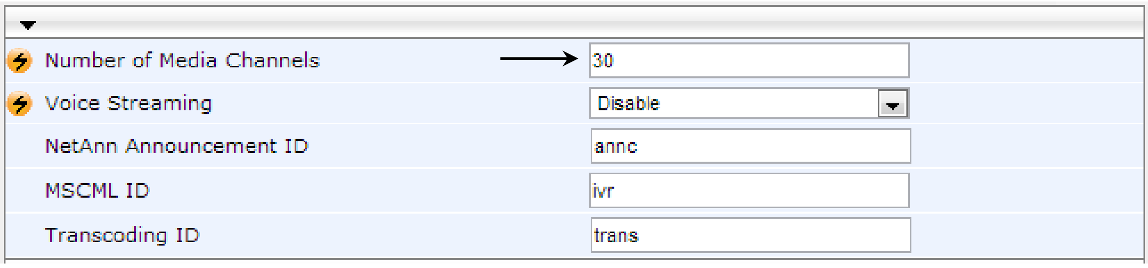
This step describes how to configure the maximum number of required IP media channels. The number of media channels represents the number of DSP channels that the E-SBC allocates to call sessions.

> **Note:** This step is required **only** if transcoding is required.

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

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

Figure 4-21: Configuring Number of IP Media Channels



2. In the 'Number of Media Channels' field, enter the number of media channels according to your environments transcoding calls (e.g., **30**).
3. Click **Submit**.
4. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.16 on page 72).

4.12 Step 12: Configure IP-to-IP Call Routing Rules

This step describes how to configure IP-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to another. The E-SBC selects the rule whose configured input characteristics (e.g., IP Group) match those of the incoming SIP message. If the input characteristics do not match the first rule in the table, they are compared to the second rule, and so on, until a matching rule is located. If no rule is matched, the message is rejected. The routing rules use the configured IP Groups to denote the source and destination of the call. As configured in Section 4.5 on page 45, IP Group 1 represents Lync Server 2013, IP Group 2 represents Swisscom VoIP Gate and IP Group 3 represents the MediaPack Analog Gateway(s).

For the interoperability test topology, the following IP-to-IP routing rules need to be configured to route calls between Lync Server 2013 (LAN), the MediaPack Analog Gateway and the Swisscom VoIP Gate (WAN):

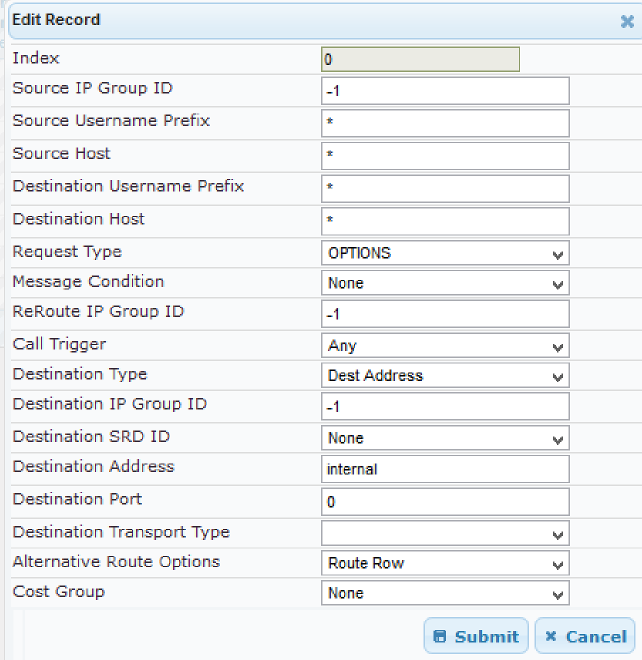
- Terminate SIP OPTIONS messages on the E-SBC that are received from the LAN
- Terminate SIP REGISTER messages on the E-SBC that are received from the connected
- Calls from Lync Server 2013 are first tested against the (internal) registration database with known analog endpoints, if no match is found the calls use an alternative route to the Swisscom VoIP Gate
- Calls from Swisscom VoIP Gate are first send to the Lync Server 2013, a 404 “not found” SIP response will re-route the call towards the MediaPack Analog Gateway,
- Calls from analog endpoints are first routed to the Lync Server 2013, a 404 “not found” SIP response will re-route the call towards the Swisscom VoIP Gate

➢ **To configure IP-to-IP routing rules:**

1. Open the IP-to-IP Routing Table page (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **IP-to-IP Routing Table**).
2. Configure a rule to terminate SIP OPTIONS messages received from the LAN:

Parameter	Value
Index	**0**
Source IP Group ID	**-1**
Request Type	**OPTIONS**
Destination Type	**Dest Address**
Destination Address	**internal**

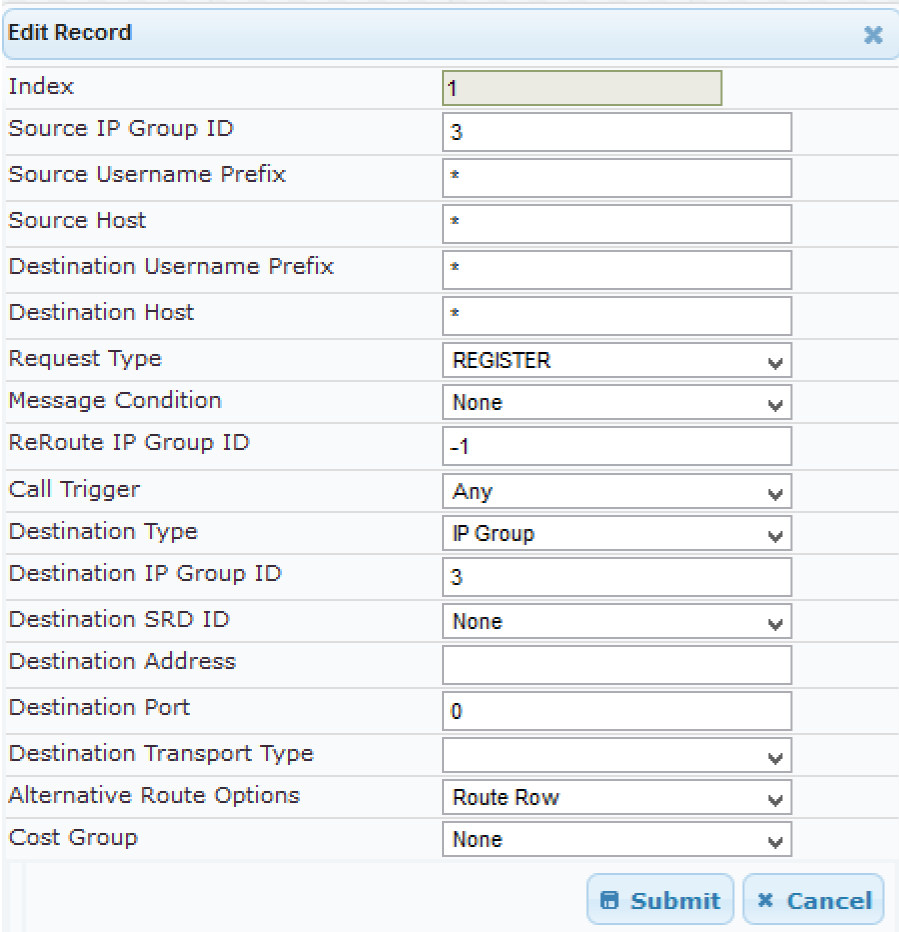
Figure 4-22: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS from LAN



3. Configure a rule to handle REGISTER messages from the MediaPack Analog gateway internally on the E-SBC:

Parameter	Value
Index	**1**
Source IP Group ID	**3**
Request Type	**REGISTER**
Destination Type	**IP Group**
Destination IP Group ID	**3**

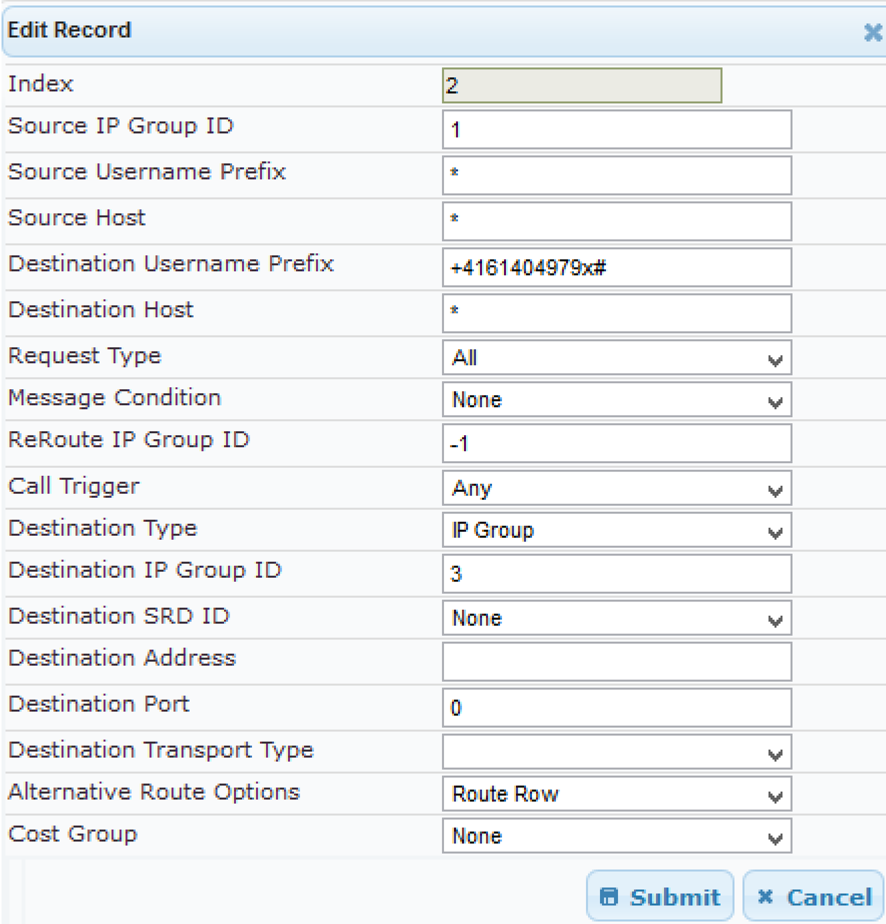
Figure 4-23: Configuring IP-to-IP Routing Rule for MediaPack REGISTER messages



4. Configure a rule to route calls from Lync Server 2013 to the analog gateway:

Parameter	Value
Index	**2**
Source IP Group ID	**1**
Destination Username Prefix	**+4161404979x#** Note: this prefix must match all analog endpoints, if required multiple rules can be used
Destination Type	**IP Group**
Destination IP Group ID	**3**

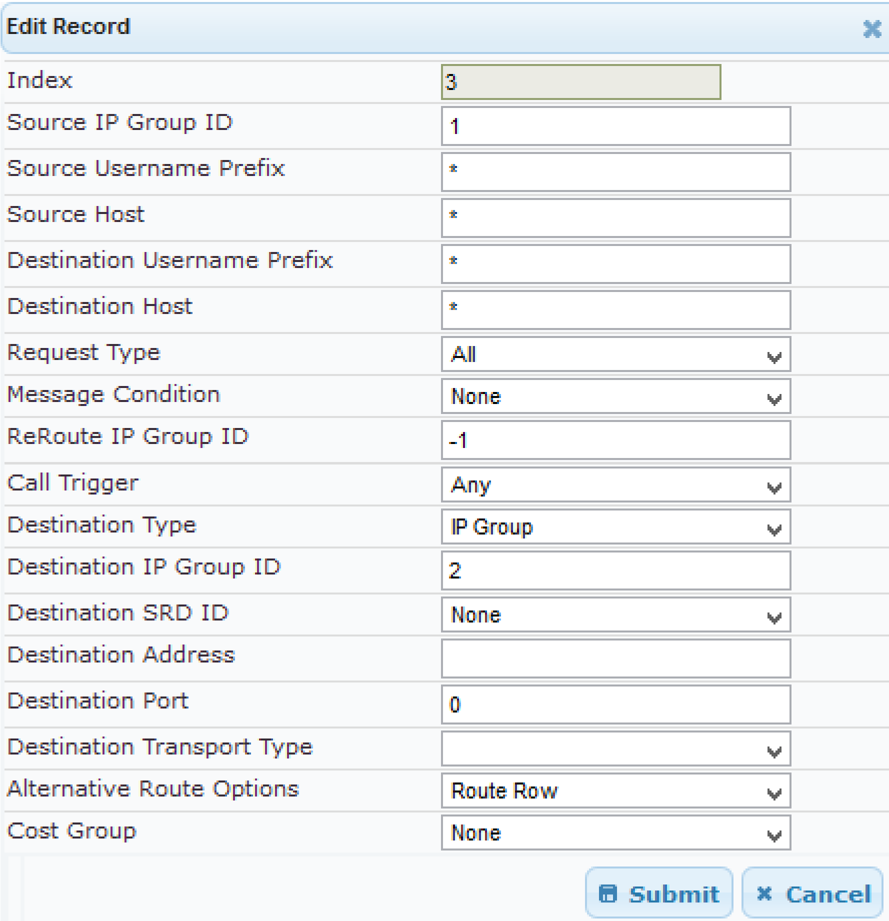
Figure 4-24: Configuring IP-to-IP Routing Rule for Lync to MediaPack



5. Configure an alternative rule to route calls from Lync Server 2013 to Swisscom VoIP Gate:

Parameter	Value
Index	**3**
Source IP Group ID	**1**
Destination Type	**IP Group**
Destination IP Group ID	**2**

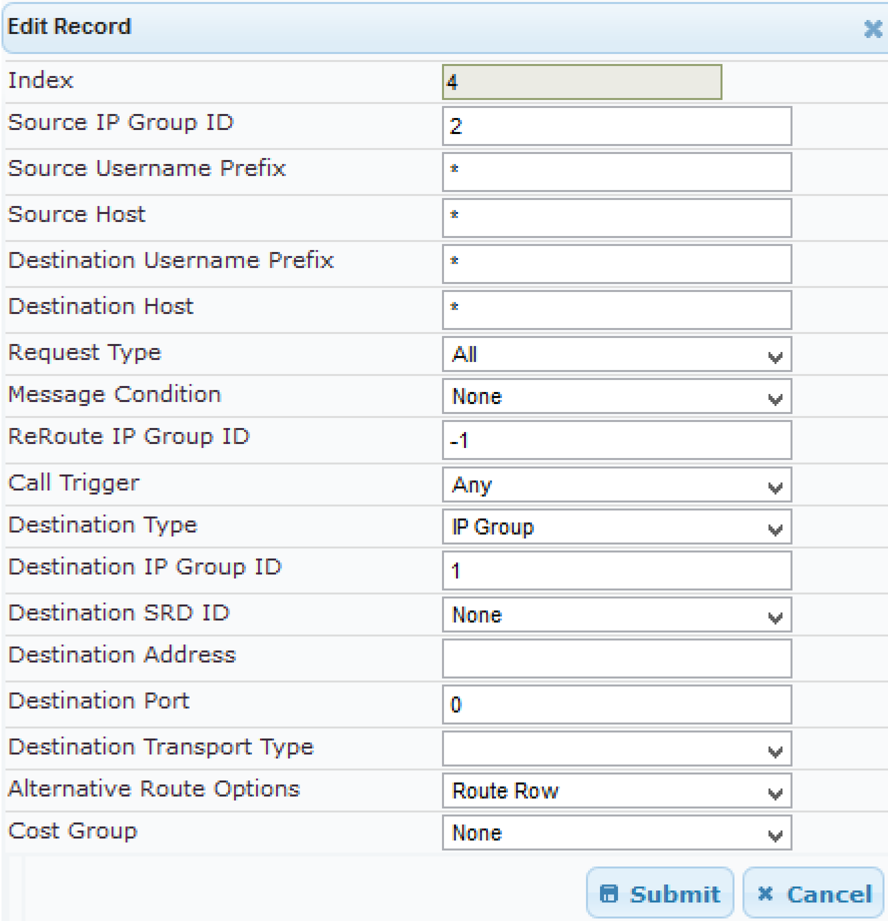
Figure 4-25: Configuring IP-to-IP Routing Rule for Lync to Swisscom VoIP Gate



6. Configure a rule to route calls from Swisscom VoIP Gate to Lync Server 2013:

Parameter	Value
Index	**4**
Source IP Group ID	**2**
Destination Type	**IP Group**
Destination IP Group ID	**1**

Figure 4-26: Configuring IP-to-IP Routing Rule for Swisscom VoIP Gate to Lync Server 2013



Edit Record

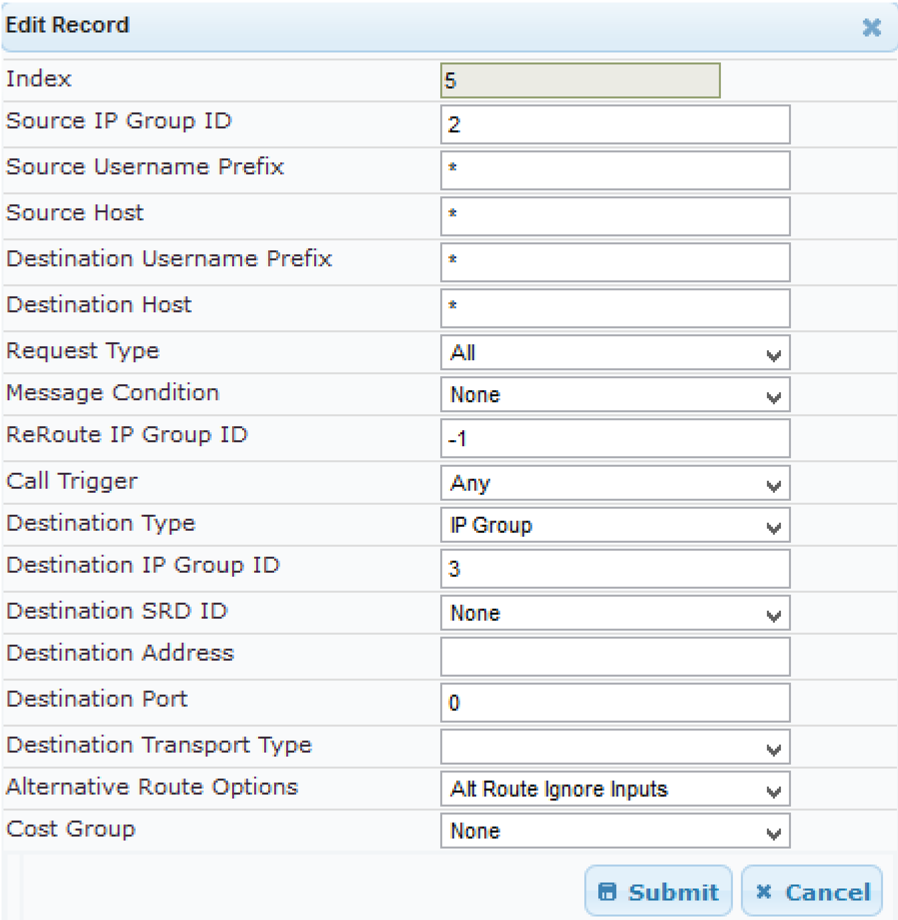
Index	4
Source IP Group ID	2
Source Username Prefix	*
Source Host	*
Destination Username Prefix	*
Destination Host	*
Request Type	All
Message Condition	None
ReRoute IP Group ID	-1
Call Trigger	Any
Destination Type	IP Group
Destination IP Group ID	1
Destination SRD ID	None
Destination Address	
Destination Port	0
Destination Transport Type	
Alternative Route Options	Route Row
Cost Group	None

Submit Cancel

7. Configure an alternative rule to route calls from Swisscom VoIP Gate to the MediaPack Analog gateway:

Parameter	Value
Index	**5**
Source IP Group ID	**2**
Destination Type	**IP Group**
Destination IP Group ID	**3**
Alternative Route Options	**Alt Route Ignore Inputs**

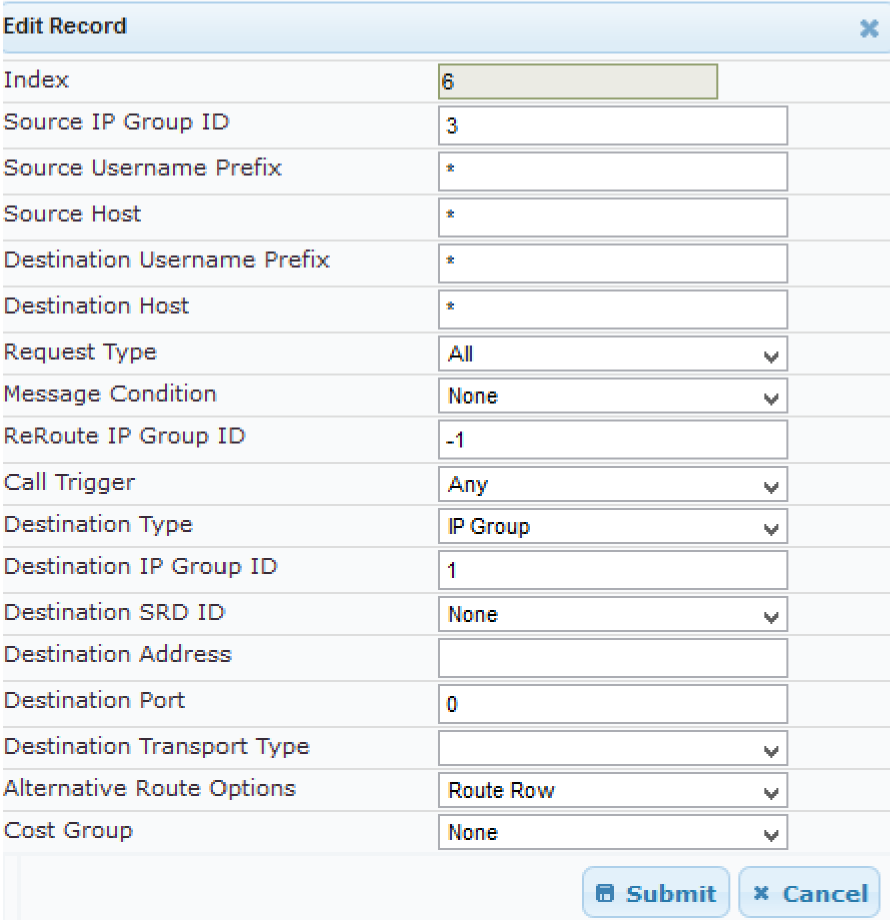
Figure 4-27: Configuring IP-to-IP Routing (alternative) Rule for Swisscom VoIP Gate to MediaPack Analog Gateway



8. Configure a rule to route calls from MediaPack to Lync Server 2013:

Parameter	Value
Index	**6**
Source IP Group ID	**3**
Destination Type	**IP Group**
Destination IP Group ID	**1**

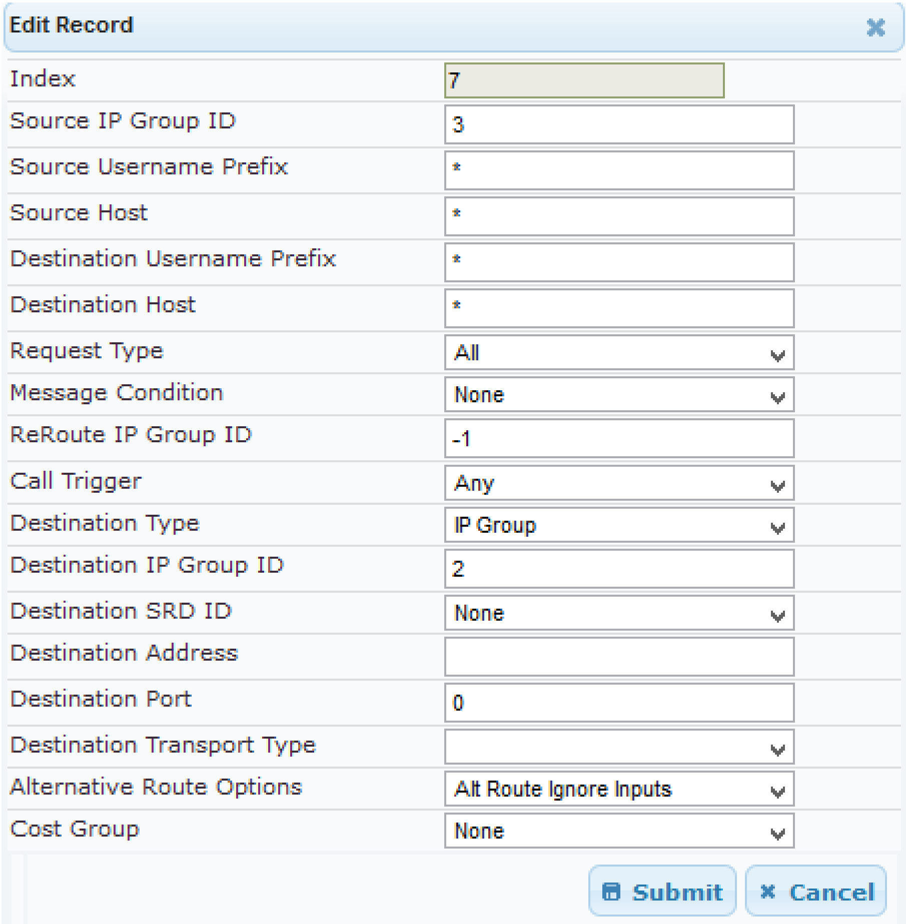
Figure 4-28: Configuring IP-to-IP Routing Rule for MediaPack to Lync Server 2013



9. Configure an alternative rule to route calls from the MediaPack analog gateway to Swisscom VoIP:

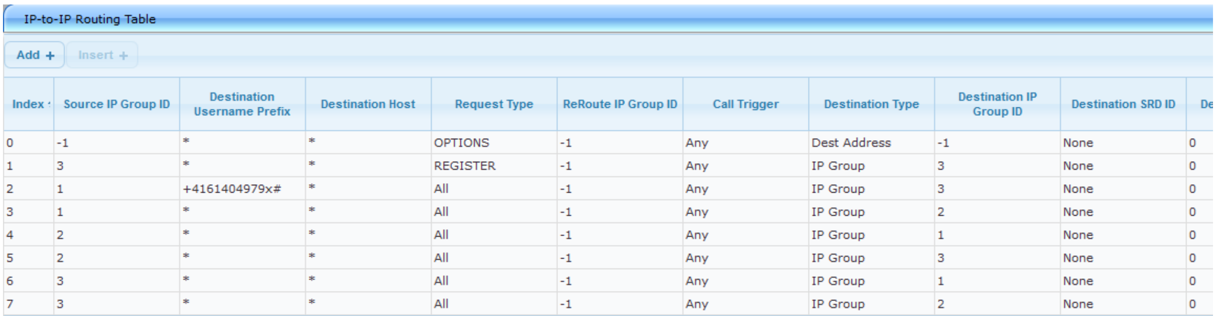
Parameter	Value
Index	**7**
Source IP Group ID	**3**
Destination Type	**IP Group**
Destination IP Group ID	**2**
Alternative Route Options	**Alt Route Ignore Inputs**

Figure 4-29: Configuring alternative IP-to-IP Routing for MediaPack to Swisscom VoIP Gate



The configured routing rules are shown in the figure below:

Figure 4-30: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table



Index	Source IP Group ID	Destination Username Prefix	Destination Host	Request Type	ReRoute IP Group ID	Call Trigger	Destination Type	Destination IP Group ID	Destination SRD ID	De
0	-1	*	*	OPTIONS	-1	Any	Dest Address	-1	None	0
1	3	*	*	REGISTER	-1	Any	IP Group	3	None	0
2	1	+4161404979x#	*	All	-1	Any	IP Group	3	None	0
3	1	*	*	All	-1	Any	IP Group	2	None	0
4	2	*	*	All	-1	Any	IP Group	1	None	0
5	2	*	*	All	-1	Any	IP Group	3	None	0
6	3	*	*	All	-1	Any	IP Group	1	None	0
7	3	*	*	All	-1	Any	IP Group	2	None	0

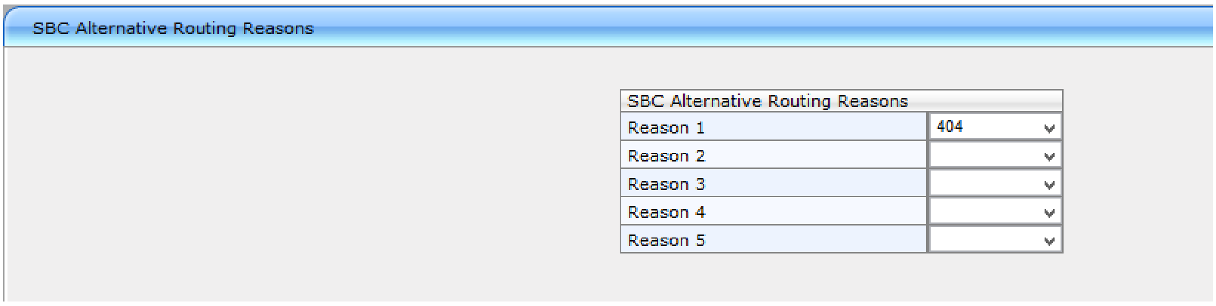
The operation of the Alternative routes included in the IP-to-IP routing table depends on correct configuration of 404 as one of the SBC alternative routing reasons. That release cause assures that the alternative routes are triggered correctly if no match is found on each of the primary routes.

➢ To configure SBC alternative routing reasons:

1. Open the Alternative Routing reasons page (**Configuration** tab > **VoIP** > **SBC** > **Routing SBC > Alternative Routing Reasons**).
2. Configure the parameters as follows:

Parameter	Value
Reason 1	**404**

Figure 4-31: Configured Alternative Routing Reasons in the SBC Alternative Routing Routing Reasons Table



4.13 Step 13: Configure IP-to-IP Manipulation Rules

This step describes how to configure IP-to-IP manipulation rules. These rules manipulate the source and / or destination number. The manipulation rules use the configured IP Groups to denote the source and destination of the call. As configured in Section 4.5 on page 45, IP Group 1 represents Lync Server 2013, and IP Group 2 represents Swisscom VoIP Gate.

Note: For this interoperability test topology, no number manipulation rules have been used. Both Lync Server 2013 and the Swisscom VoIP Gate use the +e.164 number format. Refer to the SBC manipulations chapter in the relevant AudioCodes User Manual if adjustments of the dial plan are required in your environment.

4.14 Step 14: Configure Message Manipulation Rules

This step describes how to configure SIP message manipulation rules. SIP message manipulation rules can include insertion, removal, and/or modification of SIP headers. Manipulation rules are grouped into Manipulation Sets, enabling you to apply multiple rules to the same SIP message (IP entity).

Once you have configured the SIP message manipulation rules, you need to assign them to the relevant IP Group (in the IP Group table) and determine whether they must be applied to inbound or outbound messages.

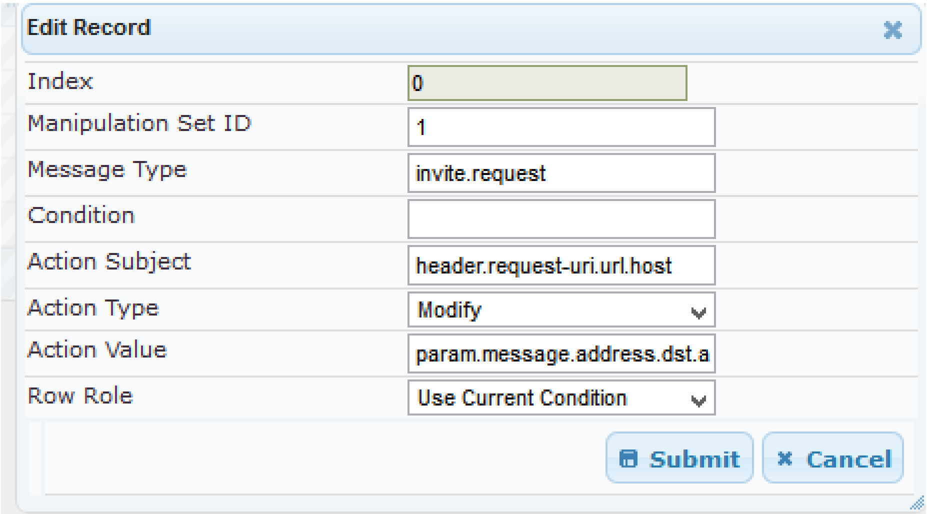
For this interoperability test topology, the host-part of request-URI, to and from header must be adjusted through message manipulation for calls towards the Lync Server 2013 (i.e., IP Group 1). A second set of manipulations is used to move a referred-by header, if it exists, into a diversion header.

➢ **To configure SIP message manipulation rule:**

1. Open the Message Manipulations page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Msg Policy & Manipulation** > **Message Manipulations**).
2. Configure the following manipulation rule (Manipulation Set 1) for use towards Lync Server 2013:

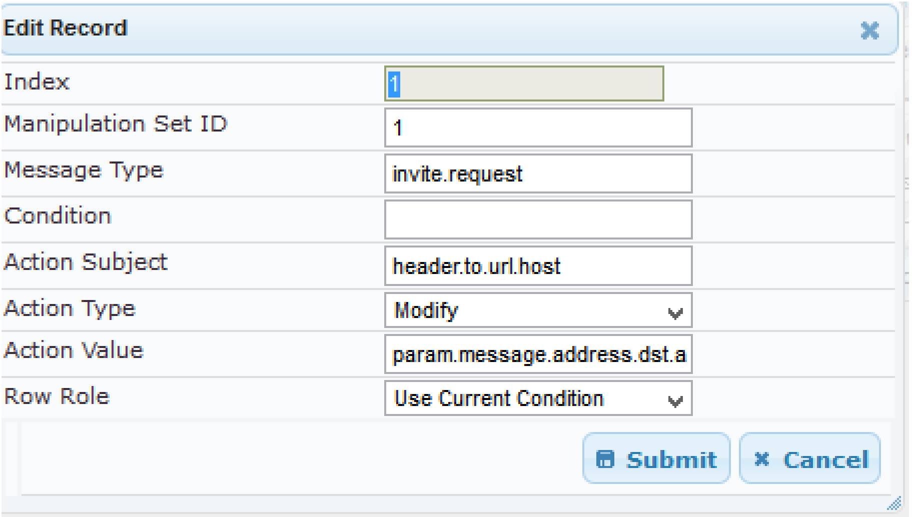
Parameter	Value
Index	**0**
Manipulation Set ID	**1**
Message Type	**invite.request**
Condition	
Action Subject	**header.request-uri.url.host**
Action Type	**Modify**
Action Value	**param.message.address.dst.address**

Figure 4-32: Configuring SIP Message Manipulation Rule #1 towards Lync Server 2013



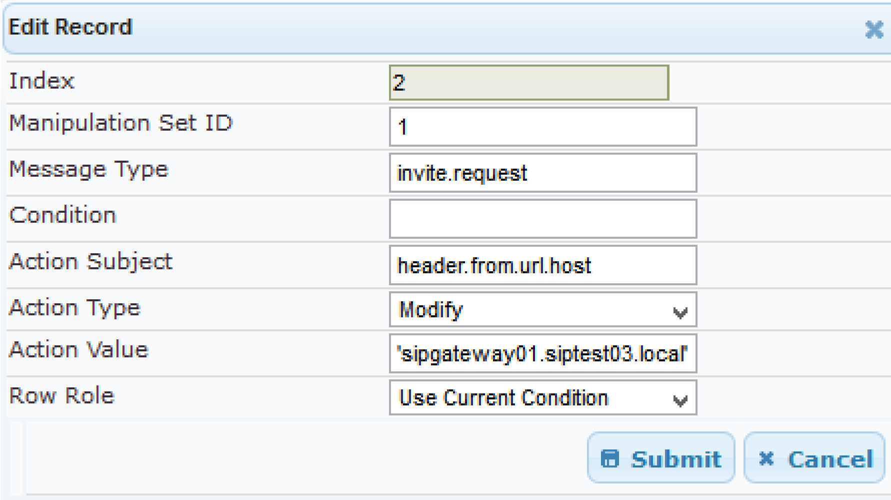
Parameter	Value
Index	**1**
Manipulation Set ID	**1**
Message Type	**invite.request**
Condition	
Action Subject	**header.to.url.host**
Action Type	**Modify**
Action Value	**param.message.address.dst.address**

Figure 4-33: Configuring SIP Message Manipulation Rule #2 towards Lync Server 2013



Parameter	Value
Index	**2**
Manipulation Set ID	**1**
Message Type	**invite.request**
Condition	
Action Subject	**header.from.url.host**
Action Type	**Modify**
Action Value	**'sipgateway01.siptest03.local'**

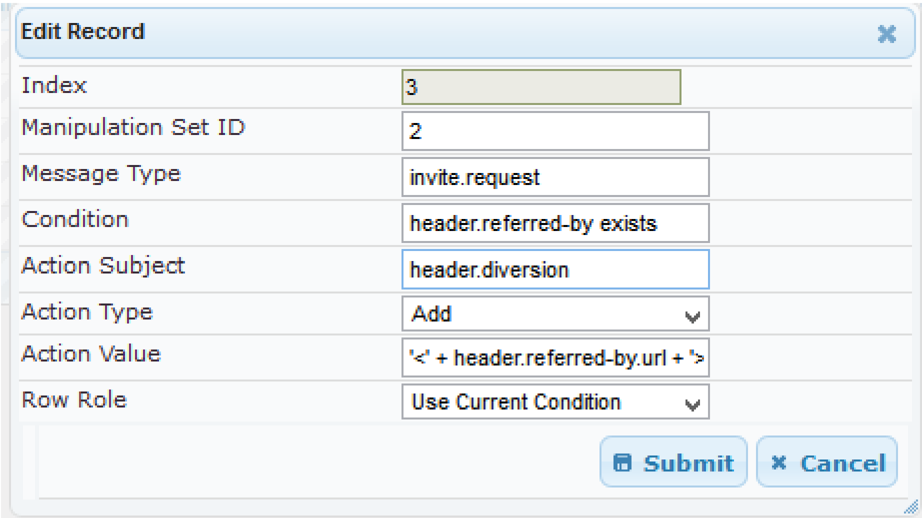
Figure 4-34: Configuring SIP Message Manipulation Rule #3 towards Lync Server 2013



3. Configure the following manipulation rule (Manipulation Set 2) for use towards the Swisscom VoIP Gate:

Parameter	Value
Index	**3**
Manipulation Set ID	**2**
Message Type	**invite.request**
Condition	**header.referred-by exists**
Action Subject	**header.Diversion**
Action Type	**Add**
Action Value	**'<'+header.referred-by.URL+'>'**

Figure 4-35: Configuring SIP Message Manipulation Rule towards the Swisscom VoIP Gate



Parameter	Value
Index	**4**
Manipulation Set ID	**2**
Message Type	**invite.request**
Condition	**header.referred-by exists**
Action Subject	**Header.referred-by**
Action Type	**Remove**
Action Value	

Figure 4-36: Configuring SIP Message Manipulation Rule #2 towards the Swisscom VoIP Gate

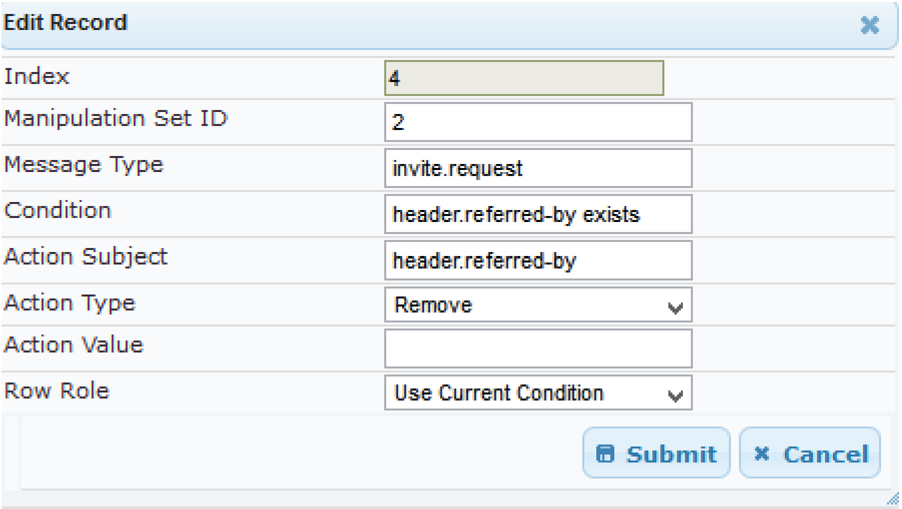
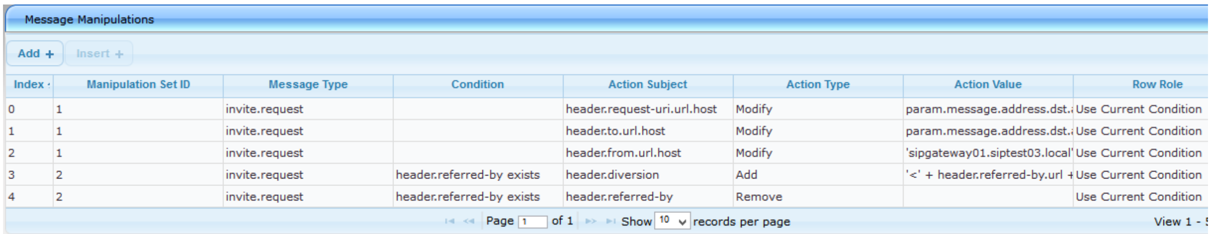


Figure 4-37: Overview of Configured SIP Message Manipulation Rules


Message Manipulations

Add + Insert +

Index	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value	Row Role
0	1	invite.request		header.request-uri.url.host	Modify	param.message.address.dst.a	Use Current Condition
1	1	invite.request		header.to.url.host	Modify	param.message.address.dst.a	Use Current Condition
2	1	invite.request		header.from.url.host	Modify	'sipgateway01.siptest03.local'	Use Current Condition
3	2	invite.request	header.referred-by exists	header.diversion	Add	'<' + header.referred-by.url +	Use Current Condition
4	2	invite.request	header.referred-by exists	header.referred-by	Remove		Use Current Condition

Page 1 of 1 Show 10 records per page View 1 - 5

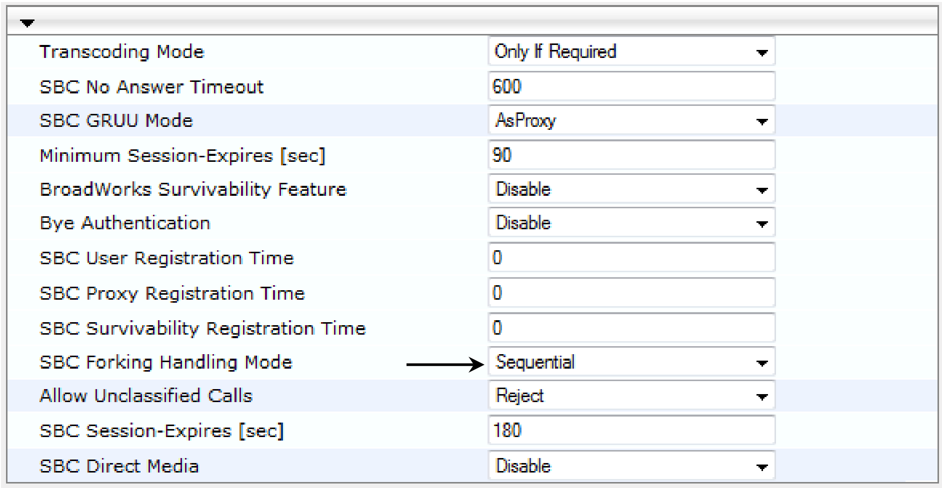
4.15 Step 15: Configure Call Forking Mode

This step describes how to configure the E-SBC's handling of SIP 18x responses received for call forking of INVITE messages. For the interoperability test topology, if 18x with SDP is received, the E-SBC opens a voice stream according to the received SDP. The E-SBC re-opens the stream according to subsequently received 18x responses with SDP or plays a ringback tone if 180 response without SDP is received. It's mandatory to set this field for the Lync Server 2013 environment.

➢ To configure call forking:

1. Open the General Settings page (**Configuration** tab > **VoIP** menu > **SBC** > **General Settings**).
2. From the 'SBC Forking Handling Mode' drop-down list, select **Sequential**.

Figure 4-38: Configuring Forking Mode



3. Click **Submit**.

4.16 Step 16: General Parameters

The following general parameters where set to non-default values on the E-SBC used for this interoperability test:

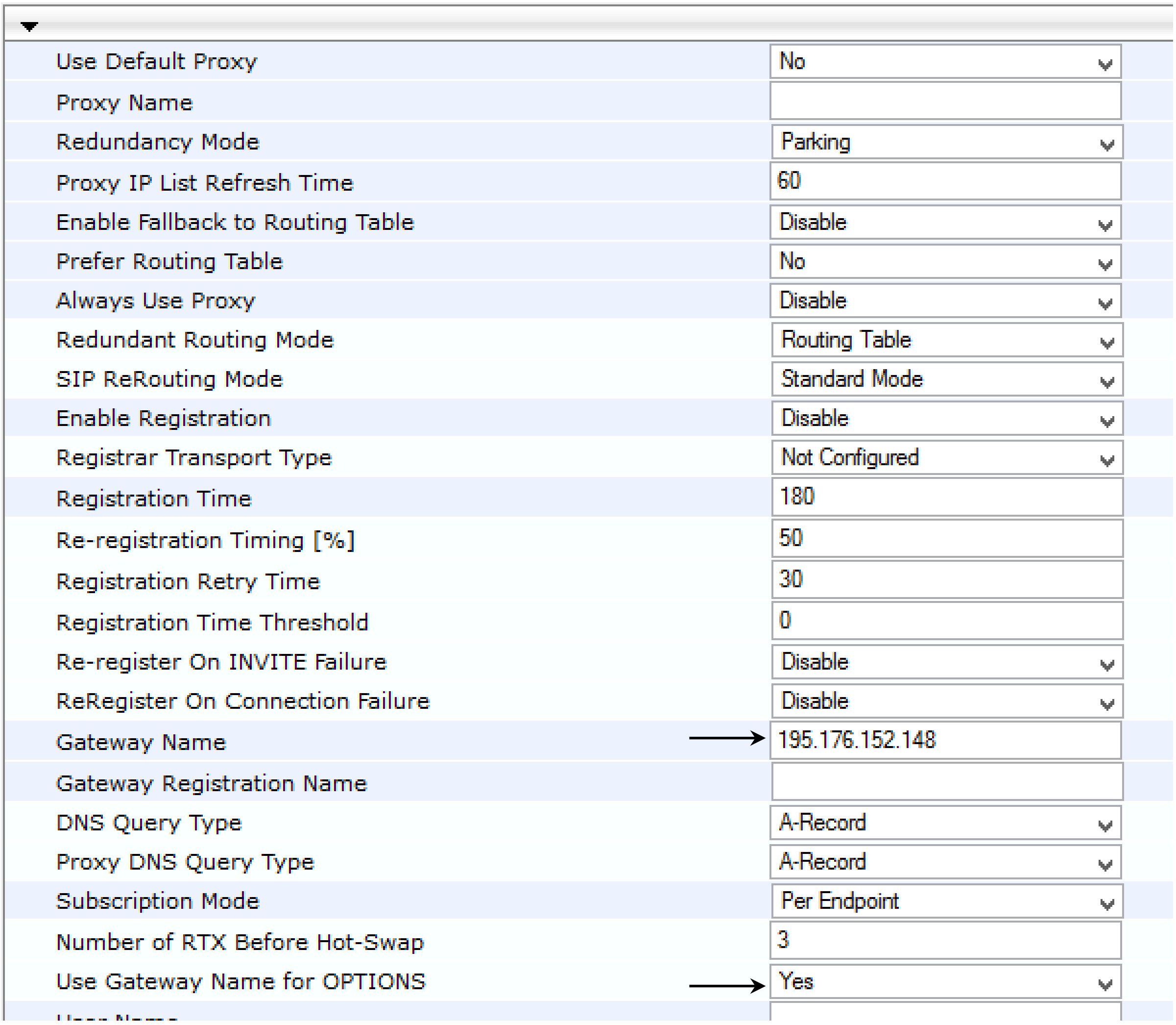
➢ **To configure Proxy and Registration:**

1. Open the Message Manipulations page (**Configuration** tab > **VoIP** menu > **SIP Definitions** Submenu > **Proxy & Registration**).
2. Configure the following Parameters:

Parameter	Value
Gateway Name	**nn.nn.nn.nn**
Use Gateway Name for Options	**Yes**

Note: The Gateway Name of the SBC is identical to the IP address of the SIP proxy which will be delivered by Swisscom.

Figure 4-39: Proxy & Registration Settings

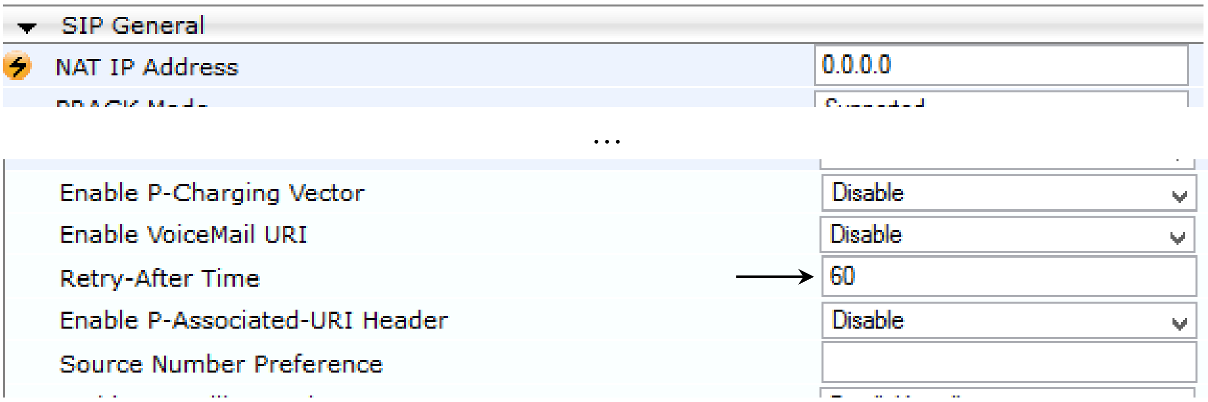


➢ To configure General SIP Parameters:

1. Open the Message Manipulations page (**Configuration** tab > **VoIP** menu > **SIP Definitions** Submenu > **General Parameters**).
2. Configure the following Parameters:

Parameter	Value
Retry-After Time	**60**

Figure 4-40: Proxy & Registration Settings



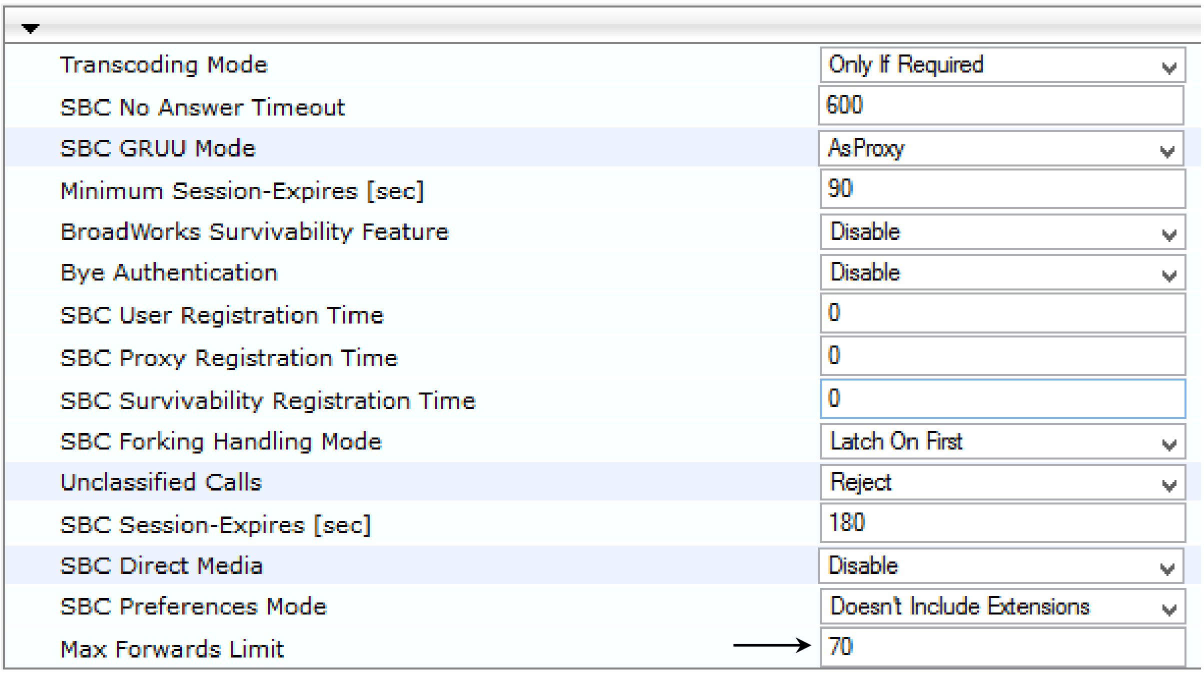
…

➢ To configure General SBC Settings:

1. Open the Message Manipulations page (**Configuration** tab > **VoIP** menu > **SBC** Submenu > **General Settings**).
2. Configure the following Parameters:

Parameter	Value
Max Forwards Limit	**70**

Figure 4-41: SBC General Settings

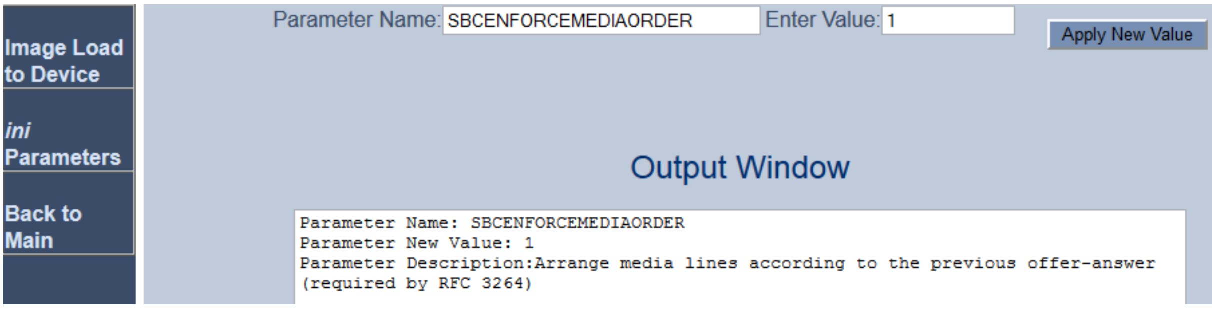


➢ To configure SBC Enforce Media Order :

1. Open the Admin page: append the case-sensitive suffix 'AdminPage' to the device's IP address in your Web browser's URL field (e.g., http://10.3.3.5/AdminPage).
2. Configure the following Parameters:

Parameter	Value
SBCENFORCEMEDIAORDER	**1**

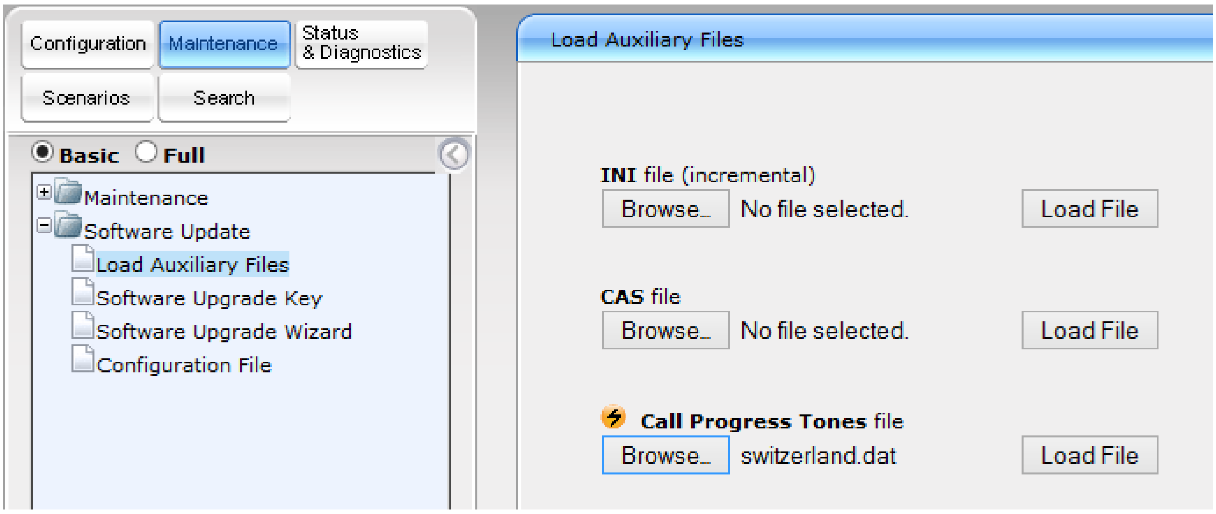
Figure 4-42: SBC General Settings



➢ To configure the Swiss Call progress tones :

1. Open the Message Manipulations page (**Maintenance** tab > **Software Update** menu > **Load Auxiliary Files**).
2. Browse for the call progress tones file on your computer and load the file in the E-SBC:

Figure 4-43: SBC General Settings



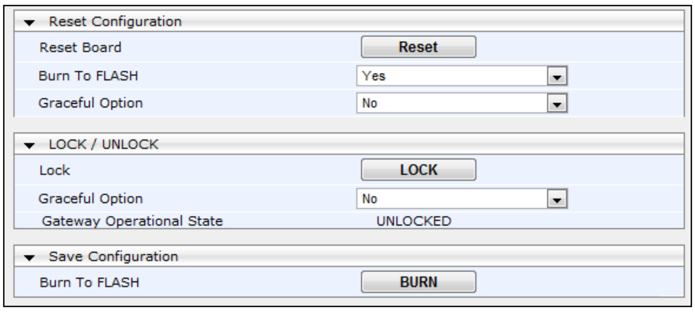
4.17 Step 17: Reset the E-SBC

After you have completed the configuration of the E-SBC described in this chapter, save ("burn") the configuration to the E-SBC's flash memory with a reset for the settings to take effect.

➢ **To save the configuration to flash memory:**

1. Open the Maintenance Actions page (**Maintenance** tab > **Maintenance** menu > **Maintenance Actions**).

Figure 4-44: Resetting the E-SBC



2. Ensure that the 'Burn to FLASH' field is set to **Yes** (default).
3. Click the **Reset** button.

5 Configuring AudioCodes Analog Gateway

This chapter provides step-by-step procedures on how to configure AudioCodes MediaPack 11x Analog Gateway for use as the fax adapter in combination with Microsoft Lync Server 2013 and the Swisscom VoIP Gate. These configuration procedures are based on the interoperability test topology described in Section 2.5 on page 11, and includes the following main areas:

- General configuration of the AudioCodes MediaPack for use with the Swisscom VoIP Gate.
- Port Configuration adjusted for analog phones or adjusted for operation with Fax machines or Modem devices.
- Registration of the MediaPack and all its FXS endpoints on the AudioCodes E-SBC
- Number manipulation for translation between user dialing behavior and the +e.164 numbering format used towards the Lync 2013 Server and Swisscom VoIP Gate

This configuration is done using the MediaPack's embedded Web server (hereafter, referred to as *Web interface*).

5.1 Step 1: IP Network Interface Configuration

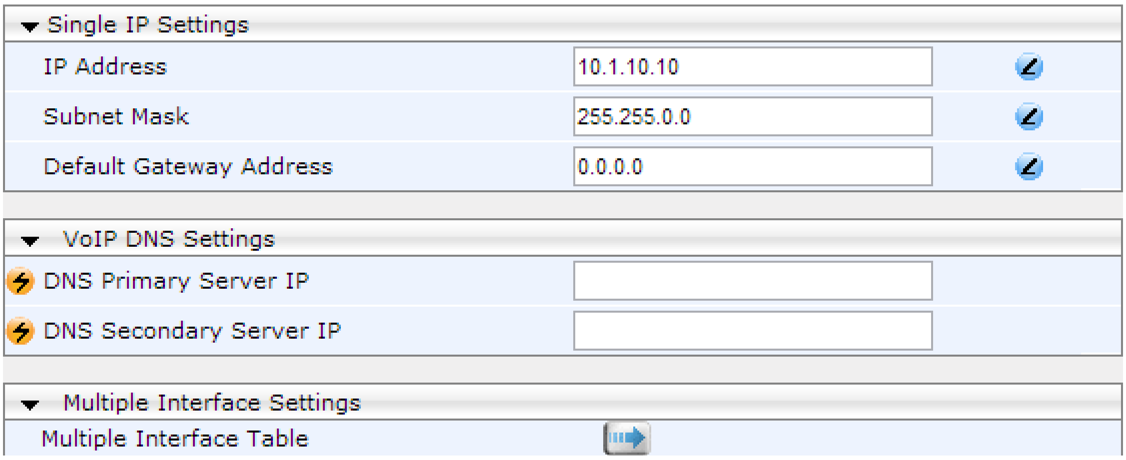
This step describes how to configure the MP11x or MP124 IP Address information. We recommend using the Multiple Interface table as this information will be backed up in the Board.ini file.

5.1.1 Step 1a: Enable the Multiple Interface Table

This step describes how to enable the Multiple Interface Table.

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **Network** > **IP Interfaces Table**).
2. Click the under the “Multiple Interface Settings” section

Figure 5-1: Enable the Multiple Interface Table



5.1.2 Step 1b: Configure Network Interface

This step describes how to configure the IP network interface.

➢ **To configure the IP network interfaces:**

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **Network** > **IP Interfaces Table**).
2. Modify the existing network interface:
   a. Select the 'Index' radio button of the **OAMP + Media + Control** table row, and then click **Edit**.
   b. Configure the interface as follows:

Parameter	Value
Application Type	**OAMP + Media + Control**
IP Address	**10.3.3.50** (IP address of MediaPack)
Prefix Length	**24** (subnet mask in bits for 255.255.255.0)
Gateway	**10.3.3.1**
VLAN ID	**1**
Interface Name	**O+M+C** (arbitrary descriptive name)
Primary DNS Server IP Address	**10.3.3.10**

Figure 5-2: Configure Network Interface

5.2 Step 2: FXS Endpoints Configuration

This step describes how to configure the analog endpoints.

➢ **To configure the analog endpoints:**

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** submenu > **Hunt Group** submenu > **Endpoint Phone Number**).
2. Configure the ports according to the telephone numbers assigned to the connected analog phone or Fax/Modem device.
   a. To assure correct operation with the ports used for analog telephones are assigned Tel Profile ID 1, Fax or Modem devices must be assigned the Tel Profile ID 2
   b. Configure the FXS ports as follows:

Parameter	Value
Channels	**1** (the port number, 1..24)
Phone Number	+41614049799 (the applicable phone number)
Trunk Group ID	**1**
Tel Profile ID	**1** (for analog phones) **2** (for modem or Fax devices)

Note: During this interoperability test all phone numbers shared a common prefix (+416140497) which has been use as part of the classification of calls made by analog devices on the E-SBC. This prefix must be adjusted accordingly in the E-SBC configuration (see to Section 4.5 on page 45).

Figure 5-3: Configure FXS endpoints

Channels	Phone Number	Trunk Group ID	Tel Profile ID
1	+41614049794	1	1
2	+41614049798	1	2
3	+41614049799	1	2

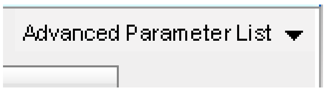
5.3 Step 3: Hunt Group Settings Configuration

The Hunt Group Settings allows you to configure the following per Hunt Group:

- Channel select method by which IP-to-Tel calls are assigned to the Hunt Group's channels.
- Registration method for registering Hunt Groups to selected Serving IP Group IDs.

➢ To configure the Hunt Group Settings:

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** submenu > **Hunt Group** submenu > **Hunt Group Settings**).
2. Change the view of the table to advanced view by clicking on the "Advanced Parameter List" button in the upper left corner

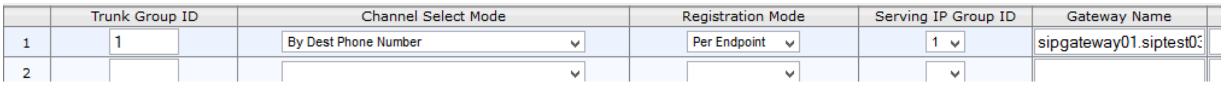


3. Configure the Hunt group settings Table as follows:

Parameter	Value
Trunk Group ID	**1**
Channel Select Mode	**By Dest Phone Number**
Registration Mode	**Per Endpoint**
Serving IP Group ID	**1**
Gateway Name	**sipgateway01.siptest03.local** (the FQDN of the ESBC, as configured in the Lync topology)

Note: Lync Server 2013 is not aware of the Analog gateway, and will use the FQDN of the PSTN gateway for all communications with the analog devices. This PSTN gateway is the AudioCodes E-SBC, configured in Chapter 5. The E-SBC will route calls for analog endpoints towards the AudioCodes MediaPack Analog Gateway, based on these registered endpoint information.

Figure 5-4: Configure Hunt Group Settings



	Trunk Group ID	Channel Select Mode	Registration Mode	Serving IP Group ID	Gateway Name
1	1	By Dest Phone Number	Per Endpoint	1	sipgateway01.siptest03
2					

5.4 Step 4: IP to Hunt Group Routing Configuration

The IP to Hunt Group Routing Table page allows you to configure the inbound call routing rules:

This table is used to route incoming IP calls to Hunt Groups. The specific channel pertaining to the Hunt Group to which the call is routed is set to “per endpoint” as determined by the Hunt Group's channel selection mode. The channel selection mode has been defined in the Hunt Group Settings table (see Section 5.3 on page 80).

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

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** submenu > **Routing** submenu > **IP to Hunt Group Routing**).
2. Configure the Hunt group settings Table as follows:

Parameter	Value
Dest. Phone Prefix	*
Source IP address	**10.3.3.5** (the IP-address of the E-SBC)
Hunt Group ID	**1**

Figure 5-5: Configure IP to Hunt Group routing

	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Hunt Group ID	Source IP Group ID
1	*		10.3.3.5		1	-1
2						
3						

5.5 Step 5: Proxy & Registration Configuration

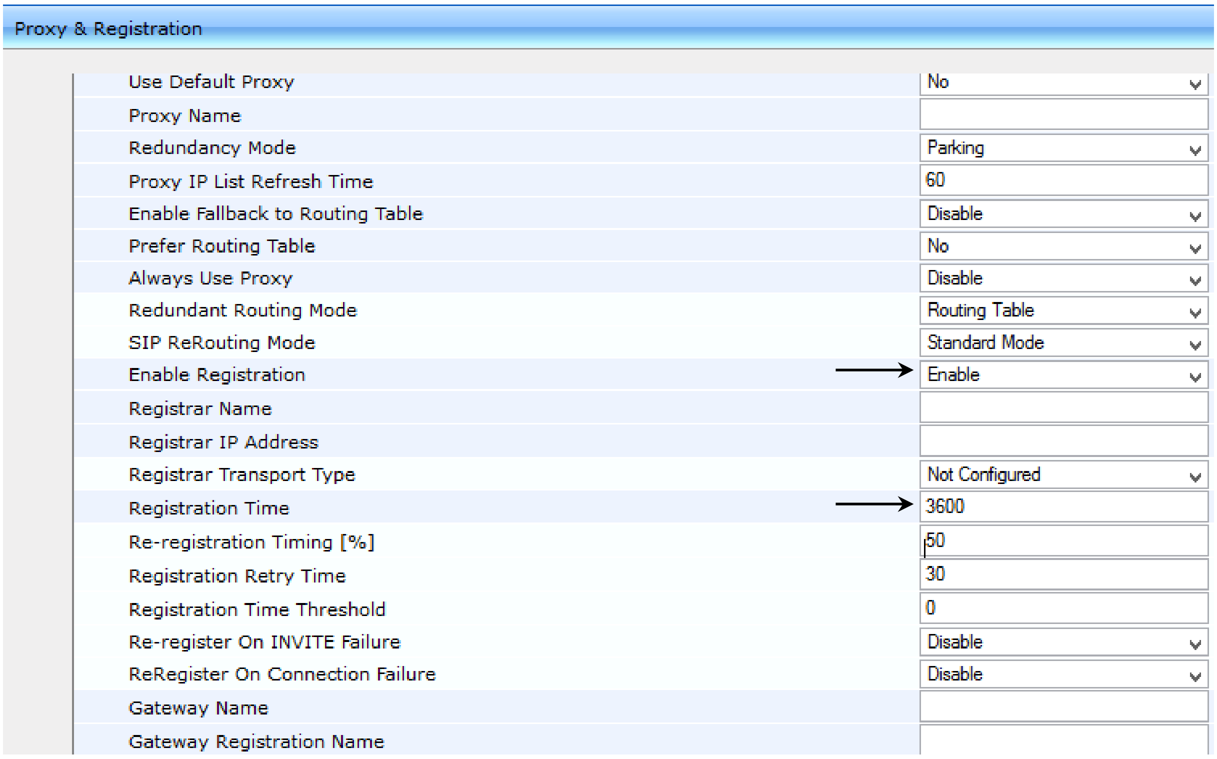
The Proxy & Registration Configuration allows you to define parameters relevant to registration, name resolution and routing towards a Proxy/Registrar.

➢ **To configure the Proxy and Registration settings:**

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **SIP Definitions** submenu > **Proxy & Registration**).
2. Configure the Proxy Sets Table as follows:

Parameter	Value
Enable Registration	**Enable**
Registration time	**3600**

Figure 5-6: Configure Proxy & Registration



5.6 Step 6: Proxy Set Configuration

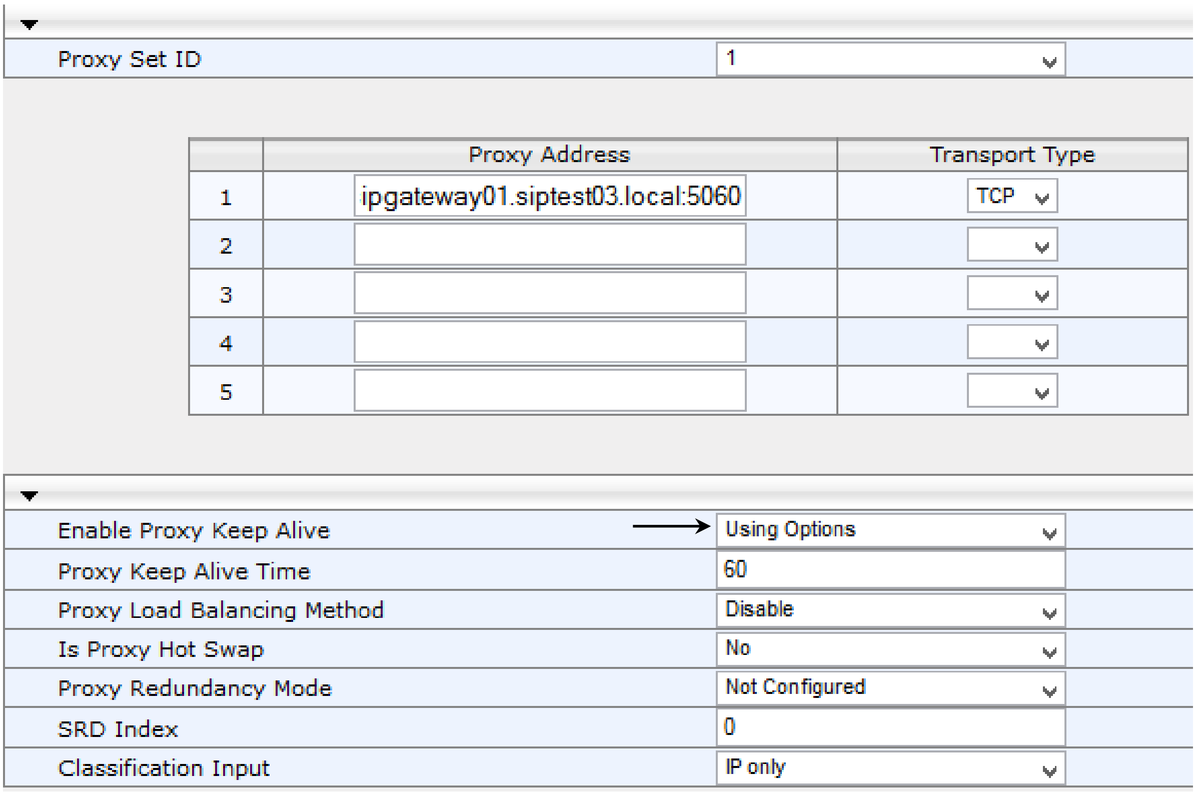
The Proxy Sets Table page allows you to define Proxy Sets. A single Proxy Set on the AudioCodes MediaPack is used, and configured with the fully qualified domain name (FQDN) of the AudioCodes E-SBC.

➢ **To configure the Proxy Sets Table:**

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **Control Network** submenu > **Proxy Sets Table**).
2. Configure the Proxy Sets Table as follows:

Parameter	Value
Proxy Set ID	**1**
Proxy Address	**sipgateway01.siptest03.local:5060** (the FQDN of the E-SBC, including the destination port)
Transport Type	**TCP**
Enable Proxy Keep Alive	**Using OPTIONS**

Figure 5-7: Configure the Proxy Set



5.7 Step 7: Coders Configuration

The Coder Group Settings page allows you to define a default coders list, and up to four additional groups of coders (termed Coder Groups). For each Coder Group, you can define up to 10 coders configured with packetization time (ptime), rate, payload type, and silence suppression. During this interoperability test the default Coders list was defined for calls without Tel Profile. Coders Group 1 was attached to Tel Profile 1, used for analog telephones and Coders Group 2 was attached to Tel Profile 2, used for fax/modem ports.

➢ **To configure the default Coders:**

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **Coders and Profiles** submenu > **Coders**).
2. Configure the Coders as follows:

Parameter	Value
Name	**G.711 A-law**
Name	**T.38**

Figure 5-8: Configure the Coders

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711A-law	20	64	8	Disabled
T.38	N/A	N/A	N/A	N/A

➢ **To configure the Coder Groups:**

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **Coders and Profiles** submenu > **Coders Group Settings**).
2. Configure the first Coder Groups as follows:

Parameter	Value
Coder Group ID	**1**
Coder Name	**G.711A-law**

Figure 5-9: Configure the Coders Group 1

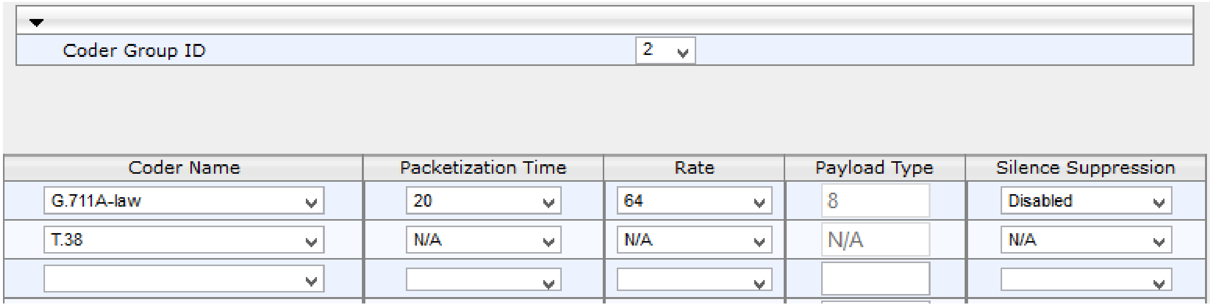
Coder Group ID	1

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711A-law	20	64	8	Disabled

3. The second coder group is used for the fax and modem endpoints. Configure it as follows:

Parameter	Value
Coder Group ID	**2**
Coder Name	**G.711A-law**
Coder Name	**T.38**

Figure 5-10: Configure the Coders Group 2



5.8 Step 8: Tel Profile Configuration

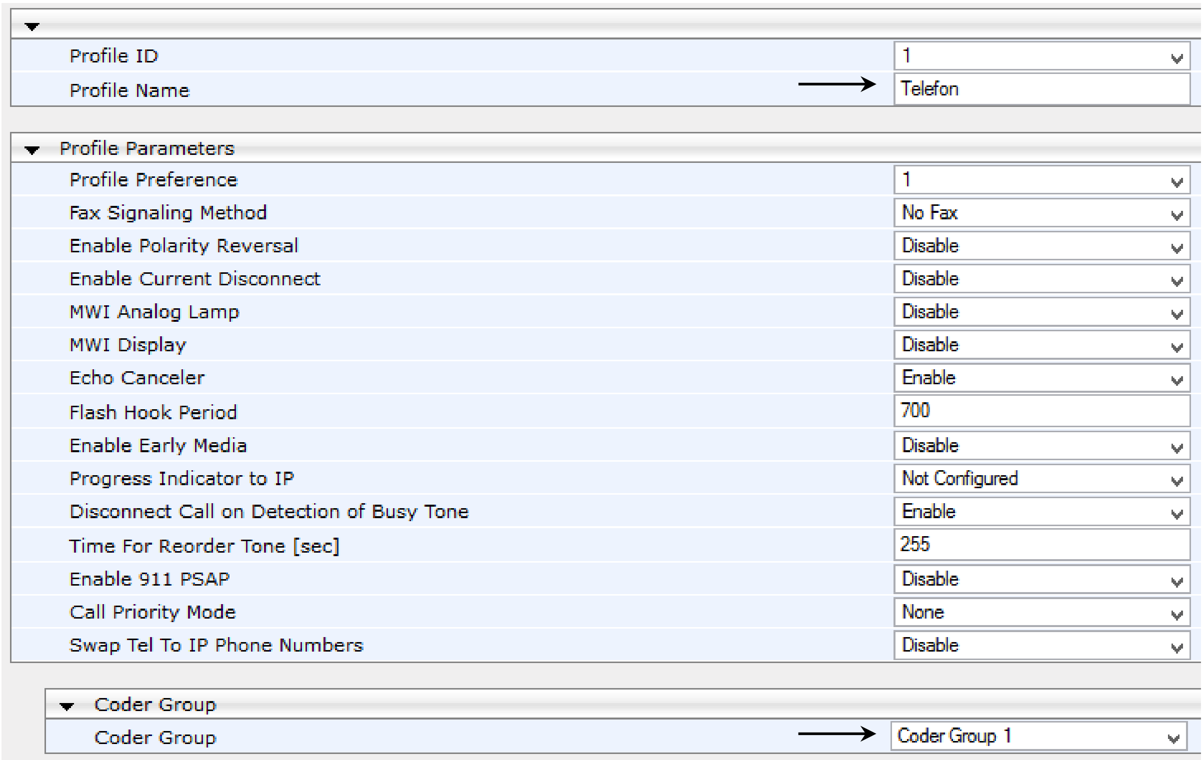
The Tel Profile Settings table allows you to define up to nine configuration profiles for Tel calls. These profiles are termed *Tel Profiles*. During this interoperability test two Tel Profile have been used, one to be applied for calls with Analog phones, a second one to be applied for calls with Fax or Modem devices.

➢ To configure the Tel Profiles:

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **Coders and Profiles** submenu > **Tel Profile Settings**).
2. Configure the first Tel Profile as follows:

Parameter	Value
Profile ID	**1**
Profile Name	**Telefon**
Coder Group	**Coder Group 1**

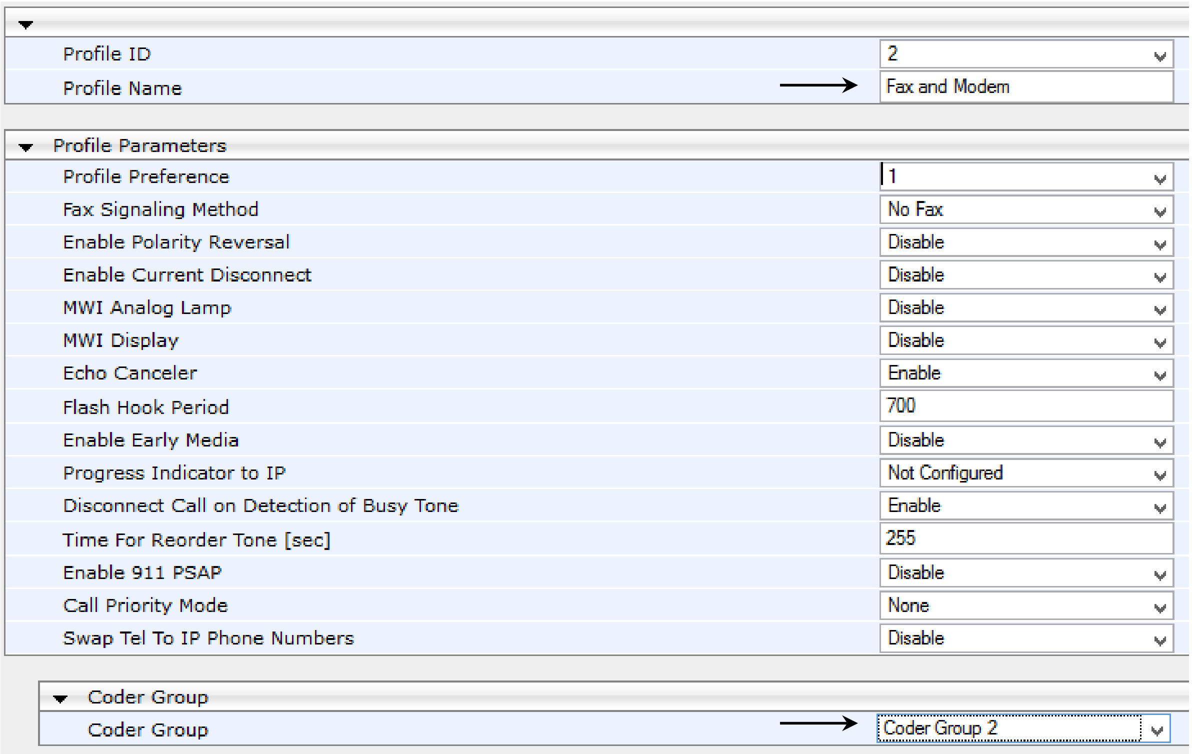
Figure 5-11: Configure the Tel Profile 1



3. Configure the Second Tel Profile as follows:

Parameter	Value
Profile ID	**2**
Profile Name	**Fax and Modem**
Coder Group	**Coder Group 2**

Figure 5-12: Configure the Tel Profile 2



5.9 Step 9 IP Profile Configuration

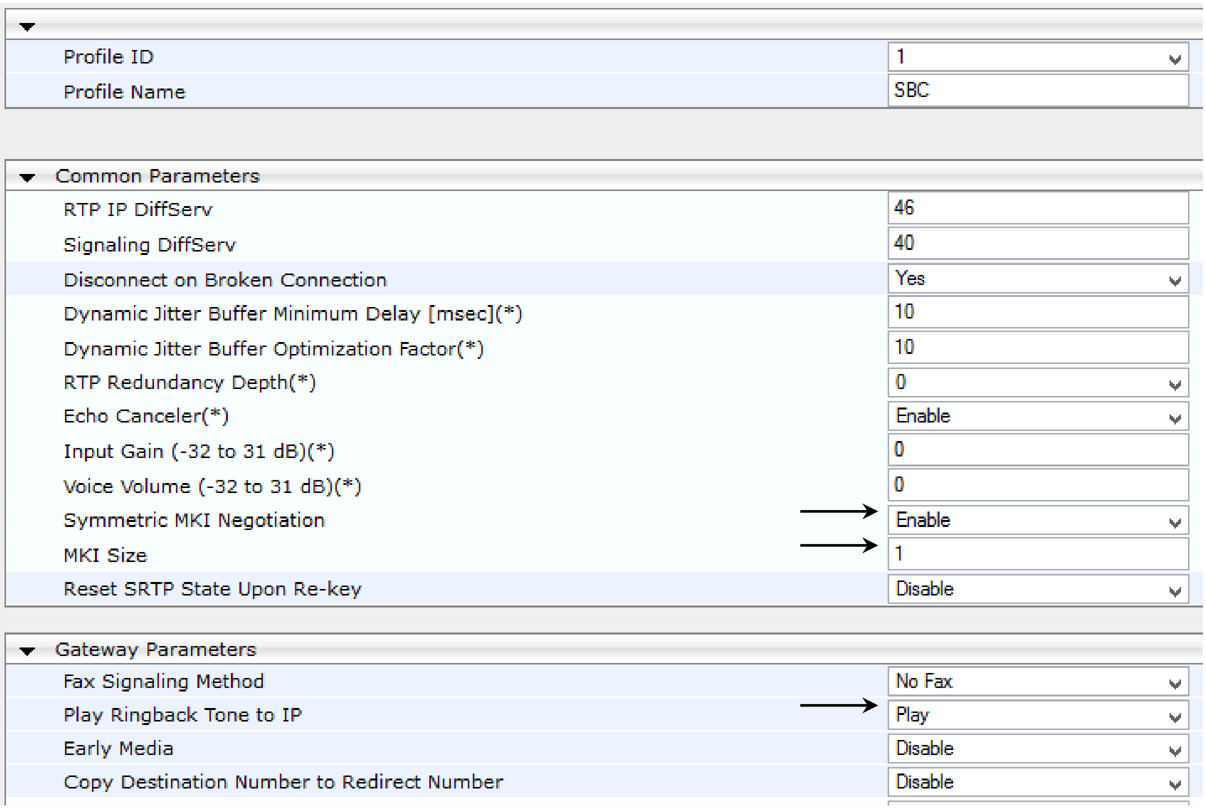
The IP Profile Settings table allows you to define up to nine *IP Profiles*. An IP Profile is a set of special call configuration behaviors relating to signaling and media (e.g., coder used) applied to specific IP calls (inbound and/or outbound). During this interoperability test one IP Profile has been used and applied for all calls between the MediaPack and the E-SBC.

➢ **To configure the IP Profiles:**

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **Coders and Profiles** submenu > **IP Profile Settings**).
2. Configure the IP Profile as follows:

Parameter	Value
Profile ID	**1**
Profile Name	**SBC**
Symmetric MKI Negotiation	**Enable**
MKI Size	**1**
Play Ringback Tone to IP	**Play**

Figure 5-13: Configure the IP Profile



5.10 Step 10 IP Group Configuration

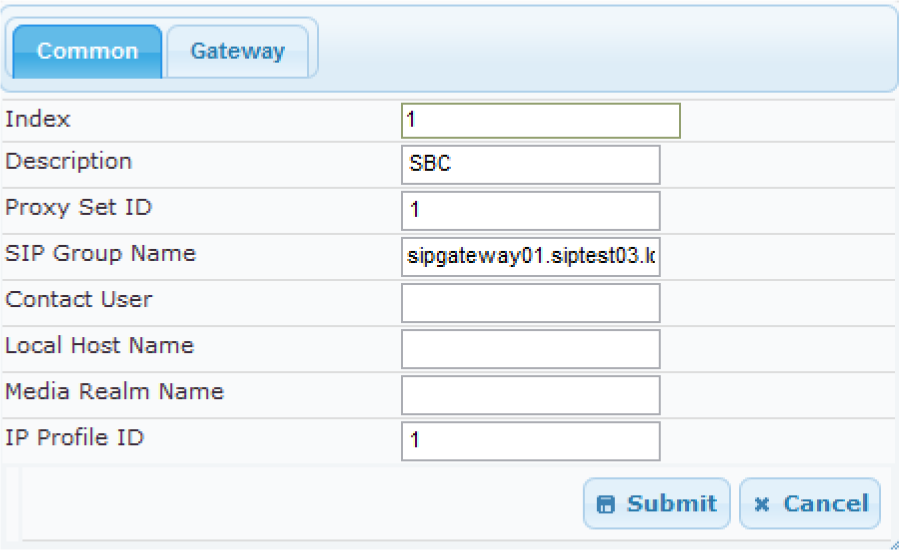
The IP Group Table page allows you to create up to nine logical IP entities called *IP Groups*. During this interoperability test one IP Profile has been used and applied for all calls between the MediaPack Analog Gateway and the E-SBC.

➢ **To configure the IP Group:**

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **Control Network** submenu > **IP Group Table**).
2. Configure the IP Group as follows:

Parameter	Value
Index	**1**
Description	**SBC**
Proxy Set ID	**1**
SIP Group Name	**sipgateway01.siptest03.local** (the FQDN of the E-SBC)
IP profile ID	**1**

Figure 5-14: Configure the IP Group



5.11 Step 11: Number Manipulations

Rules for manipulating destination and/or source telephone numbers for IP-to-Tel and Tel-to-IP calls are required. The following number manipulation tables are used as part of this interoperability test to allow correct dialing from the FXS endpoints, and permit correct presentation of the Caller Line ID on incoming calls.

- Tel-to-IP calls:
  - Destination Phone Number Manipulation Table for Tel > IP Calls
    - Convert a 00 prefix into + for international dialing
      - ✓ Destination Prefix: 00
      - ✓ Strip digits from Left: 2
      - ✓ Prefix to add: +
    - Convert a 0 prefix into +41 for national dialing
      - ✓ Destination Prefix: 0
      - ✓ Strip digits from Left: 1
      - ✓ Prefix to add: +41
    - Convert emergency and service numbers to include the +41 prefix
      - ✓ Destination Prefix: [112,117,118,143,144,145,147,161,162,163,164,187]#
      - ✓ Prefix to add: +41
      - ✓ Destination Prefix: [1811,1818,1850,1414,1415]#
      - ✓ Prefix to add: +41
      - ✓ These two manipulations apply to the following numbers:
        - → 112,117,118,143,144,145,147,161,162,163,164,187
        - → 1811,1818,1850,1414,1415
    - During this interoperability test the Lync environment used 4-digit short-dials. When used on the MediaPack the number manipulation will convert those into the complete e.164 number by adding +4161404 as the prefix
      - ✓ Destination Prefix: xxxxxx#
      - ✓ Prefix to Add: +4161404
    - Any other number formats is considered to be in national format, and converted to e.164 by adding the +41 prefix.
      - ✓ Destination Prefix: *
      - ✓ Prefix to Add: +41
- IP-to-Tel calls:
  - Source Phone Number Manipulation Table for IP > Tel Calls
    - Convert +41 to 0
    - Convert + to 00

➢ To configure the number Manipulations:

1. Open the IP Interfaces Table page (**Configuration tab** > **VoIP menu** > **GW and IP to IP submenu** > **Manipulations** > **DST number Tel->IP** Table).
2. Configure the following:

Figure 5-15: Configure Dst number Tel->IP manipulations

Index	Destination Prefix	Source Prefix	Source Trunk Group	Destination IP Group	Prefix to Add	Suffix to Add
0	00	*	-1	-1	+	
1	0	*	-1	-1	+41	
2	[112,117,118,143,144,145,147,161,162,163,164,187]#	*	-1	-1	+41	
3	[1811,1818,1850,1414,1415]#	*	-1	-1	+41	
4	xxxx#	*	-1	-1	+4161404	
5	*	*	-1	-1	+41	

Page 1 of 1 Show 10 records per page View 1

Figure 5-16: DST number Tel->IP manipulations details

Rule | Action

Index	0
Destination Prefix	00
Source Prefix	*
Source Trunk Group	-1
Destination IP Group	-1

Submit | Cancel

Rule | Action

Index	0
Stripped Digits From Left	2
Stripped Digits From Right	0
Number of Digits to Leave	255
Prefix to Add	+
Suffix to Add	
TON	Not Configured
NPI	Not Configured
Presentation	Not Configured

Submit | Cancel

Rule | Action

Index	1
Destination Prefix	0
Source Prefix	*
Source Trunk Group	-1
Destination IP Group	-1

Submit | Cancel

Rule | Action

Index	1
Stripped Digits From Left	1
Stripped Digits From Right	0
Number of Digits to Leave	255
Prefix to Add	+41
Suffix to Add	
TON	Not Configured
NPI	Not Configured
Presentation	Not Configured

Submit | Cancel

Rule | Action

Index	2
Destination Prefix	[112,117,118,143,144,145,147,161,162,1
Source Prefix	*
Source Trunk Group	-1
Destination IP Group	-1

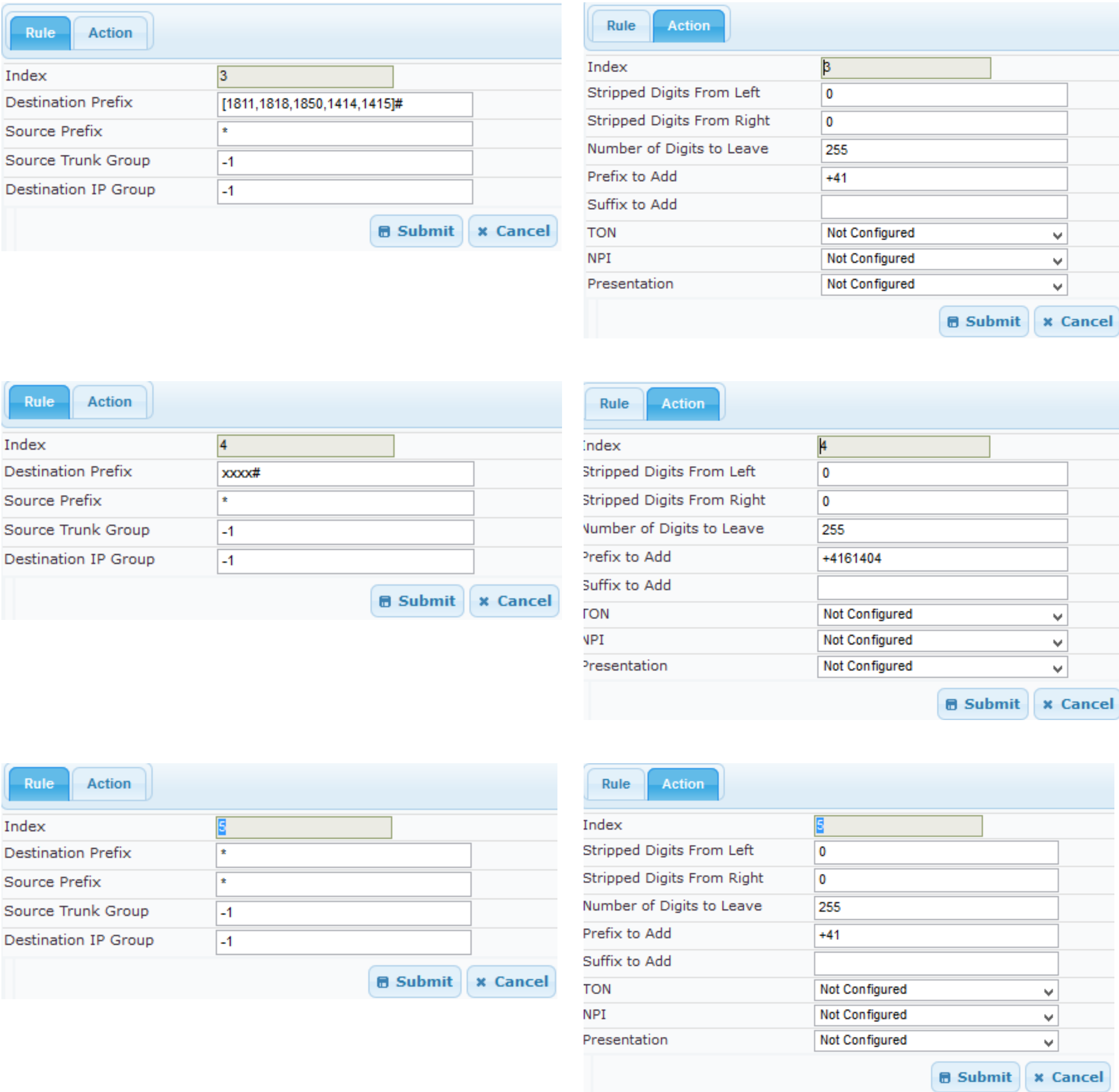
Submit | Cancel

Note: the full entry in *Destination Prefix* is:
[112,117,118,143,144,145,147,161,162,163,164,187]#

Rule | Action

Index	2
Stripped Digits From Left	0
Stripped Digits From Right	0
Number of Digits to Leave	255
Prefix to Add	+41
Suffix to Add	
TON	Not Configured
NPI	Not Configured
Presentation	Not Configured

Submit | Cancel



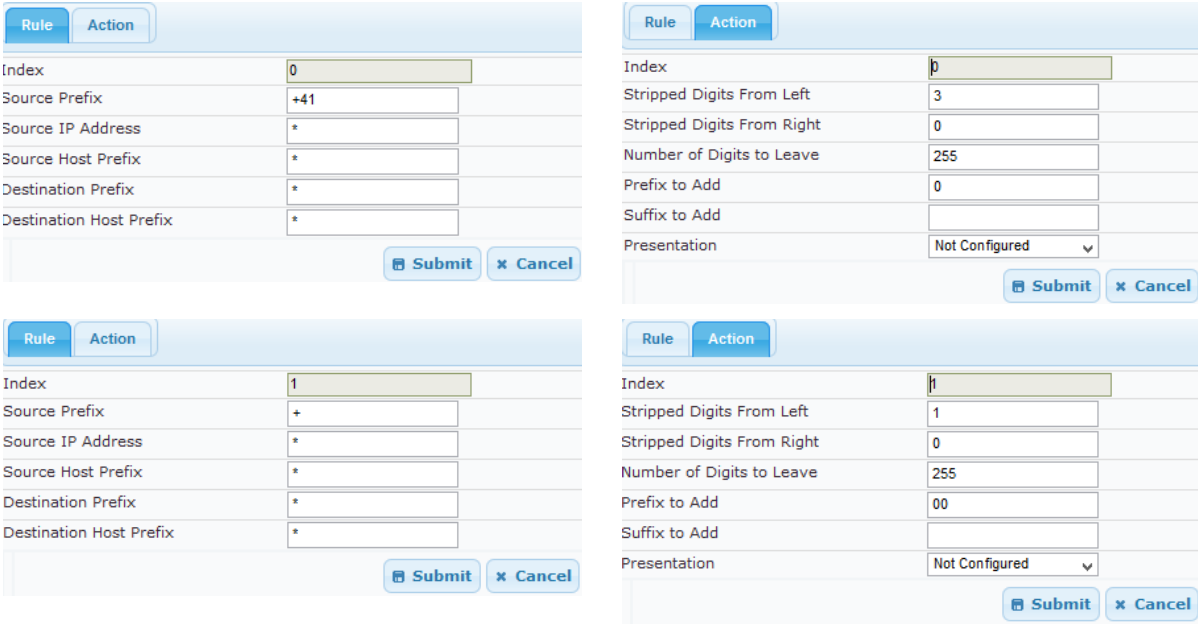
3. Open the IP Interfaces Table page (**Configuration tab** > **VoIP menu** > **GW and IP to IP submenu** > **Manipulations** > **Source number IP->Tel** Table).
4. Configure the following:

Figure 5-17: Configure Source number IP->Tel manipulations

Index	Source Prefix	Source IP Address	Source Host Prefix	Destination Prefix	Destination Host Prefix	Prefix to Add	Suffix to Add
0	+41	*	*	*	*	0	
1	+	*	*	*	*	00	

Page 1 of 1 Show 10 records per page View

Figure 5-18: Source number IP ->Tel manipulations details



5.12 Step 12: General Configuration

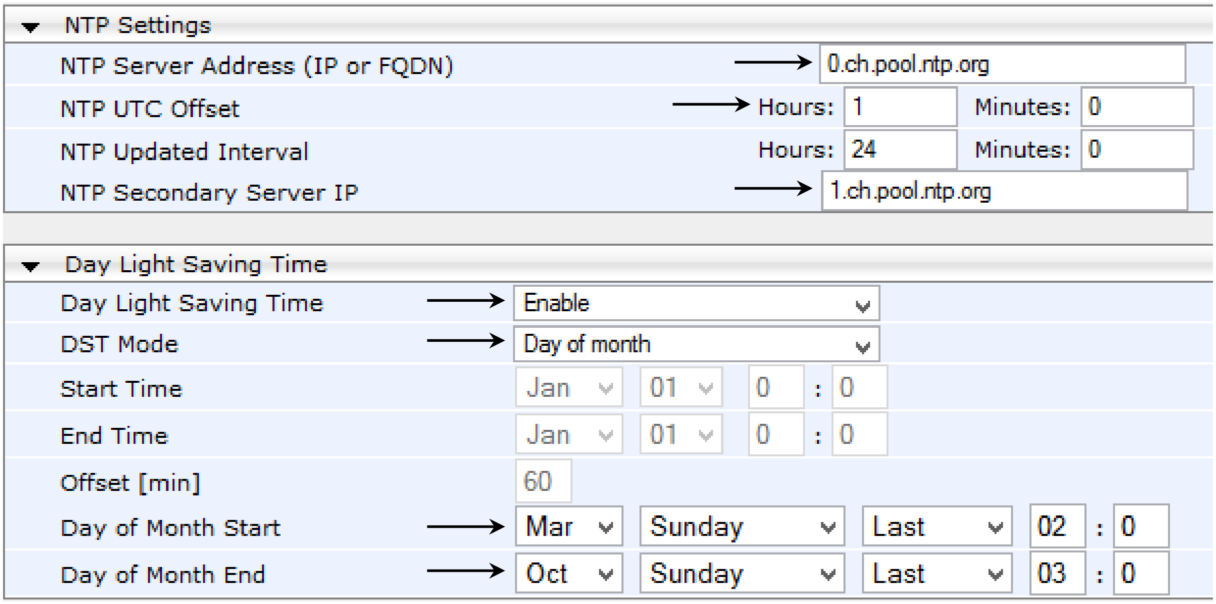
Various general configurations complete the configuration of the MediaPack Analog Gateway

➢ To configure NTP and Daylight Saving rules:

1. Open the Message Manipulations page (**Configuration** tab > **System** menu > **Application Settings**).
2. Configure the following Parameters:

Parameter	Value
NTP Server Address (IP or FQDN)	**0.ch.pool.ntp.org**
NTP UTC Offset	**1** (Hours)
NTP Secondary Server IP	**1.ch.pool.ntp.org**
Day Light Saving Time	**enable**
DST Mode	**Day of month**
Day of Month Start	**Mar Sunday Last 02 00**
Day of Month End	**Oct Sunday Last 03 00**

Figure 5-19: Application Settings

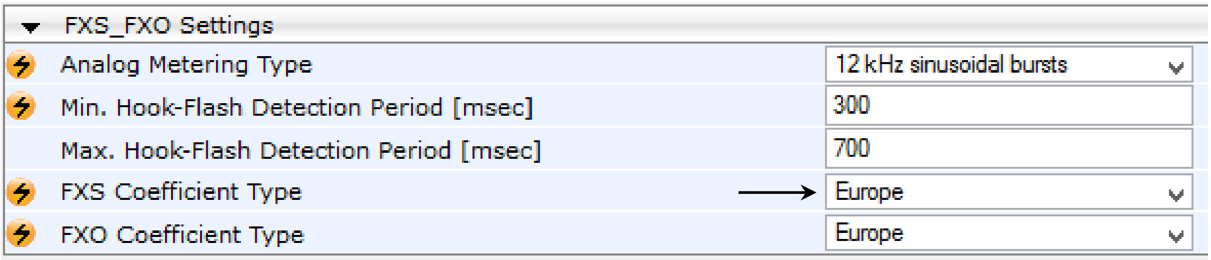


➢ To configure the Media Analog Settings:

1. Open the Analog Settings page (**Configuration** tab > **VoIP** menu > **Media** Submenu > **Analog Settings**).
2. Configure the following Parameters:

Parameter	Value
FXS Coefficient Type	**Europe**

Figure 5-20: SBC General Settings

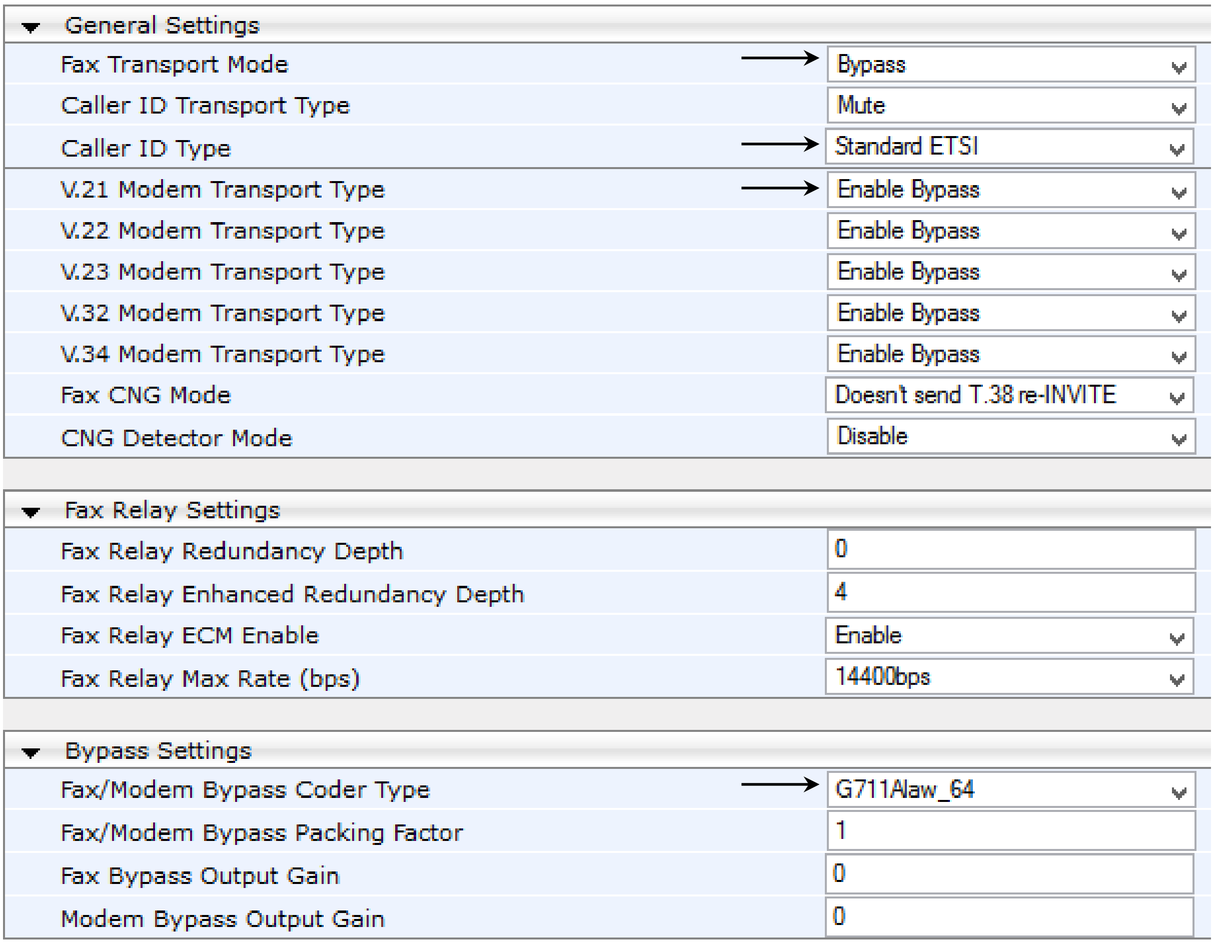


➢ To configure the FAX/Modem Settings:

1. Open the FAX/Modem Settings page (**Configuration** tab > **VoIP** menu > **Media** Submenu > **FAX/Modem/CID Settings**).
2. Configure the following Parameters:

Parameter	Value
Fax Transport Mode	**Bypass**
Caller ID Type	**Standard ETSI**
V.21 Modem Transport Type	**Enable Bypass**
FAX/Modem Bypass Coder Type	**G711Alaw_64**

Figure 5-21: FAX/Modem Settings

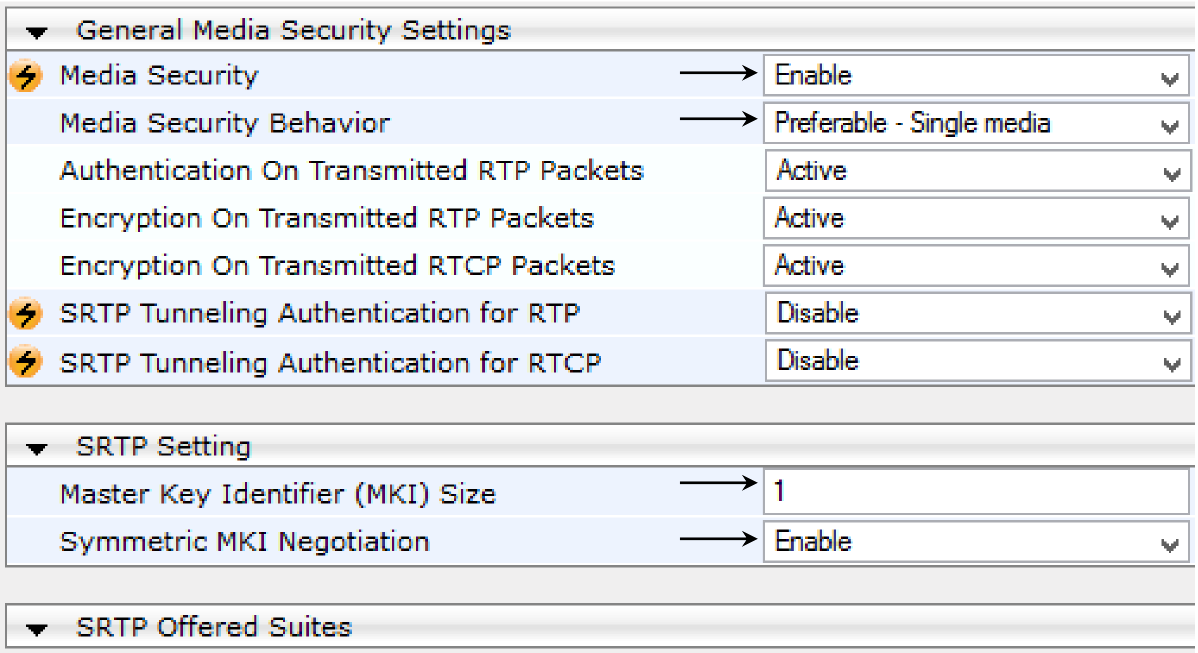


➢ **To configure the Media Security Settings:**

1. Open the Media Security Settings page (**Configuration** tab > **VoIP** menu > **Media** Submenu > **Media Security**).
2. Configure the following Parameters:

Parameter	Value
Media Security	**Enable**
Media Security Behavior	**Preferable – Single Media**
Master Key Identifier (MKI) Size	**1**
Symmetric MKI negotiation	**Enable**

Figure 5-22: FAX/Modem Settings

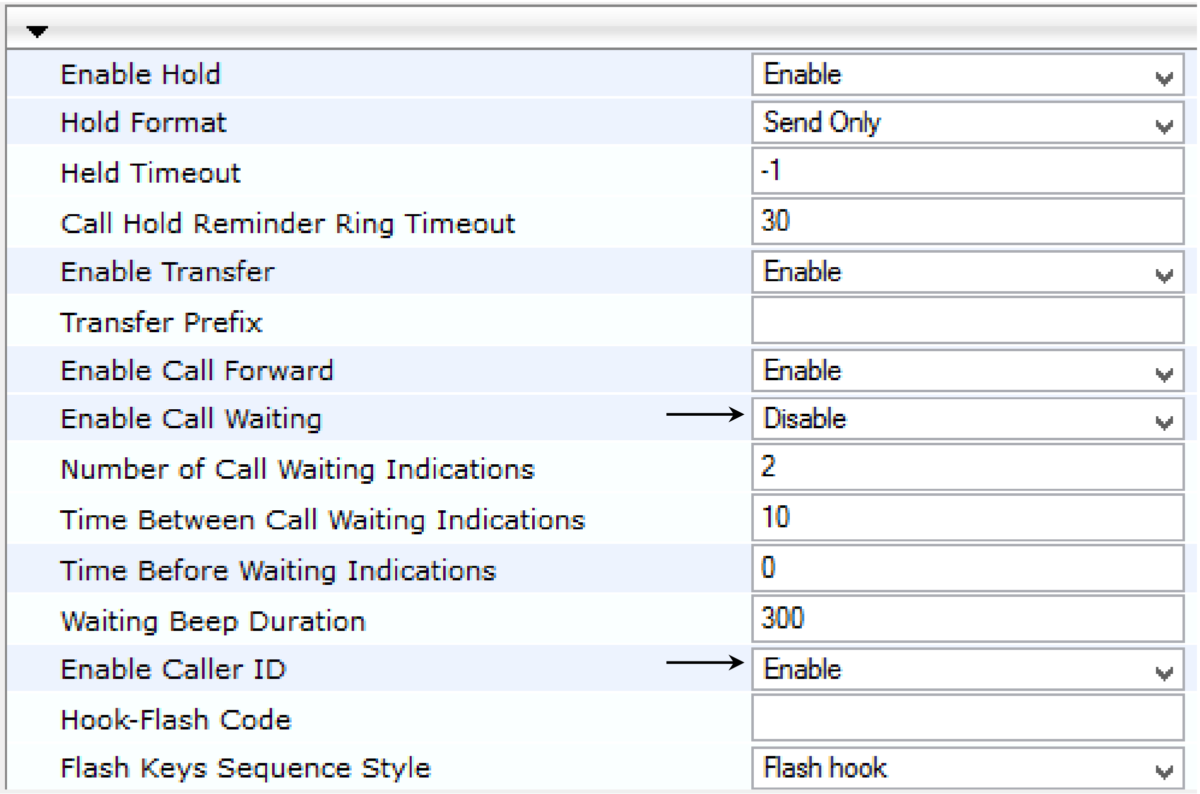


➢ **To configure the Supplementary Services:**

1. Open the Supplementary Services Page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** Submenu > **DTMF and Supplementary** Submenu > **Supplementary Services**).
2. Configure the following Parameters:

Parameter	Value
Enable Call Waiting	**Disable**
Enable Caller ID	**Enable**

Figure 5-23: Supplementary Services

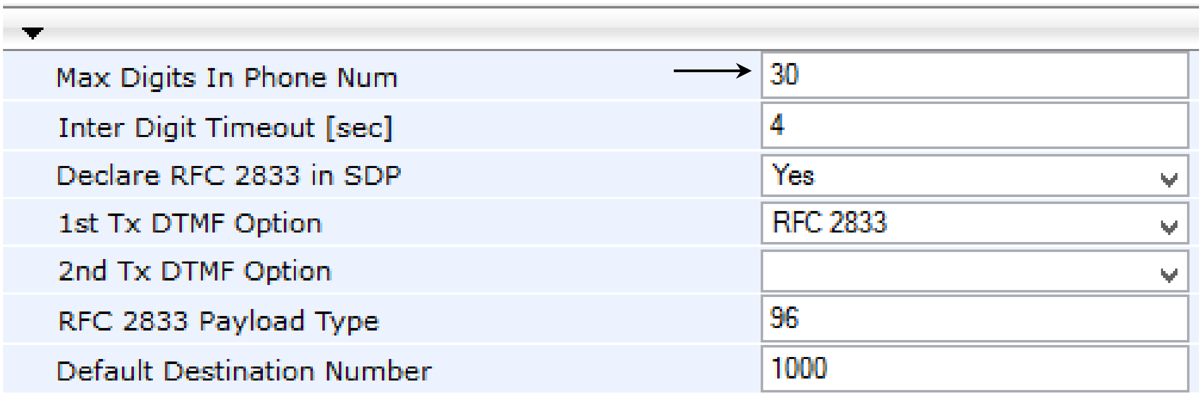


➢ **To configure the DTMF and Dialing Settings:**

1. Open the Supplementary Services Page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** Submenu > **DTMF and Supplementary** Submenu > **DTMF & Dialing**).
2. Configure the following Parameters:

Parameter	Value
Max Digits in Phone Numb	**30**

Figure 5-24: Supplementary Services

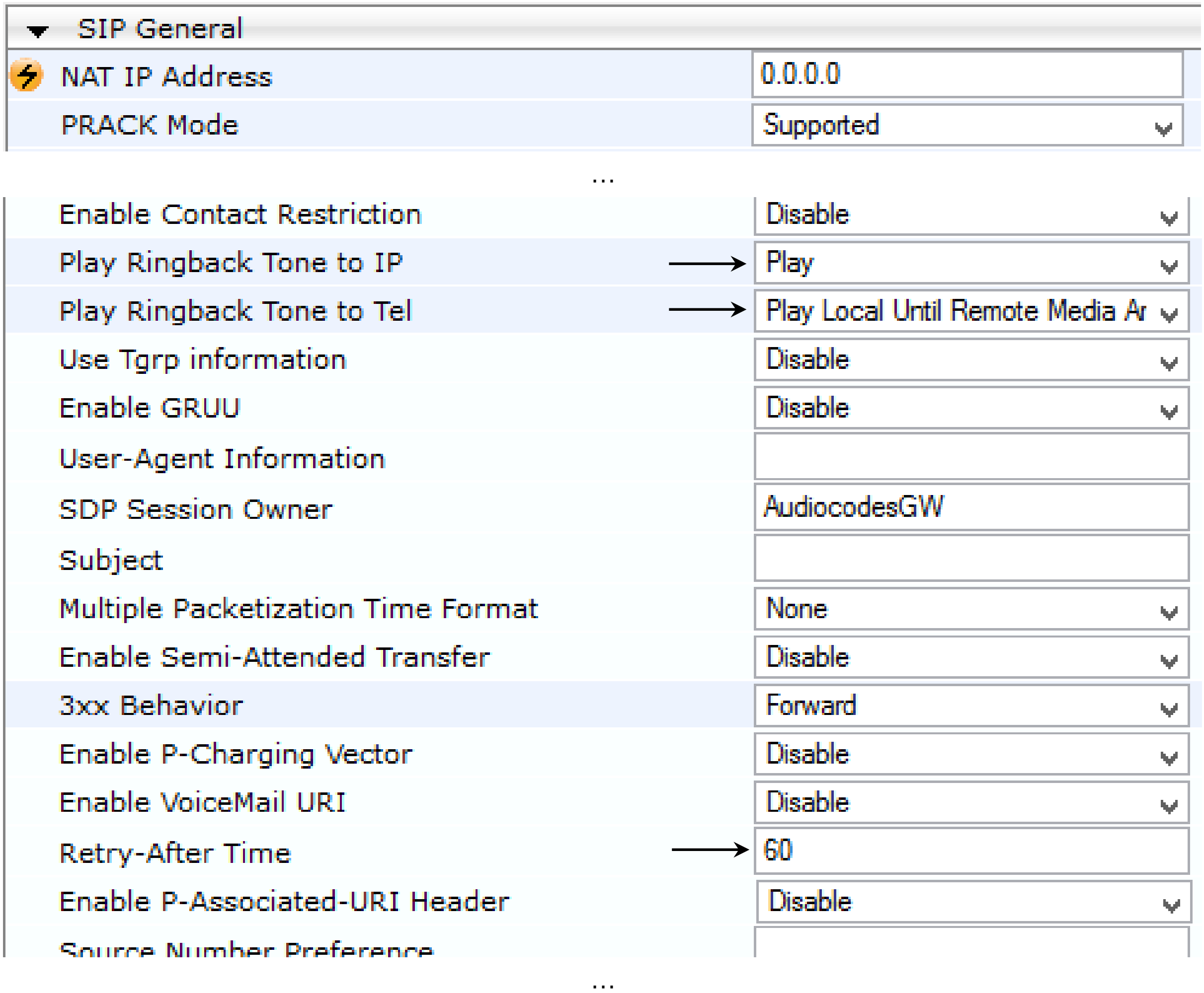


➢ **To configure the SIP General Parameters:**

1. Open the Supplementary Services Page (**Configuration** tab > **VoIP** menu > **SIP Definitions** Submenu > **General Parameters**).
2. Configure the following Parameters:

Parameter	Value
Play Ring back Tone to IP	**Enable**
Play Ring back Tone to Tel	**Play Local Until Remote Media Arrives**
Retry-After time	**60**

Figure 5-25: Supplementary Services

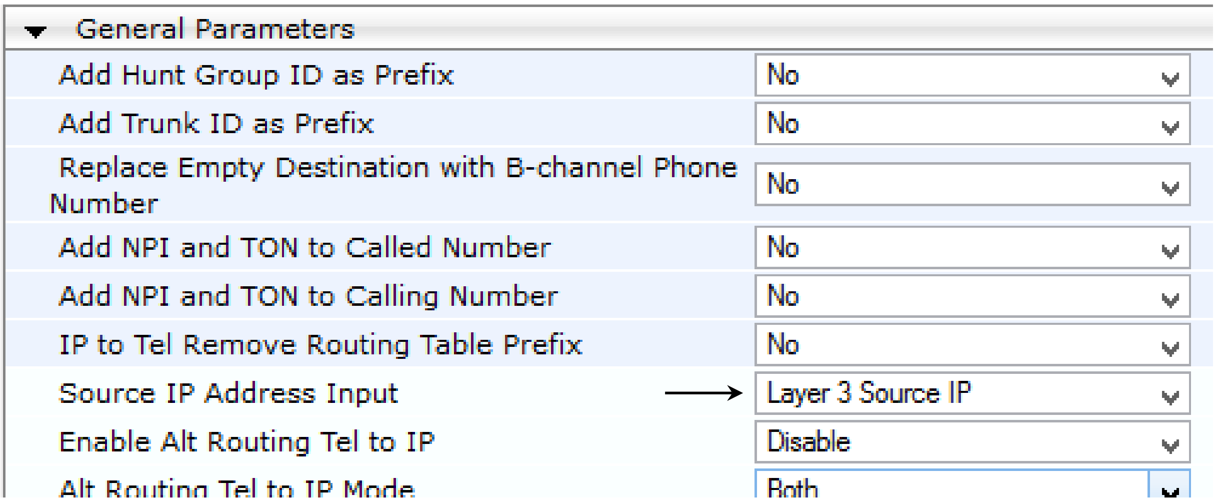


➢ To configure the Routing General Parameters:

1. Open the Supplementary Services Page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** Submenu > **Routing** Submenu > **Routing General Parameters**).
2. Configure the following Parameters:

Parameter	Value
Source IP Address Input	**Layer 3 Source IP**

Figure 5-26: Supplementary Services



➢ **To configure Enable Early 183 :**

1. Open the Admin page: append the case-sensitive suffix 'AdminPage' to the device's IP address in your Web browser's URL field (e.g., http://10.3.3.5/AdminPage).
2. Configure the following Parameters:

Parameter	Value
EnableEarly183	**1** (Enable)
BellModemTransportType	**2** (Bypass)

Figure 5-27: configuring EnableEarly183

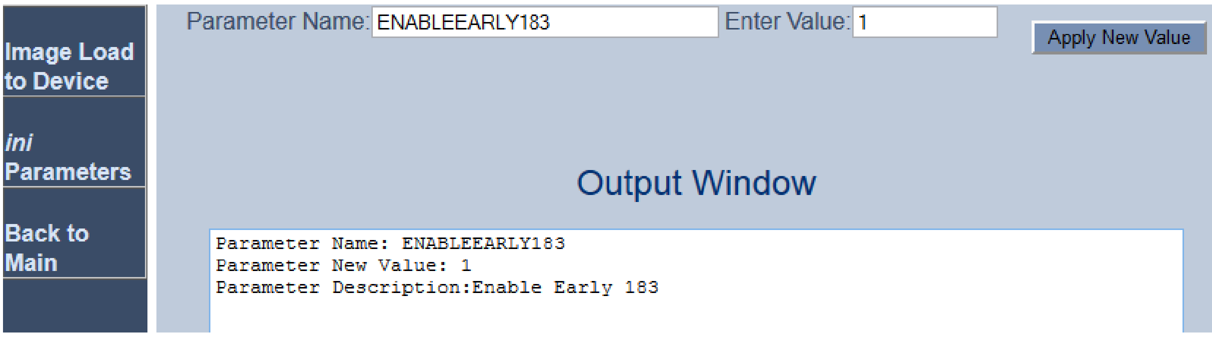
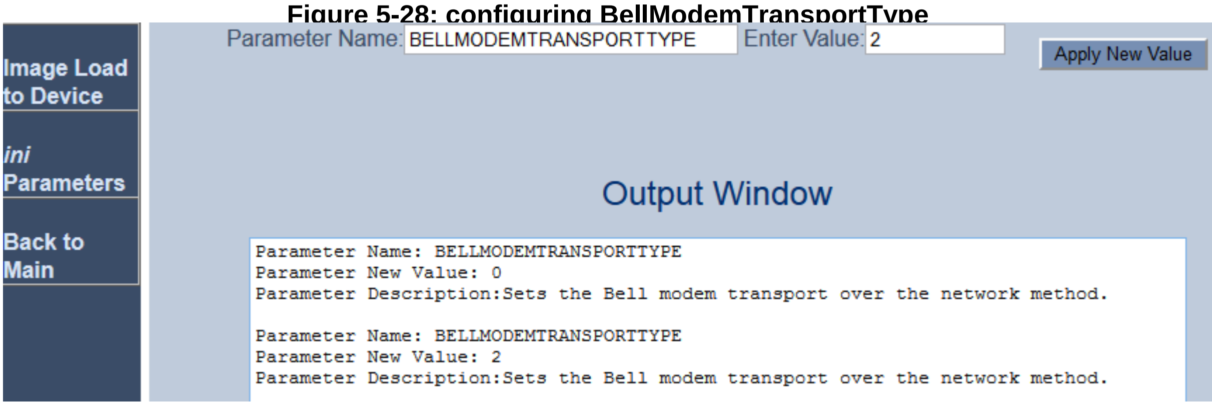


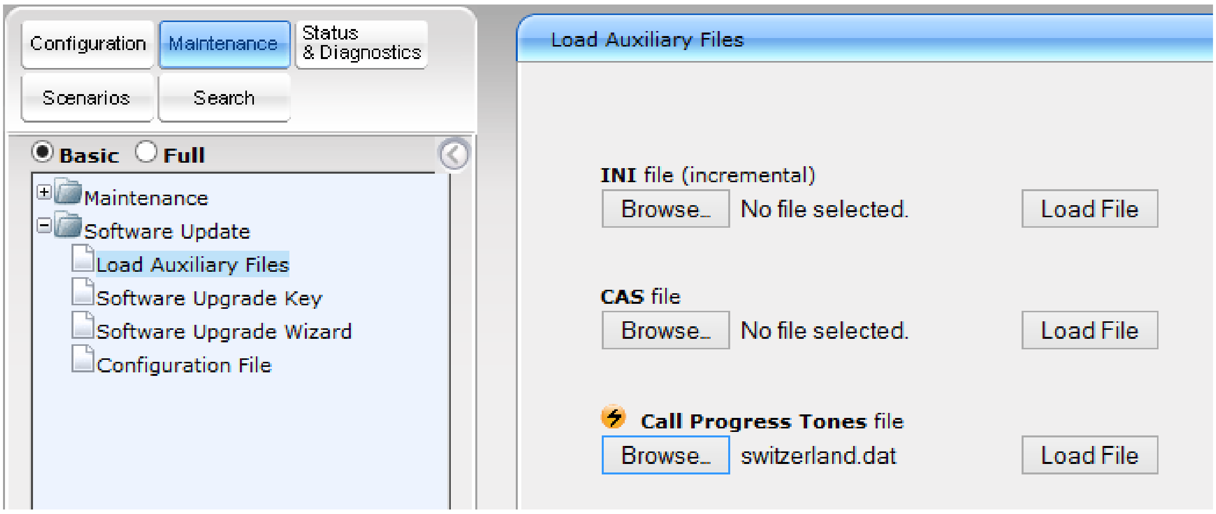
Figure 5-28: configuring BellModemTransportType



➢ **To configure the Swiss Call progress tones :**

1. Open the Message Manipulations page (**Maintenance** tab > **Software Update** menu > **Load Auxiliary Files**).
2. Browse for the call progress tones file on your computer and load the file in the E-SBC:

Figure 5-29: SBC General Settings



A AudioCodes MediaPack INI File

The *ini* configuration file of the E-SBC, corresponding to the Web-based configuration as described in Section 4 on page 33, is shown below:

> **Note:** To load and save an ini file, use the Configuration File page (**Maintenance** tab > **Software Update** menu > **Configuration File**).

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

;Board: MP-118 FXS
;Board Type: 56
;Serial Number: 3555183
;Slot Number: 1
;Software Version: 6.60A.245
;DSP Software Version: 204IM=> 660.10
;Board IP Address: 10.3.3.50
;Board Subnet Mask: 255.255.255.0
;Board Default Gateway: 10.3.3.1
;Ram size: 32M   Flash size: 8M
;Num of DSP Cores: 2  Num DSP Channels: 6
;Profile: NONE
;License Key limits aren't active full features capabilities are
available !;
;----------------------------------------------


[SYSTEM Params]

SyslogServerIP = 10.3.3.98
NTPServerUTCOffset = 3600
DayLightSavingTimeStart = '03:SUN/05:02:00'
DayLightSavingTimeEnd = '10:SUN/05:03:00'
DayLightSavingTimeEnable = 1
NTPServerIP = '0.ch.pool.ntp.org'
NTPSecondaryServerIP = '1.ch.pool.ntp.org'

[Analog Params]

FXSCountryCoefficients = 66

[Voice Engine Params]

CallerIDType = 1
FaxTransportMode = 2
V21ModemTransportType = 2
FaxBypassPayloadType = 8
```

```
BellModemTransportType = 2
ENABLEMEDIASECURITY = 1
SRTPTxPacketMKISize = 1

[SIP Params]

ENABLECALLERID = 1
MAXDIGITS = 30
ENABLECALLWAITING = 0
PLAYRBTONE2IP = 1
REGISTRATIONTIME = 3600
ISREGISTERNEEDED = 1
PLAYRBTONE2TEL = 3
GWDEBUGLEVEL = 5
MEDIASECURITYBEHAVIOUR = 3
SOURCEIPADDRESSINPUT = 1
ENABLEEARLY183 = 1
FAKERETRYAFTER = 60
ENABLESYMMETRICMKI = 1

[ InterfaceTable ]

FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_VlanID, InterfaceTable_InterfaceName,
InterfaceTable_PrimaryDNSServerIPAddress,
InterfaceTable_SecondaryDNSServerIPAddress;
InterfaceTable 0 = 6, 10, 10.3.3.50, 24, 10.3.3.1, 1, "O+M+C",
10.3.3.10, 0.0.0.0;

[ \InterfaceTable ]

[ TrunkGroup ]

FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum,
TrunkGroup_FirstTrunkId, TrunkGroup_FirstBChannel,
TrunkGroup_LastBChannel, TrunkGroup_FirstPhoneNumber,
TrunkGroup_ProfileId, TrunkGroup_LastTrunkId, TrunkGroup_Module;
TrunkGroup 0 = 1, 255, 1, 1, "+41614049794", 1, 255, 255;
TrunkGroup 1 = 1, 255, 2, 2, "+41614049798", 2, 255, 255;
TrunkGroup 2 = 1, 255, 3, 3, "+41614049799", 2, 255, 255;

[ \TrunkGroup ]


[ NumberMapTel2Ip ]

FORMAT NumberMapTel2Ip_Index = NumberMapTel2Ip_DestinationPrefix,
NumberMapTel2Ip_SourcePrefix, NumberMapTel2Ip_NumberType,
NumberMapTel2Ip_NumberPlan, NumberMapTel2Ip_RemoveFromLeft,
NumberMapTel2Ip_RemoveFromRight, NumberMapTel2Ip_LeaveFromRight,
NumberMapTel2Ip_Prefix2Add, NumberMapTel2Ip_Suffix2Add,
NumberMapTel2Ip_IsPresentationRestricted,
```

```
NumberMapTel2Ip_SrcTrunkGroupID, NumberMapTel2Ip_SrcIPGroupID,
NumberMapTel2Ip_DestIPGroupID;
NumberMapTel2Ip 0 = "00", "*", 255, 255, 2, 0, 255, "+", "", 255,
-1, -1, -1;
NumberMapTel2Ip 1 = "0", "*", 255, 255, 1, 0, 255, "+41", "", 255,
-1, -1, -1;
NumberMapTel2Ip 2 =
"[112,117,118,143,144,145,147,161,162,163,164,187]#", "*", 255,
255, 0, 0, 255, "+41", "", 255, -1, -1, -1;
NumberMapTel2Ip 3 = "[1811,1818,1850,1414,1415]#", "*", 255, 255,
0, 0, 255, "+41", "", 255, -1, -1, -1;
NumberMapTel2Ip 4 = "xxxx#", "*", 255, 255, 0, 0, 255, "+4161404",
"", 255, -1, -1, -1;
NumberMapTel2Ip 5 = "*", "*", 255, 255, 0, 0, 255, "+41", "", 255,
-1, -1, -1;

[ \NumberMapTel2Ip ]


[ SourceNumberMapIp2Tel ]

FORMAT SourceNumberMapIp2Tel_Index =
SourceNumberMapIp2Tel_DestinationPrefix,
SourceNumberMapIp2Tel_SourcePrefix,
SourceNumberMapIp2Tel_SourceAddress,
SourceNumberMapIp2Tel_SrcHost, SourceNumberMapIp2Tel_DestHost,
SourceNumberMapIp2Tel_NumberType,
SourceNumberMapIp2Tel_NumberPlan,
SourceNumberMapIp2Tel_RemoveFromLeft,
SourceNumberMapIp2Tel_RemoveFromRight,
SourceNumberMapIp2Tel_LeaveFromRight,
SourceNumberMapIp2Tel_Prefix2Add,
SourceNumberMapIp2Tel_Suffix2Add,
SourceNumberMapIp2Tel_IsPresentationRestricted;
SourceNumberMapIp2Tel 0 = "*", "+41", "*", "*", "*", 255, 255, 3,
0, 255, "0", "", 255;
SourceNumberMapIp2Tel 1 = "*", "+", "*", "*", "*", 255, 255, 1, 0,
255, "00", "", 255;

[ \SourceNumberMapIp2Tel ]


[ PstnPrefix ]

FORMAT PstnPrefix_Index = PstnPrefix_DestPrefix,
PstnPrefix_TrunkGroupId, PstnPrefix_SourcePrefix,
PstnPrefix_SourceAddress, PstnPrefix_ProfileId,
PstnPrefix_SrcIPGroupID, PstnPrefix_DestHostPrefix,
PstnPrefix_SrcHostPrefix, PstnPrefix_SrcSRDID, PstnPrefix_TrunkId,
PstnPrefix_CallSetupRulesSetId;
PstnPrefix 0 = "*", 1, "", "10.3.3.5", 0, -1, "", "", , -1, -1;

[ \PstnPrefix ]


[ ProxyIp ]
```

```
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = "sipgateway01.siptest03.local:5060", 1, 1;

[ \ProxyIp ]


[ TxDtmfOption ]

FORMAT TxDtmfOption_Index = TxDtmfOption_Type;
TxDtmfOption 0 = 4;

[ \TxDtmfOption ]


[ TrunkGroupSettings ]

FORMAT TrunkGroupSettings_Index = TrunkGroupSettings_TrunkGroupId,
TrunkGroupSettings_ChannelSelectMode,
TrunkGroupSettings_RegistrationMode,
TrunkGroupSettings_GatewayName, TrunkGroupSettings_ContactUser,
TrunkGroupSettings_ServingIPGroup,
TrunkGroupSettings_MWIInterrogationType,
TrunkGroupSettings_TrunkGroupName;
TrunkGroupSettings 0 = 1, 0, 0, "sipgateway01.siptest03.local",
"", 1, 255, "";

[ \TrunkGroupSettings ]


[ TelProfile ]

FORMAT TelProfile_Index = TelProfile_ProfileName,
TelProfile_TelPreference, TelProfile_CodersGroupID,
TelProfile_IsFaxUsed, TelProfile_JitterBufMinDelay,
TelProfile_JitterBufOptFactor, TelProfile_IPDiffServ,
TelProfile_SigIPDiffServ, TelProfile_DtmfVolume,
TelProfile_InputGain, TelProfile_VoiceVolume,
TelProfile_EnableReversePolarity,
TelProfile_EnableCurrentDisconnect,
TelProfile_EnableDigitDelivery, TelProfile_EnableEC,
TelProfile_MWIAnalog, TelProfile_MWIDisplay,
TelProfile_FlashHookPeriod, TelProfile_EnableEarlyMedia,
TelProfile_ProgressIndicator2IP, TelProfile_TimeForReorderTone,
TelProfile_EnableDIDWink, TelProfile_IsTwoStageDial,
TelProfile_DisconnectOnBusyTone, TelProfile_EnableVoiceMailDelay,
TelProfile_DialPlanIndex, TelProfile_Enable911PSAP,
TelProfile_SwapTelToIpPhoneNumbers, TelProfile_EnableAGC,
TelProfile_ECNlpMode, TelProfile_DigitalCutThrough,
TelProfile_EnableFXODoubleAnswer, TelProfile_CallPriorityMode,
TelProfile_FXORingTimeout;
TelProfile 1 = "Telefon", 1, 1, 0, 10, 10, 46, 40, -11, 0, 0, 0,
0, 0, 1, 0, 0, 700, 0, -1, 255, 0, 1, 1, 1, -1, 0, 0, 0, 0, 0, 0,
0, 0;
```

```
TelProfile 2 = "FAX & MODEM", 1, 2, 0, 10, 10, 46, 40, -11, 0, 0,
0, 0, 0, 1, 0, 0, 700, 0, -1, 255, 0, 1, 1, 1, -1, 0, 0, 0, 0, 0,
0, 0, 0;

[ \TelProfile ]


[ 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_SbcPrackMode,
IpProfile_SBCSessionExpiresMode, IpProfile_SBCRemoteUpdateSupport,
IpProfile_SBCRemoteReinviteSupport,
IpProfile_SBCRemoteDelayedOfferSupport,
IpProfile_SBCRemoteReferBehavior, IpProfile_SBCRemote3xxBehavior,
IpProfile_SBCRemoteMultiple18xSupport,
IpProfile_SBCRemoteEarlyMediaResponseType,
IpProfile_SBCRemoteEarlyMediaSupport,
IpProfile_EnableSymmetricMKI, IpProfile_MKISize,
IpProfile_SBCEnforceMKISize, IpProfile_SBCRemoteEarlyMediaRTP,
IpProfile_SBCRemoteSupportsRFC3960,
IpProfile_SBCRemoteCanPlayRingback, IpProfile_EnableEarly183,
IpProfile_EarlyAnswerTimeout, IpProfile_SBC2833DTMFPayloadType,
IpProfile_SBCUserRegistrationTime,
IpProfile_ResetSRTPStateUponRekey, IpProfile_AmdMode,
IpProfile_SBCReliableHeldToneSource, IpProfile_SBCPlayHeldTone,
IpProfile_SBCRemoteHoldFormat, IpProfile_GenerateSRTPKeys;
IpProfile 1 = "SBC", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0,
1, 0, -1, 1, 0, 2, -1, 1, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, -1, 0,
0, 0, 0, 0, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 2, 2, 1,
0, 0, 1, 0, 1, 1, 1, 0, 0, 0, 1, 1, 0, 0, 0, 0, 0, 1, 0, 0, 0;

[ \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, 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_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName;
IPGroup 1 = 0, "SBC", 1, "sipgateway01.siptest03.local", "", 0, -
1, -1, 0, -1, 0, "", 1, 1, -1, -1, -1, 0, 0, "", 0, -1, -1, "";

[ \IPGroup ]


[ 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 2 = "t38fax", 255, 255, -1, 255;

[ \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 = "t38fax", 255, 255, -1, 255;

[ \CodersGroup2 ]
```

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B AudioCodes E-SBC INI File

The *ini* configuration file of the E-SBC, corresponding to the Web-based configuration as described in Section 4 on page 33, is shown below:

Note: To load and save an ini file, use the Configuration File page (**Maintenance** tab > **Software Update** menu > **Configuration File**).

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

;Board: Mediant 1000 - MSBG
;HW Board Type: 47  FK Board Type: 67
;Serial Number: 3273006
;Slot Number: 1
;Software Version: 6.60A.257.004
;DSP Software Version: 204IM=> 660.10
;Second DSP Software Version: 624AE3=> 660.10
;Board IP Address: 10.3.3.5
;Board Subnet Mask: 255.255.255.0
;Board Default Gateway: 10.3.3.1
;Ram size: 495M   Flash size: 64M
;Num of DSP Cores: 9  Num DSP Channels: 52
;Num of physical LAN ports: 7
;Profile: NONE
;Key features:;Board Type: Mediant 1000 - MSBG ;Channel Type: RTP
DspCh=240 IPMediaDspCh=240 ;PSTN Protocols: ISDN IUA=4 CAS
;E1Trunks=4 ;T1Trunks=4 ;Security: IPSEC MediaEncryption
StrongEncryption EncryptControlProtocol ;Coders: G723 G729 GSM-FR
G727 ILBC ;DSP Voice features: IpmDetector ;IP Media: Conf VXML
VoicePromptAnnounc(H248.9) ExtVoicePrompt=1MB ;Control Protocols:
MSFT MGCP MEGACO SIP SBC=30 FEU=30;Default features:;Coders: G711
G726;

;--------- Mediant 1000 - MSBG HW components ---------
;
; Slot # : Module type : # of ports : # of DSPs
;----------------------------------------------
;      1 : Empty
;      2 : FXS         :          4 :         1
;      3 : IPMedia     :          0 :         2
;      4 : Empty
;      5 : Empty
;      6 : Empty
;----------------------------------------------


[SYSTEM Params]
```

```
NTPServerUTCOffset = 3600
DayLightSavingTimeStart = '03:SUN/05:02:00'
DayLightSavingTimeEnd = '10:SUN/05:03:00'
DayLightSavingTimeEnable = 1
NTPServerIP = '0.ch.pool.ntp.org'
NTPSecondaryServerIP = '1.ch.pool.ntp.org'

[Voice Engine Params]

ENABLEMEDIASECURITY = 1
NoOpEnable = 1
SRTPTxPacketMKISize = 1
CallProgressTonesFilename = 'switzerland.dat'

[SIP Params]

MEDIACHANNELS = 144
SIPGATEWAYNAME = ‘nn.nn.nn.nn'
USEGATEWAYNAMEFOROPTIONS = 1
MEDIASECURITYBEHAVIOUR = 1
ENABLESBCAPPLICATION = 1
FAKERETRYAFTER = 60
SBCMAXFORWARDSLIMIT = 70
ENABLESYMMETRICMKI = 1
SBCFORKINGHANDLINGMODE = 1
SBCENFORCEMEDIAORDER = 1

[ PhysicalPortsTable ]

FORMAT PhysicalPortsTable_Index = PhysicalPortsTable_Port,
PhysicalPortsTable_Mode, PhysicalPortsTable_NativeVlan,
PhysicalPortsTable_SpeedDuplex,
PhysicalPortsTable_PortDescription,
PhysicalPortsTable_GroupMember, PhysicalPortsTable_GroupStatus;
PhysicalPortsTable 0 = "GE_0_1", 1, 1, 4, "KUNDE Port #0",
"GROUP_1", "Redundant";
PhysicalPortsTable 1 = "GE_0_2", 1, 1, 4, "KUNDE Port #1",
"GROUP_1", "Active";
PhysicalPortsTable 2 = "GE_7_1", 1, 2, 4, "SWISS Port #2",
"GROUP_2", "Redundant";
PhysicalPortsTable 3 = "GE_7_2", 1, 2, 4, "SWISS Port #3",
"GROUP_2", "Active";
PhysicalPortsTable 4 = "GE_7_3", 1, 3, 4, "OAMP Port #4",
"GROUP_3", "Active";
PhysicalPortsTable 5 = "GE_7_4", 1, 3, 4, "OAMP Port #5",
"GROUP_3", "Redundant";

[ \PhysicalPortsTable ]


[ EtherGroupTable ]
```

```
FORMAT EtherGroupTable_Index = EtherGroupTable_Group,
EtherGroupTable_Mode, EtherGroupTable_Member1,
EtherGroupTable_Member2;
EtherGroupTable 0 = "GROUP_1", 2, GE_0_1, GE_0_2;
EtherGroupTable 1 = "GROUP_2", 2, GE_7_1, GE_7_2;
EtherGroupTable 2 = "GROUP_3", 2, GE_7_3, GE_7_4;

[ \EtherGroupTable ]


[ 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 1 = 6, 10, 10.3.3.5, 24, 10.3.3.1, 1, "KUNDE",
10.3.3.10, 0.0.0.0, GROUP_1;
InterfaceTable 2 = 5, 10, 10.3.4.5, 24, 10.3.4.3, 2, "SWISSCOM",
0.0.0.0, 0.0.0.0, GROUP_2;

[ \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 ]


[ IPMediaChannels ]

FORMAT IPMediaChannels_Index = IPMediaChannels_ModuleID,
IPMediaChannels_DSPChannelsReserved;
IPMediaChannels 0 = 3, 48;

[ \IPMediaChannels ]


[ CpMediaRealm ]

FORMAT CpMediaRealm_Index = CpMediaRealm_MediaRealmName,
CpMediaRealm_IPv4IF, CpMediaRealm_IPv6IF,
CpMediaRealm_PortRangeStart, CpMediaRealm_MediaSessionLeg,
CpMediaRealm_PortRangeEnd, CpMediaRealm_IsDefault;
CpMediaRealm 1 = "KUNDE_MR", KUNDE, , 6000, 150, 7490, 1;
```

```
CpMediaRealm 2 = "SWISSCOM_MR", SWISSCOM, , 7500, 150, 8990, 0;

[ \CpMediaRealm ]


[ SRD ]

FORMAT SRD_Index = SRD_Name, SRD_MediaRealm,
SRD_IntraSRDMediaAnchoring, SRD_BlockUnRegUsers,
SRD_MaxNumOfRegUsers, SRD_EnableUnAuthenticatedRegistrations;
SRD 1 = "KUNDE_SRD", "KUNDE_MR", 0, 0, -1, 1;
SRD 2 = "SWISSCOM_SRD", "SWISSCOM_MR", 0, 0, -1, 1;

[ \SRD ]


[ SBCAlternativeRoutingReasons ]

FORMAT SBCAlternativeRoutingReasons_Index =
SBCAlternativeRoutingReasons_ReleaseCause;
SBCAlternativeRoutingReasons 0 = 404;

[ \SBCAlternativeRoutingReasons ]


[ ProxyIp ]

FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = "sipt3srvlfe01.siptest03.local:5068", 1, 1;
ProxyIp 2 = "nn.nn.nn.nn:5060", 1, 2;

[ \ProxyIp ]


[ 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_SbcPrackMode,
IpProfile_SBCSessionExpiresMode, IpProfile_SBCRemoteUpdateSupport,
IpProfile_SBCRemoteReinviteSupport,
IpProfile_SBCRemoteDelayedOfferSupport,
IpProfile_SBCRemoteReferBehavior, IpProfile_SBCRemote3xxBehavior,
IpProfile_SBCRemoteMultiple18xSupport,
IpProfile_SBCRemoteEarlyMediaResponseType,
IpProfile_SBCRemoteEarlyMediaSupport,
IpProfile_EnableSymmetricMKI, IpProfile_MKISize,
IpProfile_SBCEnforceMKISize, IpProfile_SBCRemoteEarlyMediaRTP,
IpProfile_SBCRemoteSupportsRFC3960,
IpProfile_SBCRemoteCanPlayRingback, IpProfile_EnableEarly183,
IpProfile_EarlyAnswerTimeout, IpProfile_SBC2833DTMFPayloadType,
IpProfile_SBCUserRegistrationTime,
IpProfile_ResetSRTPStateUponRekey, IpProfile_AmdMode,
IpProfile_SBCReliableHeldToneSource, IpProfile_SBCPlayHeldTone,
IpProfile_SBCRemoteHoldFormat, IpProfile_GenerateSRTPKeys;
IpProfile 1 = "LYNC", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 0, 0,
0, 0, 0, -1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, -1,
0, 0, 0, 0, 0, 0, 8, 300, 400, 0, 0, 0, 1, 1, 0, 1, 3, 0, 1, 2, 0,
0, 0, 1, 0, 1, 0, 0, 0, 1, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 0, 0;
IpProfile 2 = "SWISSCOM", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 0,
0, 0, 0, 0, -1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, -
1, 0, 2, 0, 0, 0, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 2,
2, 1, 0, 0, 1, 0, 1, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 1, 0, 0,
0;

[ \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, 300, 0, 1, 1, 0, -1;
ProxySet 2 = 1, 300, 0, 0, 2, 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_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName;
IPGroup 1 = 0, "LYNC", 1, "", "", 0, -1, -1, 0, -1, 1, "KUNDE_MR",
1, 1, -1, -1, 1, 0, 0, "", 0, -1, -1, "";
IPGroup 2 = 0, "SWISSCOM", 2, "nn.nn.nn.nn", "", 0, -1, -1, 0, -1,
2, "SWISSCOM_MR", 1, 2, -1, 4, 2, 0, 0, "", 0, -1, -1, "";
IPGroup 3 = 1, "ANALOG", -1, "", "", 0, -1, -1, 0, -1, 1,
"KUNDE_MR", 0, 0, -1, -1, -1, 0, 0, "", 0, -1, -1, "";

[ \IPGroup ]


[ ConditionTable ]

FORMAT ConditionTable_Index = ConditionTable_Condition,
ConditionTable_Description;
ConditionTable 1 = "header.user-agent regex MP\-1.*/v\.6\.60A\.",
"user-agent prÃ¼fen";

[ \ConditionTable ]


[ IP2IPRouting ]

FORMAT IP2IPRouting_Index = IP2IPRouting_SrcIPGroupID,
IP2IPRouting_SrcUsernamePrefix, IP2IPRouting_SrcHost,
IP2IPRouting_DestUsernamePrefix, IP2IPRouting_DestHost,
IP2IPRouting_RequestType, IP2IPRouting_MessageCondition,
IP2IPRouting_ReRouteIPGroupID, IP2IPRouting_Trigger,
IP2IPRouting_DestType, IP2IPRouting_DestIPGroupID,
IP2IPRouting_DestSRDID, IP2IPRouting_DestAddress,
IP2IPRouting_DestPort, IP2IPRouting_DestTransportType,
IP2IPRouting_AltRouteOptions, IP2IPRouting_CostGroup;
IP2IPRouting 0 = -1, "*", "*", "*", "*", 6, , -1, 0, 1, -1, ,
"internal", 0, -1, 0, ;
IP2IPRouting 1 = 3, "*", "*", "*", "*", 2, , -1, 0, 0, 3, , "", 0,
-1, 0, ;
IP2IPRouting 2 = 1, "*", "*", "*", "*", 0, , -1, 0, 0, 3, , "", 0,
-1, 0, ;
IP2IPRouting 3 = 1, "*", "*", "*", "*", 0, , -1, 0, 0, 2, , "", 0,
-1, 1, ;
IP2IPRouting 4 = 2, "*", "*", "*", "*", 0, , -1, 0, 0, 1, , "", 0,
-1, 0, ;
IP2IPRouting 5 = 2, "*", "*", "*", "*", 0, , -1, 0, 0, 3, , "", 0,
-1, 1, ;
IP2IPRouting 6 = 3, "*", "*", "*", "*", 0, , -1, 0, 0, 1, , "", 0,
-1, 0, ;
IP2IPRouting 7 = 3, "*", "*", "*", "*", 0, , -1, 0, 0, 2, , "", 0,
-1, 1, ;

[ \IP2IPRouting ]
```

```
[ Classification ]

FORMAT Classification_Index = Classification_MessageCondition,
Classification_SrcSRDID, Classification_SrcAddress,
Classification_SrcPort, Classification_SrcTransportType,
Classification_SrcUsernamePrefix, Classification_SrcHost,
Classification_DestUsernamePrefix, Classification_DestHost,
Classification_ActionType, Classification_SrcIPGroupID;
Classification 1 = 1, 1, "", 0, 1, "+4161404979x#",
"sipgateway01.siptest03.local", "",
"sipgateway01.siptest03.local", 1, 3;

[ \Classification ]


[ SIPInterface ]

FORMAT SIPInterface_Index = SIPInterface_NetworkInterface,
SIPInterface_ApplicationType, SIPInterface_UDPPort,
SIPInterface_TCPPort, SIPInterface_TLSPort, SIPInterface_SRD,
SIPInterface_MessagePolicy, SIPInterface_TLSMutualAuthentication,
SIPInterface_TCPKeepaliveEnable,
SIPInterface_ClassificationFailureResponseType;
SIPInterface 1 = "KUNDE", 2, 0, 5060, 0, 1, , -1, 0, 500;
SIPInterface 2 = "SWISS", 2, 0, 5060, 0, 2, , -1, 0, 500;

[ \SIPInterface ]




[ MessageManipulations ]

FORMAT MessageManipulations_Index = MessageManipulations_ManSetID,
MessageManipulations_MessageType, MessageManipulations_Condition,
MessageManipulations_ActionSubject,
MessageManipulations_ActionType, MessageManipulations_ActionValue,
MessageManipulations_RowRole;
MessageManipulations 0 = 1, "invite.request", "", "header.request-
uri.url.host", 2, "param.message.address.dst.address", 0;
MessageManipulations 1 = 1, "invite.request", "",
"header.to.url.host", 2, "param.message.address.dst.address", 0;
MessageManipulations 2 = 1, "invite.request", "",
"header.from.url.host", 2, "'sipgateway01.siptest03.local'", 0;
MessageManipulations 3 = 2, "invite.request", "header.referred-by
exists", "header.diversion", 0, "'<' + header.referred-by.url +
'>'", 0;
MessageManipulations 4 = 2, "invite.request", "header.referred-by
exists", "header.referred-by", 1, "", 0;

[ \MessageManipulations ]
```

Configuration Note

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