

Microsoft® Lync™ Server 2013 and Twilio SIP Trunk using AudioCodes Mediant™ E-SBC

Version 7.0



Microsoft Partner
Gold Communications



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Notice

This document describes how to connect the Microsoft Lync Server 2013 and 8BTwilio SIP Trunk using AudioCodes Mediant E-SBC product series.

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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 8BTwilio's SIP Trunk and Microsoft's Lync Server 2013 environment.

1.1 Intended Audience

The document is intended for engineers, or AudioCodes and 8BTwilio Partners who are responsible for installing and configuring 8BTwilio's SIP Trunk 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.

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2 Component Information

2.1 AudioCodes E-SBC Version

Table 2-1: AudioCodes E-SBC Version

SBC Vendor	AudioCodes
Models	<ul style="list-style-type: none"> ▪ Mediant 500 E-SBC ▪ 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_F7.00A.013.015
Protocol	<ul style="list-style-type: none"> ▪ SIP/UDP (to the 8BTwilio SIP Trunk) ▪ SIP/TCP or TLS (to the Lync FE Server)
Additional Notes	None

2.2 Twilio SIP Trunking Version

Table 2-2: 8BTwilio Version

Vendor/Service Provider	8BTwilio
SSW Model/Service	Twilio
Software Version	
Protocol	SIP
Additional Notes	None

2.3 Microsoft Lync Server 2013 Version

Table 2-3: Microsoft Lync Server 2013 Version

Vendor	Microsoft
Model	Microsoft Lync
Software Version	Release 2013 5.0.8308.556
Protocol	SIP
Additional Notes	None

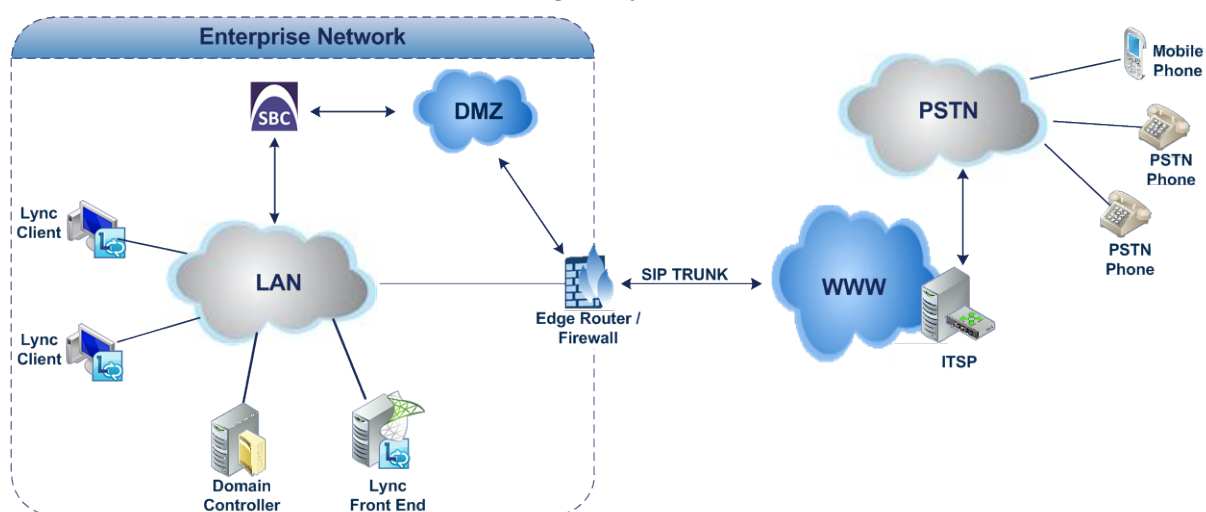
2.4 Interoperability Test Topology

The interoperability testing between AudioCodes E-SBC and 8BTwilio SIP Trunk 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 wishes to offer its employees enterprise-voice capabilities and to connect the Enterprise to the PSTN network using 8BTwilio's SIP Trunking service.
- 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 8BTwilio's SIP Trunk located in the public network.

The figure below illustrates this interoperability test topology:

Figure 2-1: Interoperability Test Topology between E-SBC and Microsoft Lync with 8BTwilio SIP Trunk



2.4.1 Environment Setup

The interoperability test topology includes the following environment setup:

Table 2-4: Environment Setup

Area	Setup
Network	<ul style="list-style-type: none"> ▪ Microsoft Lync Server 2013 environment is located on the Enterprise's LAN ▪ 8BTwilio SIP Trunk is located on the WAN
Signaling Transcoding	<ul style="list-style-type: none"> ▪ Microsoft Lync Server 2013 operates with SIP-over-TLS transport type ▪ 8BTwilio SIP Trunk operates with SIP-over-UDP transport type
Codecs Transcoding	<ul style="list-style-type: none"> ▪ Microsoft Lync Server 2013 supports a wide range of coders ▪ 8BTwilio SIP Trunk supports G.711U-law coder only
Media Transcoding	<ul style="list-style-type: none"> ▪ Microsoft Lync Server 2013 operates with SRTP media type ▪ 8BTwilio SIP Trunk operates with RTP media type

2.4.2 Known Limitations

The following limitation was observed in the Interoperability tests done for the AudioCodes E-SBC interworking between Microsoft Lync Server 2013 and 8BTwilio's SIP Trunk:

- There is a 120 seconds broken connection timeout defined on the 8BTwilio SIP Trunk. Consequently, the SIP trunk always expects to receive RTP packets. When a call is muted or placed on Hold, no packets are sent from the Microsoft Lync Server 2013 side. To resolve this issue, Force Transcoding should be enabled in the E-SBC IP Profile (see Section 4.6 on page 45).

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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.



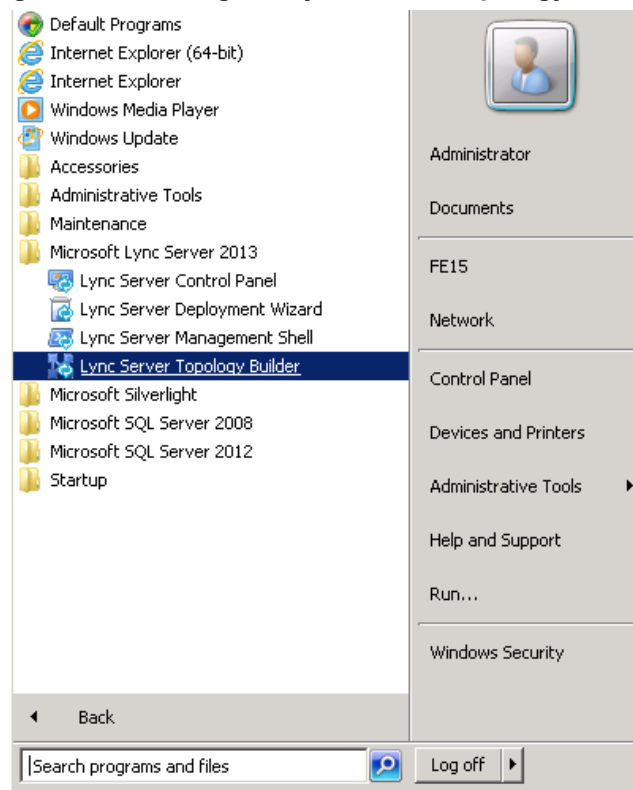
Note: Dial plans, voice policies, and PSTN usages are also necessary for Enterprise voice deployment; however, they are beyond the scope of this document.

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.

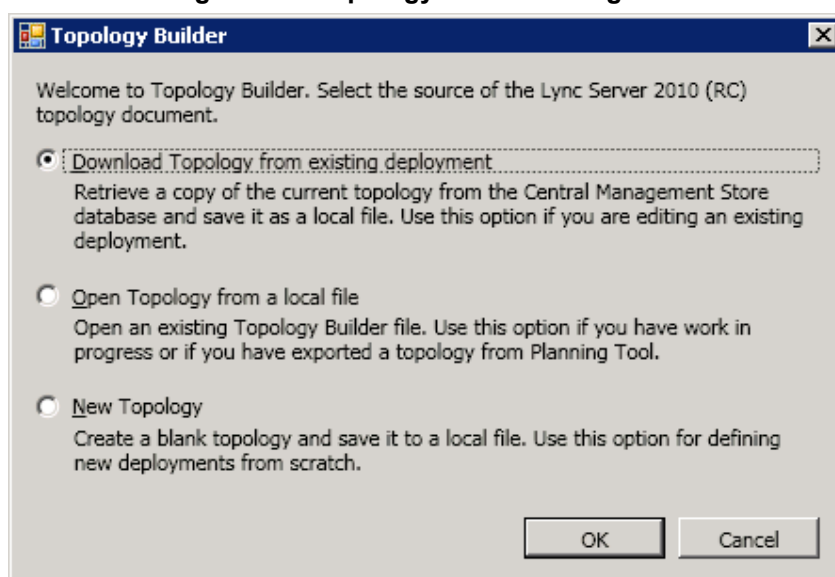
- **To configure E-SBC as IP/PSTN Gateway and associate it with Mediation Server:**
- 1. On the server where the Topology Builder is installed, start the Lync Server 2013 Topology Builder (Windows **Start** menu > **All Programs** > **Lync Server Topology Builder**), as shown below:

Figure 3-1: Starting the Lync Server Topology Builder



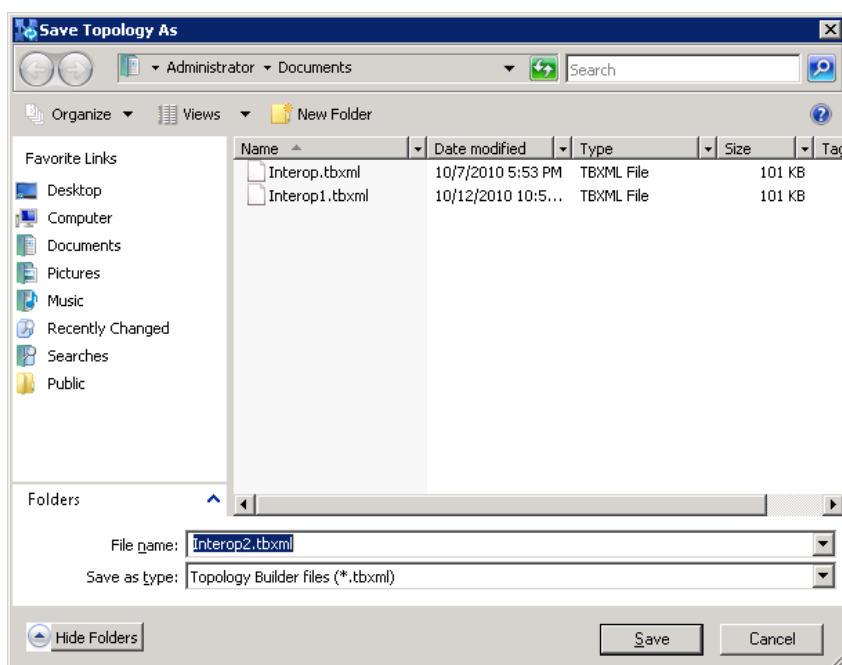
The following is displayed:

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:

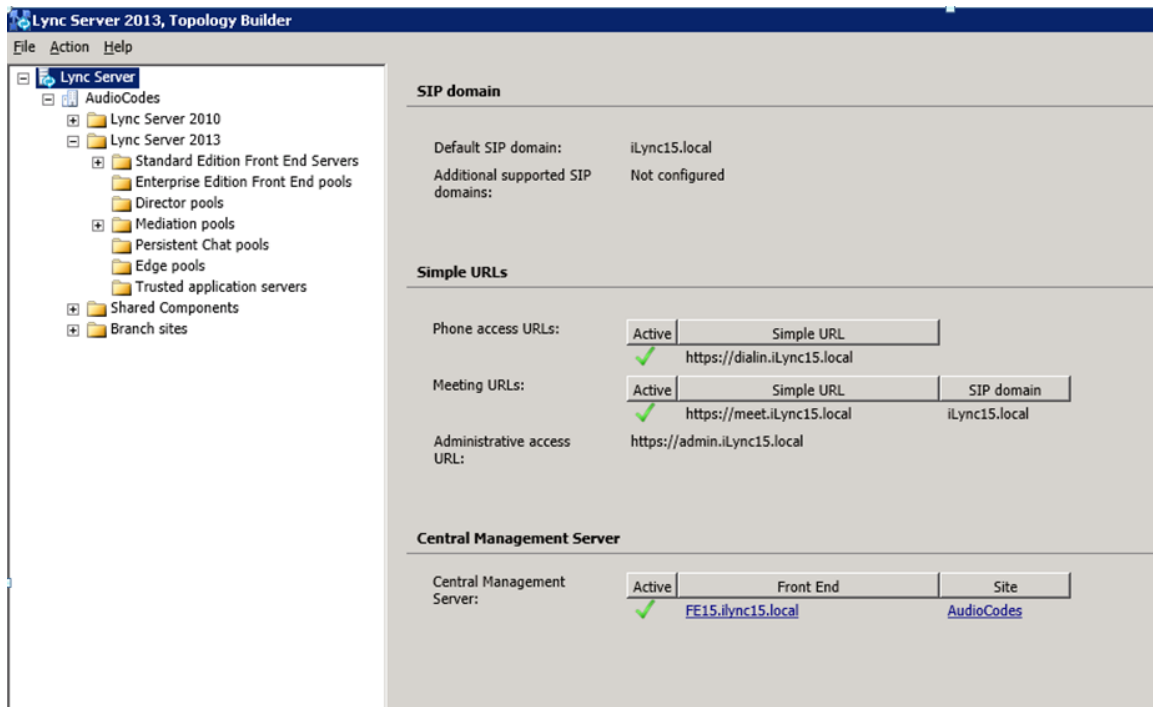
Figure 3-3: Save Topology Dialog Box



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.

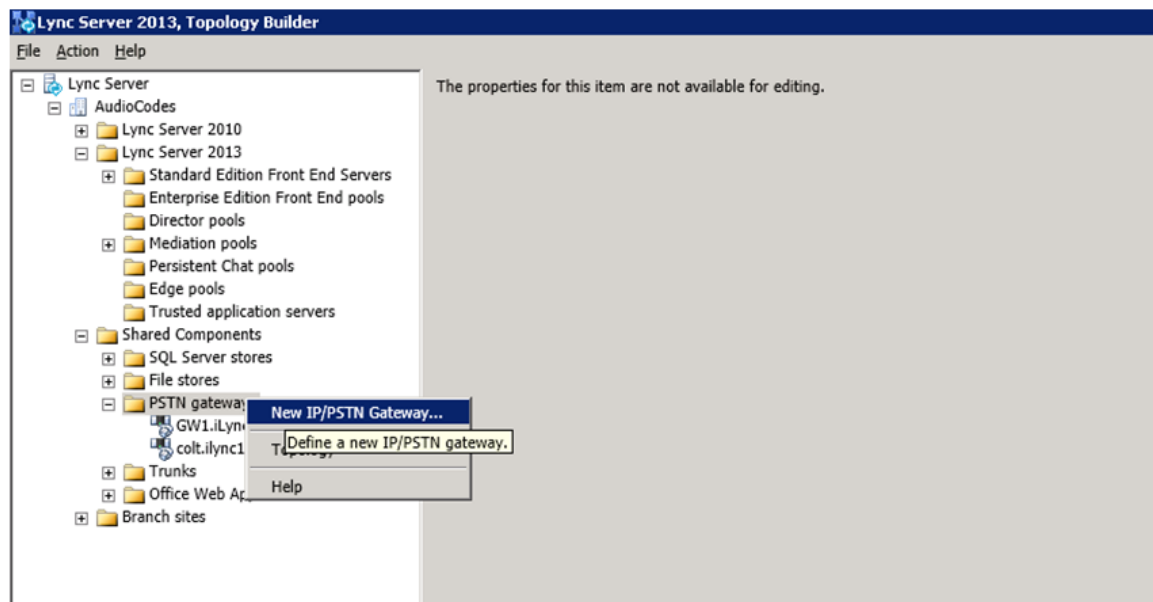
The Topology Builder screen with the downloaded Topology is displayed:

Figure 3-4: Downloaded Topology



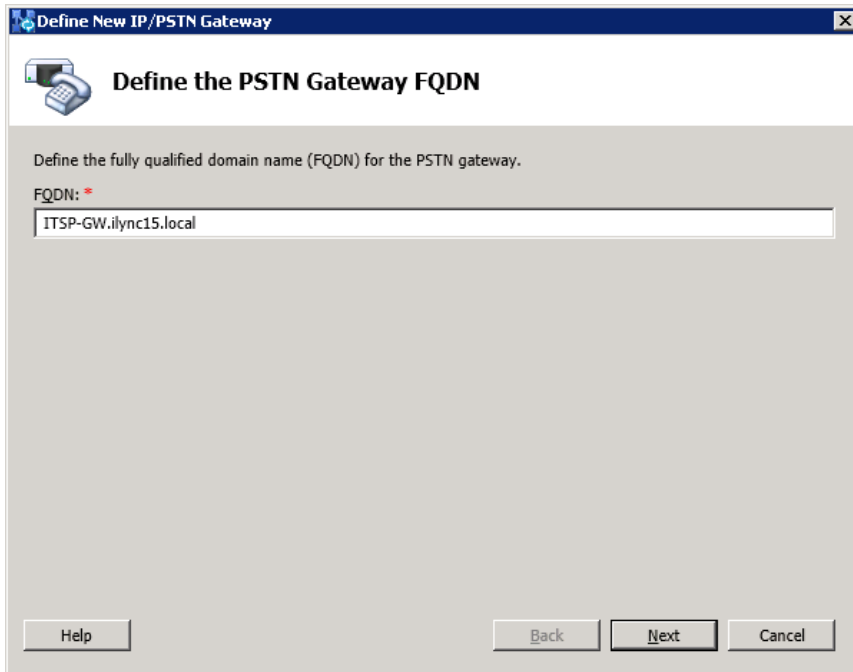
- Under the **Shared Components** node, right-click the **PSTN gateways** node, and then from the shortcut menu, choose **New IP/PSTN Gateway**, as shown below:

Figure 3-5: Choosing New IP/PSTN Gateway



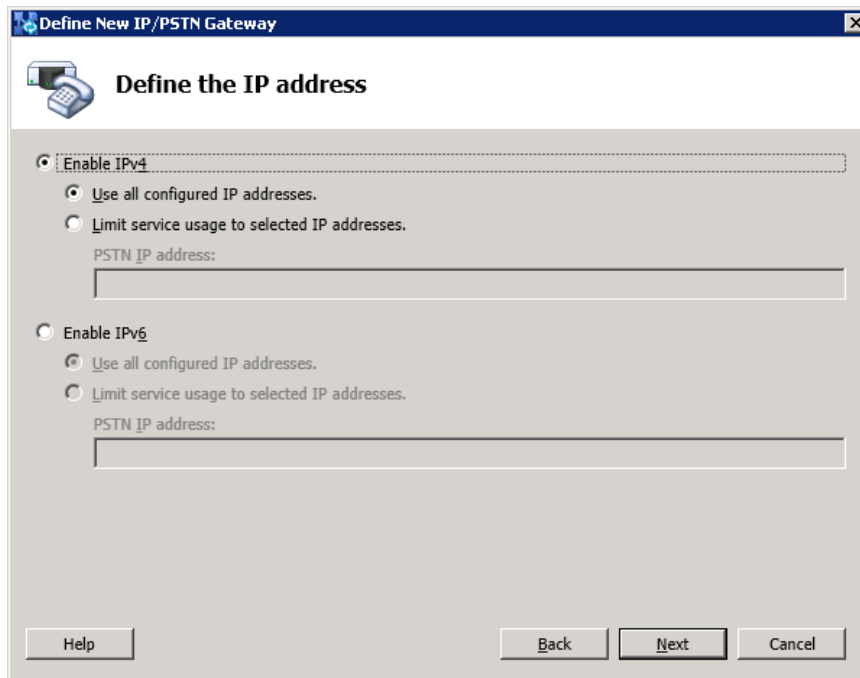
The following is displayed:

Figure 3-6: Define the PSTN Gateway FQDN



5. Enter the Fully Qualified Domain Name (FQDN) of the E-SBC (e.g., **ITSP-GW.ilync15.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

Define the root trunk

Trunk name: *
ITSP-GW.ilync15.local

Listening port for IP/PSTN gateway: *
5067

SIP Transport Protocol:
TLS

Associated Mediation Server:
FE15.ilync15.local AudioCodes

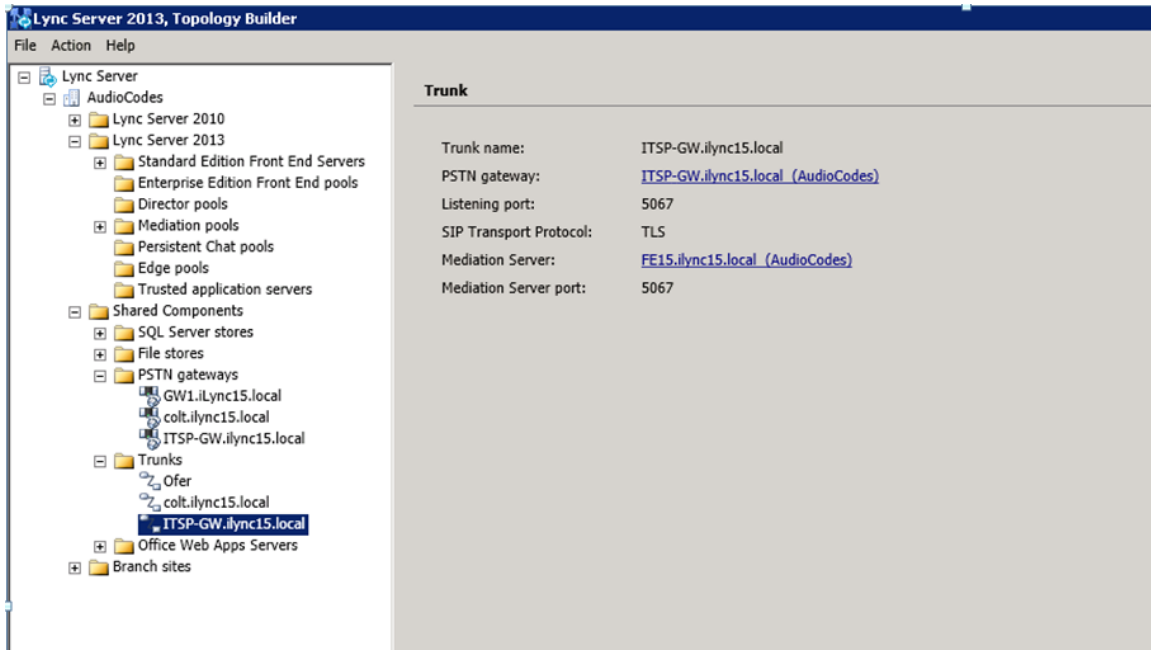
Associated Mediation Server port: *
5067

Help Back Finish Cancel

- 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., **5067**).
- In the 'SIP Transport Protocol' field, select the transport type (e.g., **TLS**) that the trunk uses.
- In the 'Associated Mediation Server' field, select the Mediation Server pool to associate with the root trunk of this PSTN gateway.
- 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., **5067**).
- Click **Finish**.

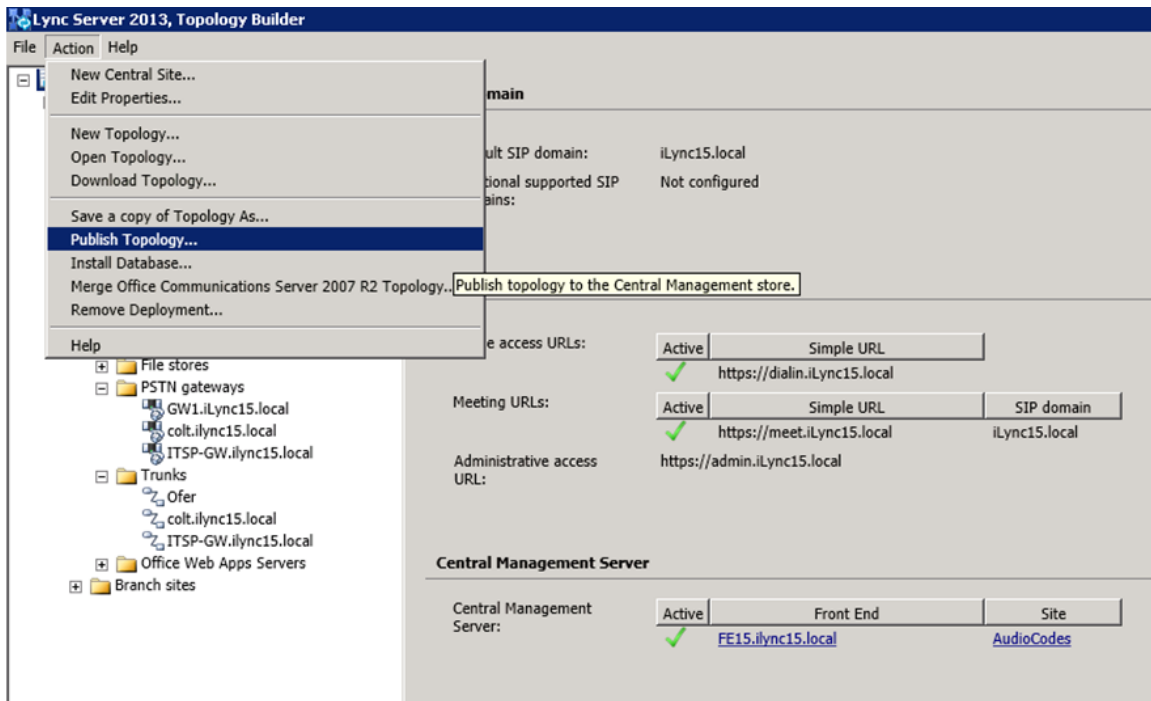
The E-SBC is added as a PSTN gateway, and a trunk is created as shown below:

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



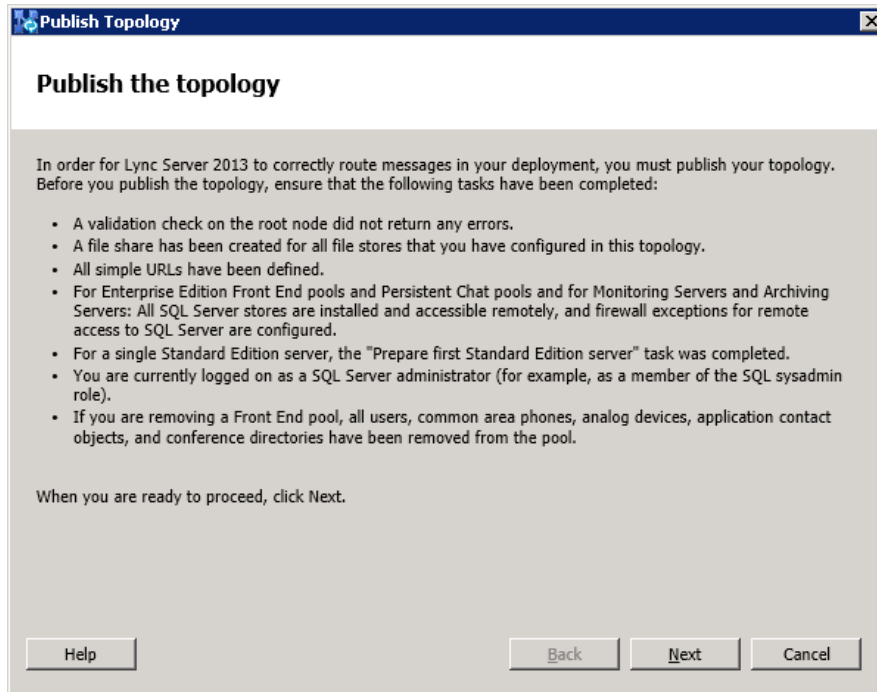
8. Publish the Topology: In the main tree, select the root node **Lync Server**, and then from the **Action** menu, choose **Publish Topology**, as shown below:

Figure 3-10: Choosing Publish Topology



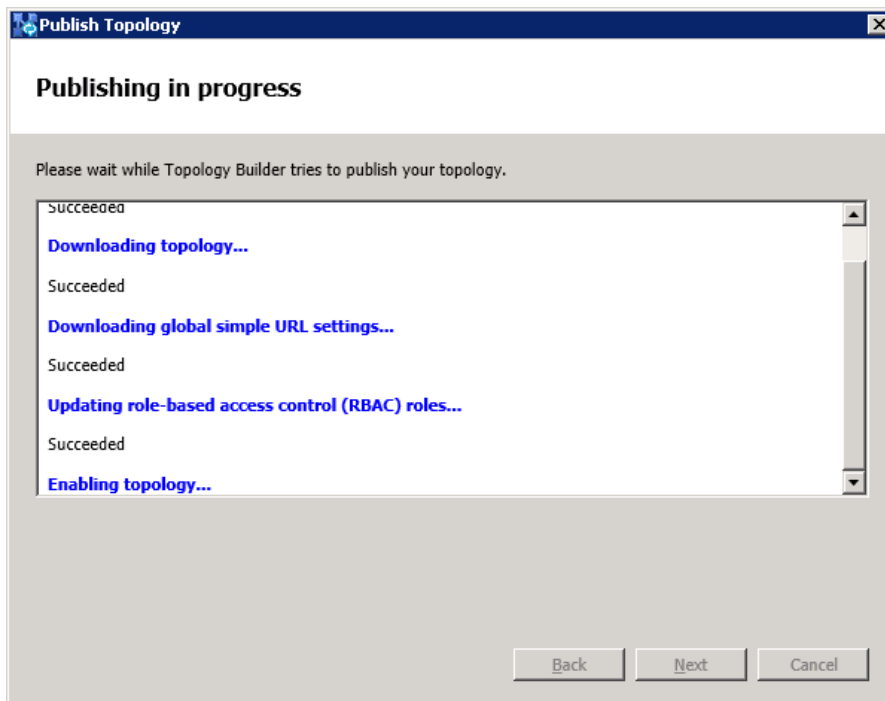
The following is displayed:

Figure 3-11: Publish the Topology



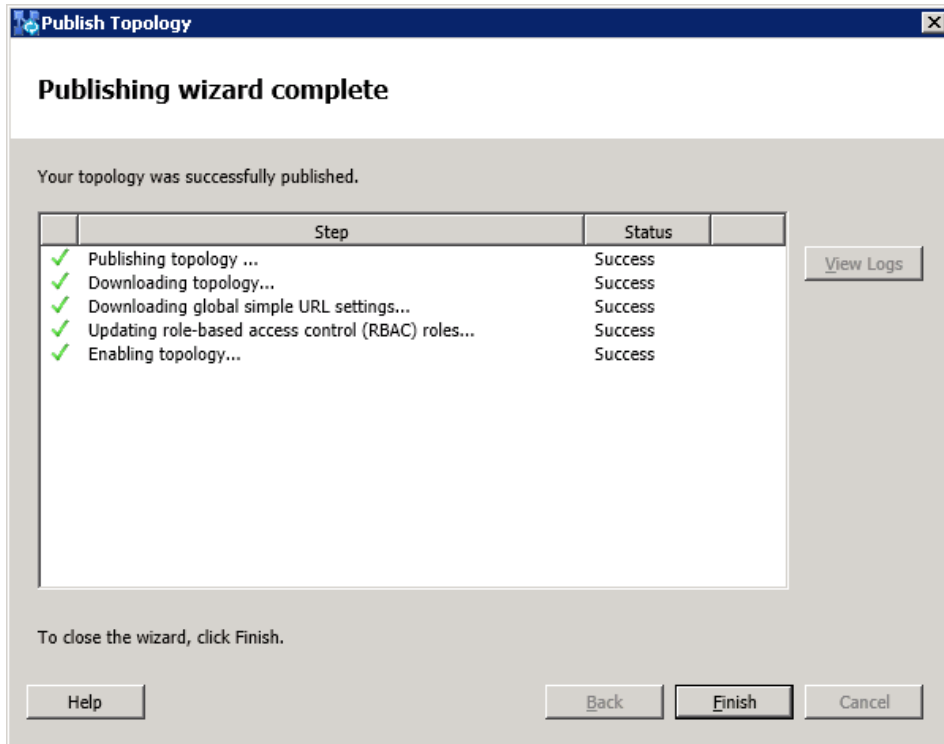
9. Click **Next**; the Topology Builder starts to publish your topology, as shown below:

Figure 3-12: Publishing in Progress



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

Figure 3-13: Publishing Wizard Complete



- 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:**

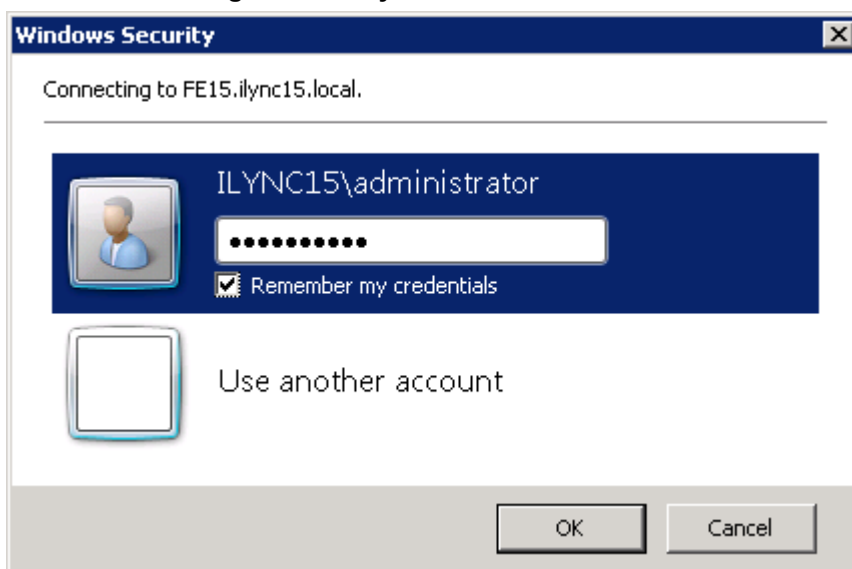
1. Start the Microsoft Lync Server 2013 Control Panel (**Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel**), as shown below:

Figure 3-14: Opening the Lync Server Control Panel



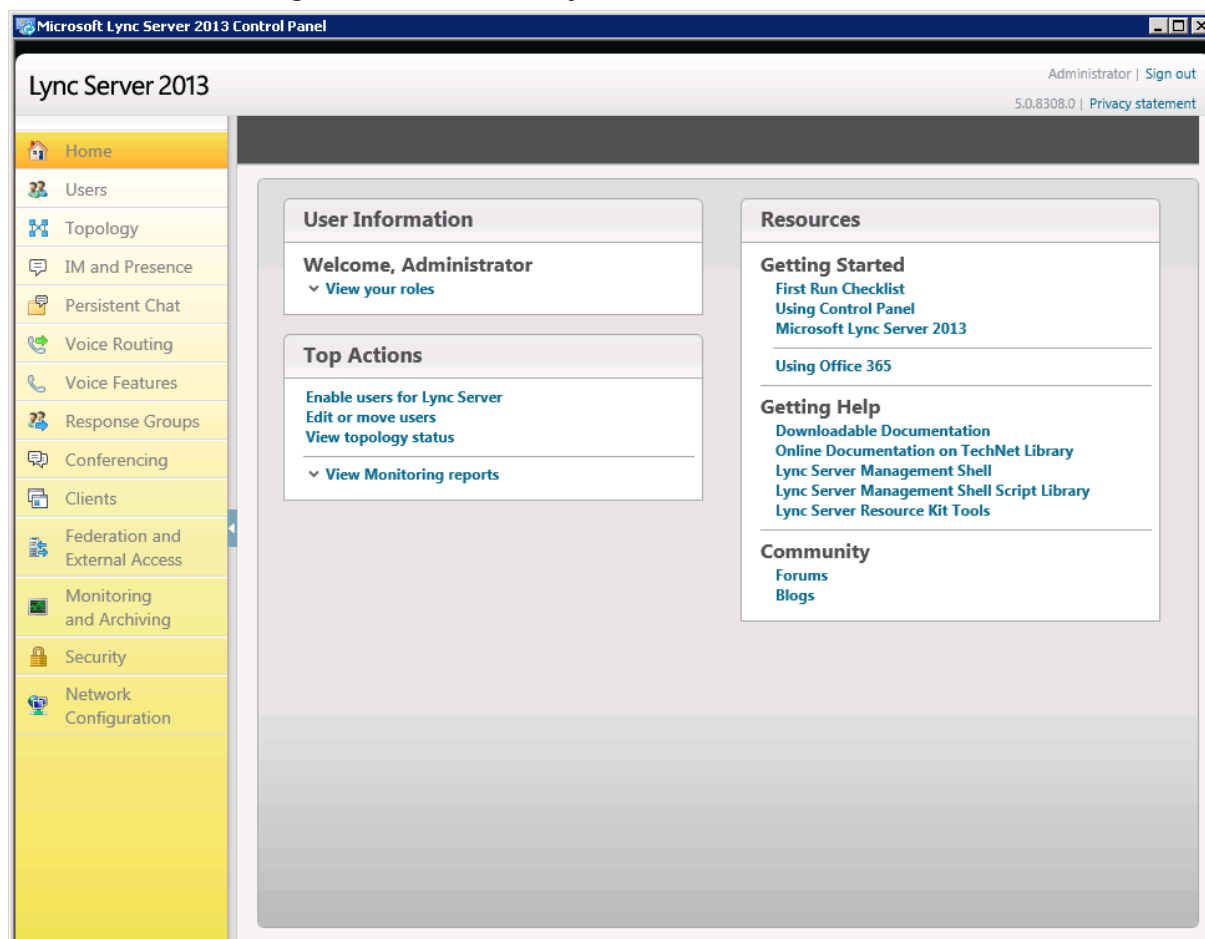
- You are prompted to enter your login credentials:

Figure 3-15: Lync Server Credentials



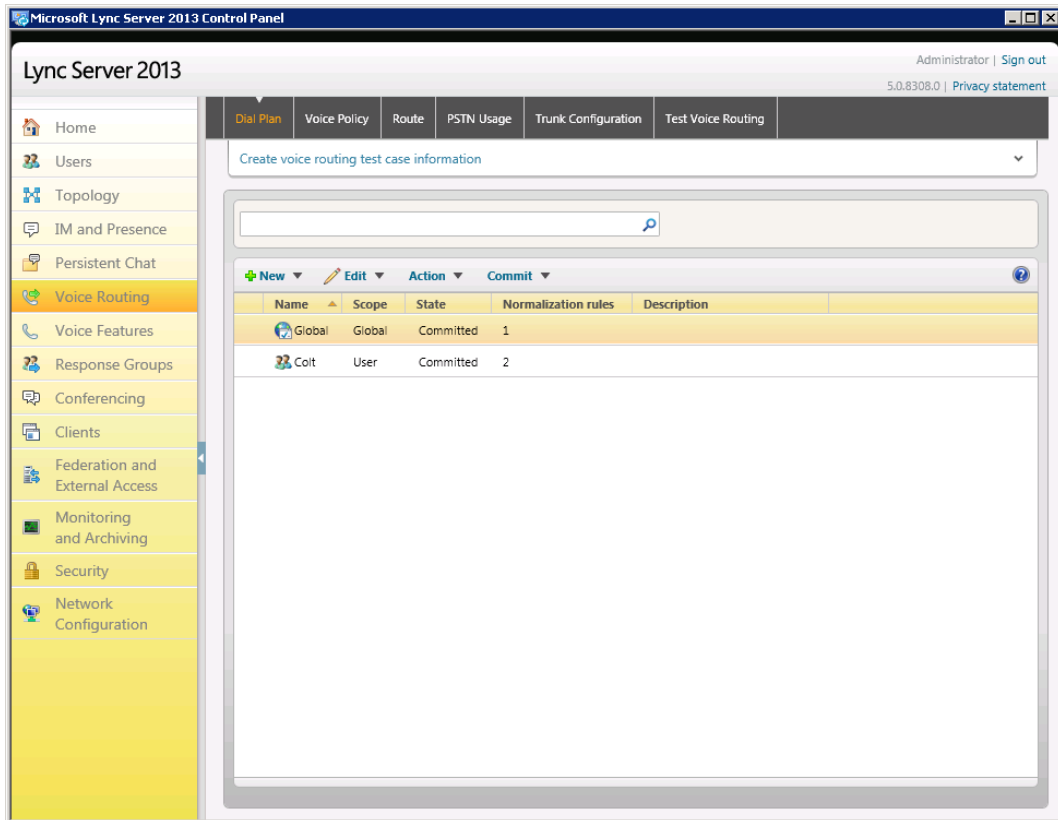
- Enter your domain username and password, and then click **OK**; the Microsoft Lync Server 2013 Control Panel is displayed:

Figure 3-16: Microsoft Lync Server 2013 Control Panel



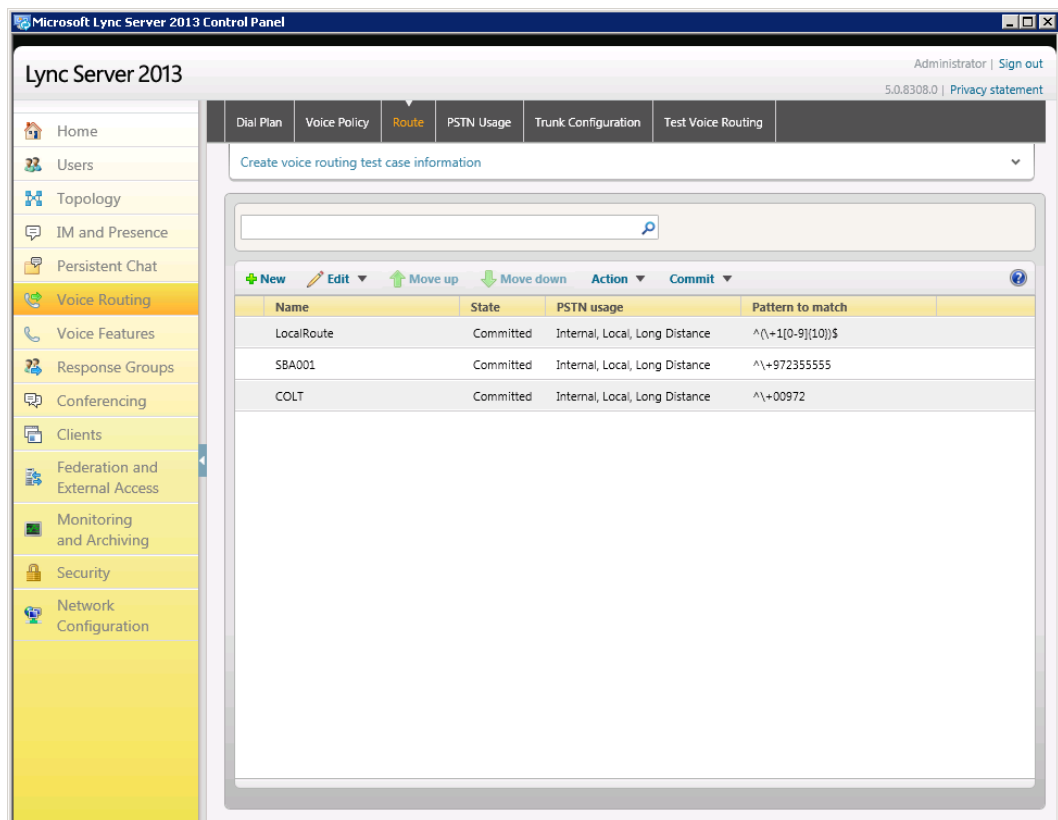
- In the left navigation pane, select **Voice Routing**.

Figure 3-17: Voice Routing Page



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

Figure 3-18: Route Tab



- Click **New**; the New Voice Route page appears:

Figure 3-19: Adding New Voice Route

The screenshot shows a 'New Voice Route' dialog box with the following fields and controls:

- Name:** SIP Trunk Route
- Description:** (empty)
- Build a Pattern to Match:**
 - Starting digits for numbers that you want to allow: *
 - Match this pattern: ^\$
- Buttons: OK, Cancel, Add, Exceptions, Remove, Edit, Reset.

- In the 'Name' field, enter a name for this route (e.g., **SIP Trunk Route**).
- 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**.

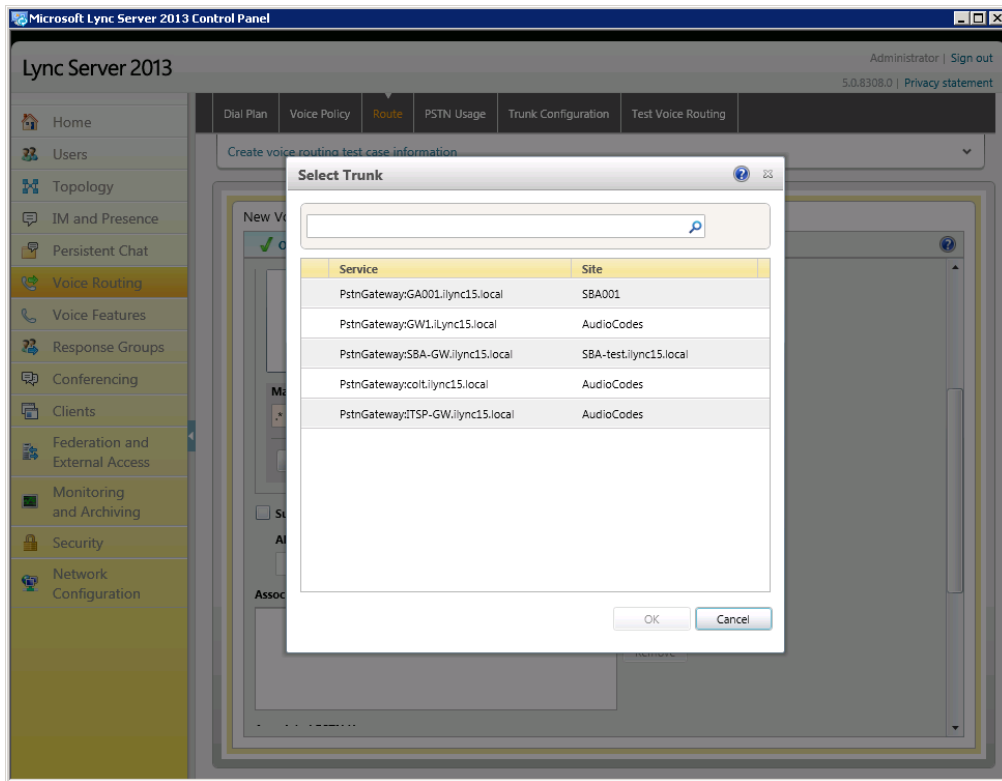
Figure 3-20: Adding New Trunk

The screenshot shows the 'Microsoft Lync Server 2013 Control Panel' with the 'New Voice Route' dialog box open. The 'Route' tab is selected in the top navigation bar. The dialog box contains the following fields and controls:

- Name:** (empty)
- Starting digits for numbers that you want to allow:** *
- Match this pattern:** ^\$
- Buttons:** OK, Cancel, Add, Exceptions, Remove, Edit, Reset.
- Supress caller ID:** (unchecked)
- Alternate caller ID:** (empty)
- Associated trunks:** (empty)

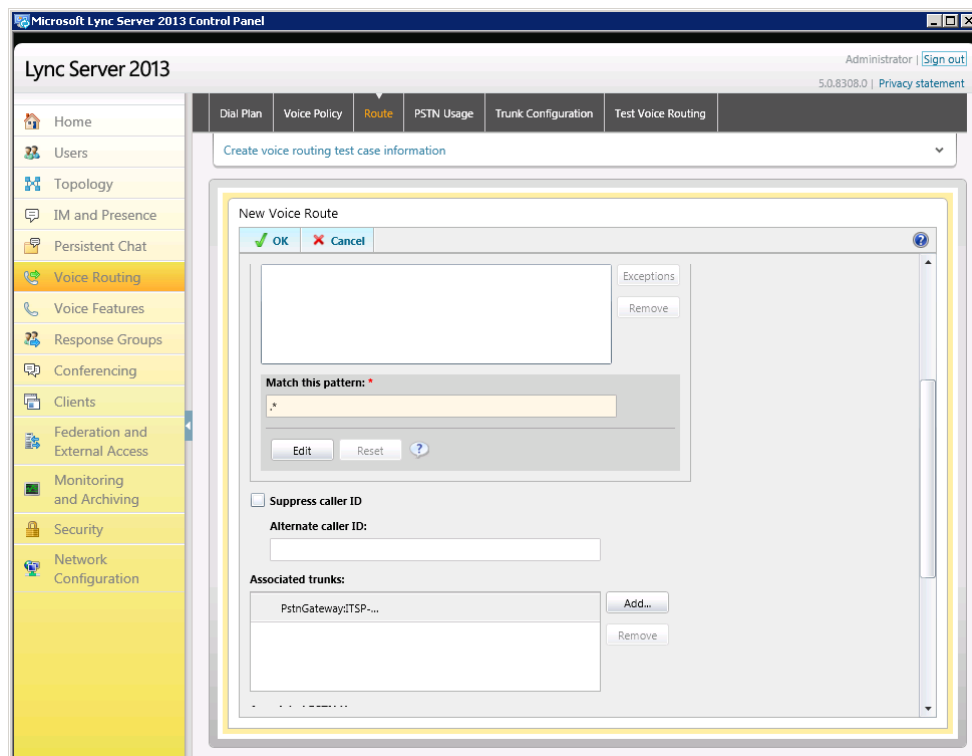
9. Associate the route with the E-SBC Trunk that you created:
 - a. Under the 'Associated Trunks' group, click **Add**; a list of all the deployed gateways is displayed:

Figure 3-21: List of Deployed Trunks



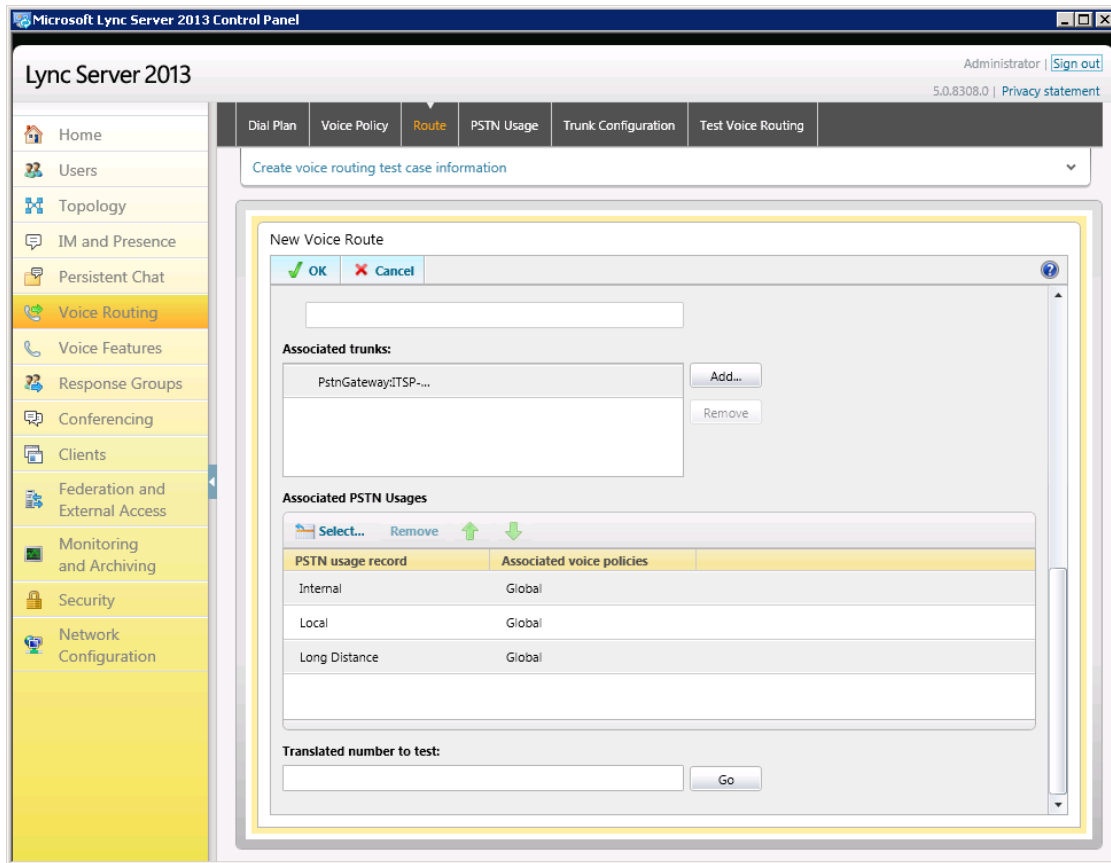
- b. Select the E-SBC Trunk you created, and then click **OK**; the trunk is added to the 'Associated Trunks' group list:

Figure 3-22: Selected E-SBC Trunk



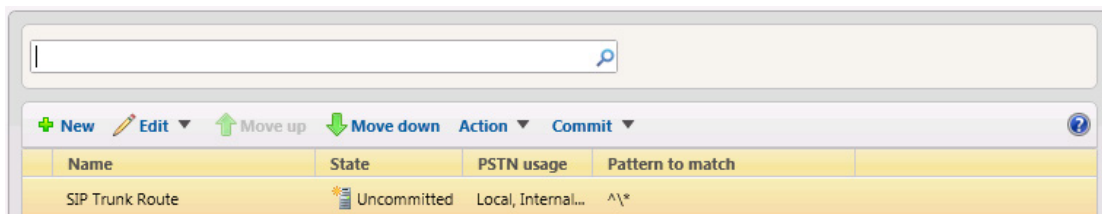
10. Associate a PSTN Usage to this route:
 - a. Under the 'Associated PSTN Usages' group, click **Select** and then add the associated PSTN Usage.

Figure 3-23: Associating PSTN Usage to Route



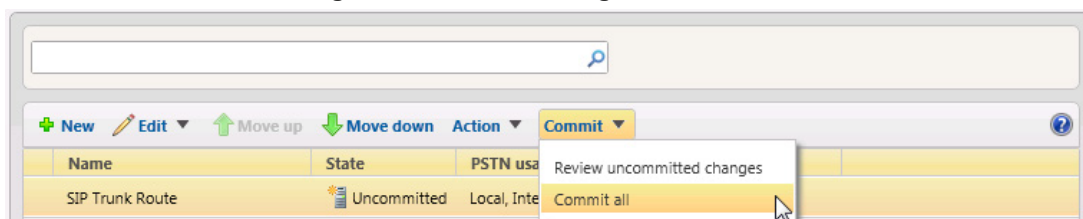
11. Click **OK** (located on the top of the New Voice Route page); the New Voice Route (Uncommitted) is displayed:

Figure 3-24: Confirmation of New Voice Route



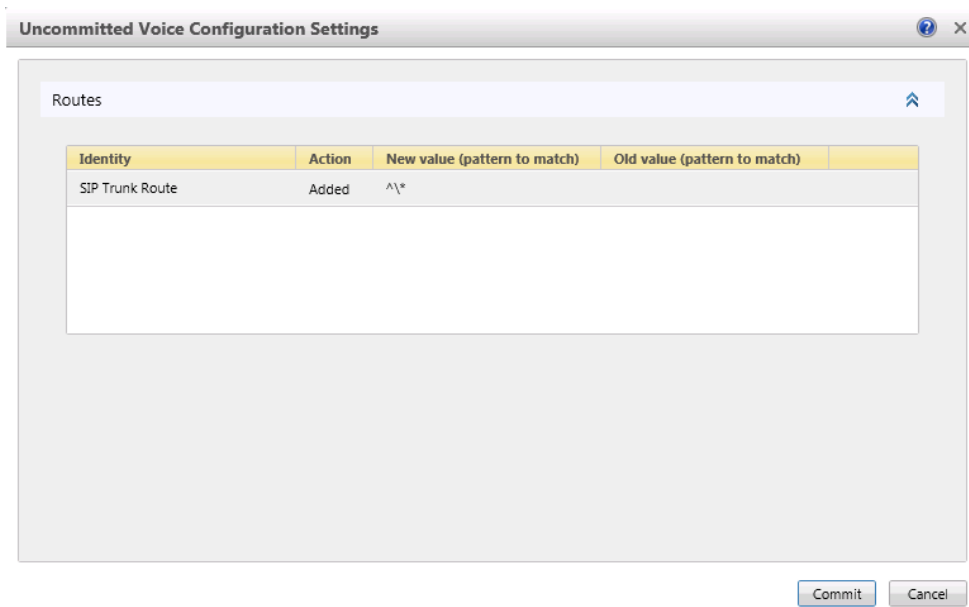
12. From the **Commit** drop-down list, choose **Commit all**, as shown below:

Figure 3-25: Committing Voice Routes



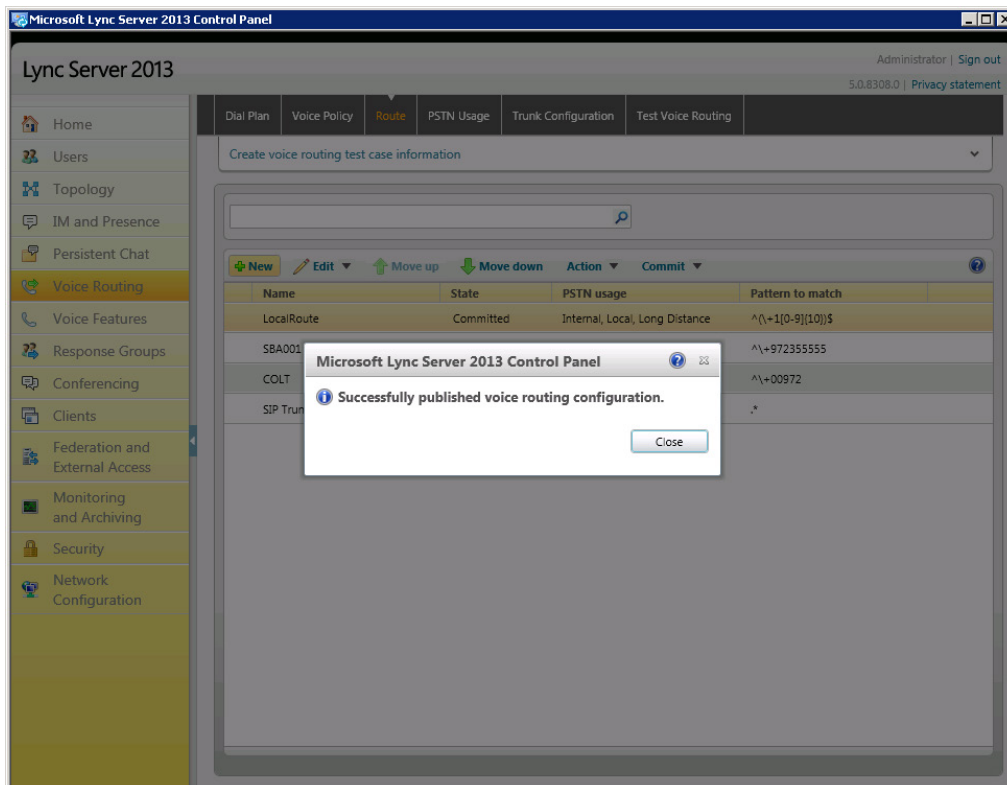
The Uncommitted Voice Configuration Settings page appears:

Figure 3-26: Uncommitted Voice Configuration Settings



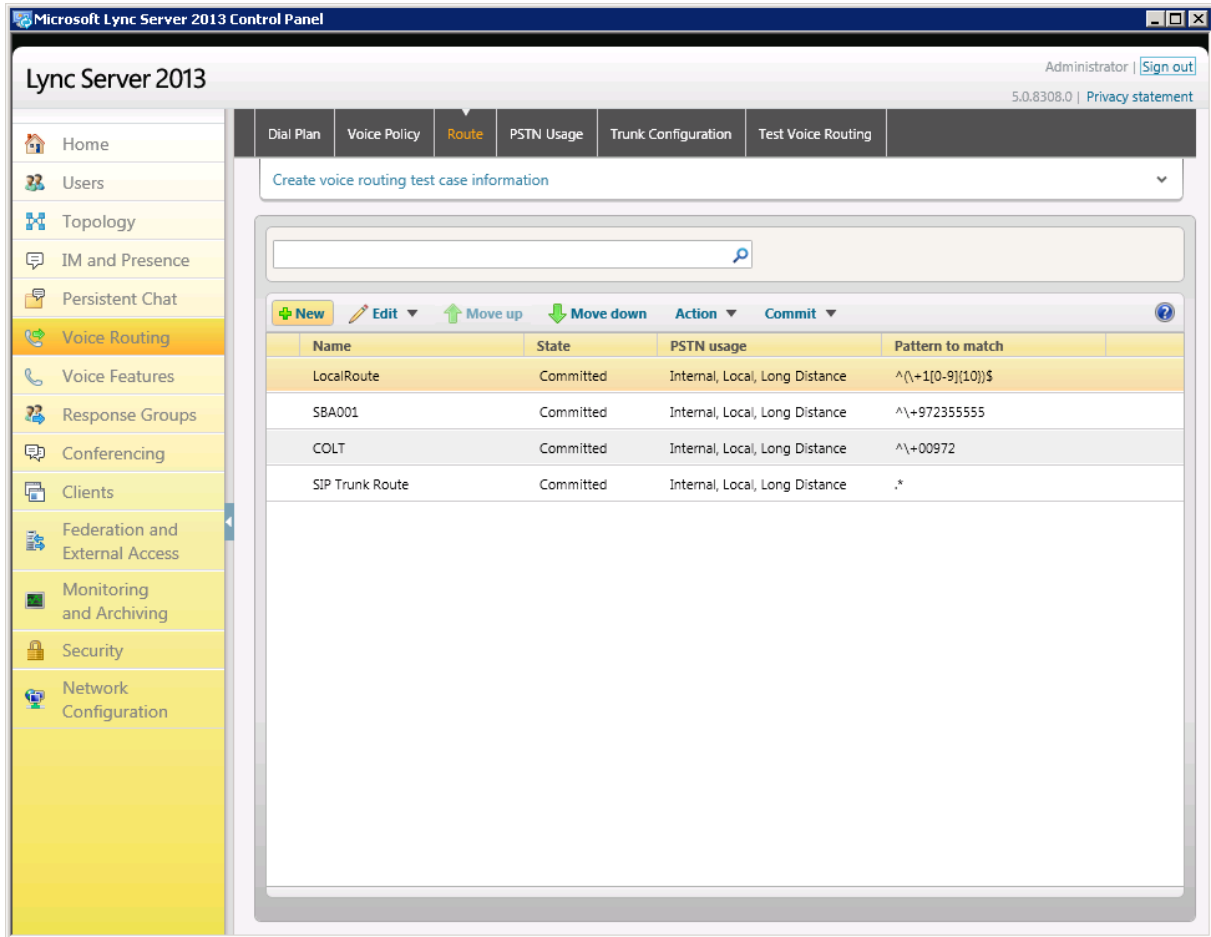
13. Click **Commit**; a message is displayed confirming a successful voice routing configuration, as shown below:

Figure 3-27: Confirmation of Successful Voice Routing Configuration



14. 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



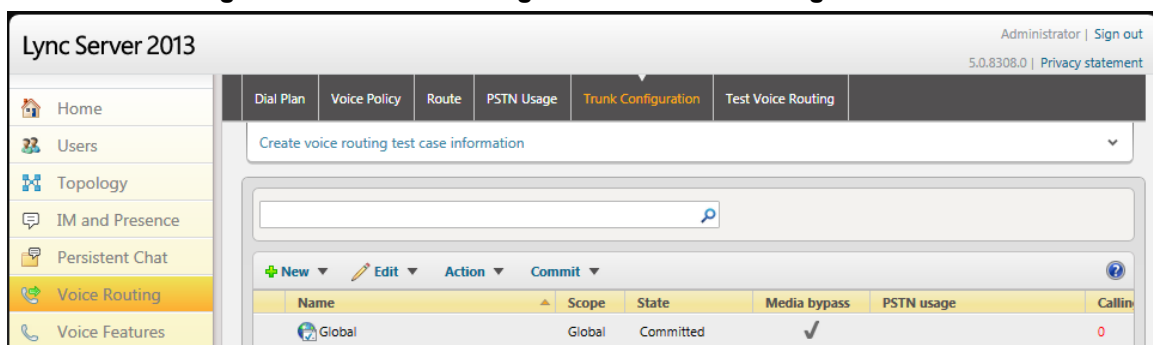
15. For ITSPs that implement a call identifier, continue with the following steps:



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 required by the 8BTwilio SIP Trunk in the P-Asserted-Identity header. The device adds this ID to the P-Asserted-Identity header in the sent INVITE message using the IP Profile (see Section 4.6 on page 45).

- 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:

Edit Trunk Configuration - Global

OK Cancel

Scope: Global

Name: *

Global

Description:

Maximum early dialogs supported:

20

Encryption support level:

Required

Refer support:

Enable sending refer to the gateway

Enable media bypass

Centralized media processing

Enable RTP latching

Enable forward call history

Enable forward P-Asserted-Identity data

Enable outbound routing failover timer

^ Associated PSTN Usages

Select... Remove ↑ ↓

- c. Select the **Enable forward call history** check box, and then click **OK**.
- d. Repeat Steps 11 through 13 to commit your settings.

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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 8BTwilio SIP Trunk. These configuration procedures are based on the interoperability test topology described in Section 2.4 on page 10, and includes the following main areas:

- E-SBC WAN interface - 8BTwilio SIP Trunking environment
- E-SBC LAN interface - Lync Server 2013 environment

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

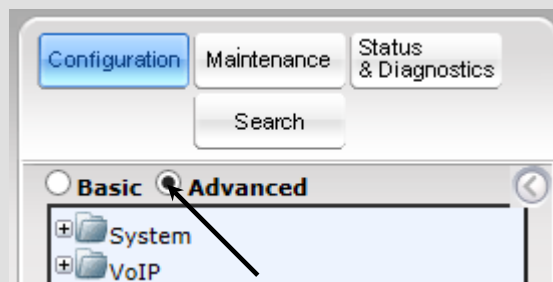
Notes:

- For implementing Microsoft Lync and 8BTwilio 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**

For more information about the Software License Key, contact your AudioCodes sales representative.

- The scope of this interoperability test and document does **not** cover all security aspects for connecting the SIP Trunk to the Microsoft Lync environment. Comprehensive security measures should be implemented per your organization's security policies. For security recommendations on AudioCodes' products, 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 Advanced-menu display mode. To do this, select the **Advanced** option, as shown below:



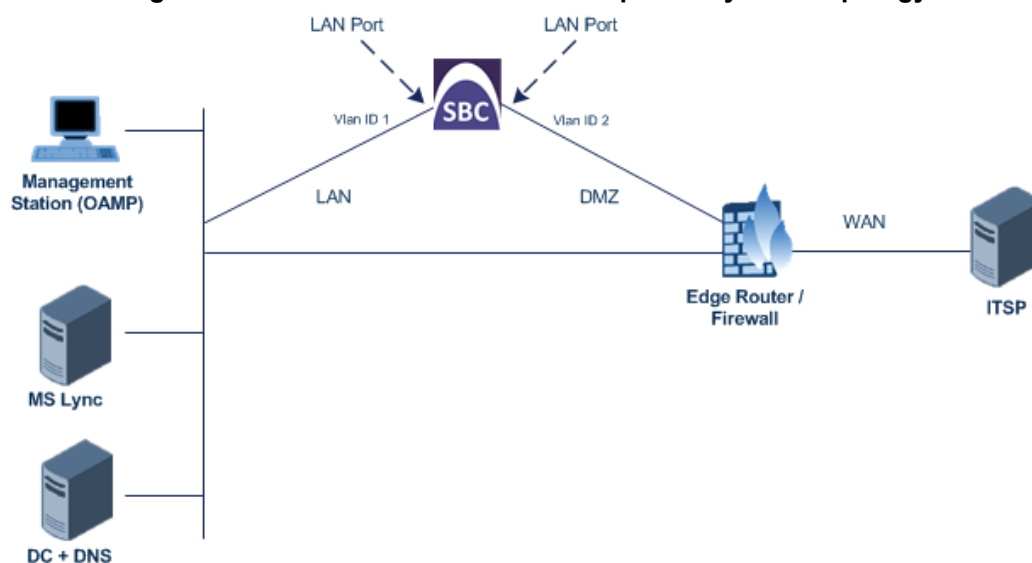
Note that when the E-SBC is reset, the Navigation tree reverts to Basic-menu display.

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
 - 8BTwilio SIP Trunk, 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 VLANs

This step describes how to define VLANs for each of the following interfaces:

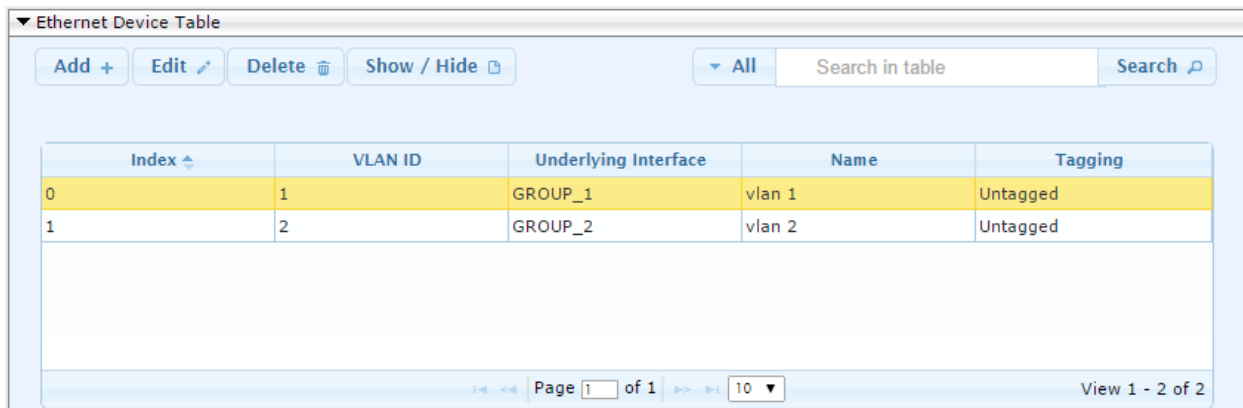
- LAN VoIP (assigned the name "Voice")
- WAN VoIP (assigned the name "WANSP")

➤ **To configure the VLANs:**

1. Open the Ethernet Device Table page (**Configuration** tab > **VoIP** menu > **Network** > **Ethernet Device Table**).
2. There will be one existing row for VLAN ID 1 and underlying interface GROUP_1.
3. Add another VLAN ID 2 for the WAN side as follows:

Parameter	Value
Index	1
VLAN ID	2
Underlying Interface	GROUP_2 (Ethernet port group)
Name	vlan 2
Tagging	Untagged

Figure 4-2: Configured VLAN IDs in Ethernet Device Table



4.1.2 Step 1b: Configure Network Interfaces

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

- LAN VoIP (assigned the name "Voice")
- WAN VoIP (assigned the name "WANSP")

➤ **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 as follows:

Parameter	Value
IP Address	10.15.17.77 (IP address of E-SBC)
Prefix Length	16 (subnet mask in bits for 255.255.0.0)
Default Gateway	10.15.0.1
Interface Name	Voice (arbitrary descriptive name)
Primary DNS Server IP Address	10.15.25.1
Underlying Device	vlan 1

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	195.189.192.158 (WAN IP address)
Prefix Length	25 (for 255.255.255.128)
Default Gateway	195.189.192.129 (router's IP address)
Interface Name	WANSP
Primary DNS Server IP Address	80.179.52.100
Secondary DNS Server IP Address	80.179.55.100
Underlying Device	vlan 2

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

The configured IP network interfaces are shown below:

Figure 4-3: Configured Network Interfaces in IP Interfaces Table

▼ Interface Table

Index	Interface Name	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Primary DNS	Secondary DNS	Underlying Device
0	Voice	OAMP + Medi	IPv4 Manual	10.15.17.77	16	10.15.0.1	10.15.25.1	0.0.0.0	vlan 1
1	WANSP	Media + Cont	IPv4 Manual	195.189.192.158	25	195.189.192.129	80.179.52.100	80.179.55.100	vlan 2

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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 4.15 on page 77).

4.3 Step 3: 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 > **VoIP Network** > **Media Realm Table**).
2. Add a Media Realm for the LAN interface. You can use the default Media Realm (Index 0), but modify it as shown below:

Parameter	Value
Index	0
Media Realm Name	MRLan (descriptive name)
IPv4 Interface Name	Voice
Port Range Start	6000 (represents lowest UDP port number used for media on LAN)
Number of Media Session Legs	100 (media sessions assigned with port range)

Figure 4-5: Configuring Media Realm for LAN

The screenshot shows a web-based configuration window titled "Edit Row" with a close button (X) in the top right corner. The window contains the following fields and values:

- Index: 0
- Name: MRLan
- IPv4 Interface Name: Voice (dropdown menu)
- Port Range Start: 6000
- Number Of Media Session Legs: 100
- Port Range End: 6990
- Default Media Realm: No (dropdown menu)
- QoS Profile: None (dropdown menu)
- BW Profile: None (dropdown menu)

At the bottom of the dialog, there are two buttons: "Save" and "Cancel".

3. Configure a Media Realm for WAN traffic:

Parameter	Value
Index	1
Media Realm Name	MRWan (arbitrary name)
IPv4 Interface Name	WANSP
Port Range Start	7000 (represents lowest UDP port number used for media on WAN)
Number of Media Session Legs	100 (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

Index	Name	IPv4 Interface Name	Port Range Start	Number Of Media Session Legs	Port Range End	Default Media Realm
0	MRLan	Voice	6000	100	6990	No
1	MRWan	WANSP	7000	100	7990	No

4.4 Step 4: 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 > **VoIP Network** > **SIP Interface Table**).
2. Add a SIP Interface for the LAN interface. You can use the default SIP Interface (Index 0), but modify it as shown below:

Parameter	Value
Index	0
Interface Name	Lync (see Note below)
Network Interface	Voice
Application Type	SBC
TLS Port	5067
TCP and UDP	0
Media Realm	MRLan

3. Configure a SIP Interface for the WAN:

Parameter	Value
Index	1
Interface Name	Twilio (see Note below)
Network Interface	WANSP
Application Type	SBC
UDP Port	5060
TCP and TLS	0
Media Realm	MRWan

The configured SIP Interfaces are shown in the figure below:

Figure 4-8: Configured SIP Interfaces in SIP Interface Table

Index	Name	SRD	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	Encapsulati Protocol	Media Realm
0	Lync	DefaultSR	Voice	SBC	0	0	5067	No encapsula	MRLan
1	Twilio	DefaultSR	WANSP	SBC	5060	0	0	No encapsula	MRWan



Note: Unlike in previous software releases where configuration entities (e.g., SIP Interface, Proxy Sets, and IP Groups) were associated with each other using table row indices, Version 7.0 uses the string **names** of the configuration entities. Therefore, it is recommended to configure each configuration entity with meaningful names for easy identification.

4.5 Step 5: 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
- 8BTwilio SIP Trunk

The Proxy Sets will be later applying to the VoIP network by assigning them to IP Groups.

➤ To configure Proxy Sets:

1. Open the Proxy Sets Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table**).
2. Add a Proxy Set for the Lync Server 2013. You can use the default Proxy Set (Index 0), but modify it as shown below:

Parameter	Value
Proxy Set ID	0
Proxy Name	Lync (see note in Section 4.4)
SBC IPv4 SIP Interface	Lync
Proxy Keep Alive	Using Options
Redundancy Mode	Homing
Load Balancing Method	Round Robin
Proxy Hot Swap	Enable
TLS Context Name	default

Figure 4-9: Configuring Proxy Set for Microsoft Lync Server 2013

Parameter	Value
Index	0
SRD	DefaultSRD
Name	Lync
Gateway IPv4 SIP Interface	None
SBC IPv4 SIP Interface	Lync
Proxy Keep-Alive	Using OPTIONS
Proxy Keep-Alive Time [sec]	60
Redundancy Mode	Homing
Proxy Load Balancing Method	Round Robin
DNS Resolve Method	
Proxy Hot Swap	Enable
Keep-Alive Failure Responses	
Classification Input	IP Address only
TLS Context Name	default

Buttons: Save, Cancel

3. Configure a Proxy Address Table for Proxy Set for Lync Server 2013:
 - a. Go to **Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table** > **Proxy Address Table**.

Parameter	Value
Index	0
Proxy Address	FE15.ilync15.local:5067 (Lync Server 2013 IP address / FQDN and destination port)
Transport Type	TLS

Figure 4-10: Configuring Proxy Address for Microsoft Lync Server 2013

Index	0
Proxy Address	FE15.ilync15.local:5067
Transport Type	TLS

Buttons: Add, Cancel

4. Configure a Proxy Set for the 8BTwilio SIP Trunk:

Parameter	Value
Proxy Set ID	1
Proxy Name	Twilio (see note in Section 4.4)
SBC IPv4 SIP Interface	Twilio
Proxy Keep Alive	Using Options
DNS Resolve Method	SRV

Figure 4-11: Configuring Proxy Set for 8BTwilio SIP Trunk

The screenshot shows a configuration window titled "Edit Row" with a close button (X) in the top right corner. The window contains the following parameters and values:

- Index: 1
- SRD: DefaultSRD
- Name: Twilio
- Gateway IPv4 SIP Interface: None
- SBC IPv4 SIP Interface: Twilio
- Proxy Keep-Alive: Using OPTIONS
- Proxy Keep-Alive Time [sec]: 60
- Redundancy Mode: (empty dropdown)
- Proxy Load Balancing Method: Disable
- DNS Resolve Method: SRV
- Proxy Hot Swap: Disable
- Keep-Alive Failure Responses: (empty text field)
- Classification Input: IP Address only
- TLS Context Name: None

At the bottom of the window, there are "Save" and "Cancel" buttons.

- a. Configure a Proxy Address Table for Twilio Proxy Set:
- b. Go to **Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table** > **Proxy Address Table**.

Parameter	Value
Index	0
Proxy Address	ilync15.pstn.twilio.com:5060 (FQDN and destination port)
Transport Type	UDP

Figure 4-12: Configuring Proxy Address for

The 'Edit Row' dialog box contains the following fields:

- Index: 0
- Proxy Address: ilync15.pstn.twilio.com
- Transport Type: UDP

Buttons: Save, Cancel

The configured Proxy Sets are shown in the figure below:

Figure 4-13: Configured Proxy Sets in Proxy Sets Table

Proxy Sets Table

▼ All

Index ↕	Name	SRD	Gateway IPv4 SIP Interface	SBC IPv4 SIP Interface	Proxy Keep-Alive Time [sec]	Redundancy Mode	Proxy Hot Swap
0	Lync	DefaultSRD (# None)		Lync	60	Homing	Enable
1	Twilio	DefaultSRD (# None)		Twilio	60		Disable

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4.6 Step 6: 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 secure mode using SRTP and TLS
- 8BTwilio SIP trunk - to operate in non-secure mode using RTP and UDP

➤ To configure IP Profile for the Lync Server 2013:

1. Open the IP Profile Settings page (**Configuration** tab > **VoIP** > **Coders and Profiles** > **IP Profile Settings**).
2. Click **Add**.
3. Click the **Common** tab, and then configure the parameters as follows:

Parameter	Value
Index	1
Name	Lync (see note in Section 4.4)
Symmetric MKI	Enable
MKI Size	1
Reset SRTP State Upon Re-key	Enable
Generate SRTP keys mode:	Always

Figure 4-14: Configuring IP Profile for Lync Server 2013 – Common Tab

The screenshot shows the 'Add Row' dialog box with the following parameters and values:

- Index: 1
- Tab: Common
- Name: Lync
- Dynamic Jitter Buffer Minimum Delay [msec]: 10
- Dynamic Jitter Buffer Optimization Factor: 10
- Jitter Buffer Max Delay [msec]: 300
- RTP IP DiffServ: 46
- Signaling DiffServ: 40
- Silence Suppression: Disable
- RTP Redundancy Depth: 0
- Echo Canceler: Line
- Disconnect on Broken Connection: Enable
- Input Gain (-32 to 31 dB): 0
- Voice Volume (-32 to 31 dB): 0
- Media IP Version: Only IPv4

Buttons: Add, Cancel

4. Click the **SBC Signaling** tab, and then configure the parameters as follows:

Parameter	Value
PRACK Mode	Optional (required, as Twilio SIP Trunk does not support PRACK)
Remote Update Support	Supported Only After Connect
Remote re-INVITE Support	Supported Only With SDP
Remote Delayed Offer Support	Not Supported
Remote REFER Mode	Handle Locally (required, as Lync Server 2013 does not support receipt of SIP REFER)
Remote 3xx Mode	Handle Locally (required, as Lync Server 2013 does not support receipt of SIP 3xx responses)
Remote Early Media RTP Detection Behavior	By Media (required, as Lync Server 2013 does not send RTP immediately to remote side when it sends a SIP 18x response)

Figure 4-15: Configuring IP Profile for Lync Server 2013 – SBC Signaling Tab

Index: 1

Common | GW | **SBC Signaling** | SBC Media

PRACK Mode: Optional

P-Asserted-Identity Header Mode: As Is

Diversion Header Mode: As Is

History-Info Header Mode: As Is

Session Expires Mode: Transparent

Remote Update Support: Supported Only Aft

Remote re-INVITE: Supported only with

Remote Delayed Offer Support: Not Supported

User Registration Time: 0

NAT UDP Registration Time: -1

NAT TCP Registration Time: -1

Remote REFER Mode: Handle Locally

Remote Replaces Mode: Standard

Save Cancel

- Click the **SBC Media** tab, and then configure the parameters as follows:

Parameter	Value
Allowed Audio Coders	Coders Group 0 (in order to ensure that voice sent to the 8BTwilio SIP Trunk uses the G.711U-law coder only)
SBC Media Security Mode	SRTP
Enforce MKI Size	Enforce
RTCP Mode	Generate Always (required, as Twilio SIP Trunk does not send RTCP packets in active and in hold calls, and in these cases, Microsoft Lync 2013 will terminate the call with network problems as the cause)

Figure 4-16: Configuring IP Profile for Lync Server 2013 – SBC Media Tab

The screenshot shows the 'Edit Row' configuration window for the SBC Media tab. The 'Index' is set to 1. The 'SBC Media' tab is selected, showing the following parameters and their values:

- Transcoding Mode: Only If Required
- Extension Coders: None
- Allowed Audio Coders: Coders Group 0
- Allowed Coders Mode: Restriction
- Allowed Video Coders: None
- Allowed Media Types: (empty field)
- SBC Media Security Mode: SRTP
- Media Security Method: SDES
- Enforce MKI Size: Enforce
- SDP Remove Crypto Lifetime: No
- RFC 2833 Mode: As Is
- Alternative DTMF Method: As Is
- RFC 2833 DTMF Payload Type: 0
- Fax Coders: None

Buttons for 'Save' and 'Cancel' are visible at the bottom right of the window.

- **To configure an IP Profile for the 8BTwilio SIP Trunk:**
- 1. Click **Add**.
- 2. Click the **Common** tab, and then configure the parameters as follows:

Parameter	Value
Index	2
Profile Name	Twilio (see note in Section 4.4)

Figure 4-17: Configuring IP Profile for 8BTwilio SIP Trunk – Common Tab

Edit Row [Close]

Index:

Common
 GW
 SBC Signaling
 SBC Media

Name:

Dynamic Jitter Buffer Minimum Delay [msec]:

Dynamic Jitter Buffer Optimization Factor:

Jitter Buffer Max Delay [msec]:

RTP IP DiffServ:

Signaling DiffServ:

Silence Suppression:

RTP Redundancy Depth:

Echo Canceler:

Broken Connection Mode:

Input Gain (-32 to 31 dB):

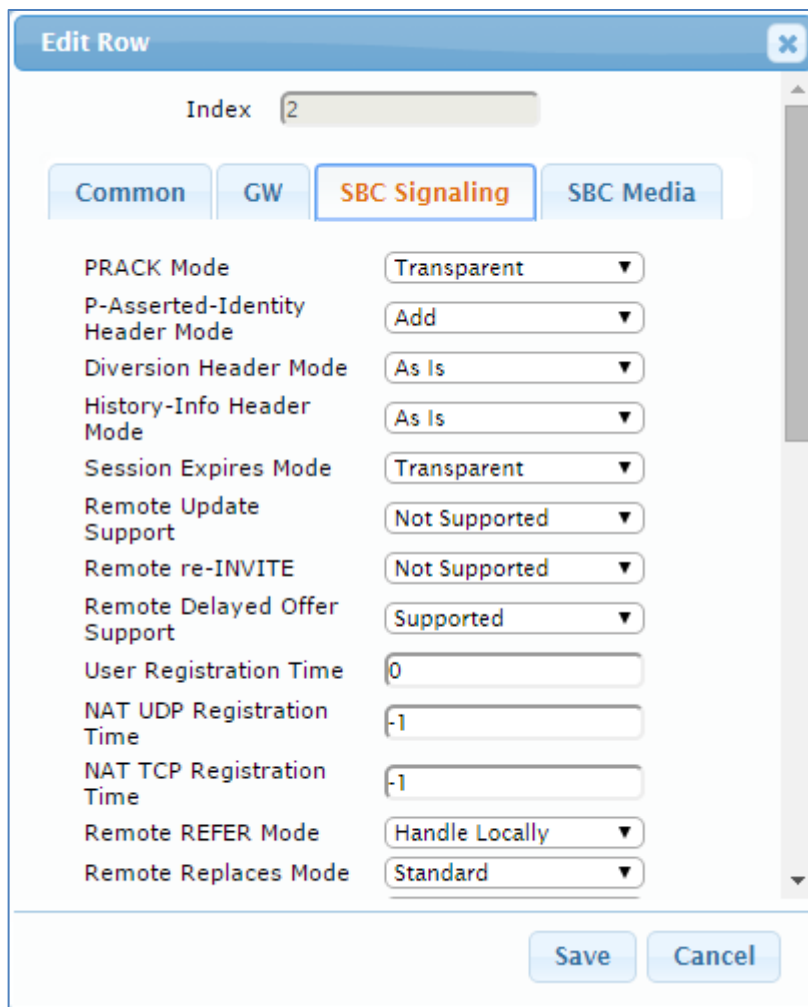
Voice Volume (-32 to 31 dB):

Media IP Version:

- Click the **SBC Signaling** tab, and then configure the parameters as follows:

Parameter	Value
P-Asserted-Identity Header Mode	Add (required for anonymous calls)
Remote Update Support	Not Supported
Remote re-INVITE Support	Not Supported
Remote REFER Behavior	Handle Locally (E-SBC handles / terminates incoming REFER requests instead of forwarding them to SIP Trunk)
Play RBT To Transferee	Yes
Remote Hold Format	Not Supported

Figure 4-18: Configuring IP Profile for 8BTwilio SIP Trunk – SBC Signaling Tab



4. Click the **SBC Media** tab, and then configure the parameters as follows:

Parameter	Value
Transcoding Mode	Force (required as there is a 120 seconds broken connection timeout defined on the Twilio SIP Trunk. Consequently, the SIP trunk always expects to receive RTP packets. When a call is muted or placed on Hold, no packets are sent from the Microsoft Lync Server 2013 side).

Figure 4-19: Configuring IP Profile for 8BTwilio SIP Trunk – SBC Media Tab

The screenshot shows the 'Edit Row' dialog box for configuring SBC Media parameters. The 'Index' is set to 2. The 'SBC Media' tab is selected. The parameters and their values are as follows:

Parameter	Value
Transcoding Mode	Force
Extension Coders	None
Allowed Audio Coders	None
Allowed Coders Mode	Restriction
Allowed Video Coders	None
Allowed Media Types	
SBC Media Security Mode	RTP
Media Security Method	SDES
Enforce MKI Size	Don't enforce
SDP Remove Crypto Lifetime	No
RFC 2833 Mode	As Is
Alternative DTMF Method	As Is
RFC 2833 DTMF Payload Type	0
Fax Coders	None

Buttons: Save, Cancel

4.7 Step 7: 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. 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
- 8BTwilio SIP Trunk located on WAN

➤ To configure IP Groups:

1. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).
2. Add an IP Group for the Lync Server 2013. You can use the default IP Group (Index 0), but modify it as shown below:

Parameter	Value
Index	0
Name	Lync (see note in Section 4.4)
Type	Server
Proxy Set	Lync
IP Profile	Lync
Media Realm	MRLan
SIP Group Name	ilync15.pstn.twilio.com (according to ITSP requirement)

3. Configure an IP Group for the 8BTwilio SIP Trunk:

Parameter	Value
Index	1
Name	Twilio (see note in Section 4.4)
Type	Server
Proxy Set	Twilio
IP Profile	Twilio
Media Realm	MRWan
SIP Group Name	ilync15.pstn.twilio.com (according to ITSP requirement)

The configured IP Groups are shown in the figure below:

Figure 4-20: Configured IP Groups in IP Group Table

Index	Name	SRD	Type	SBC Operator Mode	Proxy Set	IP Profile	Media Realm	SIP Group Name	Classify By Proxy Set	Inbound Message Manipulation Set	Outbound Message Manipulation Set
0	Lync	Default	Server	Not Config	Lync	Lync	MRLan	ilync15.pst	Enable	-1	-1
1	Twilio	Default	Server	Not Config	Twilio	Twilio	MRWan	ilync15.pst	Enable	-1	-1

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4.8 Step 8: Configure Allowed Coder

This step describes how to configure an Allowed Coders Group to ensure that voice sent to the 8BTwilio SIP Trunk uses the G.711U-law coder only. Note that this Allowed Coders Group ID was assigned to the IP Profile belonging to the 8BTwilio SIP Trunk (see Section 4.6 on page 45).

➤ **To set a preferred coder for the SIP Trunk:**

1. Open the Allowed Coders Group page (**Configuration** tab > **VoIP** menu > **SBC** > **Allowed Audio Coders Group**).
2. Configure an Allowed Coder as follows:

Parameter	Value
Allowed Audio Coders Group ID	0
Coder Name	G.711

Figure 4-21: Configuring Allowed Coders Group for SIP Trunk

The screenshot shows a web interface for configuring an Allowed Audio Coders Group. At the top, there is a dropdown menu for 'Allowed Audio Coders Group ID' with the value '0' selected. Below this, there is a section for 'Coder Name' with a dropdown menu showing 'G.711U-law' selected. There is also an empty dropdown menu below it.

3. Click **Submit**.

4.9 Step 9: SIP TLS Connection Configuration

This section describes how to configure the E-SBC for using a TLS connection with the Lync Server 2013 Mediation Server. This is essential for a secure SIP TLS connection.

4.9.1 Step 9a: 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 necessary for validating certificates of remote parties.

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

1. Open the Application Settings page (**Configuration** tab > **System** > **Application Settings**).
2. In the 'NTP Server Address' field, enter the IP address of the NTP server (e.g., **10.15.25.1**).

Figure 4-22: Configuring NTP Server Address

NTP Settings	
NTP Server Address (IP or FQDN)	<input type="text" value="10.15.25.1"/>
NTP Updated Interval	Hours: <input type="text" value="24"/> Minutes: <input type="text" value="0"/>
NTP Secondary Server Address (IP or FQDN)	<input type="text"/>
NTP Authentication Key Identifier	<input type="text" value="0"/>
NTP Authentication Secret Key	<input type="text"/>

3. Click **Submit**.

4.9.2 Step 9b: Configure a Certificate

This step describes how to exchange a certificate with Microsoft Certificate Authority (CA). The certificate is used by the E-SBC to authenticate the connection with Lync Server 2013.

The procedure involves the following main steps:

- a. Generating a Certificate Signing Request (CSR).
- b. Requesting Device Certificate from CA.
- c. Obtaining Trusted Root Certificate from CA.
- d. Deploying Device and Trusted Root Certificates on E-SBC.

➤ **To configure a certificate:**


1. Open the TLS Contexts page (**Configuration** tab > **System** menu > **TLS Contexts**).
2. In the TLS Contexts table, select the required TLS Context index row (usually default index 0 will be used), and then click the **TLS Context Certificates**  button, located at the bottom of the TLS Contexts page; the Context Certificates page appears.
3. Under the **Certificate Signing Request** group, do the following:
 - a. In the 'Subject Name [CN]' field, enter the E-SBC FQDN name (e.g., **ITSP-2.ilync15.local**).
 - b. Fill in the rest of the request fields according to your security provider's instructions.
4. Click the **Create CSR** button; a textual certificate signing request is displayed in the area below the button:

Figure 4-23: Certificate Signing Request – Creating CSR

▼ Certificate Signing Request

Subject Name [CN]	ITSP-GW.ilync15.local
Organizational Unit [OU] (optional)	
Company name [O] (optional)	
Locality or city name [L] (optional)	
State [ST] (optional)	
Country code [C] (optional)	

Create CSR

After creating the CSR, copy the text below (including the BEGIN/END lines) and send it to your Certification Authority for signing.

```

-----BEGIN CERTIFICATE REQUEST-----
MIIBXzCBYQIBADAgMR4wHAYDVQQDExVJVFNQLUdXLm1seW5jMTUubG9jYWwz8w
DQYJKoZIhvcNAQEBBQADgY0AMIGJAoGBAKkobC9QmE0XA0vaTrki0on0LVrwNsC1
3TMgncMVxdp9/BCXyygT2W1vz0NGUsypa7w2DKKkxr8x2A9sGLXwy0ZCyB49U1pDF
DJV8IldUfT8qL9d9V64e3Z004I1hweZSn4hHdAfGy0S6e91JhFw/USUD6/bNygQz
52203jtjXKmdAgMBAAGgADANBgkqhkiG9w0BAQQFAAQBqBLqe880JGrmEzPu5Q1
pRGiOuEQ4Pr6PL+JKghii6UpLmHEwixTedayzNh7b2yQgFYxiVWmX2JwrvXaCp5Y
8z8hOCZxV/E4MrR2s8bYb6bqxeteAXs+VwxgKObb4pSFfGLc82+dZUcODAB0wZFv
nxSEcPACRnZittF/GgW+A4AoMQ==
-----END CERTIFICATE REQUEST-----
    
```

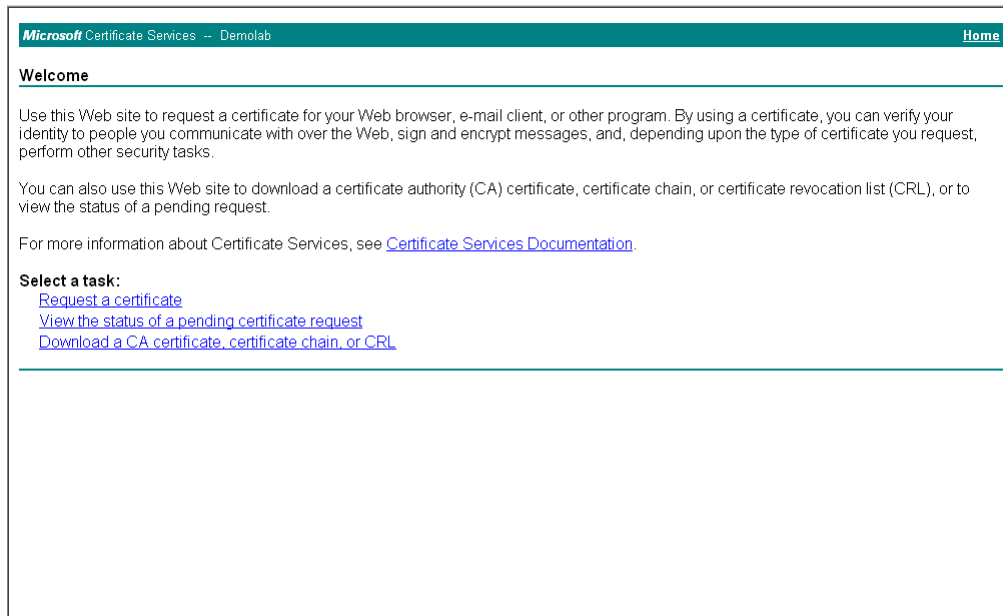


Note: The value entered in this field must be identical to the gateway name configured in the Topology Builder for Lync Server 2013 (see Section 3.1 on page 13).

5. Copy the CSR from the line "**-----BEGIN CERTIFICATE**" to "**END CERTIFICATE REQUEST-----**" to a text file (such as Notepad), and then save it to a folder on your computer with the file name, *certreq.txt*.

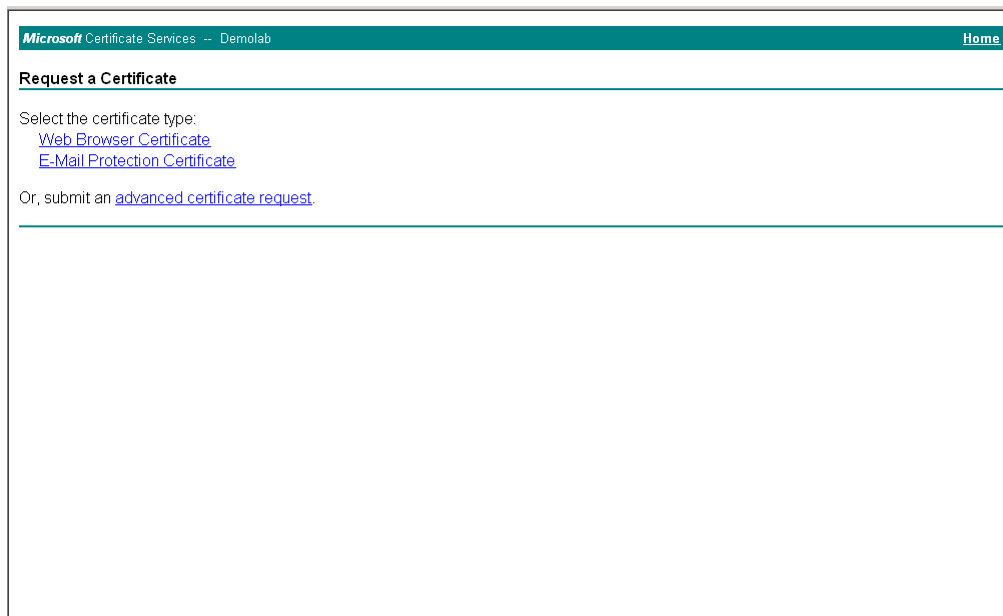
6. Open a Web browser and navigate to the Microsoft Certificates Services Web site at <http://<certificate server>/CertSrv>.

Figure 4-24: Microsoft Certificate Services Web Page



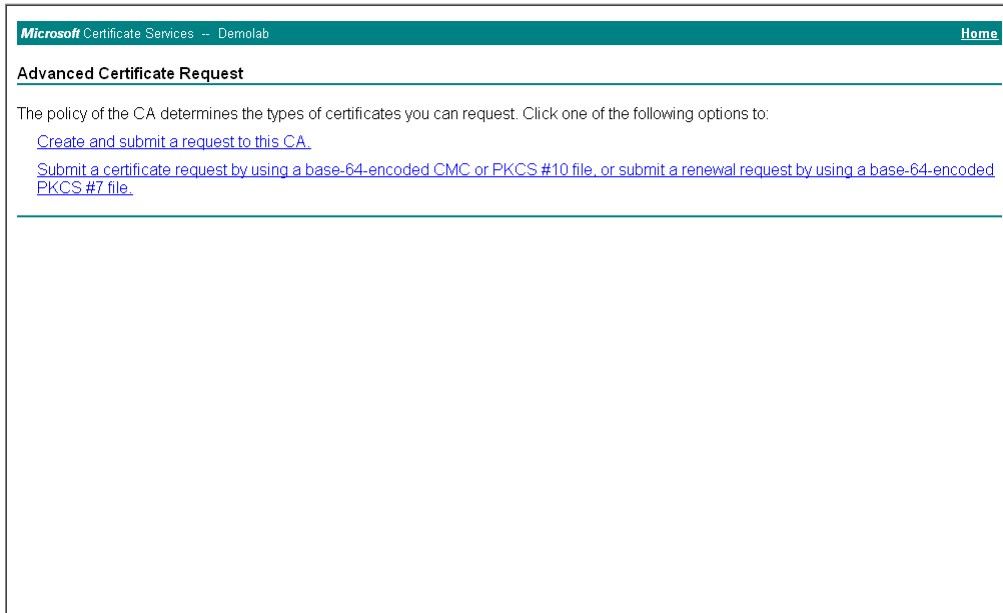
7. Click **Request a certificate**.

Figure 4-25: Request a Certificate Page



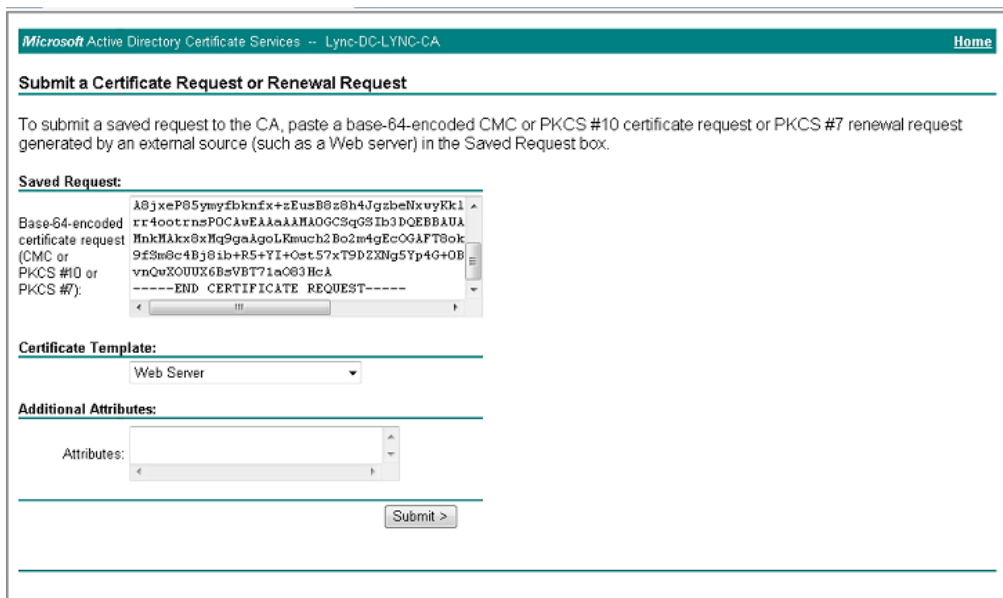
8. Click **advanced certificate request**, and then click **Next**.

Figure 4-26: Advanced Certificate Request Page



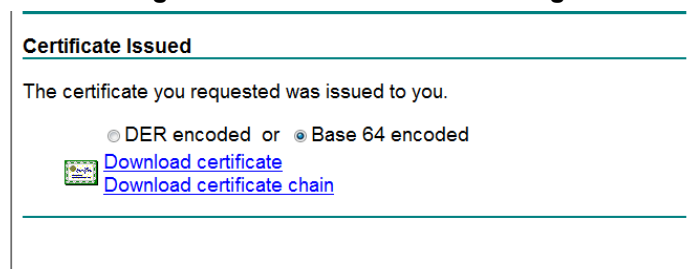
9. Click **Submit a certificate request ...**, and then click **Next**.

Figure 4-27: Submit a Certificate Request or Renewal Request Page



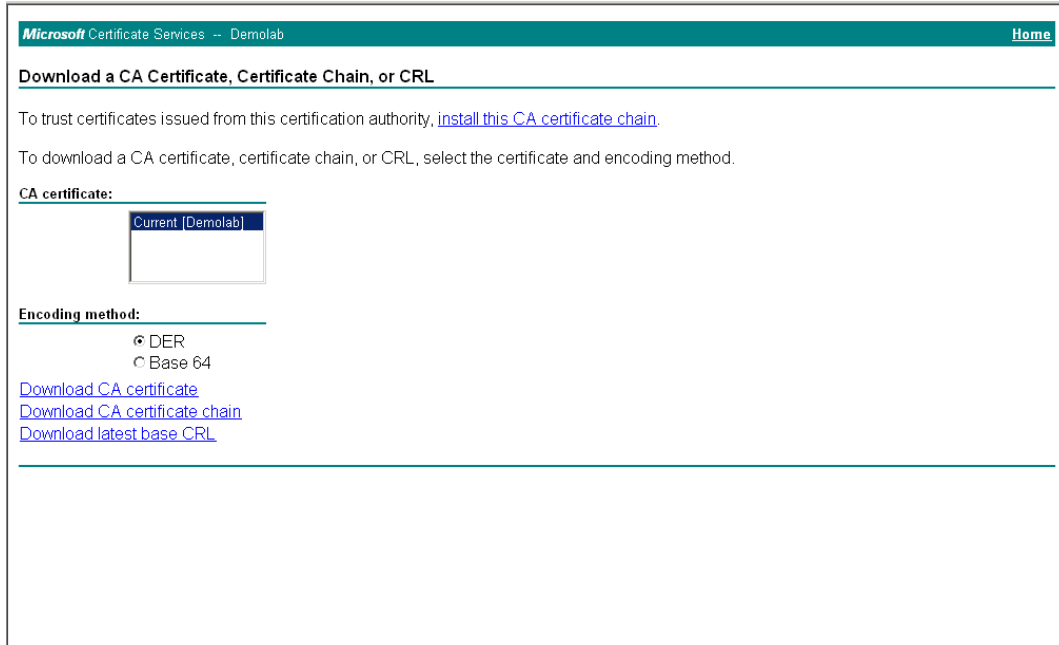
10. Open the *certreq.txt* file that you created and saved in Step 5, and then copy its contents to the 'Saved Request' field.
11. From the 'Certificate Template' drop-down list, select **Web Server**.
12. Click **Submit**.

Figure 4-28: Certificate Issued Page



13. Select the **Base 64 encoded** option for encoding, and then click **Download certificate**.
14. Save the file as *gateway.cer* to a folder on your computer.
15. Click the **Home** button or navigate to the certificate server at <http://<Certificate Server>/CertSrv>.
16. Click **Download a CA certificate, certificate chain, or CRL**.

Figure 4-29: Download a CA Certificate, Certificate Chain, or CRL Page



Microsoft Certificate Services -- Demolab Home

Download a CA Certificate, Certificate Chain, or CRL

To trust certificates issued from this certification authority, [install this CA certificate chain](#).

To download a CA certificate, certificate chain, or CRL, select the certificate and encoding method.

CA certificate:

Current [Demolab]

Encoding method:

DER
 Base 64

[Download CA certificate](#)
[Download CA certificate chain](#)
[Download latest base CRL](#)

17. Under the 'Encoding method' group, select the **Base 64** option for encoding.
18. Click **Download CA certificate**.
19. Save the file as *certroot.cer* to a folder on your computer.


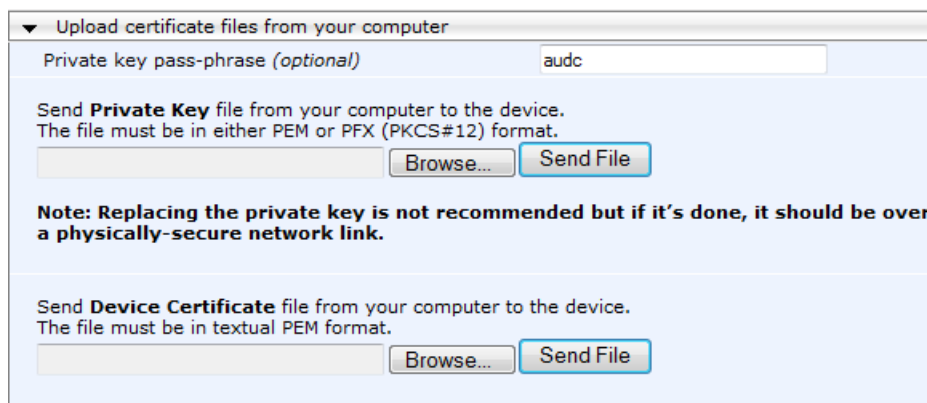
20. In the E-SBC's Web interface, return to the **TLS Contexts** page and do the following:
 - a. In the TLS Contexts table, select the required TLS Context index row (usually default index 0 will be used), and then click the **TLS Context Certificates**  button, located at the bottom of the TLS Contexts page; the Context Certificates page appears.
 - b. Scroll down to the **Upload certificates files from your computer** group, click the **Browse** button corresponding to the 'Send Device Certificate...' field, navigate to the *gateway.cer* certificate file that you saved on your computer in Step 14, and then click **Send File** to upload the certificate to the E-SBC.

Figure 4-30: Upload Device Certificate Files from your Computer Group




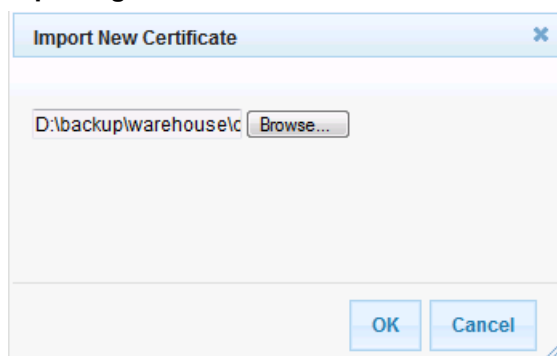
- c. In the E-SBC's Web interface, return to the **TLS Contexts** page.
- d. In the TLS Contexts table, select the required TLS Context index row, and then click the **TLS Context Trusted-Roots Certificates**  button, located at the bottom of the TLS Contexts page; the Trusted Certificates page appears.
- e. Click the **Import** button, and then select the certificate file to load.

Figure 4-31: Importing Root Certificate into Trusted Certificates Store



21. Click **OK**; the certificate is loaded to the device and listed in the Trusted Certificates store.
22. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.15 on page 77).

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 4.6 on page 45).

➤ **To configure media security:**

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

Parameter	Value
Media Security	Enable

Figure 4-32: Configuring SRTP

General Media Security Settings		
Media Security	Enable	▼
Aria Protocol Support	Disable	▼
Media Security Behavior	Mandatory	▼
Authentication On Transmitted RTP Packets	Active	▼
Encryption On Transmitted RTP Packets	Active	▼
Encryption On Transmitted RTCP Packets	Active	▼
SRTP Tunneling Authentication for RTP	Disable	▼
SRTP Tunneling Authentication for RTCP	Disable	▼

3. Click **Submit**.
4. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.15 on page 77).

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. For Interoperability with Twilio SIP Trunk it is required because forced transcoding enabled in order to deal with Broken Connection Timeout on Twilio SIP trunk issue.

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

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

Figure 4-33: Configuring Number of Media Channels

⚡ Number of Media Channels	30
----------------------------	----

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 [4.15](#) on page [77](#)).

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 4.7 on page 44, IP Group 1 represents Lync Server 2013, and IP Group 2 represents 8BTwilio SIP Trunk.

For the interoperability test topology, the following IP-to-IP routing rules need to be configured to route calls between Lync Server 2013 (LAN) and 8BTwilio SIP Trunk (WAN):

- Terminate SIP OPTIONS messages on the E-SBC that are received from the LAN
- Calls from Lync Server 2013 to 8BTwilio SIP Trunk
- Calls from 8BTwilio SIP Trunk to Lync Server 2013

➤ **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:
 - a. Click **Add**.
 - b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	0
Name	Terminate OPTIONS (arbitrary descriptive name)
Source IP Group	Lync
Request Type	OPTIONS

Figure 4-34: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS from LAN – Rule Tab

c. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Destination Type	Dest Address
Destination Address	internal

Figure 4-35: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS from LAN – Action Tab

3. Configure a rule to route calls from Lync Server 2013 to 8BTwilio SIP Trunk:
 - a. Click **Add**.
 - b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	1
Route Name	Lync to ITSP (arbitrary descriptive name)
Source IP Group	Lync

Figure 4-36: Configuring IP-to-IP Routing Rule for Lync to ITSP – Rule tab

The screenshot shows a configuration window titled "Add Row" with a close button (X) in the top right corner. Below the title bar, there are two input fields: "Index" with the value "1" and "Routing Policy" with a dropdown menu showing "Default_SBCRouting". Below these are two tabs: "Rule" (highlighted in orange) and "Action". Under the "Rule" tab, there are several configuration fields: "Name" (text input: "Lync to ITSP"), "Alternative Route Options" (dropdown: "Route Row"), "Source IP Group" (dropdown: "Lync"), "Request Type" (dropdown: "All"), "Source Username Prefix" (text input: "*"), "Source Host" (text input: "*"), "Destination Username Prefix" (text input: "*"), "Destination Host" (text input: "*"), "Message Condition" (dropdown: "None"), "Call Trigger" (dropdown: "Any"), and "ReRoute IP Group" (dropdown: "Any"). At the bottom right of the form area is a link labeled "Classic View". At the very bottom of the window are two buttons: "Add" and "Cancel".

c. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Destination Type	IP Group
Destination IP Group	Twilio
Destination SIP Interface	Twilio

Figure 4-37: Configuring IP-to-IP Routing Rule for Lync to ITSP – Action tab

The screenshot shows the 'Edit Row' dialog box with the following configuration:

- Index: 1
- Routing Policy: Default_SBCRouting
- Tab: Action
- Destination Type: IP Group
- Destination IP Group: Twilio
- Destination SIP Interface: Twilio
- Destination Address: (empty)
- Destination Port: 0
- Destination Transport Type: (empty)
- Call Setup Rules Set ID: -1
- Group Policy: None
- Cost Group: None

Buttons: Save, Cancel, Classic View

4. To configure rule to route calls from 8BTwilio SIP Trunk to Lync Server 2013:
 - a. Click **Add**.
 - b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	2
Route Name	ITSP to Lync (arbitrary descriptive name)
Source IP Group	Twilio

Figure 4-38: Configuring IP-to-IP Routing Rule for ITSP to Lync – Rule tab

c. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Destination Type	IP Group
Destination IP Group	Lync
Destination SIP Interface	Lync

Figure 4-39: Configuring IP-to-IP Routing Rule for ITSP to Lync – Action tab

✕
Edit Row

Index

Routing Policy Default_SBCRouting

Rule
Action

Destination Type IP Group

Destination IP Group Lync

Destination SIP Interface Lync

Destination Address

Destination Port

Destination Transport Type

Call Setup Rules Set ID

Group Policy None

Cost Group None

[Classic View](#)

Save
Cancel

The configured routing rules are shown in the figure below:

Figure 4-40: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table

Index	Name	Routing Policy	Alternative Route Options	Source IP Group	Request Type	Source Username Prefix	Destination Username Prefix	Destination Type	Destination IP Group	Destination SIP Interface	Destination Address
0	OPTIONS ter	Default_SBC	Route Row	Lync	OPTIONS	*	*	Dest Address:	None	None	internal
1	Lync to ITSP	Default_SBC	Route Row	Lync	All	*	*	IP Group	Twilio	Twilio	
2	ITSP to Lync	Default_SBC	Route Row	Twilio	All	*	*	IP Group	Lync	Lync	



Note: The routing configuration may change according to your specific deployment topology.

4.13 Step 13: 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.

➤ **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 a new manipulation rule (Manipulation Set 4) for 8BTwilio SIP Trunk. This rule applies to messages sent to the 8BTwilio SIP Trunk IP Group in a call forwarding scenario. This rule replaces the host part of the SIP History-Info Header with the value that was configured in the 8BTwilio SIP Trunk IP Group.

Parameter	Value
Index	0
Name	Change Host of History-Info.0
Manipulation Set ID	4
Message Type	invite.request
Condition	header.history-info.0 regex (.*)@(.*)(;user=phone)(.*)
Action Subject	header.history-info.0
Action Type	Modify
Action Value	\$1+\$2+param.ipg.dst.host+\$4+\$5

Figure 4-41: Configuring SIP Message Manipulation Rule 0 (for 8BTwilio SIP Trunk)

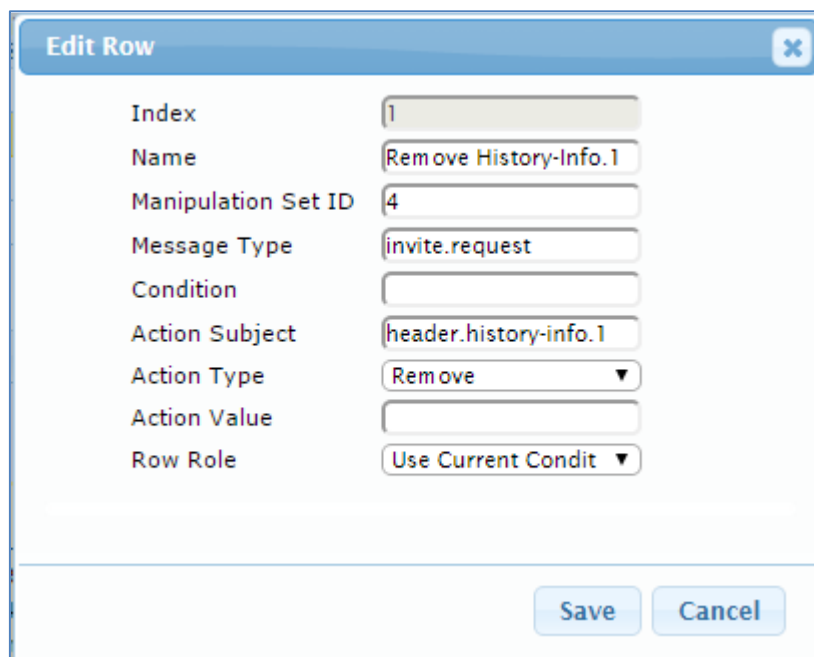
Edit Row
✕

Index	<input type="text" value="0"/>
Name	<input type="text" value="Change Host of History-"/>
Manipulation Set ID	<input type="text" value="4"/>
Message Type	<input type="text" value="invite.request"/>
Condition	<input type="text" value="header.history-info.0 re"/>
Action Subject	<input type="text" value="header.history-info.0"/>
Action Type	<input type="text" value="Modify"/>
Action Value	<input type="text" value="\$1+\$2+param.ipg.dst.h"/>
Row Role	<input type="text" value="Use Current Condit"/>

3. Configure another manipulation rule (Manipulation Set 4) for the 8BTwilio SIP Trunk. This rule also applies to messages sent to the 8BTwilio SIP Trunk IP Group in a call forwarding scenario. This rule removes SIP History-Info.1 Header.

Parameter	Value
Index	1
Name	Remove History-Info.1
Manipulation Set ID	4
Message Type	invite.request
Action Subject	header.history-info.1
Action Type	Remove

Figure 4-42: Configuring SIP Message Manipulation Rule 1 (for 8BTwilio SIP Trunk)



Edit Row
✕

Index	<input type="text" value="1"/>
Name	<input type="text" value="Remove History-Info.1"/>
Manipulation Set ID	<input type="text" value="4"/>
Message Type	<input type="text" value="invite.request"/>
Condition	<input type="text"/>
Action Subject	<input type="text" value="header.history-info.1"/>
Action Type	<input style="border-bottom: none; border-top: none; border-left: none; border-right: none; background-color: #f0f0f0; width: 100%;" type="text" value="Remove"/>
Action Value	<input type="text"/>
Row Role	<input style="border-bottom: none; border-top: none; border-left: none; border-right: none; background-color: #f0f0f0; width: 100%;" type="text" value="Use Current Condit"/>

4. Configure another manipulation rule (Manipulation Set 4) for the 8BTwilio SIP Trunk. This rule applies to messages sent to the 8BTwilio SIP Trunk IP Group in a call transfer scenario. This rule replaces the host part of the SIP Referred-by Header with the value that was configured in the 8BTwilio SIP Trunk IP Group.

Parameter	Value
Index	2
Name	Change Referred-by Host
Manipulation Set ID	4
Message Type	invite.request
Condition	header.referred-by exists
Action Subject	header.referred-by.url.host
Action Type	Modify
Action Value	param.ipg.dst.host

Figure 4-43: Configuring SIP Message Manipulation Rule 2 (for 8BTwilio SIP Trunk)

Edit Row
✕

Index	<input type="text" value="2"/>
Name	<input type="text" value="Change Referred-by Host"/>
Manipulation Set ID	<input type="text" value="4"/>
Message Type	<input type="text" value="invite.request"/>
Condition	<input type="text" value="header.referred-by exists"/>
Action Subject	<input type="text" value="header.referred-by.url.h"/>
Action Type	<input type="text" value="Modify"/>
Action Value	<input type="text" value="param.ipg.dst.host"/>
Row Role	<input type="text" value="Use Current Condit"/>

Figure 4-44: Configured SIP Message Manipulation Rules

Index	Name	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value	Row Role
0	Change Host of History-Info.0	4	invite.request	header.history-info.	header.history-i	Modify	\$1+\$2+param.ip	Use Current Conc
1	Remove History-Info.1	4	invite.request		header.history-i	Remove		Use Current Conc
2	Change Referred-by Host	4	invite.request	header.referred-by	header.referred-	Modify	param.ipg.dst.hc	Use Current Conc

The table displayed below includes SIP message manipulation rules which are bound together by commonality via the Manipulation Set ID 4 which are executed for messages sent to the 8BTwilio SIP Trunk IP Group. These rules are specifically required to enable proper interworking between 8BTwilio SIP Trunk and Lync Server 2013. Refer to the *User's Manual* for further details on the full capabilities of header manipulation.

SIP Message Manipulation Rules

Rule Index	Rule Description	Reason for Introducing Rule
0	This rule applies to messages sent to the SIP Trunk IP Group in a call forwarding scenario. This rule replaces the host part of the SIP History-Info Header with the value, configured in the 8BTwilio SIP Trunk IP Group.	To introduce Topology Hiding in the Call Forward scenarios, the host part of the SIP History-Info Header should be replaced with the value that was configured in the SIP Trunk IP Group.
1	This rule also applies to messages sent to the SIP Trunk IP Group in a call forwarding scenario. This rule removes the SIP History-Info.1 Header.	To introduce Topology Hiding in the Call Forward scenarios, the SIP History-Info.1 Header should be removed.
2	This rule applies to messages sent to the SIP Trunk IP Group in a call transfer scenario. This replaces the host part of the SIP Referred-by Header with the value, configured in the 8BTwilio SIP Trunk IP Group.	To introduce Topology Hiding in the Call Transfer scenarios, the host part of the SIP Referred-by Header should be replaced with the value that was configured in the SIP Trunk IP Group.

5. Assign Manipulation Set ID 4 to the SIP trunk IP Group:
 - a. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).
 - b. Select the row of the 8BTwilio SIP trunk IP Group, and then click **Edit**.
 - c. Click the **SBC** tab.
 - d. Set the 'Outbound Message Manipulation Set' field to **4**.

Figure 4-45: Assigning Manipulation Set 4 to the 8BTwilio SIP Trunk IP Group

The screenshot shows the 'Edit Row' dialog box with the 'SBC' tab selected. The 'Index' field is set to 2 and the 'SRD' dropdown is set to 'DefaultSRD'. The 'SBC' tab is active, and the 'Outbound Message Manipulation Set' field is set to 4. Other fields include 'SBC Operation Mode' (Not Configured), 'Classify By Proxy Set' (Enable), 'SBC Client Forking Mode' (Sequential), 'Inbound Message Manipulation Set' (-1), 'Msg Man User Defined String1' and 'String2' (empty), 'Registration Mode' (User Initiates Regis'), 'Max. Number of Registered Users' (-1), 'Authentication Mode' (User Authenticates), 'Authentication Method List' (empty), and 'Username' (empty). 'Save' and 'Cancel' buttons are at the bottom right.

- e. Click **Submit**.

4.14 Step 14: Miscellaneous Configuration

This section describes miscellaneous E-SBC configuration.

4.14.1 Step 14a: Configure Classification Table

This step shows how to configure the E-SBC Classification Table. For the interoperability test topology with the 8BTwilioSIP Trunk, the **Proxy** FQDN Address is configured in the Proxy Set Table and all outgoing calls are routed to this **Proxy** FQDN Address. However incoming calls from Twilio may arrive from different global locations (Twilio has local servers in main global regions). Consequently, it's necessary to also allow SIP messages to be received from these different local 8BTwilio servers.

➤ **To configure Classification Table:**

1. Open the Classification Table page (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **Classification Table**).
2. Click **Add**.
3. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Classification Name	Europe (arbitrary descriptive name)
Source SIP Interface	Twilio
Source Host	sip.ie1.twilio.com (Europe local FQDN)

Figure 4-46: Classification Table Page – Rule Tab

Edit Row
✕

Index

SRD

Rule

Action

Name

Source SIP Interface

Source IP Address

Source Transport Type

Source Port

Source Username Prefix

Source Host

Destination Username Prefix

Destination Host

Message Condition

[Classic View](#)

- Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Action Type	Allow
Source IP Group	Twilio
IP Profile	Twilio

Figure 4-47: Classification Table Page – Action Tab

- Click **Submit**.

Figure 4-48: Example of Classification Table

Inde	Name	SRD	Source SIP Interface	Source Username Prefix	Source Host	Destination Username Prefix	Destination Host	Action Type	Source IP Group
0	Europe	DefaultSRD	Twilio	*	sip.ie1.twilio.com	*	*	Allow	Twilio
1	North America	DefaultSRD	Twilio	*	sip.us1.twilio.com	*	*	Allow	Twilio

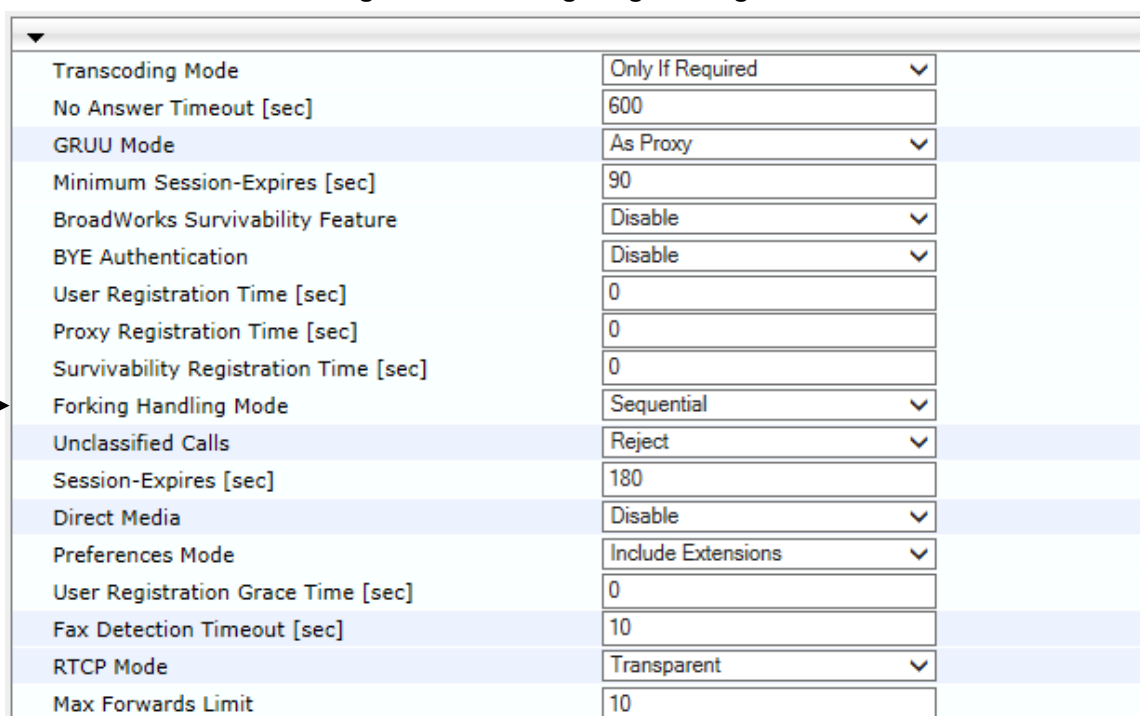
4.14.2 Step 14b: Configure Call Forking Mode

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 a SIP 18x response 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 ring-back tone if a 180 response without SDP is received. It is 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 4-49: Configuring Forking Mode



Transcoding Mode	Only If Required
No Answer Timeout [sec]	600
GRUU Mode	As Proxy
Minimum Session-Expires [sec]	90
BroadWorks Survivability Feature	Disable
BYE Authentication	Disable
User Registration Time [sec]	0
Proxy Registration Time [sec]	0
Survivability Registration Time [sec]	0
Forking Handling Mode	Sequential
Unclassified Calls	Reject
Session-Expires [sec]	180
Direct Media	Disable
Preferences Mode	Include Extensions
User Registration Grace Time [sec]	0
Fax Detection Timeout [sec]	10
RTCP Mode	Transparent
Max Forwards Limit	10

3. Click **Submit**.

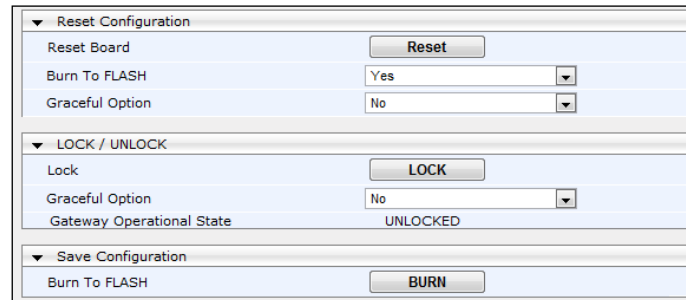
4.15 Step 15: Reset the E-SBC

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 4-50: Resetting the E-SBC



The screenshot displays a web-based configuration interface for the E-SBC. It is organized into three main sections, each with a dropdown arrow on the left:

- Reset Configuration:** This section contains three rows. The first row is 'Reset Board' with a 'Reset' button. The second row is 'Burn To FLASH' with a dropdown menu set to 'Yes'. The third row is 'Graceful Option' with a dropdown menu set to 'No'.
- LOCK / UNLOCK:** This section contains two rows. The first row is 'Lock' with a 'LOCK' button. The second row is 'Graceful Option' with a dropdown menu set to 'No'. Below this row, the text 'Gateway Operational State' is followed by 'UNLOCKED'.
- Save Configuration:** This section contains one row: 'Burn To FLASH' with a 'BURN' button.

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

This page is intentionally left blank.

A AudioCodes INI File

The *ini* configuration file of the E-SBC, corresponding to the Web-based configuration as described in Section 4 on page 31, 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 800 E-SBC
;HW Board Type: 69  FK Board Type: 72
;Serial Number: 2265355
;Slot Number: 1
;Software Version: 7.00A.013.015
;DSP Software Version: 5014AE3_R => 700.32
;Board IP Address: 10.15.17.77
;Board Subnet Mask: 255.255.0.0
;Board Default Gateway: 10.15.0.1
;Ram size: 369M  Flash size: 64M  Core speed: 300Mhz
;Num of DSP Cores: 3  Num DSP Channels: 60
;Num of physical LAN ports: 12
;Profile: NONE
;;Key features;;Board Type: 72 ;Channel Type: DspCh=60 IPMediaDspCh=60
;HA ;QOE features: VoiceQualityMonitoring MediaEnhancement ;DATA
features: ;Security: IPSEC MediaEncryption StrongEncryption
EncryptControlProtocol ;DSP Voice features: RTCP-XR ;Coders: G723 G729
G728 NETCODER GSM-FR GSM-EFR AMR EVRC-QCELP G727 ILBC EVRC-B AMR-WB G722
EG711 MS_RTA_NB MS_RTA_WB SILK_NB SILK_WB SPEEX_NB SPEEX_WB OPUS_NB
OPUS_WB ;E1Trunks=2 ;T1Trunks=2 ;FXSPorts=4 ;FXOPorts=4 ;BRITrunks=4 ;IP
Media: Conf VXML ;Control Protocols: MGCP SIP SASurvivability SBC=60 MSFT
CLI TRANSCODING=60 FEU=100 TestCall=100 EMS LAD=20 ;Default
features;;Coders: G711 G726;

;----- HW components-----
;
; Slot # : Module type : # of ports
;-----
;      1 : BRI           : 4
;      2 : FXS           : 4
;      3 : FALC56       : 1
;-----

[SYSTEM Params]

SyslogServerIP = 10.15.17.100
EnableSyslog = 1
;NTPServerIP_abs is hidden but has non-default value
NTPServerUTCOffset = 7200
;VpFileLastUpdateTime is hidden but has non-default value
NTPServerIP = '10.15.25.1'
```

```
;LastConfigChangeTime is hidden but has non-default value
;PM_gwINVITEDialogs is hidden but has non-default value
;PM_gwSUBSCRIBEDialogs is hidden but has non-default value
;PM_gwSBCRegisteredUsers is hidden but has non-default value
;PM_gwSBCMediaLegs is hidden but has non-default value
;PM_gwSBCTranscodingSessions is hidden but has non-default value

[BSP Params]

PCMLawSelect = 3
UdpPortSpacing = 10
EnterCpuOverloadPercent = 99
ExitCpuOverloadPercent = 95

[Analog Params]

[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]

[MEGACO Params]

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

[PSTN Params]

[SS7 Params]

[Voice Engine Params]

ENABLEMEDIASECURITY = 1
CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]

UserProductName = 'Mediant 800 E-SBC'
WebLogoText = 'Twilio'
UseWeblogo = 1
;UseLogoInWeb is hidden but has non-default value
UseProductName = 1
HTTPSCipherString = 'RC4:EXP'
;HTTPSCertFileName is hidden but has non-default value
;HTTPSRootFileName is hidden but has non-default value

[SIP Params]
```

```
MEDIACHANNELS = 30
GWDEBUGLEVEL = 5
;ISPRACKREQUIRED is hidden but has non-default value
ENABLESBCAPPLICATION = 1
MSLDAPPRIMARYKEY = 'telephoneNumber'
SBCPREFERENCESEMODE = 1
SBCFORKINGHANDLINGMODE = 1
ENERGYDETECTORCMD = 587202560
ANSWERDETECTORCMD = 10486144
;GWAPPCONFIGURATIONVERSION is hidden but has non-default value

[SCTP Params]

[IPsec Params]

[Audio Staging Params]

[SNMP Params]

[ PhysicalPortsTable ]

FORMAT PhysicalPortsTable_Index = PhysicalPortsTable_Port,
PhysicalPortsTable_Mode, PhysicalPortsTable_SpeedDuplex,
PhysicalPortsTable_PortDescription, PhysicalPortsTable_GroupMember,
PhysicalPortsTable_GroupStatus;
PhysicalPortsTable 0 = "GE_1", 1, 4, "User Port #0", "GROUP_1", "Active";
PhysicalPortsTable 1 = "GE_2", 1, 4, "User Port #1", "GROUP_1",
"Redundant";
PhysicalPortsTable 2 = "GE_3", 1, 4, "User Port #2", "GROUP_2", "Active";
PhysicalPortsTable 3 = "GE_4", 1, 4, "User Port #3", "GROUP_2",
"Redundant";
PhysicalPortsTable 4 = "FE_5_1", 0, 4, "User Port #4", "None", " ";
PhysicalPortsTable 5 = "FE_5_2", 0, 4, "User Port #5", "None", " ";
PhysicalPortsTable 6 = "FE_5_3", 0, 4, "User Port #6", "None", " ";
PhysicalPortsTable 7 = "FE_5_4", 0, 4, "User Port #7", "None", " ";
PhysicalPortsTable 8 = "FE_5_5", 1, 4, "User Port #8", "GROUP_5",
"Active";
PhysicalPortsTable 9 = "FE_5_6", 1, 4, "User Port #9", "GROUP_5",
"Redundant";
PhysicalPortsTable 10 = "FE_5_7", 1, 4, "User Port #10", "GROUP_6",
"Active";
PhysicalPortsTable 11 = "FE_5_8", 1, 4, "User Port #11", "GROUP_6",
"Redundant";

[ \PhysicalPortsTable ]

[ EtherGroupTable ]

FORMAT EtherGroupTable_Index = EtherGroupTable_Group,
EtherGroupTable_Mode, EtherGroupTable_Member1, EtherGroupTable_Member2;
EtherGroupTable 0 = "GROUP_1", 2, "GE_1", "GE_2";
EtherGroupTable 1 = "GROUP_2", 2, "GE_3", "GE_4";
EtherGroupTable 2 = "GROUP_3", 0, "", "";
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EtherGroupTable 3 = "GROUP_4", 0, "", "";
EtherGroupTable 4 = "GROUP_5", 2, "FE_5_5", "FE_5_6";
EtherGroupTable 5 = "GROUP_6", 2, "FE_5_7", "FE_5_8";
EtherGroupTable 6 = "GROUP_7", 0, "", "";
EtherGroupTable 7 = "GROUP_8", 0, "", "";
EtherGroupTable 8 = "GROUP_9", 0, "", "";
EtherGroupTable 9 = "GROUP_10", 0, "", "";
EtherGroupTable 10 = "GROUP_11", 0, "", "";
EtherGroupTable 11 = "GROUP_12", 0, "", "";

[ \EtherGroupTable ]

[ DeviceTable ]

FORMAT DeviceTable_Index = DeviceTable_VlanID,
DeviceTable_UnderlyingInterface, DeviceTable_DeviceName,
DeviceTable_Tagging;
DeviceTable 0 = 1, "GROUP_1", "vlan 1", 0;
DeviceTable 1 = 2, "GROUP_2", "vlan 2", 0;

[ \DeviceTable ]

[ InterfaceTable ]

FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_InterfaceName, InterfaceTable_PrimaryDNSServerIPAddress,
InterfaceTable_SecondaryDNSServerIPAddress,
InterfaceTable_UnderlyingDevice;
InterfaceTable 0 = 6, 10, 10.15.17.77, 16, 10.15.0.1, "Voice",
10.15.25.1, 0.0.0.0, "vlan 1";
InterfaceTable 1 = 5, 10, 195.189.192.158, 25, 195.189.192.129, "WANSP",
80.179.52.100, 80.179.55.100, "vlan 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 ]

[ WebUsers ]

FORMAT WebUsers_Index = WebUsers_Username, WebUsers_Password,
WebUsers_Status, WebUsers_PwAgeInterval, WebUsers_SessionLimit,
WebUsers_SessionTimeout, WebUsers_BlockTime, WebUsers_UserLevel,
WebUsers_PwNonce;
    
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WebUsers 0 = "Admin",
"$1$LE0VGBxUAQFSUAJXUQANXwoPDwtaeSNwInB2c3B+eihzKSgvfDIzMDI1YGc0YWwhub2h1P
GpUVwdVB1NSBgpRXV4=", 1, 0, 2, 15, 60, 200,
"62caced25276f6d59432fcac295a1346";
WebUsers 1 = "User",
"$1$frwcHLO4tOHmvOKy7Oiys7m5vrzbzpqfyoKL0r6v7q/iv/P35kpmUwcXBkZWYy5iaz8+Wm
NGBgoPXhdTRi4yDj94=", 1, 0, 2, 15, 60, 50,
"e124fc45691a62316416e055a60edb6f";

[ \WebUsers ]

[ TLSContexts ]

FORMAT TLSContexts_Index = TLSContexts_Name, TLSContexts_TLSVersion,
TLSContexts_ServerCipherString, TLSContexts_ClientCipherString,
TLSContexts_OcspEnable, TLSContexts_OcspServerPrimary,
TLSContexts_OcspServerSecondary, TLSContexts_OcspServerPort,
TLSContexts_OcspDefaultResponse;
TLSContexts 0 = "default", 1, "RC4:EXP", "ALL:!ADH", 0, 0.0.0.0, 0.0.0.0,
2560, 0;

[ \TLSContexts ]

[ 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_SBCAllowedMediaTypes, IpProfile_SBCAllowedCodersGroupID,
IpProfile_SBCAllowedVideoCodersGroupID, IpProfile_SBCAllowedCodersMode,
IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior,
IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCAssertIdentity,
IpProfile_AMDSensitivityParameterSuit, IpProfile_AMDSensitivityLevel,
IpProfile_AMDMaxGreetingTime, IpProfile_AMDMaxPostSilenceGreetingTime,
IpProfile_SBCDiversionsMode, 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,

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IpProfile_GenerateSRTPKeys, IpProfile_SBCPlayHeldTone,
IpProfile_SBCRemoteHoldFormat, IpProfile_SBCRemoteReplacesBehavior,
IpProfile_SBCSDPptimeAnswer, IpProfile_SBCPreferredPTime,
IpProfile_SBCUseSilenceSupp, IpProfile_SBCRTPRedundancyBehavior,
IpProfile_SBCPlayRBTToTransferee, IpProfile_SBCRTCPMode,
IpProfile_SBCJitterCompensation,
IpProfile_SBCRemoteRenegotiateOnFaxDetection,
IpProfile_JitterBufMaxDelay,
IpProfile_SBCUserBehindUdpNATRegistrationTime,
IpProfile_SBCUserBehindTcpNATRegistrationTime,
IpProfile_SBCSDPHandleRTCPAttribute,
IpProfile_SBCRemoveCryptoLifetimeInSDP, IpProfile_SBCIceMode,
IpProfile_SBCRTCPMux, IpProfile_SBCMediaSecurityMethod,
IpProfile_SBCHandleXDetect, IpProfile_SBCRTCPFeedback,
IpProfile_SBCRemoteRepresentationMode, IpProfile_SBCKeepVIAHeaders,
IpProfile_SBCKeepRoutingHeaders, IpProfile_SBCKeepUserAgentHeader,
IpProfile_SBCRemoteMultipleEarlyDialogs,
IpProfile_SBCRemoteMultipleAnswersMode, IpProfile_SBCDirectMediaTag,
IpProfile_SBCAdaptRFC2833BWTtoVoiceCoderBW;
IpProfile 1 = "Lync", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 0,
-1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, "", 0, -1, 0, 1, 0,
0, 0, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 1, 0, 1, 1, 0, 3, 2, 1, 0, 1,
1, 1, 1, 1, 0, 1, 0, 0, 0, 0, 1, 0, 1, 1, 0, 0, 0, 0, 0, 0, 1, 0,
0, 300, -1, -1, 0, 0, 0, 0, 0, 0, 0, -1, -1, -1, -1, -1, 0, "", 0;
IpProfile 2 = "Twilio", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0,
0, -1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, "", -1, 0, 1, "", -1, -1, 0, 2,
0, 0, 1, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 0, 0, 1, 3, 0, 1, 0,
1, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 0, 5, 0, 0, 0, 0, 0, 1, 0,
0, 0, 300, -1, -1, 0, 0, 0, 0, 0, 0, 0, -1, -1, -1, -1, -1, 0, "", 0;

[ \IpProfile ]

[ CpMediaRealm ]

FORMAT CpMediaRealm_Index = CpMediaRealm_MediaRealmName,
CpMediaRealm_IPv4IF, CpMediaRealm_IPv6IF, CpMediaRealm_PortRangeStart,
CpMediaRealm_MediaSessionLeg, CpMediaRealm_PortRangeEnd,
CpMediaRealm_IsDefault, CpMediaRealm_QoeProfile, CpMediaRealm_BWProfile;
CpMediaRealm 0 = "MRLan", "Voice", "", 6000, 100, 6990, 1, "", "";
CpMediaRealm 1 = "MRWan", "WANSP", "", 7000, 100, 7990, 0, "", "";

[ \CpMediaRealm ]

[ SBCRoutingPolicy ]

FORMAT SBCRoutingPolicy_Index = SBCRoutingPolicy_Name,
SBCRoutingPolicy_LCREnable, SBCRoutingPolicy_LCRAverageCallLength,
SBCRoutingPolicy_LCRDefaultCost, SBCRoutingPolicy_LdapServerGroupName;
SBCRoutingPolicy 0 = "Default_SBCRoutingPolicy", 0, 1, 1, "";

[ \SBCRoutingPolicy ]

[ SRD ]

FORMAT SRD_Index = SRD_Name, SRD_BlockUnRegUsers, SRD_MaxNumOfRegUsers,
SRD_EnableUnAuthenticatedRegistrations, SRD_SharingPolicy,
SRD_UsedByRoutingServer, SRD_SBCOperationMode,
SRD_SBCRegisteredUsersClassificationMethod, SRD_SBCRoutingPolicyName;
SRD 0 = "DefaultSRD", 0, -1, 1, 0, 0, 0, -1, "Default_SBCRoutingPolicy";
    
```

```

[ \SRD ]

[ SIPInterface ]

FORMAT SIPInterface_Index = SIPInterface_InterfaceName,
SIPInterface_NetworkInterface, SIPInterface_ApplicationType,
SIPInterface_UDPPort, SIPInterface_TCPSPort, SIPInterface_TLSPort,
SIPInterface_SRDName, SIPInterface_MessagePolicyName,
SIPInterface_TLSContext, SIPInterface_TLSMutualAuthentication,
SIPInterface_TCPKeepaliveEnable,
SIPInterface_ClassificationFailureResponseType,
SIPInterface_PreClassificationManSet, SIPInterface_EncapsulatingProtocol,
SIPInterface_MediaRealm, SIPInterface_SBCDirectMedia,
SIPInterface_BlockUnRegUsers, SIPInterface_MaxNumOfRegUsers,
SIPInterface_EnableUnAuthenticatedRegistrations,
SIPInterface_UsedByRoutingServer;
SIPInterface 0 = "Lync", "Voice", 2, 0, 0, 5067, "DefaultSRD", "",
"default", -1, 0, 500, -1, 0, "MRLan", 0, -1, -1, -1, 0;
SIPInterface 1 = "Twilio", "WANSP", 2, 5060, 0, 0, "DefaultSRD", "",
"default", -1, 0, 500, -1, 0, "MRWan", 0, -1, -1, -1, 0;

[ \SIPInterface ]

[ ProxySet ]

FORMAT ProxySet_Index = ProxySet_ProxyName,
ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap,
ProxySet_SRDName, ProxySet_ClassificationInput, ProxySet_TLSContextName,
ProxySet_ProxyRedundancyMode, ProxySet_DNSResolveMethod,
ProxySet_KeepAliveFailureResp, ProxySet_GWIPv4SIPInterfaceName,
ProxySet_SBCIPv4SIPInterfaceName, ProxySet_SASIPv4SIPInterfaceName,
ProxySet_GWIPv6SIPInterfaceName, ProxySet_SBCIPv6SIPInterfaceName,
ProxySet_SASIPv6SIPInterfaceName;
ProxySet 0 = "Lync", 1, 60, 1, 1, "DefaultSRD", 0, "default", 1, -1, "",
"", "Lync", "", "", "", "";
ProxySet 1 = "Twilio", 1, 60, 0, 0, "DefaultSRD", 0, "", -1, 1, "", "",
"Twilio", "", "", "", "", "";

[ \ProxySet ]

[ IPGroup ]

FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Name, IPGroup_ProxySetName,
IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_SipReRoutingMode,
IPGroup_AlwaysUseRouteTable, IPGroup_SRDName, IPGroup_MediaRealm,
IPGroup_ClassifyByProxySet, IPGroup_ProfileName,
IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet, IPGroup_OutboundManSet,
IPGroup_RegistrationMode, IPGroup_AuthenticationMode, IPGroup_MethodList,
IPGroup_EnablesSBCCClientForking, IPGroup_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName, IPGroup_Username,
IPGroup_Password, IPGroup_UUIFormat, IPGroup_QOEProfile,
IPGroup_BWProfile, IPGroup_MediaEnhancementProfile,
IPGroup_AlwaysUseSourceAddr, IPGroup_MsgManUserDef1,
IPGroup_MsgManUserDef2, IPGroup_SIPConnect, IPGroup_SBCPSAPMode,
IPGroup_DTLSContext, IPGroup_CreatedByRoutingServer,
IPGroup_UsedByRoutingServer, IPGroup_SBCOperationMode,
IPGroup_SBCRouteUsingRequestURIPort;

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IPGroup 0 = 0, "Lync", "Lync", "ilync15.pstn.twilio.com", "", -1, 0,
"DefaultSRD", "MRlan", 1, "Lync", -1, -1, -1, 0, 0, "", 0, -1, -1, "",
"", "$1$gQ==", 0, "", "", "", 0, "", "", 0, 0, "", 0, 0, -1, 0;
IPGroup 1 = 0, "Twilio", "Twilio", "ilync15.pstn.twilio.com", "", -1, 0,
"DefaultSRD", "MRWan", 1, "Twilio", -1, -1, 4, 0, 0, "", 0, -1, -1, "",
"", "$1$gQ==", 0, "", "", "", 0, "", "", 0, 0, "", 0, 0, -1, 0;

[ \IPGroup ]

[ ProxyIp ]

FORMAT ProxyIp_Index = ProxyIp_ProxySetId, ProxyIp_ProxyIpIndex,
ProxyIp_IpAddress, ProxyIp_TransportType;
ProxyIp 0 = "0", 0, "FE15.ilync15.local:5067", 2;
ProxyIp 1 = "1", 0, "ilync15.pstn.twilio.com:5060", 0;

[ \ProxyIp ]

[ IP2IPRouting ]

FORMAT IP2IPRouting_Index = IP2IPRouting_RouteName,
IP2IPRouting_RoutingPolicyName, IP2IPRouting_SrcIPGroupName,
IP2IPRouting_SrcUsernamePrefix, IP2IPRouting_SrcHost,
IP2IPRouting_DestUsernamePrefix, IP2IPRouting_DestHost,
IP2IPRouting_RequestType, IP2IPRouting_MessageConditionName,
IP2IPRouting_ReRouteIPGroupName, IP2IPRouting_Trigger,
IP2IPRouting_CallSetupRulesSetId, IP2IPRouting_DestType,
IP2IPRouting_DestIPGroupName, IP2IPRouting_DestSIPInterfaceName,
IP2IPRouting_DestAddress, IP2IPRouting_DestPort,
IP2IPRouting_DestTransportType, IP2IPRouting_AltRouteOptions,
IP2IPRouting_GroupPolicy, IP2IPRouting_CostGroup;
IP2IPRouting 0 = "OPTIONS termination", "Default_SBCRoutingPolicy",
"Lync", "*", "*", "*", "*", 6, "", "Any", 0, -1, 1, "", "", "internal",
0, -1, 0, 0, "";
IP2IPRouting 1 = "Lync to ITSP", "Default_SBCRoutingPolicy", "Lync", "*",
"*, "*", "*", 0, "", "Any", 0, -1, 0, "Twilio", "Twilio", "", 0, -1, 0,
0, "";
IP2IPRouting 2 = "ITSP to Lync", "Default_SBCRoutingPolicy", "Twilio",
"*, "*", "*", "*", 0, "", "Any", 0, -1, 0, "Lync", "Lync", "", 0, -1, 0,
0, "";

[ \IP2IPRouting ]

[ Classification ]

FORMAT Classification_Index = Classification_ClassificationName,
Classification_MessageConditionName, Classification_SRDName,
Classification_SrcSIPInterfaceName, Classification_SrcAddress,
Classification_SrcPort, Classification_SrcTransportType,
Classification_SrcUsernamePrefix, Classification_SrcHost,
Classification_DestUsernamePrefix, Classification_DestHost,
Classification_ActionType, Classification_SrcIPGroupName,
Classification_DestRoutingPolicy, Classification_IpProfileName;
Classification 0 = "Europe", "", "DefaultSRD", "Twilio", "", 0, -1, "*",
"sip.iel.twilio.com", "*", "", 1, "Twilio", "", "Twilio";
Classification 1 = "North America", "", "DefaultSRD", "Twilio", "", 0, -
1, "*", "sip.us1.twilio.com", "*", "*", 1, "Twilio", "", "Twilio";
    
```

```
[ \Classification ]

[ CodersGroup0 ]

FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce,
CodersGroup0_CoderSpecific;
CodersGroup0 0 = "g711Ulaw64k", 20, 0, -1, 0, "";

[ \CodersGroup0 ]

[ AllowedCodersGroup0 ]

FORMAT AllowedCodersGroup0_Index = AllowedCodersGroup0_Name;
AllowedCodersGroup0 0 = "g711Ulaw64k";

[ \AllowedCodersGroup0 ]

[ MessageManipulations ]

FORMAT MessageManipulations_Index =
MessageManipulations_ManipulationName, MessageManipulations_ManSetID,
MessageManipulations_MessageType, MessageManipulations_Condition,
MessageManipulations_ActionSubject, MessageManipulations_ActionType,
MessageManipulations_ActionValue, MessageManipulations_RowRole;
MessageManipulations 0 = "Change Host of History-Info.0", 4,
"invite.request", "header.history-info.0 regex
(.*) (@) (.*) (;user=phone) (.*)", "header.history-info.0", 2,
"$1+$2+param.ipg.dst.host+$4+$5", 0;
MessageManipulations 1 = "Remove History-Info.1", 4, "invite.request",
"", "header.history-info.1", 1, "", 0;
MessageManipulations 2 = "Change Referred-by Host", 4, "invite.request",
"header.referred-by exists", "header.referred-by.url.host", 2,
"param.ipg.dst.host", 0;

[ \MessageManipulations ]

[ GwRoutingPolicy ]

FORMAT GwRoutingPolicy_Index = GwRoutingPolicy_Name,
GwRoutingPolicy_LCREnable, GwRoutingPolicy_LCRAverageCallLength,
GwRoutingPolicy_LCRDefaultCost, GwRoutingPolicy_LdapServerGroupName;
GwRoutingPolicy 0 = "GwRoutingPolicy", 0, 1, 1, "";

[ \GwRoutingPolicy ]

[ ResourcePriorityNetworkDomains ]

FORMAT ResourcePriorityNetworkDomains_Index =
ResourcePriorityNetworkDomains_Name,
ResourcePriorityNetworkDomains_Ip2TelInterworking;
ResourcePriorityNetworkDomains 1 = "dsn", 1;
ResourcePriorityNetworkDomains 2 = "dod", 1;
ResourcePriorityNetworkDomains 3 = "drsn", 1;
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ResourcePriorityNetworkDomains 5 = "uc", 1;  
ResourcePriorityNetworkDomains 7 = "cuc", 1;  
  
[ \ResourcePriorityNetworkDomains ]
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