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Interoperability Guide

Integrating an ADTRAN eSBC with Lync Server 2013

This interoperability guide provides instructions for integrating an ADTRAN enterprise session border controller (eSBC) and the Microsoft Lync Server 2013 using a Session Initiation Protocol (SIP) trunk to provide a connection to the service provider network. This guide includes the description of the network application, verification summary, and example individual device configurations for the ADTRAN eSBC and the Lync Server products.

This guide consists of the following sections:

Overview

Service providers are increasingly using SIP trunks to provide Voice over IP (VoIP) services to customers. ADTRAN eSBC products provide features that normalize the SIP signaling and media between the customer’s PBX and the service provider’s SBC and softswitch server. In this application, the ADTRAN eSBC gateway operates as a SIP back-to-back user agent (B2BUA). One Ethernet interface on the ADTRAN eSBC provides the wide area network (WAN) connection to the service provider network and terminates the service provider SIP trunk. A second Ethernet interface connects the customer's local area network (LAN) and provides a SIP trunk connection to the IP PBX for VoIP applications. *Figure 1* illustrates the use of the ADTRAN eSBC in a typical network deployment.

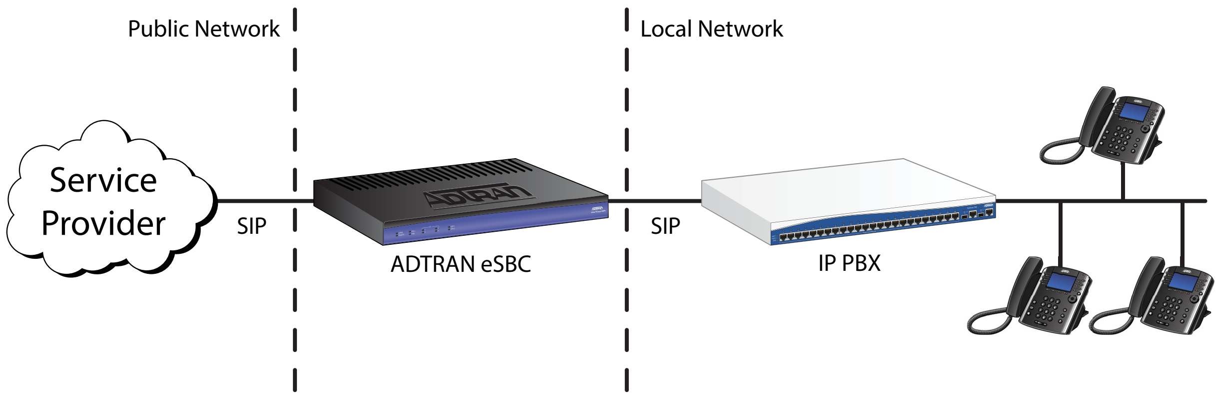


Figure 1. ADTRAN eSBC in the Network

Interoperability

The network topology shown in *Figure 2 on page 3* was used for interoperability verification between the ADTRAN eSBC and the Lync Server. The configuration is a typical SIP trunking application, in which the ADTRAN eSBC gateway Ethernet interface provides the Ethernet WAN connection to the service provider network. It should be noted that the WAN connection is not limited to Ethernet. A second Ethernet interface connects to the customer LAN. The Lync Server connects to the customer LAN. Two SIP trunks are configured on the ADTRAN eSBC gateway: one to the service provider SIP network and the second to the Lync Server. The ADTRAN eSBC gateway operates as a SIP B2BUA, and outbound and inbound calls to the public switched telephone network (PSTN) are routed through the ADTRAN eSBC.

The Lync Server supports various phone types (including digital, H.323, and SIP IP phones). The phones register locally to the Lync Server. Dial plan configuration routes external calls through the SIP trunk to the ADTRAN eSBC gateway.

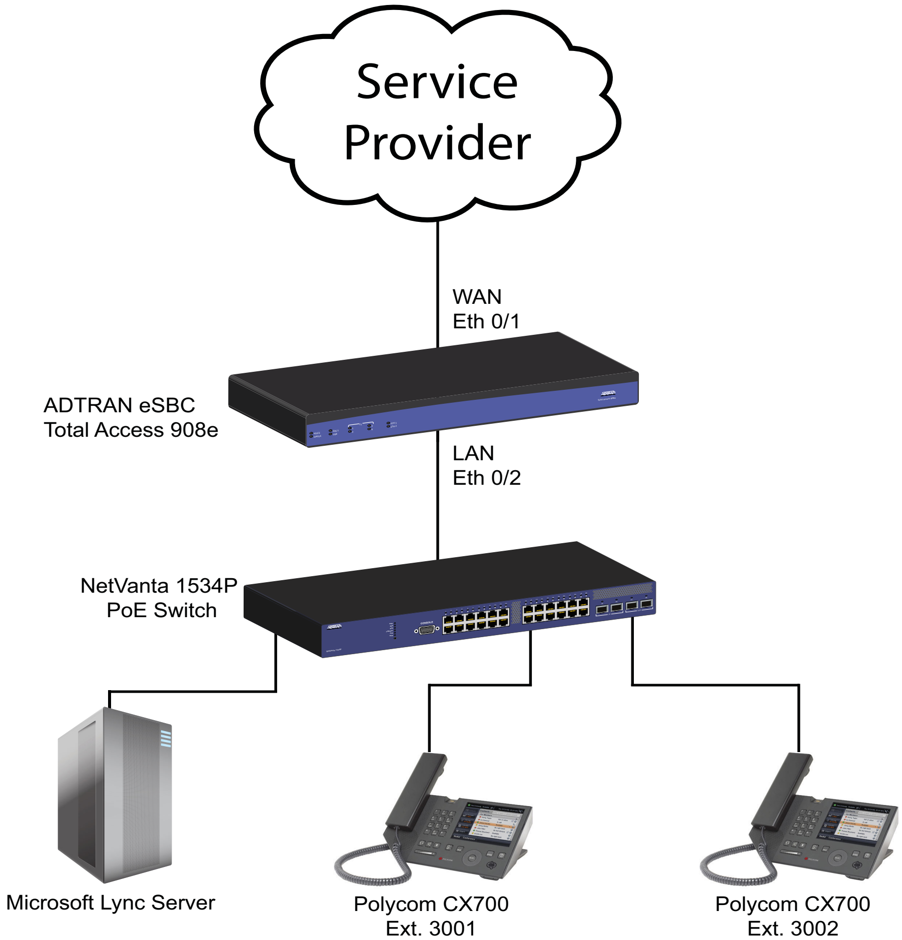


Figure 2. Network Topology for Verification

Hardware and Software Requirements and Limitations

Interoperability with Lync Server is available on ADTRAN products with the eSBC feature code as outlined in the *AOS Feature Matrix*, available online at ADTRAN's Support Forum, https://supportforums.adtran.com. The test equipment, testing parameters, and associated caveats are described in the following sections.

Equipment and Versions

The following table outlines the equipment and firmware versions used in verification testing.

Table 1. Verification Test Equipment and Firmware Versions

Product	Firmware Version
ADTRAN Total Access 908e with eSBC	R10.9.6
Lync Server	2013 Standard Edition
Polycom CX700 OCS/Lync Edition Phone	4.0.7577.4451

Configuring the ADTRAN eSBC

The following sections describe the key configuration settings required for this solution. These settings are implemented using the ADTRAN Operating System (AOS) command line interface (CLI) .

To configure the ADTRAN eSBC for interoperability with the Lync Server using the CLI, follow these steps:

Step 1: Access the eSBC CLI

The AOS unit can be managed using the console port, Hypertext Transfer Protocol (HTTP), HTTP Secure (HTTPS), Telnet, and Secure Shell (SSH). Most of the initial configuration is performed through the console port or Telnet session. Accessing the AOS unit is described in this step.

To access the CLI on your AOS unit, follow these steps:

1. Boot up the unit.
2. Telnet to the unit (**telnet** <*ip address*>), for example:

   telnet 10.10.10.1.

   > NOTE: *If during the unit's setup process you have changed the default IP address (10.10.10.1), use the configured IP address.*

3. Enter your user name and password at the prompt.

   > NOTE: *The AOS default user name is **admin** and the default password is **password**. The default enable password is **password**. If your product no longer has the default user name and passwords, contact your system administrator for the appropriate user name and passwords.*

4. Enable your unit by entering **enable** at the prompt as follows:

   >**enable**

5. If configured, enter your Enable mode password at the prompt.
6. Enter the unit's Global Configuration mode as follows:

   #**configure terminal**
   (config)#

Step 2: Configure the Basic Network Settings

Basic network configuration includes setting up two Ethernet interfaces, one for the Ethernet LAN interface to the Lync Server, and the second for the Ethernet WAN interface to the service provider. Both interfaces are configured using the **ip address** *<ipv4 address> <subnet mask>* and **media-gateway ip primary** commands. The **ip address** command configures a static IPv4 address for the interface, and the **media-gateway** command is required on the interface for SIP and Realtime Transport Protocol (RTP) media traffic.

> NOTE: *When configuring the basic network settings, use the IP address information supplied by the service provider.*

Enter the commands from the Ethernet interface configuration mode as follows:

For the WAN:
(config)#**interface ethernet 0/1**
(config-eth 0/1)#**description WAN**
(config-eth 0/1)#**ip address 203.0.113.2 255.255.255.252**
(config-eth 0/1)#**media-gateway ip primary**
(config-eth 0/1)#**no shutdown**

For the LAN:
(config)#**interface ethernet 0/2**
(config-eth 0/2)#**description LAN**
(config-eth 0/2)#**ip address 192.168.1.1 255.255.255.0**
(config-eth 0/2)#**media-gateway ip primary**
(config-eth 0/2)#**no shutdown**

Step 3: Enable SIP over TCP

Lync Server uses Transport Control Protocol (TCP) for SIP messaging. Consequently, SIP over TCP must be enabled on the ADTRAN eSBC. Use the **sip tcp** command from the Global Configuration mode to enable SIP over TCP.

Enter the command as follows:

(config)#**sip tcp**

Step 4: Configure Global Voice Modes for Local Handling

Configure the ADTRAN eSBC to use the local mode for call forwarding and transfer handling. By default, both of these functions are handled by the network. To change these settings, use the **voice transfer-mode local** and **voice forward-mode local** commands. Enter these commands from the Global Configuration mode. By using the **local** parameter, both commands specify allowing the unit to handle call forwarding and transfers locally.

Enter the commands as follows:

(config)#**voice transfer-mode local**
(config)#**voice forward-mode local**

Step 5: Enable Media Anchoring

Media anchoring is an eSBC feature that forces RTP traffic through the ADTRAN eSBC gateway. Minimum configuration for media anchoring includes enabling the feature using the **ip rtp media-anchoring** command from the Global Configuration mode. The RTP symmetric filter works in conjunction with media anchoring to filter nonsymmetric RTP packets. The RTP symmetric filter is enabled using the **ip rtp symmetric-filter** command.

Enter the commands as follows:

(config)#**ip rtp symmetric-filter**
(config)#**ip rtp media-anchoring**

Step 6: Configure the Service Provider SIP Trunk

The first of two voice trunks that must be configured is the SIP trunk to the service provider from the ADTRAN eSBC. Check with your service provider for any specific requirements beyond those listed in this document. Your service provider will provide you with the IP addresses or fully qualified domain name (FQDN) and possibly the port numbers for their SIP server. They may also provide a backup or secondary SIP server.

The trunk is created using the **voice trunk** <*Txx*> **type sip** command. The **description** <*text*> command is used to label the trunk. The **sip-server primary** <*ip address* | *hostname*> command is used to define the host name or IP address of the primary server to which the trunk sends SIP messages.

Enter the commands as follows:

(config)#**voice trunk T01 type sip**
(config-T01)#**description PROVIDER**
(config-T01)#**sip-server primary sip.example.com**

Step 7: Configure the PBX SIP Trunk

The second voice trunk that must be configured is the SIP trunk to the Lync Server from the ADTRAN eSBC. The **voice trunk** <*Txx*> **type sip** command is used to define a new SIP trunk and activate the Voice Trunk Configuration mode for the individual trunk. From the Voice Trunk Configuration mode, you can provide a descriptive name for the trunk and define the IP address (or host name) of the PBX. The **description** <*text*> command is used to label the trunk. The **sip-server-primary** <*ip address* | *hostname*> command is used to set the server address to the Lync Server LAN IP address.

In addition, the Lync Server must control call transfers. This is accomplished using the **transfer-mode-network** command in the trunk's configuration. The **grammar from host local** command is used to specify that the IP address of the interface is used in the SIP From header for outbound messages.

Enter the commands as follows:

(config)#**voice trunk T11 type sip**
(config-T02)#**description PBX**
(config-T02)#**sip-server primary 192.168.1.2 tcp**
(config-T02)#**transfer-mode network**
(config-T02)#**grammar from host local**

Step 8: Configure a Trunk Group for the Service Provider

After configuring the two SIP trunks, configure an individual trunk group for the service provider trunk account. The previously created trunks are added to the trunk group, which is then used to assign outbound call destinations (local calls, long distance calls, etc.). A cost is also assigned to each **accept** template in the trunk group.

Use the **voice grouped-trunk** *<name>* command to create a trunk group and to enter the Voice Trunk Group Configuration mode. The **trunk** *<Txx>* command adds an existing trunk to the trunk group, so that outbound calls can be placed out that particular trunk. The *<Txx>* parameter specifies the trunk identity where *xx* is the trunk ID number.

Use the **accept** *<pattern>* command to specify number patterns that are accepted for routing calls out of the trunk. Use the **no** form of this command to remove a configured dial pattern. The *<pattern>* parameter is specified by entering a complete phone number or using wildcards to help define accepted numbers.

Valid characters for templates are as follows:

0 - 9	Match the exact digit(s) only
X	Match any single digit 0 through 9
N	Match any single digit 2 through 9
M	Match any single digit 1 through 8
$	Match any number string dialed
[]	Match any digit in the list within the brackets (for example, [1,4,6])
,()	Formatting characters that are ignored but allowed
-	Use within brackets to specify a range, otherwise ignored

The following are example template entries using wildcards:

1) NXX-XXXX	Match any 7-digit number beginning with 2 through 9
2) 1-NXX-NXX-XXXX	Match any number with a leading 1, then 2 through 9, then any 2 digits, then 2 through 9, then any 6 digits
3) 555-XXXX	Match any 7-digit number beginning with 555
4) XXXX$	Match any number with at least 5 digits
5) [7,8]$	Match any number beginning with 7 or 8
6) 1234	Match exactly 1234

Some template number rules:

1. All brackets must be closed with no nesting of brackets and no wildcards within the brackets.
2. All brackets can hold digits and commas, for example: [1239]; [1,2,3,9]. Commas are implied between numbers within brackets and are ignored.
3. Brackets can contain a range of numbers using a hyphen, for example: [1-39]; [1-3,9].
4. The $ wildcard is only allowed at the end of the template, for example: 91256$; XXXX$.

Enter the commands as follows:

(config)#**voice grouped-trunk PROVIDER**
(config-PROVIDER)#**trunk T01**
(config-PROVIDER)#**accept NXX-NXX-XXXX cost 0**
(config-PROVIDER)#**accept 1-NXX-NXX-XXXX cost 0**
(config-PROVIDER)#**accept 011-$ cost 0**
(config-PROVIDER)#**accept 411 cost 0**
(config-PROVIDER)#**accept 611 cost 0**
(config-PROVIDER)#**accept 911 cost 0**

Step 9: Configure a Trunk Group for the Lync Server

After configuring a trunk group for the service provider, create a trunk group for the Lync Server trunk account. Create the trunk group using the **voice grouped-trunk** *<name>* command. Add an existing trunk to the trunk group using the **trunk** *<Txx>* command. The outbound allowed calls are defined using the **accept** *<pattern>* command and are assigned a cost using the **cost** parameter, as described in *Step 8: Configure a Trunk Group for the Service Provider on page 7*.

Enter the commands as follows:

(config)#**voice grouped-trunk PBX**
(config-PBX)#**trunk T11**
(config-PBX)#**accept $ cost 0**

Step 10: Configure the Double reINVITE Preference

The **sip prefer double-reinvite** command is used from the Global Configuration mode to specify that a double reINVITE is preferred globally for all calls in the system. Calls that typically require a double reINVITE are forwarded calls and attended transfers. When these calls connect, a double reINVITE is initiated.

By default, the system is configured so that double reINVITEs are preferred. If a transfer call involves a SIP trunk operating in the local transfer mode, a double reINVITE is executed regardless of this preference setting. To avoid extra SIP messaging in situations where it is not necessary, set this feature to not prefer double reINVITEs by entering the **no** form of the **sip prefer double-reinvite** command from the Global Configuration mode.

Enter the command as follows:

(config)#**no sip prefer double-reinvite**

Step 11: Configure SIP Privacy (Optional)

The ADTRAN eSBC supports SIP user privacy using the Privacy and P-Asserted-Identity (PAI) SIP headers. The **sip privacy** command is used from the Global Configuration mode to enable SIP privacy support. The **trust-domain** command is used from the Voice SIP Trunk Configuration mode to connect the trunk to a trusted domain and enable PAI support, adding security measures for user's identity and privacy.

Enter the commands as follows:

(config)#**sip privacy**
(config)#**voice trunk T01 type sip**
(config-T01)#**trust-domain**
(config)#**voice trunk T11 type sip**
(config-T02)#**trust-domain**

ADTRAN eSBC Sample Configuration

The following example configuration demonstrates a typical installation of an ADTRAN eSBC configured as the SIP trunking gateway between a Lync Server and a service provider.

NOTE *The configuration parameters entered in this example are sample configurations only, and only pertain to the configuration of the SIP trunking gateway functionality. This application should be configured in a manner consistent with the needs of your particular network. CLI prompts have been removed from the configuration example. This configuration should not be copied without first making the necessary adjustments to ensure it will function properly in your network.*

NOTE *For details on each configuration option, refer to the AOS Command Reference Guide and other IP business gateway/eSBC configuration guides on the ADTRAN support forums at https://supportforums.adtran.com.*

```
!
interface eth 0/1
  description WAN
  ip address 203.0.113.2 255.255.255.252
  media-gateway ip primary
  no shutdown
!
interface eth 0/2
  description LAN
  ip address 192.168.1.1 255.255.255.0
  media-gateway ip primary
  no shutdown
!
ip route 0.0.0.0 0.0.0.0 203.0.113.1
!
sip
sip udp 5060
sip tcp 5060
```

```
!
voice feature-mode network
voice transfer-mode local
voice forward-mode local
!
voice trunk T01 type sip
  description PROVIDER
  sip-server primary sip.example.com
  trust-domain
!
voice trunk T11 type sip
  description PBX
  sip-server primary 192.168.1.2 tcp
  trust-domain
  grammar from host local
  transfer-mode network
!
voice grouped-trunk PROVIDER
  trunk T01
  accept NXX-NXX-XXXX cost 0
  accept 1-NXX-NXX-XXXX cost 0
  accept 011-$ cost 0
  accept 411 cost 0
  accept 611 cost 0
  accept 911 cost 0
!
voice grouped-trunk PBX
  trunk T11
  accept $ cost 0
!
sip privacy
!
no sip prefer double-reinvite
!
ip rtp symmetric-filter
ip rtp media-anchoring
!
```

Configuring the Lync Server

The following section describes the Lync Server system configuration to support SIP trunking interoperability with the ADTRAN eSBC. The Lync Server is configured using a custom Microsoft Silverlight app installed with Lync Server that runs on Windows Server. Topology Builder (typically installed on the Lync Server) is also required to reconfigure the topology.This section is divided into general system settings and trunk configurations (both group and individual) needed to support SIP trunking.

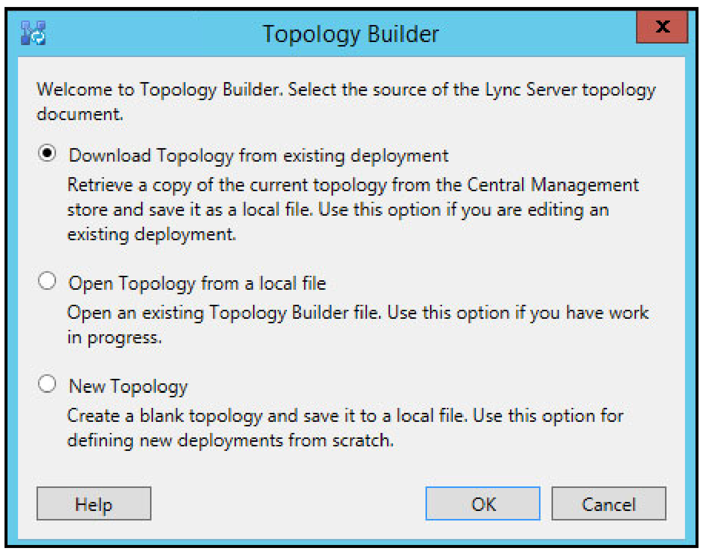
To configure the Lync Server SIP trunk, follow these steps:

Step 1: Create the PSTN Gateway Instance to the ADTRAN eSBC on page 11
Step 2: Configure the Normalization Rules on page 15
Step 3: Assign the DID to the User on page 17
Step 4: Configure Outbound Calling on page 17
Step 5: Configure the ADTRAN eSBC Trunk on page 19
Step 6: Restart the Front-End and Mediation Services on page 19

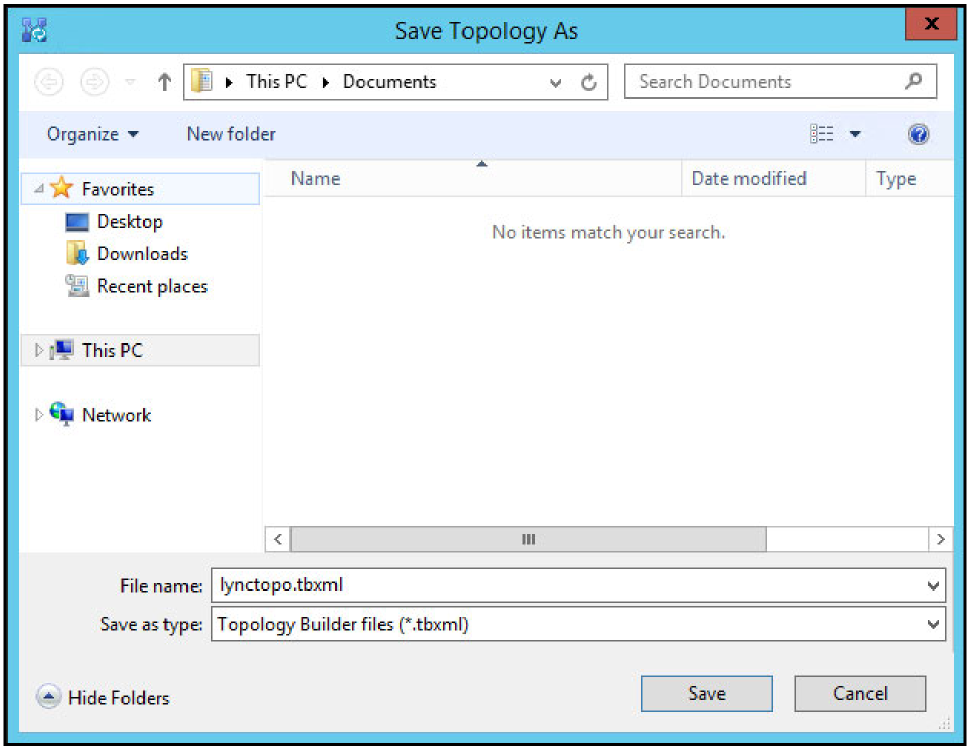
Step 1: Create the PSTN Gateway Instance to the ADTRAN eSBC

To create a PSTN gateway instance on the Lync server to the ADTRAN eSBC follow these steps:

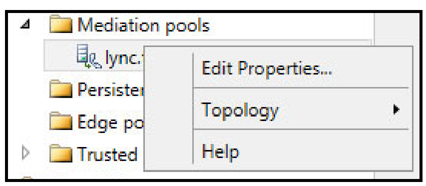
1. On the server where the Lync Server is installed, launch **Topology Builder** from the **Start** screen. When the welcome prompt appears, select the **Download Topology from existing deployment** radio button, then select **OK**.



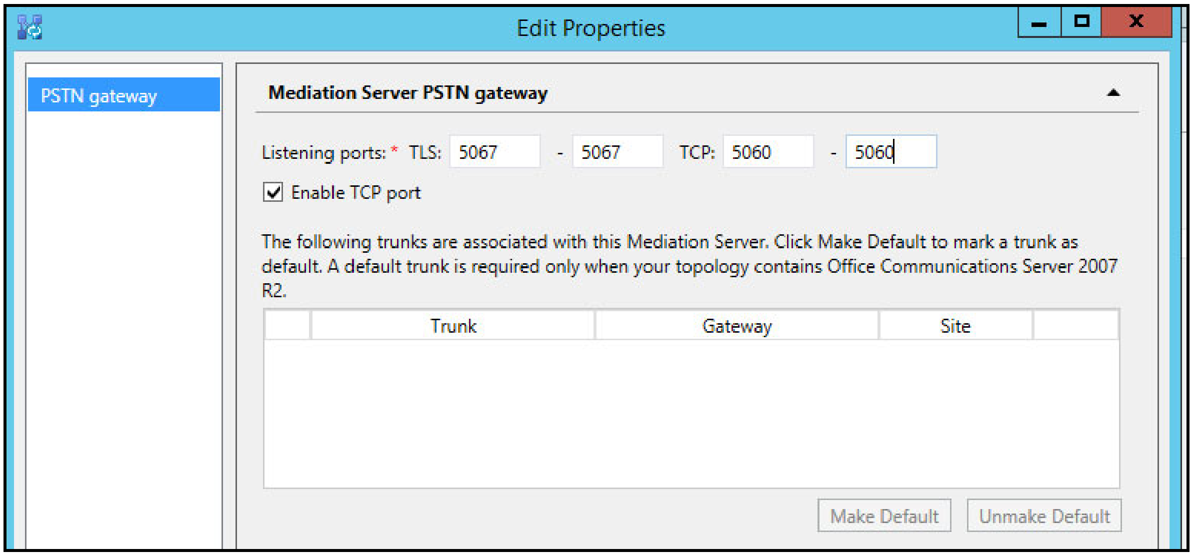
2. When prompted, navigate to the desired directory on your local drive, and select **Save** to save the topology.



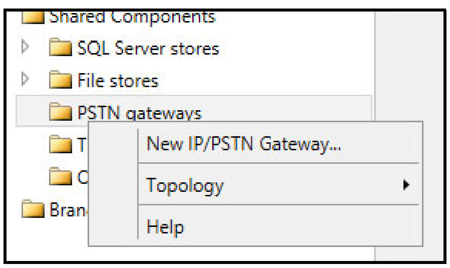
3. In the left pane of **Topology Builder**, navigate to **Lync Server** > ***(Site name)*** > **Lync Server 2013** > **Mediation pools**. Right click on the mediation server and select **Edit Properties**.



4. In the **Edit Properties** menu perform the following:
   a. Select the **Enable TCP port** check box.
   b. In both **TCP** fields enter **5060**.
   c. Select **OK**.



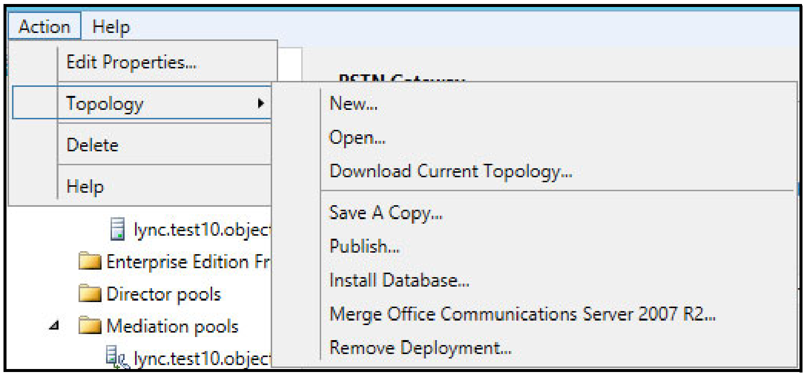
5. In the left pane of **Topology Builder**, navigate to **Lync Server** > ***(Site name)*** > **Shared Components** > **PSTN gateways**. Right-click on **PSTN gateways** and select **New IP/PSTN Gateway**.



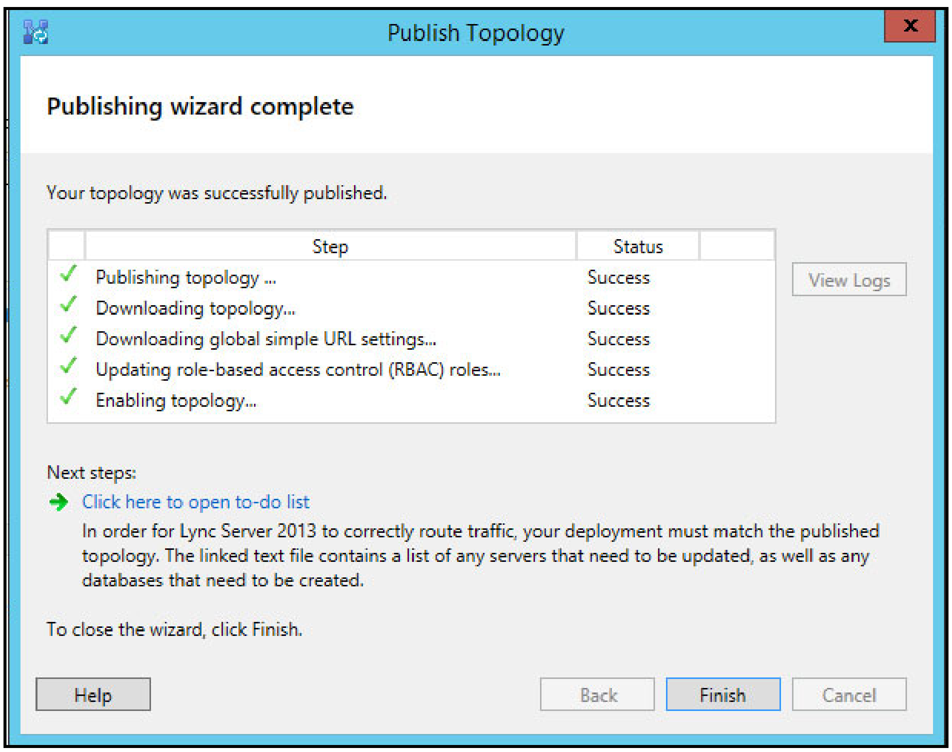
6. In the **Define New IP/PSTN Gateway** wizard, perform the following:
   a. In the **FQDN** field of the **Define the PSTN Gateway FQDN** menu, enter the LAN IP address of the ADTRAN eSBC gateway, for example: **192.168.1.1**. Then, select **Next**.
   b. In the **Define the IP address** menu, select the **Enable IPv6 radio** button, then select the **Use all configured IP addresses** radio button beneath. Select **Next**.
   c. In the **Trunk Name** field of the **Define the root trunk** menu, enter a unique name to as the trunk name to the ADTRAN eSBC. In the **Listening port for IP/PSTN gateway** field, enter **5060**. Select

TCP from the **SIP Transport Protocol** drop-down list. In the **Associated Mediation Server port** field, enter **5060**. Then, select **Finish**.

7. To apply the changes to the gateway configuration, you must publish the topology to the Lync Server. From the **Action** menu in **Topology Builder**, select **Topology > Publish**.



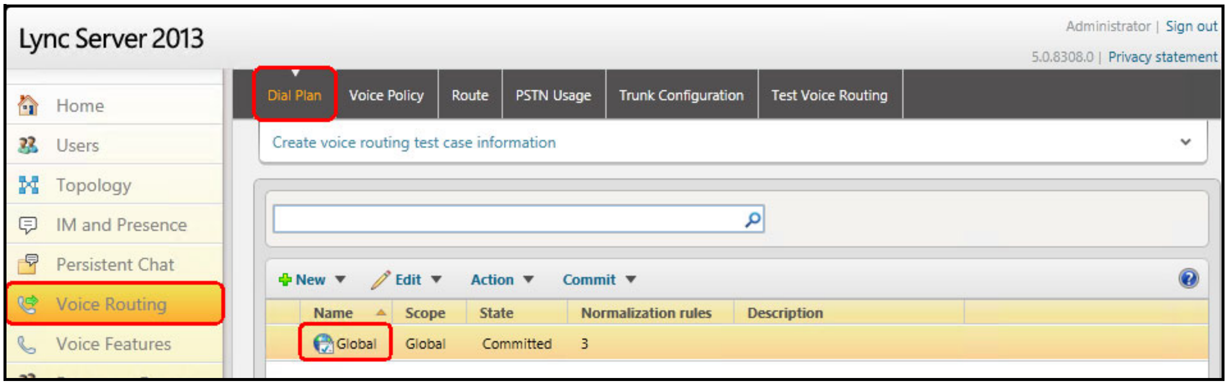
8. When the **Publish Topology** menu appears, select **Next** and wait for the wizard to apply the changes. When the publishing wizard has finished publishing the topology, select **Finish** to close the wizard.



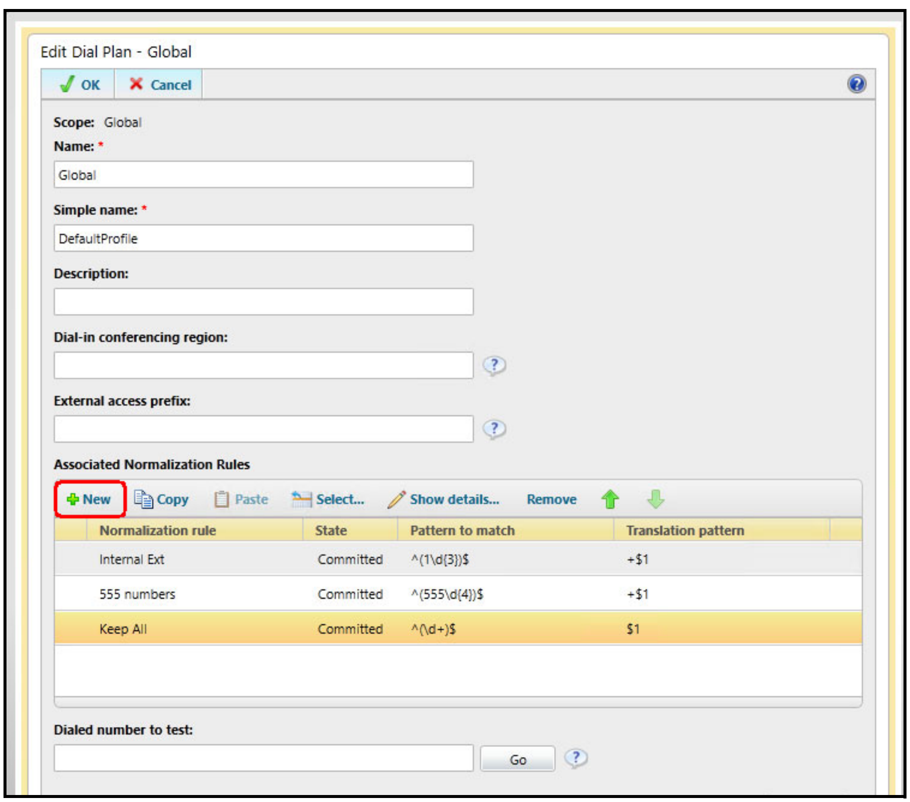
Step 2: Configure the Normalization Rules

Direct inward dialing (DID) normalization rules must be configured on the Lync Server for each DID in the deployment. At a minimum, a rule must be defined for the main site DID. To configure the DID normalization rules for the Lync Server, follow these steps:

1. Launch the **Lync Server 2013 Control Panel** from the Windows Server Start screen.
2. Log in as the active directory user assigned to the **CsAdministrator** group role.
3. In the control panel, select **Voice Routing** from the left pane, and select the **Dial Plan** tab.



4. Double click on the **Global** dial plan. The **Edit Dial Plan - Global** menu will appear.
5. In the **Edit Dial Plan - Global** menu, select **New** under **Associated Normalization Rules**. The **New Normalization Rule** menu will appear.



6. In the **New Normalization** menu, you will configure a normalization rule for inbound DID manipulation, and route the inbound DID to a specified extension. For example, if your DID is **8015553104** then this will need to be converted to **8015553104;ext=**<*user ext*> when routing through the dial plan. Using this example, perform the following:

   a. In the **Name** field, enter a unique name for the normalization rule.

   b. In the **Starting Digits** field, enter the DID.

   c. Use the **Length** drop-down list to select **Exactly**, and enter **10** in the adjacent field.

   d. In the **Digits to remove** field, enter **0**.

   e. Clear the **Digits to add** field.

   f. Clear the **Internal Extension** check box.

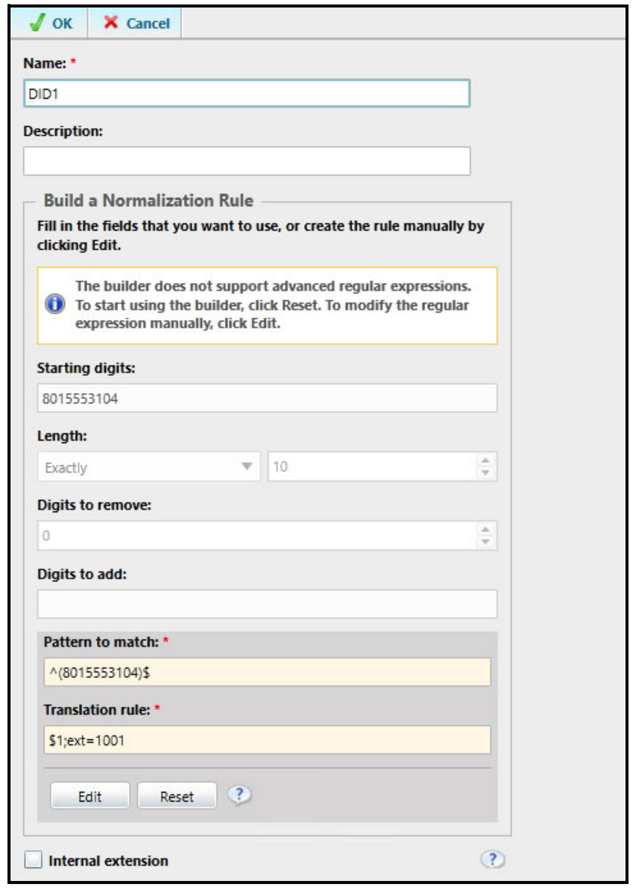
   g. Select **Edit** under the **Pattern to match** and **Translation rule** fields.

   h. In the **Pattern to match** field, enter the match pattern in the format **^(**<*DID number*>**)$**, where <*DID number*> is the DID number to be normalized. For example: **^(8015553104)$**.

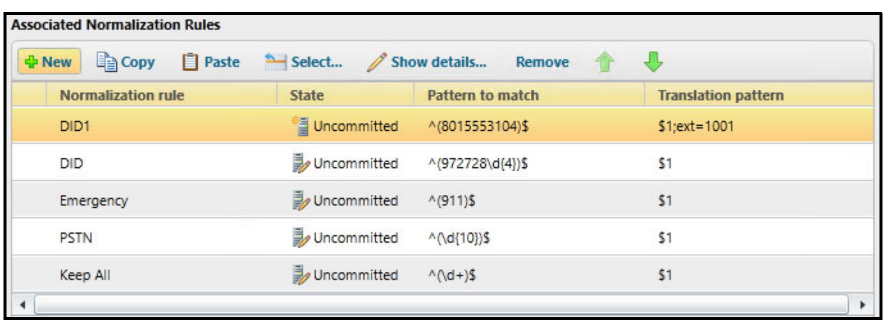
   i. In the **Translation rule** field, enter the translation rule in the format **$1;ext=**<*extension*>, where <*extension*> is the internal extension to route the DID. For example: **$1;ext=1001**.

   j. Select **OK** to add the normalization rule.

   k. The normalization rule will be added, and the **Edit Dial Plan - Global** menu will appear.



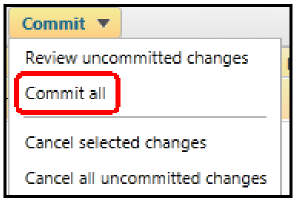
7. In the **Associated Normalization Rules** section of the **Edit Dial Plan - Global** menu, select the rule you just added, and use the green up arrow to move the rule to the top of the list.



8. Repeat Steps 1 through 7 for each DID.

9. Select **OK** to apply the rule changes. The **Dial Plan** menu will appear.

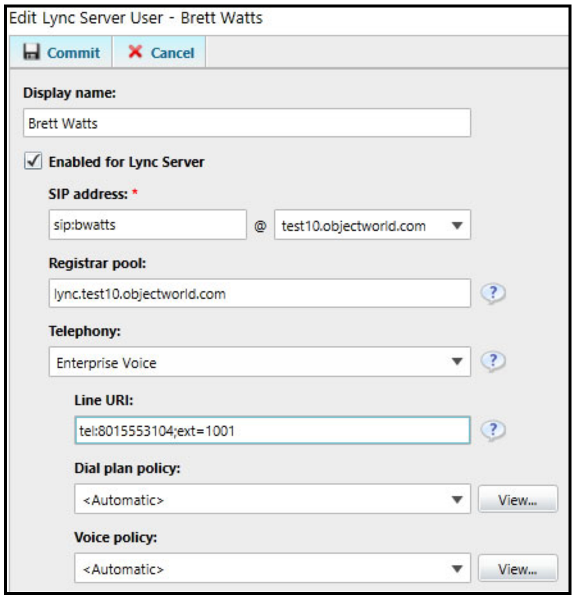
10. In the **Dial Plan** menu, use the **Commit** drop-down list to select **Commit all**.



Step 3: Assign the DID to the User

After configuring the normalization rules, the DID numbers can be assigned to users. Before proceeding to this step, ensure you have existing users enabled on the Lync Server for each DID. To assign a DID number to a user, follow these steps.

1. Select **Users** from the left pane of the Lync Server control panel, then select the **User Search** tab. The **User Search** menu will appear.
2. In the **User Search** menu, search for or browse to the user you want to assign to the DID, and double-click on the user. The **Edit Lync Server User** menu will appear.
3. In the **Edit Lync Server User** menu perform the following:
   a. Use the **Telephony** drop-down list to select **Enterprise Voice**.
   b. In the **Line URI** field, enter a DID and user extension in the format: **tel:**<*DID number*>**;ext=**<*extension*>, where <*DID number*> is the DID number for the user and <*extension*> is the extension for the user, for example: **tel:8015553104;ext=1001**.
   c. Select **Commit** to apply the changes.

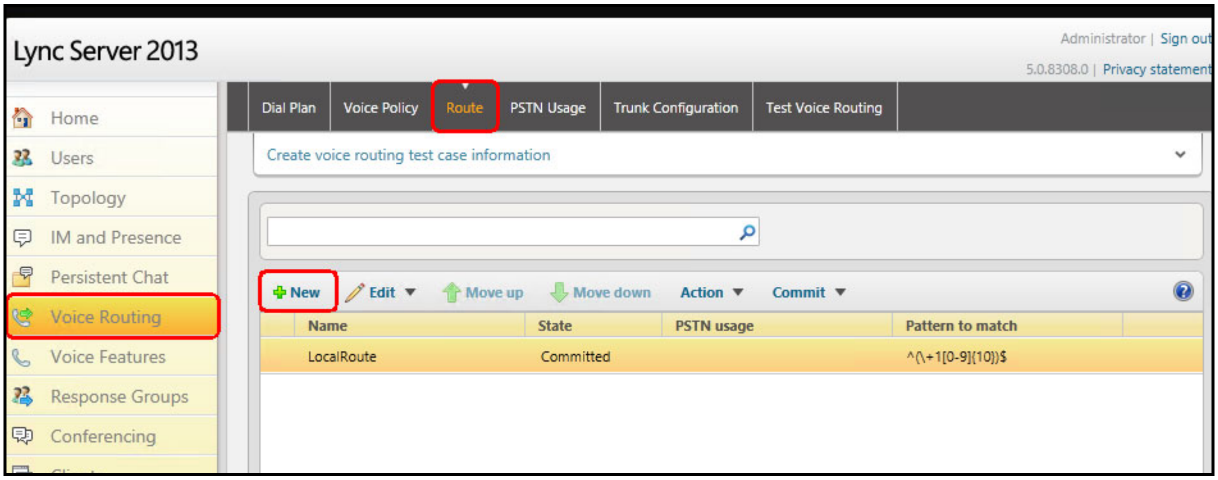


Step 4: Configure Outbound Calling

To route outbound calls to the ADTRAN eSBC and the service provider, you must add a voice route for 10-digit numbers. Additional routes can be added as needed by adding match patterns. To configure a 10-digit match pattern for outbound calling, follow these steps:

1. Select **Voice Routing** from the left pane of the Lync Server control panel, then select the **Route** tab. The **Route** menu will appear.

2. In the **Route** menu, select **New**. The **New Voice Route** menu will appear.



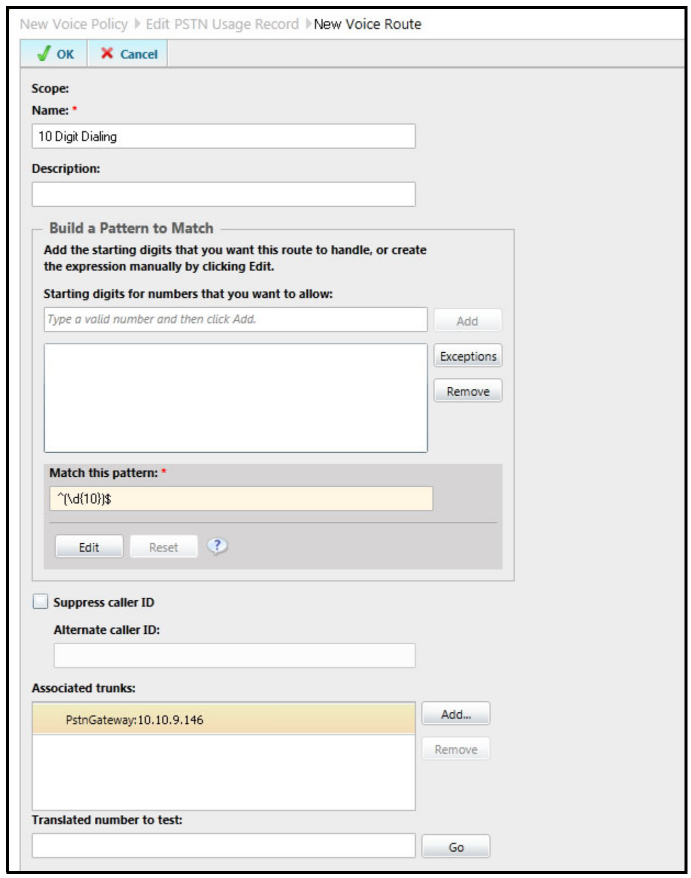
3. In the **New Voice Route** window, perform the following:

   a. In the **Name** field, enter a unique name for the voice route.

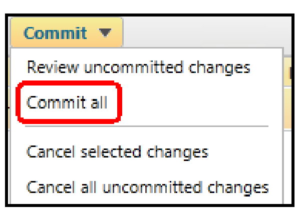
   b. In the **Match this pattern** field, enter **^(\d{10})$**.

   c. In the **Associated trunks** section, select the **Add** button. Then, select the PSTN trunk created in *Step 1: Create the PSTN Gateway Instance to the ADTRAN eSBC on page 11*.

   d. Select **OK** to save the changes.



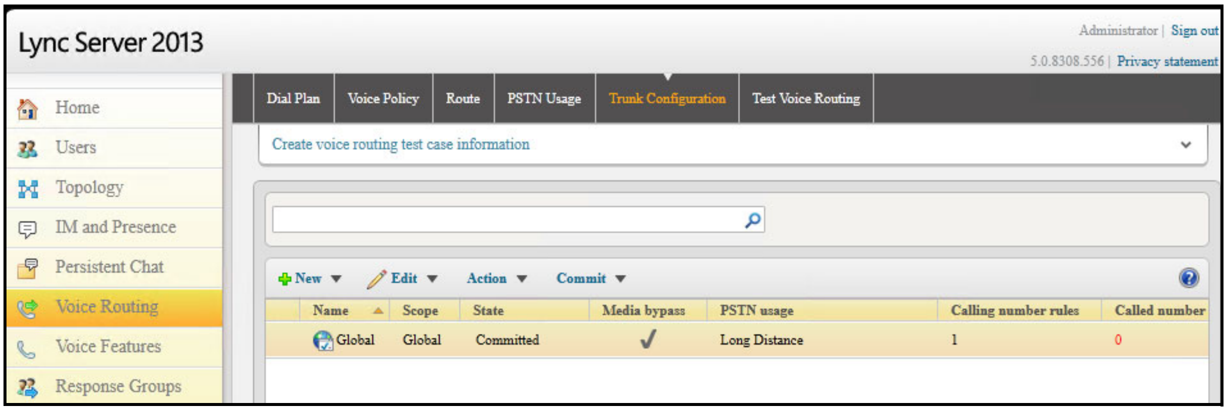
4. In the **Route** menu, use the **Commit** drop-down list to select **Commit all**.



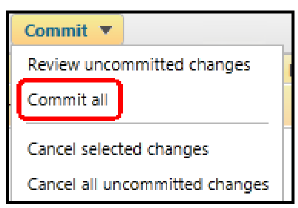
Step 5: Configure the ADTRAN eSBC Trunk

The ADTRAN eSBC trunk will need to be modified to support transfers via third-party call control and media bypass support. To do this, use the following steps:

1. Select **Voice Routing** from the left pane of the Lync Server control panel, then select the **Trunk Configuration** tab. The **Trunk Configuration** menu will appear.
2. In the **Trunk Configuration** menu, select the ADTRAN eSBC SIP trunk; this will likely be the **Global** trunk. Then, select **Edit** > **Show Details**. The **Edit Trunk Configuration** menu will appear.



3. In the **Edit Trunk Configuration** menu, use the **Refer support** drop-down list to select **Enable refer using third-party call control**.
4. Select the **Enable media bypass** check box.
5. Select the **Centralized media processing** check box.
6. Select **OK** to apply the changes.
7. In the **Trunk Configuration** menu, use the **Commit** drop-down list to select **Commit all**.



Step 6: Restart the Front-End and Mediation Services

After configuring the Lync Server, the Front-end and Mediation services must be restarted to update the changes. These services can be restarted using the Windows Services snap-in.

Additional Resources

There are additional resources available to aid in configuring your ADTRAN eSBC unit. Many of the topics discussed in this guide are complex and require additional understanding, such as using the CLI, eSBC in AOS, and ANI/DNIS substitution. The documents listed in *Table 2* are available online on ADTRAN's Support Forum at https://supportforums.adtran.com.

Table 2. Additional ADTRAN Documentation

Feature	Document Title
All AOS Commands Using the CLI	*AOS Command Reference Guide*
ANI and DNIS Substitution	*Enhanced ANI/DNIS Substitution in AOS*
eSBC Product Overview	*Session Border Controllers in AOS*
Media Anchoring	*Configuring Media Anchoring in AOS*
PSTN connectivity in Lync 2013	*Planning for PSTN connectivity in Lync Server 2013*

