

Mediant 1000 E-SBC

# **Configuration Note**

## Connecting Verizon Business SIP Trunk to Microsoft® Lync Server Using AudioCodes Mediant 1000 E-SBC

Document #: LTRT-31900



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**Reader's Notes** 

### Notice

This document describes the procedure for integrating the Verizon Internet Telephony Service Provider (ITSP) SIP Trunk with Microsoft® Lync Server using AudioCodes Mediant 1000 E-SBC.

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### **Abbreviations and Terminology**

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



**Note:** Throughout this guide, the term *gateway* refers to AudioCodes' Mediant 1000 E-SBC device.



**Reader's Notes** 

### 1 Introduction

This document is intended for incumbent IP telephony customers who wish to successfully integrate their Microsoft® Lync environments with a Verizon Business SIP Trunk, using the AudioCodes' E-SBC device.

Microsoft® Unified Communications implements the Microsoft® Lync platforms to combine Enterprise voice, instant messaging, enhanced presence, audio-video conferencing, and e-mail into a familiar, integrated communications experience. Enterprise voice is a Microsoft® Lync server fully featured VoIP solution, which includes connectivity with the telephony network.

For Enterprises wishing to fully utilize their deployed Microsoft® Lync server and not only communicate over IP within the Enterprise, but also outside the Enterprise, a SIP trunk provided by an Internet Telephony Service Provider (ITSP) (such as Verizon IP Trunking service) for connection to the traditional PSTN network, is one of the best possible solutions. Unlike traditional telephony, where bundles of physical wires are delivered from the PSTN service provider to a business, a SIP trunk allows a company to replace these traditional fixed PSTN lines with PSTN connectivity using a SIP Trunking service provider on the Internet.

Connecting Microsoft® Lync directly to a SIP trunk poses various implementation issues. Some ITSPs require SIP user registration, which Microsoft® Lync does not support. In addition, some ITSPs implement SIP over UDP, while Microsoft® Lync implements SIP over TCP or TLS. However, AudioCodes E-SBC devices are capable of providing the interface between the Microsoft® Lync and the ITSP's network, thereby allowing the support for all required functionalities.

Verizon Business has introduced new Internet protocol-based capabilities for its Contact Center Services and VoIP portfolio to help businesses enhance customerservice operations and leverage the benefits of VoIP. IP Trunking enables companies who have already invested in the Microsoft Lync system, to now connect on as few as one converged access lines for both internal and external traffic.



**Note:** This document is relevant for the Microsoft® Lync environment, as well as for the Microsoft Office Communication Server R2 environment.



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

To facilitate the understanding of the configuration setup, the procedures described in this document are based on an example scenario as described in Section 2 on page 11.



**Reader's Notes** 

## 2 Example Scenario

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

- An Enterprise has a deployed Microsoft® Lync server 2010 in its private network for enhanced communication within the company.
- The enterprise decides to offer its employees Enterprise voice and to connect the company to the PSTN network using the Verizon Business SIP Trunking service.

The setup requirements are as follows:

- While the Microsoft® Lync Server 2010 environment is located on the Enterprise's Local Area Network (LAN), the Verizon Business SIP Trunks are located on the WAN.
- Microsoft® Lync Server 2010 works with the TLS transport type, while the Verizon Business SIP trunk works on the SIP over UDP transport type.
- Microsoft® Lync Server 2010 supports G.711a/ulaw, while Verizon Business SIP Trunk supports also G.729 coder type.
- Support for call forwarding
- Support for early media handling

The AudioCodes **Mediant 1000B** E-SBC media gateway, which enables smooth integration for supporting all the required functionalities, was used to implement this solution.

The Mediant 1000 E-SBC is a networking E-SBC device that combines multiple service functions, such as a Media Gateway, Session Border Controller (SBC), Data Router and Firewall, LAN switch, WAN access, Stand Alone Survivability (SAS) and an integrated general-purpose server. The E-SBC device offers enhanced dialing plans and voice routing capabilities along with SIP-to-SIP mediation, allowing enterprises to implement SIP Trunking services (IP-to-IP call routing) and IP-based Unified Communications, as well as flexible PSTN and legacy PBX connectivity.

A similar configuration is implemented in the case where the customer uses the Microsoft® Office Communication Server 2007 R2.

Figure 2-1 below illustrates the above example scenario.



#### Figure 2-1: Microsoft Lync and Verizon Business SIP Trunking



**Reader's Notes** 

### 3 Configuring Lync Server 2010

This section describes how to configure the Lync Server 2010 to operate with the Mediant 1000 E-SBC. This section describes the following procedures:

- Configuring the Mediant 1000 E-SBC as a 'PSTN Gateway'. See Section 3.1 on page 13.
- 2. Associating the 'PSTN Gateway' with the Mediation Server. See Section 3.2 on page 18.
- **3.** Configuring a 'Route' to utilize the SIP trunk connected to the Mediant 1000 E-SBC. See Section 3.3 on page 24.



**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 Mediant 1000 E-SBC as a 'PSTN Gateway'

This section describes how to configure the Mediant 1000 E-SBC as a PSTN Gateway.

To configure the Mediant 1000 E-SBC as a PSTN Gateway and associating it with the Mediation Server:

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



#### Figure 3-1: Starting the Lync Server Topology Builder

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The following screen is displayed:

Figure 3-2: Topology Builder Options

🔛 Topology Builder 🛛 🗙
Welcome to Topology Builder. Select the source of the Lync Server 2010 (RC) topology document.
Download Topology from existing deployment Retrieve a copy of the current topology from the Central Management Store database and save it as a local file. Use this option if you are editing an existing deployment.
Open Topology from a local file Open an existing Topology Builder file. Use this option if you have work in progress or if you have exported a topology from Planning Tool.
New Topology Create a blank topology and save it to a local file. Use this option for defining new deployments from scratch.
OK Cancel

 Choose 'Download Topology from the existing deployment and click OK. You are prompted to save the Topology which you have downloaded.

Figure 3-3: Save Topology



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

The Topology Builder screen with the topology downloaded is displayed.

Kunc Server 2010 (RC), Topology Builder		
Eile Action View Help		
Lync Server 2010 (RC)		Actions
🛨 🔃 Interop	SIP domain	Lync Server 2010 (RC) 🔺
	Default STP domains Occur14 local	🔢 New Central Site
	Additional supported SIP Not configured	Edit Properties
	domains:	New Topology
		Open Topology
	Simple URLs	Download Topology
		Save a copy of Topology
	Phone access URLs: Active Simple URL	Publish Topology
	https://dialin.Ocsw14.local	Install Database
	Meeting URLs: Active Simple URL SIP domain	Merge 2007 or 2007 R2 T
	https://meet.Ocsw14.local Ocsw14.local	Remove Deployment
	Administrative access Not configured URL:	View
		👔 Help
	Central Management Server	-
	Central Management fe-ocsw14.ocsw14.local (Interop) Server:	
,		,

Figure 3-4: Downloaded Topology

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



見) upg Controls 2010 (DC) Topology Puildor		
File Action View Help		
Lync Server 2010 (RC)	The properties for this item are upavailable for editing.	Actions
🖃 🔃 Interop		PSTN gateways
Enterprise Edition Front End peoels		New IP/PSTN Gateway
🕀 🧰 Director pools		
A/V Conferencing pools      SOL stores		View
Mediation pools      DETM esteur		M Leih
Monitoring		
Archiving S     Topology	•	
+ Get Edge pools	•	
	T	
1		
	2010/DC	, , D-
😼 scare   🧑 📾 📼 -   🙀 Lyne Serve	C 2010 (KL	🔄 🕪 🕓

#### Figure 3-5: New IP/PSTN Gateway

#### Figure 3-6: Define New IP/PSTN Gateway

Define New IP/PSTN Gateway	×
Gateway F <u>O</u> DN or IP Address *	
E-SBC.OCSW14.local	
Listening port for IP/PSTN gateway: *	
5067	
Sip Transport Protocol: O ICP O TLS	
Help OK Cancel	

5. Enter the FQDN of the Mediant 1000 E-SBC (i.e. 'E-SBC.OCSW14.local') and click **OK**.

Note that the listening port for the Gateway is '5067' and the transport type is 'TLS'.

The Mediant 1000 E-SBC is now added as a 'PSTN Gateway'.

🔀 Lync Server 2010 (RC), Topology Builder						
Elle Action Yiew Help						
< 🔿 🖄 📰  🖬						
by Lync Server 2010 (RC)	DETHIC I		Ac	tions		
Interop     Shandard Edition Event End Services	PSTN Gateway	4	E-:	SBC.OCSW14.local		
Standard Edition Front End pools				Edit Properties		
Director pools	Gateway FQDN or IP Address:	E-SBC.OCSW14.local		Topology 🕨		
A/V Conferencing pools      Sol stores	Listening port:	5067		View		
	SIP Transport Protocol:	TLS		Delete		
Mediation pools     PSTN cohemans	Alternate media IP	Not configured				
	address:	Met annual de		Help		
	Mediation Server	Not associated				
E-SBC. OCSW14.local						
Archiving Servers						
Edge pools     Tructed application servers						
Trusted application servers     E    Branch sites						
J						

### Figure 3-7: PSTN Gateway

# AudioCodes

### 3.2 Associating the 'PSTN Gateway' with the Mediation Server

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

### > To associate the PSTN Gateway with the Mediation Server:

1. Right-click on the **Mediation Server** to use with the E-SBC (i.e. Mediation2.OCSW14.local) and choose **Edit Properties**.

Kunc Server 2010 (RC), Topology Builder						
File Action View Help						
🗢 🔿 🔰 🖬 🛿 🖬						
🎝 Lync Server 2010 (RC)				Actions		
Interop     Charlend Edition Succe End Community	General			Mediation2.ocsw14.local		
Gandard Edition Front End Servers     Edition Front End pools     Director pools	FQDN:	Mediation2.ocsw14.local		New Server		
A/V Conferencing pools	Associations			Tanalani		
Get stores      File stores	Edge pool (for media):	Not associated		Topology •		
E image of the second secon	Note: To view the federation	route, use the site property page		View •		
	Note: To view the rederation	riote, use the site property page.		💢 Delete		
	Next hop selection		•			
Med     Topology     E-S8     Monitorir     View     Archiving     Delete	Next hop pool:	fe-ocsw14.ocsw14.local (Interop)				
	Mediation Server PSTN gate	eway	<b>^</b>			
	TLS listening port:	5067				
	TCP listening port:	Not configured				
	PSTN Gateways:	Default Gateway Site				
Edit the properties for this pool.						
7Start   🏈 🚋 🔲 🛛 🕅 🎇 Lync Server 2010 (RC						

#### Figure 3-8: Associating Mediation Server with PSTN Gateway

The following screen is displayed:

🔡 Edit Properties		_ 🗆 X
General	Note: To view or change the federation route, use the site property page.	<b>^</b>
PSTN gateway	Next hop selection	•
	Next hop pool:	
	fe-ocsw14.ocsw14.local Interop	
	Mediation Server PSTN gateway	.
		-
	Enable TCP port	
	The following gateways are not associated with any Mediation Server. Click Add to associate them with this Mediation Server.	
	Gateway Site	
	Med-gw.ocsw14.local     Interop       E-SBC.OCSW14.local     Interop	
	The following gateways are associated with this mediation server. Click New to define a new gateway and add it to the list.	.
	Gateway Site	
	New Remove	
	Make Default	
		•
Help	ОК	Cancel

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

2. In the top-left corner, choose **PSTN gateway** and in the Mediation Server PSTN gateway pane, mark the E-SBC gateway (i.e. 'E-SBC.OCSW14.local') and click **Add** to associate it with this Mediation Server.

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



Edit Properties		
	Note: To view or change the federation route, use the site property page	
General Next hop PSTN gateway	Next hop selection	
	Next has peels	
	fe-ocsw14.ocsw14.local Interop	
	Mediation Server PSTN gateway	
	Listening ports: * TLS: 5067 ICP:	
	<b>Enable TCP port</b>	
	Mediation Server.	
	Gateway     Site       Med-gw.ocsw14.local     Interop	
	The following gateways are associated with this mediation server. Click New to define a new gateway and add it	
	to the list. Click Remove to remove a gateway from the list.	
	Gateway Site	
	Remove	
	Make Default	
		•

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

In the Mediation Server PSTN gateway pane, the E-SBC Gateway that you associated with the Mediation Server is displayed with an adjacent Green  $\checkmark$ .

3. Click OK.

Image: Set of the set o	Klypc Server 2010 (RF) Topology Builder				
Image: Serie 2010 (RC)         Image: Interop         <	File Action View Help				
Image: Server 2010 (RC)       General       Attoms         Image: Standard Edition Front End Servers       FQDN:       Mediation2.occw14.local         Image: Standard Edition Front End Servers       FGDN:       Standard Edition2.occw14.local         Image: Standard Edition Front End Servers       FGDN:       Mediation2.occw14.local (Interop)         Image: Standard Edition Forcer PSTN gateway       Image: Standard Edition Forcer PSTN gateway       Image: Standard Edition2.         Image: Standard Edition Forcer PSTN gateways:       Image: Standard Edition					
B Monitoring Servers B Getape pools B Getap	Linc Server 2010 (RC)     Linc Server 2	General FQDN: Associations Edge pool (for media): Note: To view the federativ Next hop selection Next hop pool:	Mediation2.ocsw14.local Not associated nn route, use the site property page. fe-ocsw14.ocsw14.local (Interop)		Actions       Mediation2.ocsw14.local       New Server       Edit Properties       Topology       View       View       Delete       Image: Properties       Help
E-SBC.OCSW14.local Interop	BMonitoring Servers     JAchiving Servers     JEdge pools     JEdge pools     JTrusted application servers     JB reanch sites	Mediation Server PSTN ga TLS listening port: TCP listening port: PSTN Gateways:	teway 5067 Not configured Default Gateway	Site	
			E-SBC.OCSW14.local	Interop	

Figure 3-11: Media Server PSTN Gateway Association Properties

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

👆 Lyı	c Server 2010 (RC), Topology Builder			
File	Action View Help			
	New Central Site Edit Properties New Topology Open Topology Download Topology Save a copy of Topology As	SIP domain Default SIP domain: Additional supported SIP	Ccsw14.local Not configured	Actions Lync Server 2010 (RC)  New Central Ste Edit Properties
	Publish Topology Install Database Merge 2007 or 2007 R2 Topology Remove Deployment Help	domains: Simple URLs		New Topology Open Topology Download Topology Save a copy of Topology
E E E E	Way gw01.ocsw14.local     Way gw01.ocsw14.local     Wed-gw.ocsw14.local     Execution CSW14.local     Montoring Servers     Archiving Servers     Archiving Servers     Cdge pools     Trusted application servers     Branch sites	Phone access URLs: Meeting URLs: Administrative access URL:	Active         Simple URL           Image: Active         https://deln.Ocsw14.local           Active         Simple URL         SIP domain           Image: Active         Simple URL         Ocsw14.local           Mot configured         Ocsw14.local         Ocsw14.local	Publish Topology Install Database Merge 2007 or 2007 R2 T Remove Deployment View
		Central Management Serv Central Management Server:	rer  fe-ocsw14.ocsw14.local (Interop)	
Publish	topology to the Central Management Store.			
🛛 Sta	rt 🛛 🏉 🚠 💻 🛛 🛛 🔀 Lync Server	2010 (RC		🖑 Խ 📑 🕼

Figure 3-12: Publishing Topology

# AudioCodes

The Publish Topology screen is displayed.



Publish Topology	×
Publish the topology	
<ul> <li>In order for Lync Server 2010 (RC) to correctly route messages in your deployment, you must publish your topology. Before you publish the topology, ensure that the following tasks have been completed:</li> <li>A validation check on the root node did not return any errors.</li> <li>A file share has been created for all file stores that you have configured in this topology.</li> <li>All simple URLs have been defined.</li> <li>For Enterprise Edition Front End pools and for Monitoring Servers and Archiving Servers: All SQL stores are installed and accessible remotely; firewall exceptions for remote access to SQL Server are configured.</li> <li>For a single Standard Edition server: The task "Prepare first Standard Edition server" was run.</li> <li>You are currently logged on as a SQL administrator, for example, as a member of the SQL sysadmin role.</li> <li>If you are removing a Front End pool, all users, common area phones, analog devices, application contact objects, and conference directories have been removed from the pool.</li> </ul>	
Help Back Next Cancel	

#### 5. Click Next.

The Topology Builder attempts to publish your topology.



Pu	iblish Topology	×
	Publishing in progress	
	Please wait while Topology Builder tries to publish your topology.	
	Publishing topology	_
	Succeeded	
	Downloading topology	
	Succeeded	
	Downloading global simple URL settings.	
	Succeeded	
	Enabling topology	•
	<u>B</u> ack <u>N</u> ext	Cancel

Wait until the publish topology process has ended successfully.



Publish Topology				×
Publishing wizard complete				
Your topology was successfully published.				
Step	Status			
Publishing topology	Success			View Logs
Downloading topology	Success			
<ul> <li>Downloading global simple URL settings.</li> </ul>	Success			
<ul> <li>Enabling topology</li> </ul>	Success			
To close the wizard, click Finish.				
Help		Back	<u>F</u> inish	Cancel

6. Click Finish.

# 

### 3.3 Configuring the 'Route' on the Lync Server 2010

This section describes how to configure a 'Route' on the Lync server and associate it with the E-SBC PSTN gateway.

### To configure the 'route' on the Lync server:

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



Figure 3-16: Lync Server Control Panel

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



#### Figure 3-17: Lync Server Credentials

The CSCP Home page is displayed.

Home         Susers         Topology         I Mand Presence         Voice Routing         Voice Reatures         Response Groups         Conferencing         Clients         External User Access         Monitoring and Archiving         Security         Network Configuration		Lync Server 2010			Administrator   Sign o 4.0.745
<ul> <li>Users</li> <li>User Information</li> <li>Velcome, Administrator</li> <li>View your roles</li> <li>Voice Routing</li> <li>Voice Features</li> <li>Response Groups</li> <li>Conferencing</li> <li>Clients</li> <li>External User Access</li> <li>Monitoring and Archiving</li> <li>Security</li> <li>Network Configuration</li> </ul>	<b>•</b>	Home			
Welcome, Administrator         Voice Routing         Voice Routing         Voice Features         Response Groups         Conferencing         Clients         External User         Access         Monitoring and Archiving         Security         Network Configuration	8 	Users	User Information	Resources	
Voice Features         Response Groups         Conferencing         Clients         External User Access         Monitoring and Archiving         Security         Network Configuration	• •	IM and Presence Voice Routing	Welcome, Administrator  View your roles	Getting Started First Run Checklist Using Control Panel	
Conferencing Clients External User Access Monitoring and Archiving Security Network Configuration	•	Voice Features Response Groups	Top Actions	Getting Help	
External User Access   Monitoring and Archiving   Security   Network Configuration	)	Conferencing	Enable users for Lync Server Edit or move users View topology status	Online Documentation on TechNet Lib Lync Server Management Shell Lync Server Management Shell Script I	rary Library
Monitoring and Archiving Security Network Configuration	1	External User Access	View Monitoring Server reports	Community	
Security Network Configuration		Monitoring and Archiving		Blogs	
Network Configuration		Security			
		Network Configuration			

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



Ni 🐺	crosoft Lync Server 2010	Control Panel								
	Microsoft"								Administrat	or   Sign out
Ø.	Lync Server 20	10								4.0.7457.0
	Home	Dial Plan	Voice Policy	Route P	STN Usage	e Trunk Cont	īguration Test Voice F	touting		
33	Users	Create	voice routing tes	st case inforn	nation					*
N P	Topology IM and Presence						٩			
ণ্ড	Voice Routing	+ Nev	v 🔻 🧪 Edit 🔻	Action <b>▼</b>	Commit	•				0
S	Voice Features	Na	me		Scope	State	Normalization rules	Description		
23	Response Groups	C	Global		Global	Committed	1			
Ð	Conferencing	Ð	LAB2		Site	Committed	1			
r 🖶	Clients	Đ	SBA-LAB3		Site	Committed	2			
	External User Access	•								
-	Monitoring and Archiving									
•	Security									
9	Network Configuration									
J										

### Figure 3-19: Voice Routing Option

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

icrosoft Lync Server 2010 (	Control Panel				
	0				Administrator   9
Lync Server 201	.0				4.
Home	Dial Plan Voice Policy Route	PSTN Usage Tru	nk Configuration	Test Voice Routing	
Users	Create voice routing test case i	information			
Topology					
IM and Presence				<b>م</b>	
Voice Routing	♣ New	e up 🕹 Move down	Action V Com	mit 🔻	
Voice Features	Name	State	PSTN usage	Pattern to match	
Response Groups	LocalRoute	Committed	Internal, Local	^((\+1[0-9]{10}))(\+972)(\+011))	
Conferencing					
Clients					
External User Access					
Monitoring and Archiving					
Security					
Network Configuration					

#### Figure 3-20: Route Option

- 5. In the content area toolbar, click
- 6. In the Build a Pattern to Match pane, fill in a Name for this route (i.e SIP Trunk Route) and a Pattern to Match for the phone numbers you wish this route to handle. In this example, the pattern to match is '\*', which implies "to match all numbers".
- 7. Click Add.

# AudioCodes

Ni 🐺	crosoft Lync Server 2010 C	iontrol Panel	
-	Microsoft*		Administrator   Sign out
Ø.	Lync Server 2010	0	4.0.7457.0
	Home	Dial Plan         Voice Policy         Route         PSTN Usage         Trunk Configuration         Test Voice Routing	
33	Users	Create voice routing test case information	*
м	Topology		
Ģ	IM and Presence	New Voice Route	
ণ্ড	Voice Routing	V OK Cancel	0
C	Voice Features	Name:*	
23	Response Groups	SIP Trunk Route	
Ð	Conferencing	Description:	
6	Clients	Build a Pattern to Match	
4	External User Access	Add the starting digits that you want this route to handle, or create the expression manually by clicking Edit.	
	Monitoring and Archiving	Starting digits for numbers that you want to allow:       Type a valid number and then click Add.   Add	
1	Security	Exceptions	
9	Network Configuration	Remove	
		Match this pattern:*	
		Edit Reset 🥐	-

Figure 3-21: Adding New Voice Route

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

A list of all the deployed Gateways is displayed.

rvice Sit tnGateway:gw01.ocsw14.local Int toGateway:SRA	te terop
tryce Sit	te terop
tnGateway:gw01.ocsw14.local Int	terop
toGateway/SBA. ow OCSW14 local M	
uloateway.soA-gw.oc.sw14.local Mi.	1К
tnGateway:SBA-gw2.OCSW14.local M.	2К
tnGateway:sba-gw03.ocsw14.local M2	2K-Test
tnGateway:ofer-gw.ocsw14.local of	er
tnGateway:GW-LAB2.ocsw14.local LA	82
tnGateway:GW-LAB3.ocsw14.local SB	A-LAB3
tnGateway:Med-gw.ocsw14.local Int	terop
tnGateway:E-SBC.OCSW14.local Int	terop

Figure 3-22: List of Deployed Gateways

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

Figure	3-23:	Selecting	the	E-SBC	Gateway
--------	-------	-----------	-----	-------	---------

Ni 🌄	crosoft Lync Server 2010	Control Panel	
	Microsoft		Administrator   Sign out
	Lync Server 20	10	4.0.7457.0
	Home	Dial Plan Voice Policy Route PSTN Usage Trunk Configuration Test Voice Routing	
33	Users	Create voice routing test case information	*
×	Topology		
Ģ	IM and Presence	New Voice Route	
હ	Voice Routing	V Cancel	0
C	Voice Features	Edit Reset ?	<b>^</b>
23	Response Groups		
₽	Conferencing	Suppress caller ID	
6	Clients	Alternate caller ID:	
盐	External User	Associated gateways:	
	Access	PstnGateway:E-SBC.OCSW14.local Add	
-	Monitoring and Archiving	Remove	
4	Security		
9	Network	Associated PSTN Usages	
	comgulation	Select Remove 👚 🦊	
		PSTN usage record Associated voice policies	
			· · ·

# 

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

Figure	3-24:	Associating	<b>PSTN</b>	Usage t	o E-SBC	Gateway
	•	/ 1000 0 1 a 111 1 g		Jougs .		Catomay

	Microsoft*		Administrator   Sign ou
Ø.	Lync Server 20	10	4.0.7457
<b>^</b>	Home	Dial Plan Voice Policy Route PSTN Usage Trunk Configuration Test Voice Routing	
33	Users	Create voice routing test case information	*
М	Topology		
Ģ	IM and Presence	New Voice Route	
હ	Voice Routing	V Cancel	0
C	Voice Features	Associated gateways:	<b>^</b>
23	Response Groups	Remove	
Ð	Conferencing		
Ē.	Clients		
該	External User	Associated PSTN Usages	
	Access	Select Remove 👚 🐥	
	Monitoring and Archiving	PSTN usage record Associated voice policies	
Д	Security	Internal 😨 Global	
	Network		
Ŷ	Configuration	Long Distance 💮 Global	
		Translated number to test:	
		Go	
			▼

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

88 Mie	rosoft Lync Server 2010	Control Panel							
2	Lync Server 201	.0							Administrator   Sign out 4.0.7457.0
	Home	Dial Plan	Voice Policy	Route PSTI	N Usage Tru	ink Configuration	Test Voice Routing		
33	Users	Create v	oice routing tes	st case informat	ion				*
×	Topology								
Ģ	IM and Presence						<mark>م</mark>		
¢	Voice Routing	🗣 New	🧪 Edit 🔻 🥤	🏫 Move up 🚽	Move down	Action <b>▼</b> Com	mit 🔻		0
S	Voice Features	Nam	e	S	tate	PSTN usage	Pattern to match		
23	Response Groups	Local	Route	C	ommitted	Internal, Local	^((\+1[0-9]{10}))(\+97	2) (\+011))	
Ð	Conferencing	SIP T	runk Route	1	Uncommittee	d Internal, Local	*		
P	Clients								
ii ja	External User Access								
	Monitoring and Archiving								
1	Security								
9	Network Configuration								

Figure 3-25: Confirmation of New Voice Route

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

Figure 3-26: Committing Voice Routes

3 Mic	crosoft Lync Server 2010	trol Panel	
×		Administrator	Sign o
æ.	Lync Server 20		4.0.745
•	Home	Dial Plan Voice Policy Route PSTN Usage Trunk Configuration Test Voice Routing	
3	Users	Create voice routing test case information	≽
X	Topology		
P	IM and Presence	٩	
3	Voice Routing		?
6	Voice Features	Name State PSTN usa Review uncommitted changes	
2	Response Groups	LocalRoute Committed Internal, L Commit all [1])	
Þ	Conferencing	SIP Trunk Route Uncommitted Internal, L Cancel selected changes	
3	Clients	Cancel all uncommitted changes	
	External User Access		
	Monitoring and Archiving		
1	Security		
2	Network Configuration		



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

	guration Setting:	5		
outes				*
Identity	Action	New value (pattern to match)	Old value (pattern to match)	
SIP Trunk Route	Modified		1	

Figure 3-27: Uncommitted Voice Configuration Settings

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



Microsoft Lync Server 2010 Control Panel	0	×
O Successfully published voice routing configuration.		
	Close	

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

Figure 3-29: Voice Routing Screen Displaying Committed Routes

🌄 Mic	rosoft Lync Server 2010	ontrol Panel			
	Lvnc Server 20	)			Administrator   Sign out
		Dial Plan Voice Policy Route PSTN	Usage Trunk Configuration	Test Voice Routing	4.0.7457.0
	Home	Create voice reuting text case informatic			× .
33	Users	Create voice routing test case informatic	n		
м	Topology				
Ð	IM and Presence				
ę	Voice Routing	🕈 New 🧪 Edit 🔻 👚 Move up 👃	Move down Action <b>T</b> Cor	nmit 🔻	0
C	Voice Features	Name Sta	te PSTN usage	Pattern to match	
23	Response Groups	LocalRoute Co	nmitted Internal, Local.	^((\+1[0-9]{10}) (\+972) (\+011))	
Ð	Conferencing	SIP Trunk Route Co	nmitted Internal, Local.	*	
	Clients				
B	External User Access				
	Monitoring and Archiving				
	Security				
9	Network Configuration				



**Reader's Notes** 

### 4 **Configuring AudioCodes Gateway**

This section provides step-by-step procedures for configuring AudioCodes' gateway. These procedures are based on the setup example described in Section 2 on page 11.

The steps for configuring the gateway can be summarized as follows:

- **Step 1**: Configure IP Addresses (see Section 4.1 on page 36).
- **Step 2**: Enable the SBC Capabilities (see Section 4.2 on page 42).
- **Step 3**: Configure the Number of Media Channels (see Section 4.3 on page 43).
- **Step 4**: Configure the Proxy Sets (see Section 4.4 on page 44).
- **Step 5**: Configure the IP Groups (see Section 4.5 on page 46)
- **Step 6**: Define SIP TLS Transport Type (see Section 4.6 on page 48).
- Step 7: Configure Secure Real-Time Transport Protocol (SRTP) (see Section 4.7 on page 56).
- **Step 8**: Configure the Voice Coders (see Section 4.8 on page 57)
- Step 9: Define Silence Suppression and Comfort Noise (see Section 4.8.1 on page 58).
- **Step 10**: Configure IP Profile Settings (see Section 4.9 on page 59).
- **Step 11:** Configure IP Profile for Call Forwarding (see Section 4.9.1 on page 61).
- **Step 12**: Configure IP-to-IP Routing Setup (see Section 4.10 on page 63).
- **Step 13**: Configure Number Manipulation (see Section 4.11 on page 66).
- **Step 14**: Configuring SIP General Parameters (see Section 4.12 on page 72).
- **Step 15**: Defining Reasons for Alternative Routing (see Section 4.13 on page 75).

The procedures described in this section are performed using the gateway's Webbased management tool (i.e., embedded Web server). Before you begin configuring the gateway, ensure that the Web interface's Navigation tree is in full menu display mode (i.e., the **Full** option on the Navigation bar is selected), as displayed below:

Figure 4-1: Web Interface Showing Basic/Full Navigation Tree Display



### 4.1 Step 1: Configure IP Addresses

This section describes how to configure the IP addresses of the E-SBC devices' LAN and WAN interfaces.

### 4.1.1 LAN and WAN Interface Separation

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

### Notes:



- The VoIP and Management interface must be in the same subnet as the data-routing interface as shown in the figure below.
- When operating with both VoIP and data-routing functionalities, it is recommended to define the Default Gateway IP address for the VoIP network interface in the same subnet and with the same VLAN ID as the IP address for the data-routing LAN interface as shown in the figure below.




#### 4.1.1.1 Configuring the LAN IP Addresses

This section describes how to assign the LAN IP addresses.

### To assign a LAN VoIP and Management IP address using the Web interface:

- Open the 'IP Settings' page (Configuration tab > VoIP menu > Network submenu > IP Settings).
- Select the 'Index' radio button corresponding to the Application Type "OAMP + Media + Control (i.e., VoIP and management interface), and then click Edit.
- **3.** Configure the new IP address and prefix length so that it corresponds to your network IP addressing scheme (e.g., 10.8.6.86).
- 4. Configure additional IP interfaces, if required.

Index	Application Type	IP Addres	IS Length	Gateway	VLAN ID	Interface Name	Primary DNS Server IP Address	Secondary DN IP Addre
0 🖲	OAMP + Media + Control 🔹	10.8.6.86	16	10.8.6.85	1	Voice	0.0.0.0	0.0.0.0
							<u></u>	
			*					
			WAN Inter	ace Name	Not Co	onfigured	-	

### Figure 4-3: Multiple Interface Table Page

- 5. Click **Apply**, and then **Done** to apply and validate settings. If validation fails, the E-SBC device does not reboot.
- 6. Save your settings to flash memory and reset the E-SBC device.

#### To define the E-SBC device's LAN data-routing IP address:

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

Figure 4-4: Connections Page
------------------------------

Name	Status	Action
LAN switch	1 Ports Connected	1
WAN Ethernet	Cable Disconnected	1
LAN switch VLAN 1	Connected	1 22
ew Connection		4

- 3. Click the Edit A icon corresponding to the "LAN Switch VLAN 1" connection, and then click the Settings tab.
- **4.** In the 'IP Address' and 'Subnet Mask' fields, enter the required IP address (e.g., 10.8.6.85) and subnet respectively, and then click **OK**.

(	
Device Name: Status:	eth0.1 Connected
Schedule:	Always 👻
Network:	LAN 🗸
Connection Type:	Ethernet
Physical Address:	00:90:8f:22:2e:31
MTU:	Automatic 💙 1500
Underlying Connection:	LAN switch
Internet Protocol	Use the Following IP Address
IP Address:	10 .8 .6 .85
Subnet Mask:	255 .0 .0

#### Figure 4-5: Defining LAN Data-Routing IP Address

#### 4.1.1.2 Assigning WAN IP Addresses

This section describes how to assign the WAN IP addresses.

#### To assign a WAN IP address:

- 1. Cable the E-SBC device to the WAN network (i.e., ADSL or Cable modem), using the WAN port.
- 2. Access the E-SBC device's Web interface with the Voice and Management IP address.
- Access the 'Settings' page (Configuration tab > Data menu > WAN Access > Settings tab).

Name: Status:	Conn	Ethernet			
IP Address:	195	.189	. 192	.133	
Subnet Mask:	255	255	255	128	
Default Gateway:	63	.97	. 104	. 62	
Primary DNS Server:	0	.0	.0	.0	
Secondary DNS Server:	0	.0	.0	.0	
Click here for Advanced Settin	igs				

#### Figure 4-6: Configuring the WAN IP Address

**4.** From the 'Connection Type' drop-down list, select the required connection type for the WAN, and then configure the IP address (e.g., 195.189.192.133).

#### To assign a WAN interface for VoIP traffic:

- 1. Select the WAN interface.
- Open the 'Multiple Interface Table' page (Configuration tab > VoIP menu > Network submenu > IP Settings).

#### Figure 4-7: Selecting WAN Interface for VoIP Traffic in Multiple Interface Table Page

Index	Application Type	IP Addre	ss Prefo	Gateway	VLAN ID	Interface Name	Primary DNS Server I Address	P Secondary D IP Addr
0 0	OAMP + Media + Control +	10.8.6.86	16	10.8.6.85	1	Voice	0.0.0.0	0.0.0.0
						12.		
					_			
			WAN Inter	face Name	WAN Ethernet		- 0	

- 3. From the 'WAN Interface Name' drop-down list, select the WAN interface for VoIP traffic.
- 4. Click Done, and then reset the E-SBC device for your setting to take effect.

# 4.1.2 Single LAN Interface

This section describes how to configure IP addresses when a single LAN interface is used to connect to the Verizon Business SIP Trunk. In this configuration, the internal data-routing capabilities of the E-SBC device are not used as shown in the figure below. As a consequence, you must disable the internal data-routing interface as described in the procedure below.

#### Figure 4-8: Single LAN Interface





Note: When operating in LAN VoIP-only mode, do not use the E-SBC device's WAN port.

#### To operate the E-SBC device as a LAN VoIP gateway only:

- 1. Disconnect the network cable from the WAN port and then connect one of the E-SBC device's LAN ports to the network.
- 2. Disable or remove the data-routing IP network interface:
  - Access the 'Connections' page (Configuration tab > Data menu > Data System > Connections).
  - Delete the "LAN Switch VLAN 1" connection by clicking the corresponding
     Remove Statution, and then clicking OK to confirm deletion.

#### Figure 4-9: Removing Data-Routing Connection Interface

Name	Status	Action
🔈 LAN switch	1 Ports Connected	1
WAN Ethernet	Cable Disconnected	1
LAN switch VLAN 1	Connected	\ 🗶 🔫
ew Connection		

- 3. Configure VoIP IP network interfaces in the 'Multiple Interface' table (Configuration tab > VoIP menu > Network > IP Settings).
  - In the 'Multiple Interface' table, define a single IP network interface for application types "OAMP + Media + Control".

#### Figure 4-10: Multiple Interface Table

I	ndex	Application Type	IP Address	Prefix Length	Gateway	VLAN ID	Interface Name
C	$\circ$	OAMP + Media + Control	10.8.6.86	16	10.8.0.1	1	Voice

• Click **OK** to save settings.

# 4.2 Step 2: Enable the SIP IP2IP Application

This step describes how to enable the gateway's SIP IP2IP application.

#### To enable the SIP IP2IP application:

 Open the 'Application Enabling' page (Configuration tab > VolP menu > Applications Enabling > Applications Enabling).

Figure	4-11:	Applicatio	n Enabling

▼		
🗲 Enable SAS	Disable	•
🗲 Enable SBC Application	Disable	<b>-</b> 🖉
🗲 Enable IP2IP Application	Enable	2

 From the 'Enable IP2IP Application' drop-down list, select "Enable". Reset with BURN to FLASH is required.



**Note:** To enable the IP2IP capabilities on the AudioCodes gateway, your gateway must be loaded with the feature key that includes the IP2IP feature and also the E-SBC device must be running SIP version 6.2 or later.

# 4.3 Step 3: Configure the Number of Media Channels

In order to reform the coder transcoding, you need to define DSP channels. The number of media channels represents the number of digital signaling processors (DSP) channels that the gateway allocates to IP-to-IP calls (the remaining DSP channels can be used for PSTN calls). Two IP media channels are used per IP-to-IP call. The maximum number of media channels available on the gateway is 120 (i.e., up to 60 IP-to-IP calls).

#### To configure the number of media channels:

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

					Basic Paramet
•					
4	Number of Media Channels	2	120		2
4	Voice Streaming	$\smile$	Disable	*	
	NetAnn Announcement ID		annc		
	MSCML ID		ivr		
	Transcoding ID		trans		
•	Conference				
	Conference ID		conf		
	Beep on Conference		Enable	*	
	Enable Conference DTMF Clamping		Enable	*	
	Enable Conference DTMF Reporting		Disable	*	

#### Figure 4-12: IP Media Channels Settings

2. In the 'Number of Media Channels parameter, enter "120" to enable up to 60 IPto-IP calls with transcoding. Click **Apply New Value**.

# 4.4 Step 4: Configure the Proxy Sets

This step describes how to configure the Proxy Sets. The Proxy Sets represent the IP addresses (or FQDN), which are required for communicating with the entities in the network:

- Proxy Set ID #1 is assigned with the IP address of Verizon Business SIP Trunk.
- Proxy Set ID #2 is assigned with the IP address of Lync Mediation server.

These Proxy Sets are later assigned to IP Groups (see Section 4.5 on page 46).

#### To configure proxy sets:

- Open the 'Proxy Sets Table' page (Configuration tab > VolP menu > Control Network > Proxy Sets Table).
- 2. Configure the Proxy Set for Verizon Business SIP Trunk:

From the 'Proxy Set ID' drop-down list, select "1".

- a. In the 'Proxy Address' column, enter the IP address or FQDN of the Verizon Business SIP Trunk and the listening port of the Verizon Business SIP Trunk.
- **b.** From the 'Transport Type' drop-down list, corresponding to the IP address entered above, select "UDP".

#### Figure 4-13: Proxy Set ID 1 for Verizon Business SIP Trunk

▼ Pro	XV Se	et ID	1	•
	~y 50	. 10		•
		Prox	y Address	Transport Type
	1	63.97.104.62:50	)72	UDP 👻 🗲
	2			
	3			
	4			-
	5	7		•

3. Configure the Proxy Set for the Lync Mediation Server:

From the 'Proxy Set ID' drop-down list, select "2".

- **a.** In the 'Proxy Address' column, enter the IP address or the FQDN and the listening port of the Lync Mediation Server.
- **b.** From the 'Transport Type' drop-down list corresponding to the IP address entered above, select "TCP" or "TLS" depending on the deployed Mediation Server Transport Type.

-							
Proxy S	Se	t ID	2			•	
[		Proxy	Address	Transı Typ	port e		
	1	10.64.2.23:5068		TLS	•		-3
	2				•		
	3				•		
	4				•		
1	5				•		

Figure 4-14: Proxy Set ID 2 for Lync Mediation Server

# AudioCodes

# 4.5 Step 5: Configure the IP Groups

This step describes how to create IP groups. Each IP group represents a SIP entity in the gateway's network. You need to create IP groups for the following entities:

- 1. Verizon Business SIP Trunk
- 2. Lync Server 2010 Mediation Server

These IP groups are later used by the IP2IP application for routing calls.

### > To configure IP Groups:

- Open the 'IP Group Table' page (Configuration tab > VoIP menu > Control Network> IP Group Table).
- 2. Define IP Group **#1** for the Verizon ITSP as follows:
  - a. IP Group Index '1'
  - b. Type: "SERVER"
  - c. Description: arbitrary name. (e.g., "Verizon")
  - **d.** Proxy Set ID: "1" (represents the IP address, configured in Section 4.4 on page 44, for communicating with this IP Group).
  - e. SIP Group Name: The SIP Request-URI host name used in INVITE messages sent to the IP Group, or the host name in the From header of INVITE messages received from the IP Group. Enter the WAN IP address.
  - f. IP Profile ID: "1": Different IP profile is used for the Verizon Business SIP Trunk and the Mediation Server. See Section 4.9 on page 59.

Index	<u>2a</u>	▶ 1
<ul> <li>Common Parameters</li> </ul>		
Туре	<b>2b</b> —	SERVER -
Description		Verizion
Proxy Set ID	<mark>2d</mark> —	▶ 1
SIP Group Name	2e	▶ 174.46.0.189
Contact User		
SRD		0
Media Realm		
IP Profile ID	<mark>2</mark> f	▶ 1
<ul> <li>Gateway Parameters</li> </ul>		
Always Use Route Table		No 👻
Routing Mode		Not Configured -
ore e l' M l		0. I I

#### Figure 4-15: IP Group 1 Table

Index	3a	▶ 2	-
Common Parameters			
Туре	<mark>3b</mark> —	► SERVER	•
Description		OCS	<b>←</b> 3c
Proxy Set ID	3d	→ 2	•
SIP Group Name	<mark>3e</mark>	E-SBC.ocsw14.local	2
Contact User	<u> </u>		
SRD		0	
Media Realm			•
IP Profile ID	<mark>(3f)</mark>	→ 2	•
Gateway Parameters			
Always Use Route Table		No	<b>•</b>
Routing Mode		Not Configured	•
SIP Re-Routing Mode		Standard	•
Enable Survivability		Disable	-

Figure 4-16: IP Group 2 Table Page

- 3. Define IP Group #2 for Mediation Server as follows:
  - a. Select IP Group Index '2':
  - b. Type: "SERVER"
  - c. Description: <Free Description> (e.g., "Lync Mediation Server")
  - d. Proxy Set ID: "2"
  - e. SIP Group Name: The SIP Request-URI host name used in INVITE messages sent to the IP Group, or the host name in the From header of INVITE messages received from the IP Group. Enter the Gateway Name.
  - f. IP Profile ID: "2" (see Section 4.9 on page 59).

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# 4.6 Step 6: Define SIP TLS Transport Type

This section describes how to configure AudioCodes gateways for implementing a TLS connection with the Mediation Server.

### 4.6.1 Configure NTP and DNS Server

The procedure below describes how to configure the NTP Server IP address or FQDN and the Domain Name System (DNS) servers.

#### > To configure NTP servers:

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

#### Figure 4-17: Application Settings

▼ NTP Settings	
NTP Server IP Address	10.198.210.62
NTP UTC Offset	Hours: 0 Minutes: 0
NTP Updated Interval	Hours: 24 Minutes: 0

2. Define the NTP server's IP address so that it corresponds to your network environment.

#### > To configure DNS servers:

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

#### Figure 4-18:DNS Settings

🗲 DNS Primary Server IP	10.198.210.16	<b>←_2</b>
🗲 DNS Secondary Server IP	157.54.14.178	

- In the 'DNS Primary Server IP' and 'DNS Secondary Server IP' fields, set the primary and secondary DNS server's IP addresses with the IP address of your DNS server.
- 3. Click the **Submit** button to save your changes.
- 4. Save the changes to flash memory, by clicking **Burn** button on the toolbar. The changes take effect after the restart operation.

## 4.6.2 Configure the Gateway Name

The procedure below describes how to configure the Media Gateway name.

### **>** To configure the Media Gateway name:

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

Proxy	& Registration		
			Basic Parameter List 🔺
	Enable Registration	Disable	× ^
	Registration Time	180	
	Re-registration Timing [%]	50	
	Registration Retry Time	30	
	Registration Time Threshold	0	
	Re-register On INVITE Failure	Disable	•
	ReRegister On Connection Failure	Disable	•
	Gateway Name	E-SBC.ocsw14.local	<b>←</b> <u></u> <u></u> <u></u>
	Gateway Registration Name		
	DNS Query Type	A-Record	✓
	Proxy DNS Query Type	A-Record	✓
	Subscription Mode	Per Endpoint	✓
	Number of RTX Before Hot-Swap	3	
	Use Gateway Name for OPTIONS	No	A 1 - 1     A 1      A 1
	User Name		
	Password	Default_Passwd	
	Register Sul	Un-Register	

#### Figure 4-19: Proxy & Registration

2. In the 'Gateway Name' field, assign a unique FQDN name to the Media Gateway within the domain, for example, 'E-SBC.OCSW14.local'.

# 4.6.3 Configure a Certificate

The procedure below describes how to exchange a certificate with the Microsoft Certificate Authority. The certificate is used by the E-SBC device to authenticate the connection with the management PC (the PC used to manage the E-SBC using the embedded Web server).

### > To configure a certificate:

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

Certificate Signing Request         Subject Name E-SBC.OCSW14.local         Generate CSR         Generate CSR         Copy the certificate signing request and send it to your Certification Authority for signing.        BEGIN CERTIFICATE REQUEST         MILEXDCBxgTBADAdMRswCQYDVQQDBxJFLVNCQySPQ1NXMTQubC9jYWwwg28wDQYJ         KoZIhvcNAQBBBQADgYOAMIGJAoGBAKCIAX6d0e1kK002nzjHurw0Py/D2v211Vje         /4Hpw80mmsYnrD4NoqkuwHKVThwbeQZDV23B3uQ04yHhGYnxwdvrgwK20eend5d         wwq866YHksHvwHj/AhjmQWukWQYPcv4WjkCaGexSJXnF60VgWIgYZDc006zbq         SMm/41n1AgMBAAGgADANBgkqhkiG9w0BAQQFAA0BgQCEkBTD0Ja00uk47+dmuqkU         Sp4/vKgAq94jWulty+s2dbA2IqCydv8XwdBJ5vo76j0L5V81Sh2e4+3gLf2JCStL         B3c7sSTuV6bvhU0gRwvpgvKq14bejiSCQ5hiwotcySyghiAStkLkH76KmacvzzB         yU4gpvug0eMktT23B5Hygg=        END CERTIFICATE REQUEST         Press the button "Generate self-signed" to create a self-signed certificate using the subject name provided above.	Certificates
Press the button "Generate self-signed" to create a self-signed certificate using the subject name provided above.	Certificate Signing Request Subject Name E-SBC.OCSW14.local Generate CSR Generate CSR Copy the certificate signing request and send it to your Certification Authority for signing. BEGIN CERTIFICATE REQUEST MIIEXDCBxgIBADAdMRswGQYDVQQDExJFLVWCQy5PQ1NXMTQubG9jYWwwg28wDQYJ Ko2IhvcNAQEBBQADgYOAMICJAoGBAKCiAX6d0eItkKC0ZnzjHurw0Py/D2w21IVje /4Hpw80mmsYnrD4NoqkuwHkvThwbeQ2Dv23B3uQ04yuHnGYnzw4wrqwKz0exnd5d wxwq566YMk8hYwJHj/AhjnMQWmKWQrPcW4WjkCaGsX5JXnF60VgWIgY2Dt006zbq 8Mm/4In1AgMBAAGgADANBgkqhkiG9w0BAQQFAA0BgQCEkETD0Ja00uk47+dmuqkU Sp4/vKgAq94jVwIcy+s2dbA2IqCydv8XwdBJSvo76j0L5V8ISn2e++3gLf2JCstL B3t7s5TuV6bvhU0gKmvpgvKq14bejiSG1Q5hiwotcy5yqhiAS+k1kM76KmacvzzB vU4qpvud0eMktT23B5HMgq==
Press the button "Generate self-signed" to create a self-signed certificate using the subject name provided above.	·····END CERTIFICATE REQUEST·····
Important the lease of the two strates and the two the dovice will be out at convice	Press the button "Generate self-signed" to create a self-signed certificate using the subject name provided above.

Figure 4-20: Certificates Page

- 2. In the 'Subject Name' field, enter the Media Gateway name as configured in the previous section (see Section 4.6.2 on page 49), and then click **Generate CSR**; a Certificate request is generated.
- **3.** Copy the CSR (from the line "----BEGIN CERTIFICATE" to "END CERTIFICATE REQUEST----") to a text file (such as Notepad), and then save it to a folder on your PC as *certreq.txt*.

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

🗿 Microsoft Certificate Services - Microsoft Internet Explorer	_ 8 ×
Eile Edit <u>Vi</u> ew Favorites <u>T</u> ools <u>H</u> elp	
🔇 Back + 🛞 - 🖹 🙆 🏠 🔎 Search 📌 Favorites 🔗 🍰 - 😓 🗔 📙 鑬 🖄	
Address 🕘 http://10.15.4.201/certsrv/	🔁 Go 🛛 Links 👌
Microsoft Certificate Services Demolab	<u>Home</u>
Welcome	
	y your I request,
You can also use this Web site to download a certificate authority (CA) certificate, certificate chain, or certificate revocation list (CR view the status of a pending request.	L), or to
For more information about Certificate Services, see Certificate Services Documentation.	
Select a task: Request a certificate	
View the status of a pending certificate request	
Download a CA certificate chain, of CRL	
	r
	ernet

Figure 4-21: Microsoft Certificate Services Web Page

5. Click the link Request a Certificate.

#### Figure 4-22: Request a Certificate Page

Microsoft Certificate Services - Microsoft Internet Explorer	
<u>Eile Edit Vi</u> ew Favorites <u>T</u> ools <u>H</u> elp	🥂 🔍 🖉
😋 Back 🔹 📀 👻 😰 🏠 🔎 Search 🤺 Favorites 🤣 😥 🌭 💹 👻 🔛 🐇	
Address 🕘 http://10.15.4.201/certsrv/certrgus.asp	🗾 🔁 Go 🛛 Links 🎽
Microsoft Certificate Services Demolab	Home
Request a Certificate	
Select the certificate type:	
Web Browser Certificate	
E-Mail Protection Certificate	
Or submit an advanced certificate request	
or, submit an <u>advanced certificate reguest</u> .	
	🕘 Internet

# AudioCodes

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

Figure 4-23: Advanced Certificate Request Page



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

•		
🕘 Microsoft Active	Directory Certificate Services - Microsoft Internet Explorer	X
File Edit View F	avorites Tools Help	7
G Back 🔹 🕥	🔹 😰 🏠 🔎 Search 🤺 Favorites 🤣 🔗 - چ 🚍 🚭 🦓	
Address 🕘 http://10.	15.4.50/certsrv/certrqxt.asp 💽 🕤 Lincs	»
Microsoft Active	Directory Certificate Services OCSR2-CA Home	^
Submit a Certi	ficate Request or Renewal Request	
To submit a sav PKCS #7 renev box.	red request to the CA, paste a base-64-encoded CMC or PKCS #10 certificate request or val request generated by an external source (such as a Web server) in the Saved Request	
Saved Request:		
Base-64-encoded certificate request (CMC or PKCS #10 or PKCS #7):	Q1NSMi5sb2NhbAwTTONTUjJCYWRtaW5pc3RyYXRv CSqGSIb3DQEBAQUABIGAFdvCIkp5YmpE9MxrP2y/ rZwY/e+b1+3fF1AE/i8DCO2hUSOrViZoVjisLIzz W38f2bBOHIFbNAbMUuLhr/bmGaDpsmhtTASZNEH1 END NFW CERTIFICATE REQUEST	
Certificate Temp	late:	
	Web Server	
Additional Attribu	ites:	
Attributes:		
	Submit >	~
🕘 Done	🥥 Internet	

#### Figure 4-24: Submit a Certificate Request or Renewal Request Page

- 8. Open the *certreq.txt* file that you created and saved (see Step 3), and then copy its contents to the 'Base64 Encoded Certificate Request' text box.
- 9. Select "Web Server" from the **Certificate Template** drop-down box.
- 10. Click Submit.
- 11. Choose the 'Base 64' encoding option, and then click the link **Download CA** certificate.
- **12.** Save the file as 'gateway.cer' in a folder on your PC.
- 13. Navigate to the certificate Server http://<Certificate Server>/CertSrv.
- 14. Click the link Download a CA Certificate, Certificate Chain or CRL.



Figure 4-25: Download a CA Certificate, Certificate Chai	II, OF CRL Page
Microsoft Certificate Services - Microsoft Internet Explorer	X
Eile Edit View Favorites Tools Help	
🛛 🚱 Back 🔹 💮 🖌 📓 🏠 🔎 Search 🤺 Favorites 🔣 😥 🗞 💹 👻 📒 🎎 🖄	
Address 🕘 http://10.15.4.201/certsrv/certcarc.asp	🔽 🄁 Go 🛛 Links 🎽
<b>Microsoft</b> Certificate Services Demolah	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	
	-
Cone	

#### E. Download a CA Cartificate, Cartificate Chain

- **15.** Under the Encoding method group, perform the following:
  - а. Select the 'Base 64' encoding method option.
  - b. Click the link Download CA certificate.
- 16. Save the file as 'certroot.cer' in a folder on your PC.
- **17.** Navigate back to the 'Certificates' page.
- 18. In the 'Certificates' page, in the 'Server Certificate' field, click Browse and select the 'Gateway.cer' certificate file that you saved on your local disk (see Step 12), and then click Send File to upload the certificate.

 In the 'Certificates' page, in the 'Trusted Root Certificate Store' field, click Browse and select the 'Certroot.cer' certificate file that you saved on your local disk (see Step 16), and then click Send File to upload the certificate.

	Generate self-signed
	Certificate Files
Send "Server Certificate" I	file from your computer to the device
C:\Gateway.cer	Browse Send File (18)
Send "Trusted Root Certifi	cate Store" file from your computer to the device
C:\Certroot.cer	Browse Send file (19)
Send "Private Key" file fro	m your computer to the device
	Browse Send file

Figure 4-26: Certificates Page

20. Save (burn) the Media Gateway configuration and reset the Media Gateway, using the Web interface's 'Maintenance Actions' page (On the Navigation bar, click the **Management** tab, and then in the Navigation tree, select the **Management Configuration** menu, and then choose the **Maintenance** Actions).

# 4.7 Step 7: Configure Secure Real-Time Transport Protocol (SRTP)

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

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

If you choose to configure the Mediation Server to use SRTP (Required or Optional), you need to configure the Media Gateway to operate in the same manner.

### To configure the media security:

 Open the 'Media Security' page (Configuration tab > Media menu > Media Security).

dia Security	
	Basic Parameter List
🗲 Media Security	Enable 💽 🔶 🚺
Media Security Behavior	Preferable - Single media 💉 🛶 🕣
Disable Authentication On Transmitted RTP Packets	0
Disable Encryption On Transmitted RTP Packets	0
Disable Encryption On Transmitted RTCP Packets	0
✓ SRTP Setting	
🗲 SRTP Offered Suites	0 - All
Master Key Identifier (MKI) Size	1 4
	Subr

#### Figure 4-27: Media Security Page

- 2. From the 'Media Security' drop-down list, select "Enable", to enable SRTP.
- 3. From the 'Media Security Behavior' drop-down list, select:
  - "Mandatory" if Mediation Server is configured to SRTP Required
  - "Preferable-Single media" if Mediation Server is configured to SRTP Optional.
- 4. In the 'Master Key Identifier (MKI) Size' field, enter '1'.
- 5. Click Submit.
- 6. Save (burn) the E-SBC configuration and reset the Gateway.

**Note:** In order to set the 'Media Security Behavior' to the IP Profile of the Mediation Server, see the IP Profile Settings (see Section 4.9 on page 59).

# 4.8 **Step 8: Configure the Voice Coders**

Since the Mediation Server support only G.711 a/ulaw voice coders, while the ITSP SIP trunk requires G.729 coder, you need to configure two coder tables for each entity. The Coder table is associated with an IP Profile (see Section 4.9 on page 59) which is associated with the IP Group (see Section 4.5 on page 46).

#### ➢ To configure Coder Table for Mediation server:

 Open the 'Coders Table' page (Configuration tab > VolP menu > Coders and Profiles > Coder Group Settings).

▼ Coder Group ID			1 🗸 🔶 2		
Coder Name		Packetization Time	Rate	Payload Type	Silence Suppression
G.711A-law	•	20 🔻	64 💌	8	Enable
G.711U-law	•	20 👻	64 👻	0	Enable

#### Figure 4-28: Coder Group Table - Mediation Server

- 2. From the 'Coder Group ID drop-down list, select '1'.
- 3. Select the G.711A-law and G.711U-law coders, as shown in the figure above.
- 4. From 'Silence Suppression' drop-down list, select 'Enable' as shown in the figure above.

> To configure Coder Table for Verizon Business SIP Trunk:

 Open the 'Coders Table' page (Configuration tab > VolP menu > Coders and Profiles > Coder Group Settings).

#### Figure 4-29: Coder Group Table - ITSP SIP Trunk

ler Group Settings					
•					
Coder Group ID	)		2 🖌 🔶	2	
Coder Na	ime	Packetization Time	e Rate	Payload Type	Silence Suppressio
Coder Na G.729	ame 💉	Packetization Time	e Rate	Payload Type	Silence Suppressic
Coder Na G.729	ame	Packetization Time	e Rate	Payload Type 18	Silence Suppressio
G.729	ame	Packetization Tim	e Rate	Payload Type 18	Silence Suppressio

- 2. From the 'Coder Group ID drop-down list, select '2'.
- **3.** Select the G.729 coder, as shown in the figure above.

## 4.8.1 Step 9: Define Silence Suppression and Comfort Noise

Overall voice quality has been significantly improved for the Microsoft Lync 2010 environment. These improvements include suppression of typing noise during calls and improved generation of "comfort noise," which reduces hissing and smoothes over the discontinuous flow of audio packets. You may need to change the E-SBC Silence Suppression and Comfort Noise parameters to achieve this goal. Note that the Echo canceller is enabled by default.

#### To configure silence suppression parameters:

- 1. Silence Suppression is configured per coder type. (See Section 4.8 on page 57 above to enable Silence Suppression per coder.)
- Open the 'RTP/RTCP Settings' page (Configuration tab > Media menu > RTP / RTCP Settings).

RTP/RTC	CP Settings		
			Basic ParameterList 🔺
-	<ul> <li>General Settings</li> </ul>		<u> </u>
	Dynamic Jitter Buffer Minimum Delay	10	
	Dynamic Jitter Buffer Optimization Factor	10	
	RTP Redundancy Depth	0	
	Packing Factor	1	
	Basic RTP Packet Interval	Default 💌	
	RTP Directional Control	RTPTxRx 💌	
	RFC 2833 TX Payload Type	96	
	RFC 2833 RX Payload Type	96	
	RFC 2198 Payload Type	104	
	Fax Bypass Payload Type	102	
	Enable RFC 3389 CN Payload Type	Enable 💌	
4	RTP Base UDP Port	6000	
	Comfort Noise Generation Negotiation	Enable 💌	· • · · · · · · · · · · · · · · · · · ·
	Analog Signal Transport Type	Disable 🗸 🗸	
	Remote RTP Base UDP Port	0	
	RTP Multiplexing Local UDP Port	0	
			Submit

#### Figure 4-30: RTP/RTCP Settings Page

- **3.** From the 'Comfort Noise Generation Negotiation' drop-down list, select 'Enable'. This enables negotiation and usage of Comfort Noise (CN).
- 4. Click Submit.

# 4.9 Step 10: Configure IP Profile Settings

This section describes how to configure the IP Profile Settings.

### To configure IP Profile for Verizon :

 Open the 'IP Profile Settings' page (Configuration tab > VoIP menu > Coders and Profiles > IP Profile Settings).

•				
	Profile ID	(2)→	1	•
	Profile Name			
•	Common Parameters			
-	Gateway Parameters			
	Fax Signaling Method		No Fax	•
	Play Ringback Tone to IP		Don't Play	•
	Enable Early Media		Enable	•
	Copy Destination Number to Redirect Number		Disable	•
	Media Security Behavior		Disable	•
	CNG Detector Mode		Disable	•
	Modems Transport Type		Enable Bypass	•
	NSE Mode		Disable	•
	Number of Calls Limit		-1	
	Progress Indicator to IP		Not Configured	•
	Profile Preference		1	•
	Coder Group	3→	Coder Group 2	•
	Remote RTP Base UDP Port		0	
ĺ	First Tx DTMF Option		RFC 2833	•
	Second Tx DTMF Option			•
	Declare RFC 2833 in SDP		Yes	•
	Add IE In SETUP			
	AMD Sensitivity Parameter Suit		0	
	AMD Sensitivity Level		8	
	AMD Max Greeting Time		300	
	AMD Max Post Silence Greeting Time		400	Í
	Enable Hold		Enable	•

#### Figure 4-31: IP Profile Page-Verizon Server

- 2. From the 'Profile ID' drop-down list, select '1'.
- **3.** From the 'Coder Group' drop-down list, select 'Coder Group 2'.

#### > To configure IP Profile for Mediation Server:

 Open the 'IP Profile Settings' page (Configuration tab > VoIP menu > Coders and Profiles > IP Profile Settings).

#### Figure 4-32: IP Profile Page-Mediation Server

Profile ID	2→2	
Profile Name	Lync	
Common Parameters		
Gateway Parameters		
Fax Signaling Method	No Fax	
Play Ringback Tone to IP	Don't Play	
Enable Early Media	Enable	
Copy Destination Number to Redirect Number	Disable	
Media Security Behavior	3 -> Mandatory	
CNG Detector Mode	Disable	
Modems Transport Type	Enable Bypass	
NSE Mode	Disable	
Number of Calls Limit	-1	
Progress Indicator to IP	P1 = 8	
Profile Preference	1	
Coder Group	4 → Coder Group 1	
Remote RTP Base UDP Port	0	
First Tx DTMF Option	RFC 2833	
Second Tx DTMF Option		
Declare RFC 2833 in SDP	Yes	-
Add IE In SETUP		
AMD Sensitivity Parameter Suit	0	
AMD Sensitivity Level	8	
AMD Max Greeting Time	300	
AMD Max Post Silence Greeting Time	400	
Enable Hold	Enable	

- 2. From the 'Profile ID' drop-down list, select '2'.
- **3.** From the 'Media Security Behavior' drop-down list, select one of the following options:
  - "Mandatory" if Mediation Server is configured to SRTP Required
  - "Preferable-Single media" if Mediation Server is configured to SRTP Optional.
  - "Disable" if the Mediation Server is configured to SRTP disabled.
- 4. From the 'Coder Group' drop-down list, select 'Coder Group 1'.

### 4.9.1 Step 11: Configure IP Profile for Call Forwarding

One of the challenges with the integration of the Microsoft Lync 2010 server and the Verizon Business SIP Trunk is the implementation of call forwarding. Since the Microsoft Lync client forwards the call back to the SIP Trunk, it does not provide any information in the forwarded INVITE (such as Diversion header) informing that this call has been forwarded. Consequently, it is necessary to configure a special IP Profile that adds the diversion header toward the SIP trunk in the event of a call forwarding scenario.

This profile is later associated to the routing table in the event of a call forwarding scenario.

#### > To configure IP Profile for call forwarding:

 Open the 'IP Profile Settings' page (Configuration tab > VoIP menu > Coders and Profiles > IP Profile Settings).

#### Figure 4-33: IP Profile Settings for Call Forwarding "numbers"

Profile ID	(2)→	3	-	*
Profile Name				
Common Parameters				
Fax Signaling Method		No Fax	<b>•</b>	Ε
Play Ringback Tone to IP		Don't Play	•	
Enable Early Media	~	Enable	•	
Copy Destination Number to Redirect Number	(3)→	Before Manipulation	•	
Media Security Behavior		Disable	-	
CNG Detector Mode		Disable	<b>•</b>	
Modems Transport Type		Enable Bypass	-	
NSE Mode		Disable	-	
Number of Calls Limit		-1		
Progress Indicator to IP		Not Configured	-	
Profile Preference		1	-	
Coder Group	(4)→	Coder Group 2	•	-

- 2. From the 'Profile ID' drop-down list, select '3'.
- 3. From the 'Copy Destination Number to Redirect Number' drop-down list, select 'Before Manipulation'; this parameter adds the Diversion Header to the INVITE in event of a call forwarding scenario.
- 4. From the 'Coder Group' drop-down list, select 'Coder Group 2'.

- Open the 'Admin" page, by appending the case-sensitive suffix 'AdminPage' to the Media Gateway's IP address in your Web browser's URL field (e.g., http://10.15.4.15/AdminPage).
- 6. On the left pane, click *ini* Parameters.

#### Figure 4-34: Output Window

Image	Parameter Name: Enter Value: USESIPURIFORDIVERSIONHEADER	Apply New Value
Load to Device		
<i>ini</i> Parameters	Output Window	
Back to Main	Parameter Name: USESIPURIFORDIVERSIONHEADER The Value is invalid: Parameter Current Value: 1 Parameter Description:Use Tel uri or Sip uri for Diversion header	•

- 7. In the 'Parameter Name' field, enter the parameter USESIPURIFORDIVERSIONHEADER. In the 'Enter Value' field, enter "1".
- 8. Click Apply New Value.

# 4.10 Step 12: Configure IP-to-IP Routing Setup

The E-SBC's IP-to-IP call routing capabilities is performed in two stages:

- 1. Inbound IP Routing: Recognizes the received call as an IP-to-IP call, based on the call's source IP address. This stage is configured in the 'Inbound IP Routing Table'
- 2. Outbound IP Routing: Once recognized as an IP-to-IP call in the first stage (see above), the call is routed to the appropriate destination (i.e., IP address). This stage is configured in the 'Outbound IP Routing Table'.

### 4.10.1 Configure Inbound IP Routing

This step defines how to configure the E-SBC for routing inbound (i.e., received) IP-to-IP calls.

#### To configure in bound IP routing:

 Open the 'Inbound IP Routing Table' page (Configuration tab > VoIP menu > GW and IP to IP > Routing submenu > IP to Trunk Group Routing).

#### Figure 4-35: Inbound IP Routing Table Page

	<b>v</b>											
		Routing Index		1-10 🔻								
	IP To Tel Routing Mode				Route calls before manipulation 💌							
_								_				
	Dest. Host Prefix	So	ource Host Prefix	Dest. Phone Prefix	Source Pho	one Prefix	Source IP Address	->	Trunk Gro	JD ID	IP Profile ID	Source IPGroup ID
1		ocs20	010.local	*	*		10.64.2.23		-1		3	2
2				*	*		10.64.2.23		-1		2	2
3				*	*		63.97.104.62		-1		1	1
4												

- Index #1 configuration identifies all IP calls received from the Mediation Server in the event of a call forwarding Scenario as IP-to-IP calls and assigns them to the IP Group ID configured for the Mediation server:
  - 'Source Host Prefix: Enter the Lync Front end FQDN in case of call forwarding, the Source host in the incoming INVITE from the Mediation Server is the Lync Front End server FQDN, while for regular calls, the Source host is the Mediation ServerFQDN.
  - 'Dest Phone Prefix': Enter the asterisk (\*) symbol to indicate all destinations.
  - 'Source IP Address': Enter the IP address of Mediation Server.
  - 'Trunk Group ID': Enter "-1" to indicate that these calls are IP-to-IP calls.
  - 'IP Profile ID: Enter '3' to indicate that the IP Profile supports the call forwarding scenario.
  - 'Source IP Group ID': Enter "2" to assign these calls to the IP Group pertaining to the Mediation server.

# 

- Index #2 configuration identifies all IP calls received from the Mediation Server as IP-to-IP calls and assigns them to the IP Group ID configured for the Lync Mediation Server:
  - 'Dest Phone Prefix': Enter the asterisk (\*) symbol to indicate all destinations.
  - 'Source IP Address': Enter the IP address of the Mediation server.
  - 'Trunk Group ID': Enter "-1" to indicate that these calls are IP-to-IP calls.
  - 'IP Profile ID: Enter '2' indicate the IP Profile for Mediation server.
  - 'Source IP Group ID': Enter "2" to assign these calls to the IP Group pertaining to the Mediation server.
- 4. Index #3 configuration identifies all IP calls received from Verizon Business SIP Trunk as IP-to-IP calls and assigns them to the IP Group ID configured for the Verizon Business SIP Trunk:
  - 'Dest Phone Prefix': Enter the asterisk (\*) symbol to indicate all destinations.
  - Source IP Address': Enter the IP address of Verizon Business SIP Trunk.
  - 'Trunk Group ID': Enter "-1" to indicate that these calls are IP-to-IP calls.
  - 'IP Profile ID: Enter '1' indicate the IP Profile for Verizon Business SIP Trunk.
  - 'Source IP Group ID': Enter "1" to assign these calls to the IP Group pertaining to Verizon Business SIP Trunk.

# 4.10.2 Configure Outbound IP Routing

This step defines how to configure the gateway for outbound routing (i.e., sent) IP-to-IP calls.

#### To configure outbound IP routing:

 Open the 'Outbound IP Routing Table' page (Configuration tab > VoIP menu > GW and IP to IP > Routing submenu > Tel to IP Routing).

#### Figure 4-36: Outbound IP Routing Table Page

			•										
			Ro	Routing Index					-				
				To IP Rout	ting Mode			Route ca	Route calls before manipulation 🔻				
	Src. IPGroupID	Src. Host Prefix	Dest Host	Prefix	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Pr	efix >	Dest. IP Address	Port	Transport Type	Dest. IPGroup	
1	1				*	*	*				Not Configured 👻	2	
2	2				*	*	*				Not Configured 👻	1	

- Index #1 defines routing of IP calls to the Lync 2010 Mediation server. All calls received from Source IP Group ID 1 (i.e., from the Verizon Business SIP Trunk) are routed to Destination IP Group ID 2 (i.e., to Lync 2010 Mediation server):
  - 'Source IP Group ID': Select "1" to indicate received (inbound) calls identified as belonging to the IP Group configured for the Verizon Business SIP Trunk.
  - 'Dest Phone Prefix': Enter the asterisk (\*) symbol to indicate all destinations.
  - 'Source Phone Prefix': Enter the asterisk (\*) symbol to indicate all callers.
  - 'Dest IP Group ID': Select "2" to indicate the destination IP Group to where the calls must be sent, i.e., to Lync 2010 Mediation server.
- Index #2 defines the routing of IP calls to the Verizon Business SIP Trunk. All calls received from IP Group ID 2 (i.e., Lync 2010 Mediation server) are routed to Destination IP Group ID 1 (i.e., Verizon Business SIP Trunk):
  - 'Source IP Group ID': Select "2" to indicate received (inbound) calls identified as belonging to the IP Group configured for the Lync 2010 Mediation Server.
  - 'Dest Phone Prefix': Enter the asterisk (\*) symbol to indicate all destinations.
  - 'Source Phone Prefix': Enter the asterisk (\*) symbol to indicate all callers.
  - 'Dest IP Group ID': Select "1" to indicate the destination IP Group to where the calls must be sent, i.e., to the Verizon Business SIP Trunk.

# 4.11 Step 13: Configure Number Manipulation

The Manipulation Tables submenu allows you to configure number manipulation and mapping of NPI/TON to SIP messages. This submenu includes the following options:

- Dest Number IP->Tel. See Section 4.11.1 on page 67.
- Dest Number Tel->IP. See Section 4.11.1 on page 67.
- Source Number IP->Tel. See Section 4.11.2 on page 69.
- Source Number Tel->IP. See Section 4.11.2 on page 69.
- Redirect Number IP->Tel. See Section 4.11.3 on page 71.
- Redirect Number Tel->IP. See Section 4.11.3 on page 71

## 4.11.1 Configure Destination Phone Number Manipulation

This section describes how to configure the destination phone number manipulation.

#### To configure Destination Phone Number Manipulation Table for IP -> Tel Calls Table:

 Open the 'Destination Phone Number Manipulation Table for IP -> Tel calls' page (Configuration tab > VoIP menu > GW and IP to IP > Manipulations submenu > Dest Number IP >Tel).

#### Figure 4-37: Destination Phone Number Manipulation Table for IP -> Tel Calls Page

Index	Destination Prefix	Source Prefix	Source IP Address	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Numł
1 🔘	+1	*	10.64.2.23	2	0			255
2 🔘	+	*	10.64.2.23	1	0			255
3 🔘	1	*	10.64.2.23	1	0			255

- Index #1 defines destination number manipulation of IP calls from Lync 2010 Me4diation server. All calls received from IP address 10.64.2.23 (i.e., from Mediation Server) and the destination number prefix begins with '+1', Remove the '+1' from the Number.
- Index #2 defines destination number manipulation of IP calls from Lync 2010 Me4diation server. All calls received from IP address 10.64.2.23 (i.e., from Mediation Server) and the destination number prefix begins with '+', Remove the '+' from the Number.
- Index #3 defines destination number manipulation of IP calls from Lync 2010 Mediation server. All calls received from IP address 10.64.2.23 (i.e., from Mediation Server) and the destination number prefix begins with '1', Remove the '1' from the Number.

#### To configure Destination Phone Number Manipulation Table for Tel -> IP Calls Table:

 Open the 'Destination Phone Number Manipulation Table for Tel -> IP calls' page (Configuration tab > VoIP menu > GW and IP to IP > Manipulations submenu > Dest Number Tel > IP).

Figure 4-38: Destination Phone Number Manipulation Table for Tel -> IP Calls Page

	Index Source Trunk Group		Source IP Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digit Leave
	0 0	-1	1	+	•	0	0			255
	1 🔘	-1	1	1	*	0	0	+		255
ſ	2 🔘	-1	1	XXXXXXXXXXX#	•	0	0	+1		255

- Index #1 defines destination number manipulation of IP calls from Verizon Business SIP Trunk. All calls received from Source IP Group 1 (i.e., from Verizon Business SIP Trunk) and the destination number prefix begins with '+', do not perform any changes to the number.
- Index #2 defines destination number manipulation of IP calls from Verizon Business SIP Trunk. All calls received from Source IP Group 1 (i.e., from Verizon Business SIP Trunk) and the destination number prefix begins with '1', add the '+' prefix to the number.
- Index #3 defines destination number manipulation of IP calls from Verizon Business SIP Trunk. All calls received from Source IP Group 1 (i.e., from Verizon Business SIP Trunk) and the destination number length is 10 digit number, add the '+1' prefix to the number.

### 4.11.2 Configure Source Phone Number Manipulation

To configure Source Phone Number Manipulation Table for IP -> Tel Calls Table:

 Open the 'Source Phone Number Manipulation Table for IP -> Tel calls' page (Configuration tab > VoIP menu > GW and IP to IP > Manipulations submenu > Source Number IP >Tel).

Inc	dex	Destination Prefix	Source Prefix	Source IP Address	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Numt
2	0		+1	10.64.2.23	2	0			255
3	0	•	*	10.64.2.23	1	0			255
4	0	-	1	10.64.2.23	1	0			255
5	0		anonymous	10.64.2.23	20	0	7192083390		255

#### Figure 4-39: Source Phone Number Manipulation Table for IP -> Tel Calls Page

NDI	TON	Presentation
NPI	TON	Presentation
Not Configured	Not Configured	Allowed
Not Configured	Not Configured	Allowed
Not Configured	Not Configured	Allowed
Not Configured	Not Configured	Not Configured

- Index #1 defines Source number manipulation of IP calls from Lync Mediation Server. All calls received from IP address 10.62.2.23 (i.e., from Lync Mediation Server) and the source number prefix begins with '+1', remove the '+1' from the number.
- Index #2 defines Source number manipulation of IP calls from Lync Mediation Server. All calls received from IP address 10.62.2.23 (i.e., from Lync Mediation Server) and the source number prefix begins with '+', remove the '+' from the number.
- Index #3 defines Source number manipulation of IP calls from Lync Mediation Server. All calls received from IP address 10.62.2.23 (i.e., from Lync Mediation Server) and the source number prefix begins with '1', remove the '1' from the number.
- Index #4 defines Source number manipulation of anonymous calls from Lync Mediation Server. Anonymous calls received from IP address 10.62.2.23 (i.e., from Lync Mediation Server) replace the 'anonymous' caller ID with a well known number i.e. 7192083390. This manipulation is performed to create a well known number in the P-Asserted-Identity header. Without this number, the Verizon Business SIP Trunk rejects the call. See below for the Source Number manipulation Tel->IP manipulation rule that restricts the caller ID to an anonymous call.

#### To configure Source Phone Number Manipulation Table for Tel -> IP Calls Table:

 Open the 'Source Phone Number Manipulation Table for Tel -> IP calls' page (Configuration tab > VoIP menu > GW and IP to IP > Manipulations submenu > Source Number Tel > IP).

#### Figure 4-40: Source Phone Number Manipulation Table for Tel -> IP Calls Page

In	dex	Source Trunk Group	Source IP Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digit Leave
0	0	-1	1	*	+	0	0			255
1	0	-1	1:	÷	1	0	0	+		255
2	0	-1	1	•	200000000000	0	0	+1		255
4	0	-1	2		7192083390	1	0			255
					-					



- Index #1 defines Source number manipulation of IP calls from Verizon Business SIP Trunk. All calls received from Source IP Group 1 (i.e., from Verizon Business SIP Trunk) and the Source number prefix begins with '+', do not perform any changes to the number.
- Index #2 defines Source number manipulation of IP calls from Verizon Business SIP Trunk. All calls received from Source IP Group 1 (i.e., from Verizon Business SIP Trunk) and the Source number prefix begins with '1', Add a '+' as a prefix to the number.
- Index #3 defines Source number manipulation of IP calls from Verizon Business SIP Trunk. All calls received from Source IP Group 1 (i.e., from Verizon Business SIP Trunk) and the Source number length is 10 digit number, add the '+1' prefix to the number.\
- Index #4 defines Source number manipulation of anonymous calls from Verizon Business SIP Trunk. All calls received from Source IP Group 1 (i.e., from Verizon Business SIP Trunk) and the Source number is 7192083390 (which is a well known number that was replace the anonymous caller ID on the source number manipulation IP->Tel above), the presentation should be set to 'restricted'.

# 4.11.3 Configure Redirect Number Manipulation

In the event of a call forwarding scenario, a Diversion header needs to be added to the INVITE towards the Verizon Business SIP Trunk (as configured in Section 4.9.1 on page 61). In this case, the E-SBC copies the Destination number to the Redirect number and adds this number to the Diversion header. In order to have a well known number in the Diversion header (for Verizon Business SIP Trunk), a manipulation rule should be defined to replace the redirect number to a well known number.

#### To configure redirect number Tel -> IP Table:

 Open the 'Redirect Number Tel -> IP' page (Configuration tab > VoIP menu > GW and IP to IP > Manipulations submenu> Redirect Number Tel > IP).

Index	Source Trunk Group	Source IP Group	Destination Prefix	Redirect Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digit Leave
1 🔘	-1	-1	+	+	20	0	7192083390		255

#### Figure 4-41: Redirect Number Tel -> IP Page

• Index **#1** defines redirect number manipulation for the call forwarding scenario.

The redirect number is changed to a well known number i.e. 7192083390.

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# 4.12 Step 14: Configuring SIP General Parameters

This section describes how to configure the SIP general parameters.

#### > To configure general SIP parameters:

 Open the 'SIP General Parameters' page (Configuration tab > VoIP menu > SIP Definitions submenu > General Parameters).

#### Figure 4-42: SIP General Parameters Page "numbers"

•	SIP General			
4	NAT IP Address	174.46.0.189		
	PRACK Mode	Supported	•	
	Channel Select Mode	~	Cyclic Ascending	<b>~</b>
	Enable Early Media		Enable	<b>•</b>
	183 Message Behavior		Progress	•
	Session-Expires Time		0	
	Minimum Session-Expires	90		
	Session Expires Method		Re-INVITE	•
	Asserted Identity Mode 4		Adding PAsserted Identity	•
	Fax Signaling Method	No Fax	•	
	Detect Fax on Answer Tone		Initiate T.38 on Preamble	•
	SIP Transport Type	5→	TLS	<b>•</b>
	SIP UDP Local Port	5060		
	SIP TCP Local Port		5060	
	SIP TLS Local Port	(6)→	5061	
	Enable SIPS	Disable	•	
	Enable TCP Connection Reuse		Enable	•
	TCP Timeout		0	
	SIP Destination Port		5060	
	Use user=phone in SIP URL		Yes	<b>•</b>
	Use user=phone in From Header		No	<b>~</b>
	Use Tel URI for Asserted Identity		Disable	<b>~</b>
	Tel to IP No Answer Timeout		180	
	Enable Remote Party ID		Disable	•
	Add Number Plan and Type to RPI Header		Yes	<b>~</b>
	Enable History-Info Header		Disable	<b>~</b>
	Use Source Number as Display Name		No	•
	Use Display Name as Source Number		No	•
	Enable Contact Restriction		Disable	▼
	Play Ringback Tone to IP		Don't Play	▼
	Play Ringback Tone to Tel	(7)→	Play Local Until Remote Media A	▼
	Use Tgrp information	Disable	<b>▼</b>	
	Enable GRUU	Disable	<b>•</b>	
User-Agent Information				
--	-----------------------			
SDP Session Owner	AudiocodesGW			
Play Busy Tone to Tel	Don't Play 🔹			
Subject				
Multiple Packetization Time Format	None 💌			
Enable Semi-Attended Transfer	Disable 🔹			
3xx Behavior	Forward 👻			
Enable P-Charging Vector	Disable 🔹			
Enable VoiceMail URI	Disable 🔹			
Retry-After Time	0			
Enable P-Associated-URI Header	Disable 💌			
Source Number Preference				
Forking Handling Mode 8	Sequential handling 🔹			
Enable Comfort Tone	Enable 🔹			
Add Trunk Group ID as Prefix to Source	No 👻			
Fake Retry After (9)->	60			
Enable Reason Header	Enable 🔹			

- 2. In the 'NAT IP Ad dress' field, enter the Global (public) IP address of the E-SBC device to enable the static NAT between the E-SBC device and the Internet.
- From the 'Enable Early Media' drop-down list, select 'Enable' to enable early media.
- 4. From the 'Asserted Identity Mode' drop-down list, select 'Adding PAsserted Identity'.
- 5. From the 'SIP Transport Type' drop-down list, select 'TLS' in case the Mediation Server is configured to use TLS transport Type.
- 6. In the 'SIP TLS Local Port' field, enter '5061'; this port is the listening E-SBC port for TLS transport type. This port must match the transmitting port of the Mediation Server.
- 7. From 'Play Ringback Tone to Tel' drop-down list, select 'Play Local Until Remote Media Arrive'. Plays the RBT according to the received media. If a SIP 180 response is received and the voice channel is already open (due to a previous 183 early media response or due to an SDP in the current 180 response), the E-SBC device plays a local RBT if there are no prior received RTP packets. The E-SBC device stops playing the local RBT as soon as it starts receiving RTP packets. At this stage, if the E-SBC device receives additional 18x responses, it does not resume playing the local RBT.
- 8. From the 'Forking Handling Mode' drop-down list, select 'Sequential handling'; this parameter determines whether18x with SDP is received. In this case, the E-SBC device opens a voice stream according to the received SDP. The E-SBC device re-opens the stream according to subsequently received 18x responses with SDP.
- **9.** In the 'Fake Retry After' field, enter '60' sec. This parameter determines whether the E-SBC device, upon receipt of a SIP 503 response without a Retry-After header, behaves as if the 503 response included a Retry-After header and with the period (in seconds) specified by this parameter.

## AudioCodes

- Open the 'Admin" page, by appending the case-sensitive suffix 'AdminPage' to the Media Gateway's IP address in your Web browser's URL field (e.g., http://10.15.4.15/AdminPage).
- **11.** On the left pane, click *ini* Parameters.
- 12. In the 'Parameter Name' field, enter the parameter **IGNOREALERTAFTEREARLYMEDIA**. In the 'Enter Value' field, enter "1".
- 13. Click Apply New Value.

#### Figure 4-43: INI file Output Window

Image Load to Device	Parameter Name IGNOREALER TAFTEREARLYMEDIA	Enter Value.	Apply New Value
vn/ Parameters Back to Main	Outpu	It Window	
	Parameter DescriptioniInterwork of Ale	rt from ISIN to SIP	

### 4.13 Step 15: Defining Reasons for Alternative Routing

A 503 SIP response from the Mediation Server to an INVITE must cause the Media Gateway to perform a failover. For this event to occur, you need to configure the Reasons for Alternative Routing for Tel-to-IP calls to be a 503 SIP response.

#### To define SIP Reason for Alternative Routing:

 Open the 'Reasons for Alternative Routing' page (Configuration tab > VoIP menu > GW and IP to IP > Routing submenu > Alternative Routing Reasons).

#### Figure 4-44: Reasons for Alternative Routing Page

Reasons	for	Altern	ative	Routing
---------	-----	--------	-------	---------

IP to Tel Reasons	
Reason 1	~
Reason 2	~
Reason 3	~
Reason 4	~
Tel to IP Reasons	
Reason 1	503 💙
Reason 2	~
Reason 3	~
Reason 4	~

- 2. Under the Tel to IP Reasons group, for Reason 1, select '503'.
- 3. Click Submit.
- Open the 'Proxy & Registration' page (Configuration > VoIP > SIP Definitions > Proxy & Registration) and configure the 'Redundant Routing Mode' parameter to 'Proxy' as shown below.

▼	
Use Default Proxy	No 👻
Proxy Name	
Redundancy Mode	Parking -
Proxy IP List Refresh Time	60
Enable Fallback to Routing Table	Disable 👻
Prefer Routing Table	No 👻
Always Use Proxy	Disable 👻
Redundant Routing Mode	Proxy
SIP ReRouting Mode	Standard Mode 👻
Enable Registration	Disable 👻
Registration Time	180
Re-registration Timing [%]	50
Registration Retry Time	30
Registration Time Threshold	0
Re-register On INVITE Failure	Disable 👻
ReRegister On Connection Failure	Disable 👻
Gatewav Name	acow.ocs2010.local

#### Figure 4-45: 'Proxy & Registration' Page

**Reader's Notes** 



Mediant 1000 E-SBC

# **Configuration Note**

### Connecting Verizon Business SIP Trunk to Microsoft® Lync Server Using AudioCodes Mediant 1000 E-SBC



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