Enterprise Session Border Controllers

Multi-Service Business Gateways

VoIP Media Gateways

Configuration Note Connecting Microsoft® Lync™ and Verizon SIP-Trunk using AudioCodes[®] Mediant[™] E-SBC Series









Version 6.4

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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 E-SBC series, which includes the Mediant 800 E-SBC, Mediant 1000 E-SBC and Mediant 3000 E-SBC, as was performed during the certification/compliance process.

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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 *E-SBC device* refers to the E-SBC family (Mediant 800 Media Gateway and E-SBC, Mediant 800 MSBG, Mediant 1000 MSBG, Mediant 1000B Media Gateway and E-SBC, Mediant 3000 Media Gateway and E-SBC).

1 Introduction

This Configuration Note shows how to set up AudioCodes' networking device to operate with Verizon SIP Trunking Service and the Microsoft Lync Communication platform.

1.1 Intended Audience

The Note is intended as a supporting reference for certification/compliance testing and may be utilized as a configuration example by installation engineers or AudioCodes and Verizon partners installing and configuring Verizon SIP Trunking service and the Microsoft Lync Communication platform for VoIP calls using AudioCodes networking devices for superior voice quality services.

1.2 About AudioCodes Networking Devices

AudioCodes' family of Enterprise Session Border Controllers (E-SBC) enables reliable connectivity and security between enterprises and Service Providers' VoIP networks.

The E-SBC family (Mediant 800 E-SBC, Mediant 1000 E-SBC and Mediant 3000 E-SBC) provides perimeter defense as a way of protecting companies from malicious VoIP attacks; mediation for allowing the connection of any PBX and/or IP-PBX to any Service Provider; and Service Assurance for service quality and manageability.

AudioCodes E-SBC is available as an integrated solution running on top of the fieldproven Mediant Media Gateway and Multi-Service Business Gateway platforms or as a software-only solution for deployment on 3rd party hardware.

The native implementation of E-SBC functions on the AudioCodes Mediant Media Gateways and Multi-Service Business Gateways provides a host of additional capabilities that are not possible with standalone SBC appliances, such as VoIP mediation, PSTN Access, data routing, WAN access, data security, survivability, and third party value-added services applications. This enables enterprises to utilize the advantages of converged networks and eliminate the need for standalone appliances.

Mediant 800 E-SBC is based on field-proven VoIP and network services with a native host processor, allowing the creation of purpose-built multiservice appliances, providing smooth connectivity to cloud services, with integrated quality of service, SLA monitoring, security and manageability. The native implementation of SBC provides a host of additional capabilities that are not possible with standalone SBC appliances, such as VoIP mediation, PSTN access survivability, and third party value-added services applications. This enables enterprises to utilize the advantages of converged networks and eliminate the need for standalone appliances.



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



Note: The scope of this document does not serve as a replacement or alternative to the AudioCodes *Release Notes*, *User's Manual*, *Installation Guide* or *Product Reference Manual* with regards to specific aspects for connecting to the Microsoft Lync environment. Specific implementation requirements should be implemented in accordance with your organization's topology requirements and utilization requirements. For further capability understanding, see the AudioCodes *Release Notes*, *User's Manual*, *Hardware Installation Guide* and *Product Reference Manual*.

2 **Components Information**

2.1 AudioCodes E-SBC Version

Table 2-1: AudioCodes E-SBC Version

Gateway Vendor	AudioCodes
Model	The E-SBC family (Mediant 800 Media Gateway and E-SBC, Mediant 800 MSBG, Mediant 1000 MSBG, Mediant 1000B Media Gateway and E-SBC, Mediant 3000 Media Gateway and E-SBC).
Software Version	SIP_6.40A.034.005
Interface Type	SIP/IP
VoIP Protocol	SIP
Additional Notes	With Digital Trunk Module or MPM module for DSP resources MSFT – Enable working with Microsoft Lync IPSEC, MediaEncryption, StrongEncryption and EncryptControlProtocol – Enable working with TLS transport type SBC = 120 – Enable the IP-to-IP feature up to 60 sessions

2.2 AudioCodes Gateway Version

Table 2-2: AudioCodes Gateway Version

Gateway Vendor	AudioCodes
Model	AudioCodes MP-11x series.
Software Version	SIP_6.20A.060.001
Interface Type	SIP/IP
VoIP Protocol	SIP
Additional Notes	Analog device FAX gateway

2.3 Verizon SIP Trunking Version

Table 2-3: Verizon Version

Gateway Vendor	Verizon
Models	
Software Version	
VoIP Protocol	SIP
Additional Notes	None

2.4 Microsoft Lync Server 2010 Version

Table 2-4: Microsoft Lync Version

Gateway Vendor	Microsoft
Models	Microsoft Lync Server 2010
Software Version	RTM: Release 2010 4.0.7577.184
VoIP Protocol	SIP
Additional Notes	None

2.5 Mediation Server Version

Table 2-4: Microsoft Lync Version

Gateway Vendor	Microsoft
Models	Mediation Server
Software Version	RTM: Release 2010 4.0.7577.183
VoIP Protocol	SIP
Additional Notes	None

2.6 Microsoft Lync Client Version

Table 2-5: Microsoft Lync Version

Gateway Vendor	Microsoft
Models	Microsoft Lync Client
Software Version	RTM: Release 2010 4.0.7577.4051
VoIP Protocol	SIP
Additional Notes	None

2.7 Lync Server 2010 Specification

For the purposes of this testing, the following server specification was used to host the Microsoft Lync environment in a fully functioning Lync Server 2010 deployment, including Active Directory and DNS.

The Domain Controller (DC) and the Lync standard edition where installed on the OSN3 server hosted on the Mediant 1000B MSBG chassis.

OSN3 Specifications

Parameter	Specification
CPU	Intel® Core [™] 2 Duo 1.5 GHz processors L7400 with Intel 3100 Chipset (64-bit)
RAM Memory	2 G DDR2 with ECC
Hard Drives	Up to 2 hard drives (HDMX modules) 160GB
Bus/Chipset	64 Bit
L2 Cache	2 M
Interfaces	 Gigabit Ethernet USB 2.0 via Connection Module RS-232 COM
Operating System	Windows Server 2008 standard

2.8 Topology

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

- An enterprise has a deployed 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 SIP Trunking service.
- AudioCodes Session Border Controller (SBC) is used to manage the connection between the Enterprise LAN and the Verizon SIP trunk.

The "session" refers to the real-time voice session using IP SIP signaling protocol. The "border" refers to the IP to IP network border between the Microsoft Lync network in the Enterprise LAN and the Verizon SIP trunk in the public network.

Figure 2-1 below illustrates the interoperability topology between the Lync Server 2010 LAN and the Verizon SIP Trunking site.

The setup requirements are characterized as follows:

- While the Lync Server 2010 environment is located on the Enterprise's Local Area Network (LAN), the Verizon SIP Trunks are located on the WAN.
- The internal data routing capabilities of the Mediant 1000 MSBG device are used. Consequently, a separate WAN interface is configured in the LAN.
- Lync Server 2010 works with the TLS transport type, while the Verizon SIP trunk works on the SIP over UDP transport type.
- Transcoding support: Lync Server 2010 supports G.711A-law and G.711U-law coders, while the Verizon SIP Trunk also supports the G.729 coder type.
- ATA FAX Support via Analog Media Gateway.
- The AudioCodes E-SBC device can support B2BUA transcoding or nontranscoding interactions. The E-SBC device can also be configured to transparently forward fax calls without intervention (e.g. transcoding).



Support for early media handling:

2.8.1 Port Topology

This section summarizes the SIP source and destination port settings in this configuration scenario in support of call processing.

Proxy Set	Transport Protocol	Destination (Sending) Port
Microsoft Lync	TCP or	5068
	TLS	5067
Verizon	UDP	5111
Fax Gateway	UDP	5060

The Destination Port for Microsoft Lync, Verizon and Fax Gateways are configured in the Proxy Sets table. See Section 4.4 on page 61.

Transport Protocol	Source (Listening) Port
UDP	5060
ТСР	5068
TLS	5067

The Source Ports for the Microsoft Lync and Verizon proxy sets are configured in the SIP General Settings screen for the E-SBC device. See Section 4.14 on page 87.

The Source Port for the Fax Gateway is configured in the SIP General Settings for the ATA device. See Section A.4 on page 106.

3 Configuring Lync Server 2010

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

- 1. Configuring the device as an 'IP/PSTN Gateway'. See Section 3.1 on page 19.
- 2. Associating the 'IP/PSTN Gateway' with the Mediation Server. See Section 3.2 on page 23.
- **3.** Configuring a 'Route' to utilize the SIP trunk connected to the E-SBC device. See Section 3.3 on page 28.



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

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

This section describes how to configure the E-SBC device as an IP/PSTN Gateway.

- To configure the Mediant 1000 MSBG as a IP/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, and select Lync Server Topology Builder).





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	

2. Click the **Download Topology from the existing deployment** button and click **OK**; you are prompted to save the Topology which you have downloaded.

Figure 3-3: Save Topology

🌄 Save Topology As				×
Administr	ator 👻 Documents	👻 🔂 Se	earch	2
🖭 Organize 👻 📗 Views	👻 📑 New Folder			0
Favorite Links	Name A	Date modified ▼ 1 10/7/2010 5:53 PM 1 10/12/2010 10:5 1	Гуре	e v Tac 101 KB 101 KB
Computer Documents Pictures				
Music Recently Changed				
Searches				
Folders ^	•			Þ
File <u>n</u> ame: Inter	op2.tbxml			•
Save as <u>t</u> ype: Topol	ogy Builder files (*.tbxml)			•
Hide Folders			Save	Cancel //

3. Enter new File Name and **Save** (this action enables you to rollback from any

changes you make during the installation); the Topology Builder screen with the topology downloaded is displayed.

Figure 3-	4: Down	loaded	Topology
-----------	---------	--------	----------

Lync Server 2010, Topology Builder		
<u>File Action View H</u> elp		
🗢 🔿 🖬 🚺 🖬		
	SIP domain Default SIP domain: Additional supported SIP domains: Simple URLs	FE-Lync.Lync.local Not configured
 → Montoring Servers → Archiving Servers → Edge pools → Edge pools → Trusted application servers → Branch sites 	Phone access URLs: Meeting URLs: Administrative access URL: Central Management Serve	Active Simple URL Active Simple URL SIP domain Active Simple URL SIP domain Active Https://meet.FE-Lync.Lync.local https://admin.Lync.local
	Central Management Server:	FE-Lync.Lync.local (Interop)

4. Expand the Site, right-click the IP/PSTN Gateway and choose New IP/PSTN Gateway.

Figure 3-5: New IP/PSTN Gateway

🔀 Lync Server 2010 (RC), Topology Builder		_ 🗆 🗵
File Action View Help		
🗢 🔿 🔁 🖬 🛛 🖬		
Lync Server 2010 (RC) The properties for this item an	e unavailable for editing.	Actions
E 🔝 Interop		PSTN gateways
Scandard Edition Front End pervers Enterprise Edition Front End pools		New IP/PSTN Gateway
🗉 🧰 Director pools		Topology
A/V Conferencing pools Sol, stores		View b
Mediation pools		Help
PSTN gatev New IP/PSTN Gateway Monitoring		
Archiving Se Topology		
Edge pools		
Instead app Instead a		
- Hop		
		1
🈂 Start 🔰 🏀 🔜 🔢 🧏 Lync Server 2010 (RC		🦉 婉 🗟 🥼

Figure 3-6: Define New IP/PSTN Gateway

Define New IP/PSTN Gateway	>	<
Gateway FODN or IP Address *		
Verizon.Lync.local		
Listening port for IP/PSTN gateway: *		
5067		
Sip Transport Protocol: O ICP		
Help	OK Cancel	

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

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



Kultur Server 2010, Topology Builder			
File Action View Help			
(+ -> 2 💼 🛛 🖬			
Lync Server 2010	PSTN Gateway		
Interop Interop Standard Edition Front End Servers Image: Standard Edition Front End pools Image: Standard Edition Front End pools Image: Standard Edition Front End pools	Gateway FQDN or IP Address:	Verizon.Lync.local	
A/V Conferencing pools	Listening port:	5067	
🕀 🚞 SQL stores	SIP Transport Protocol:	TLS	
	Alternate media IP address:	Not configured	
 PSTN gateways ACGW.lync.local ALE-GW.lync.local ACEG.Lync.local Telenet.Lync.local Verizon.Lync.local Archiving Servers Archiving Servers Edge pools Trusted application servers Branch sites 	Mediation Server	Not associated	

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

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

> To associate the IP/PSTN Gateway with the Mediation Server:

1. Right-click the Mediation server to use with the E-SBC device (i.e. 'FE-Lync.Lync.local') and choose **Edit Properties**.

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

Lync Server 2010, Topology Builder				_ 🗆 ×
Ele Action View Help				
Lync Server 2010	Mediation Server PSTN	gateway		
Standard Edition Front End Servers Enterprise Edition Front End pools Director pools AV Conferencing pools	TLS listening port: TCP listening port: PSTN Gateways:	5067 5068	Category	1 00
SQL stores S		ACEC ALE- ACEC Teler Veriz	Gateway W.lync.local GW.lync.local S.Lync.local net.Lync.local non.Lync.local	Interop Interop Interop Interop Interop



The following screen is displayed:

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

a gatemay	Mediation Server PSTN gateway			3
	Listening ports: * TLS: 5067	TCP: 5068		
	Enable TCP port			
	The TCP port of this Mediation Server	must be enabled because a	TCP gateway depends on it.	
	The following gateways are not accord	stad with any Madiation Ca	nuar Click Add to accordate them with this	Madiation
	Server.	atou with any riculation Ser	res. Calck Add to associate them with this	- Housdon
	Gateway	Site		
	Verizon.Lync.local	Interop	Add	
	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 ga	teway from the list.		
	Gateway	Site	1 الدر	
	Gateway GW-131.Lync.local	Site Interop	New	
	GW-131.Lync.local GW-161.Lync.local	Site Interop Interop	New	

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

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

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

STN gateway	Mediation Server PSTN gateway	-
	Listening ports: * TLS: 5067 TCP: 5068	
	Enable TCP port	
	The TCP port of this Mediation Server must be enabled because a TCP gateway depends on it.	
		11000
	The following gateways are not associated with any Mediation Server. Click Add to associate them with this M Server.	lediation
	Gateway Cite	
	Glickoy	
	Add	
	Add	
	Add The following gateways are associated with this mediation server. Click New to define a new gateway and ad	d it to
	Add The following gateways are associated with this mediation server. Click New to define a new gateway and ad the list. Click Remove to remove a gateway from the list.	d it to
	Add The following gateways are associated with this mediation server. Click New to define a new gateway and ad the list. Click Remove to remove a gateway from the list. Gateway Site	d it to
	Add The following gateways are associated with this mediation server. Click New to define a new gateway and ad the list. Gateway Site Gateway Site GW-161.Lync.local Interop New	d it to
	Add The following gateways are associated with this mediation server. Click New to define a new gateway and ad the list. Click Remove to remove a gateway from the list. Gateway Site Gateway Site Gateway Site Gevi161.Lync.local Interop Verizon.Lync.local Interop Remove	d it to

3. Click OK.

Lync Server 2010, Topology Builder				
File Action View Help				
🕨 🏟 🖄 🔟 🔛 🛄				
🌡 Lync Server 2010 ⊡ 🗐 Interop	Mediation Server PSTN	gateway		4
P Standard Edition Front End Servers Description Front End pools Description Front End pools Description Pools Description Pools	TLS listening port: TCP listening port:	5067 5068		
🟵 🧰 SQL stores	PSTN Gateways:	Default	Gateway	Site
AcGution pools Mediation pools PSTN gateways ACGW.lync.local B AcGW.lync.local B AccGW.lync.local B AccGW.lync.local		ACGW ALE-G ACEG. Telenx Verizo	Jync.local W.Jync.local Lync.local et.Lync.local n.Lync.local	Interop Interop Interop Interop Interop

Figure 3-11: Media Server PSTN Gateway Association Properties

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

Eiguro	2-12.	Dubliching	Topology
rigure	J-12.	Fublishing	ropology

🏹 Lync Server 2010, Topology Builder	
Eile Action View Help	
🗢 🔿 🖄 💼 🛛 🖬	
Image: Second state Image: Second state Image: Second state New Central Site Image: Second state Edit Properties	Site
 New Topology Open Topology Download Topology Save a copy of Topology As Save a copy of Topology As Install Database Merge 2007 or 2007 R2 Topology Remove Deployment 	Name: Interop Description: City: State/Province: Country/Region Code:
	Call Admission Control Setting
	Call Admission Control: FE-Lync.Lync.local
	Site federation route assignment
	Federation: Disabled



Help

The Publish Topology screen is displayed.

Figure 3-13: Publish Topology Confirmation

Publish Topology	×
Publish the topology	
 In order for Lync Server 2010 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: A validation check on the root node did not return any errors. A file share has been created for all file stores that you have configured in this topology. All simple URLs have been defined. 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. For a single Standard Edition server: The task "Prepare first Standard Edition server" was run. You are currently logged on as a SQL administrator, for example, as a member of the SQL sysadmin role. 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. 	
When you are ready to proceed, click Next.	

5. Click **Next**; the Topology Builder attempts to publish your topology.

Back

Next

Cancel

Figure 3-14: Publish Topology Confirmation screen

Publish Topology	×
Publishing in progress	
Please wait while Topology Builder tries to publish your topology.	
Publishing topology	

 Publishing topology ...

 Downloading topology ...

 Succeeded

 Downloading global simple URL settings.

Wait until the publish topology process has ended successfully.

Figure 3-15: Publish Topology Successfully Completed

Publish Topology	X
Publishing wizard complete	
Your topology was successfully published.	

	Step	Status			
<	Publishing topology	Success			View Logs
√_	Downloading topology	Success			
<	Downloading global simple URL settings.	Success			
×.	Enabling topology	Success			

6. Click Finish.

AudioCodes

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 device 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

Connect to FE-Ly	ync.Lync.local	? ×
		A PA
Connecting to FE	-Lync.Lync.local.	
User name:	🕵 Lync\Administrato	r 💌 💷
Password:	•••••	
	Remember my pass	word
	ОК	Cancel

The CSCP Home page is displayed.

Figure 3-18: CSCP Home page

🌄 Mi	crosoft Lync Server 2010 (Control Panel	
	Lync Server 201	0	Administrator Sign ou 4.0.7577)
徛	Home		
33	Users		
N	Topology	User Information	Resources
Ģ	IM and Presence	Welcome, Administrator	Getting Started
Ç	Voice Routing	View your roles	Using Control Panel
C	Voice Features	Top Actions	Microsoft Lync Server 2010
23	Response Groups		Getting Help Downloadable Documentation
Ð	Conferencing	Edit or move users	Online Documentation on TechNet Library Lync Server Management Shell
6	Clients	View topology status	Lync Server Management Shell Script Library
詻	External User Access	View Monitoring Server reports	Community
	Monitoring and Archiving		Blogs
1	Security		
ġ	Network Configuration		

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



Z.	Lync Server 2010								
	Home	Dial Plan	Voice Policy	Route	PSTN Usage	Trunk Con	figuration Test V	oice Routing	
33	Users	Create v	oice routing te	st case info	ormation				
×	Topology								
Ð	IM and Presence						Q		
Ç	Voice Routing	+ New	🔻 🧪 Edit 🔻	Action	Commit				
S	Voice Features	Nam	e		 Scope 	State	Normalization ru	lles Desc	ription
23	Response Groups	C	ilobal		Global	Committed	2		

Figure 3-19: Voice Routing Option

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

Figure 3-20: Route Option

Ø.	Lync Server 2010						
	Home	Dial Plan	Voice Policy	Route	PSTN Usage	Trunk Configuration	Test Voice Routing
33	Users	Create voi	ce routing tes	t case info	ormation		
×	Topology						
P	IM and Presence						م
୯	Voice Routing	🕈 New 🍃	🦯 Edit 🔻 🔺	Move u	p 🔸 Move do	wn Action 🔻 Com	mit 🔻
C	Voice Features	Name			State	PSTN usage	Pattern to match
23	Response Groups	Global			Committed	Internal, Local	^\+1

- 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. Verizon 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 means "to match all numbers".
- 7. Click Add.

Figure 3-21:	Adding	New	Voice	Route
--------------	--------	-----	-------	-------

OK X Cancel	
Name:*	
Verizion	
Description:	
Build a Pattern to Match	
Add the starting digits that you want this route to handle, or create the expression manually by clicking Edit.	
Starting digits for numbers that you want to allow:	
	Add
*	Exceptions
	Remove
Match this pattern:*	
^*]
Edit Reset ?	
Suppress caller ID	
Alternate caller ID:	

8. Associate the route with the E-SBC device IP/PSTN gateway you created above; scroll down to the Associated Gateways pane and click **Add**; a list of all the deployed gateways is displayed.

Select Gateway	0 ×
	٩
Service	Site
PstnGateway:ACGW-SBA-32.Lync.local	SBA-32
PstnGateway:ALE-GW.lync.local	Interop
PstnGateway:GW-SBA-New-M2K.lync.local	SBA-New-M2K
PstnGateway:GW-SBA-M8K.lync.local	SBA-M8K
PstnGateway:Ido-01-GW.lync.local	Ido-01
PstnGateway:ACEG.Lync.local	Interop
PstnGateway:test1112.lync.local	SBA-10
PstnGateway:Telenet.Lync.local	Interop
PstnGateway:GWtest32.lync.local	SBAtest32
PstnGateway:M800TEST01-GW.lync.local	M800TEST01
PstnGateway:Verizon.Lync.local	Interop 🗸
	OK Cancel

Figure 3-22: List of Deployed Gateways

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

scriptio	1:				
Build a	Pattern	to Match —			
dd the	starting dig	gits that you wa	nt this route a Edit.	to handle, o	or create
tarting	digits for n	umbers that yo	u want to allo	ow:	
	2				Add
*					Exception
					Remove
	his pattern	•			
Match t					
Match t					
Match t ^* Edit	Re	set 🕐			

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

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

Alternate caller ID:		
PstnGateway:Verizon.Lync.local		Add Remove
Associated PSTN Usages		
Associated PSTN Usages		
Associated PSTN Usages Select Remove PSTN usage record	Associated voice policies	
Associated PSTN Usages Select Remove PSTN usage record Verizon Users	Associated voice policies	

Figure 3-24: Associating PSTN Usage to E-SBC Device

Click the OK button in the toolbar at the top of the New Voice Route pane.
 Figure 3-25: Confirmation of New Voice Route

Dial Plan	Voice Policy	Route	PSTN Usage	Trunk Configurat	ion Test Voice Routi	ng
Create v	oice routing tes	st case info	ormation			
					٩	
🗣 New	🧪 Edit 🔻 🔮	The Move u	p 🔸 Move do	own Action 🔻	Commit 🔻	
Nam	e		State	PSTN usa	ge Pattern to mat	ch
Verizi	on		🦆 Uncomr	nitted Verizon U	sers ^*	
Globa	al		🦆 Uncomr	nitted Internal, L	ocal ^\+1	

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

Dial Plan	Voice Policy	Route	PSTN Usage Tr	unk Configurat	ion Test V	oice Routing	
Create v	oice routing t	est case info	ormation				
					2		
A New	🧪 Edit 🔻	1 Move up	p 🔸 Move down	Action •	Commit 🔻		
Nam	e		State	PSTN usa	Review unc	ommitted chang	ges 📃
Verizi	on		🦆 Uncommitte	d Verizon U	Commit all	N	
Globa	I		🦆 Uncommitte	d Internal, L	Cancel selec	cted changes	

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

Figure 3-27: Uncommitted Voice Configuration Settings

Unco	mmitted Voice Configuratio	n Settings	5) ×
R	outes			*	
	Identity	Action	New value (pattern to match)	Old value (pattern	
	Verizion	Modified	\/*	^((\+1) (\+2) (\+3) (
	Global	Modified	^\+1	^\+1	

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

Figure 3-28: Voice Routing Configuration Confirmation

Microsoft Lync Server 2010 Control Panel	0	×
Successfully published voice routing configuration		
U succession, 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

Dial Plan	Voice Policy	Route	PSTN Usage	Trunk Configuration	Test Voice Routing	
Create v	oice routing te	st case info	ormation			
	•	•			<u>م</u>	
New	🧪 Edit 🔻	The move up	p 🔸 Move do	own Action V Com	P mit ▼	
New Nam	🧪 Edit 🔻 🤺	🏠 Move u	P Move do	wn Action ▼ Com PSTN usage	P mit ▼ Pattern to match	
New Nam Verizi	<mark>∕[*]Edit ▼</mark> e ion	🏠 Move u	Move do State Committed	wn Action V Com PSTN usage Verizon Users	P mit ▼ Pattern to match ^*	
3.4 Configuring the 'Dial Plan' on the Lync Server 2010

This section describes how to configure a 'Dial Plan' on the Lync server and associate it with 'Normalization' Rules.

To configure the 'Dial Plan on the Lync Server 2010:

1. In the Navigation pane, select the **Voice Routing** option.

Figure 3-30: Voice Routing Option

Ø.	Lync Server 201	0							
	Home	Dial Plan	Voice Policy	Route	PSTN Usage	Trunk Con	figuration	Test Voice Rou	ıting
33	Users	Create	voice routing te	st case info	ormation				
×	Topology								
P	IM and Presence							م	
હ	Voice Routing	🗣 New	🔻 🥖 Edit 🔻	Action	Commit	•			
C	Voice Features	Na	ne		Scope	State	Normaliza	tion rules	Description
23	Response Groups	6	Global		Global	Committed	2		

2. In the Voice Routing menu at the top of the page, select the **Dial Plan** option.

Figure 3-31: Dial Plan Option

2	Lync Server 2010				
<u>ن</u>	Home	Dial Plan Voice Policy	Route PSTN Usage	Trunk Configuration	Test Voice Routing
33	Users	Create voice routing te	est case information		
×	Topology				
Ģ	IM and Presence				<u>م</u>
હ	Voice Routing	🗣 New 🧪 Edit 🔻	The move up 🕹 Move do	own Action 🔻 Com	mit 🔻
C	Voice Features	Name	State	PSTN usage	Pattern to match
23	Response Groups	Global	Committed	Internal, Local	^\+1
	3. In the cor ♣ New ¹ Site dial	ntent area toolbar,	click 🕈 New . and	d select the Dia	Il Plan type.
	Pool dial	plan			
	User dial	plan			
	In this ex	ample, we create a	a 'User Dial Plan'.		

4. In the New Dial Plan pane, fill in a Name for this Dial Plan (i.e. Verizon Users) and create new 'Associated Normalization Rules' to match for the dialing phone numbers you wish this Dial Plan to handle.



5. Click Add.

6. For creating 'New Normalization Rules' in the content area toolbar, click the

+ New option.

Figure 3-32: New Normalization Rules Option

Associated Normalization Rules					
🕂 New 🖹 Copy 📋 Paste 🛸 Sele	ct 🧪 Show detail	sRemove 👚 🗸	ŀ		
Normalization rule	State	Pattern to match	Translation pattern		

7. In the New Normalization Rules pane, fill in a name for these Normalization rules (i.e. Verizon Users) and the Normalization rules for the phone numbers you wish this device to handle.

Figure 3-33: New Normalization Rules Option

Edit Dial Plan > New Normalization Rule

ок	🗙 Cancel			
Name:*				
Descriptio	n:			
– Build	a Normaliz	ation Rule		
Fill in th	e fields that	you want to ι	ise, or create the rule	manually by
clicking	Edit.			
Starting	digits:			
Length:				
Exactly		•	11	
Digits to	o remove:			
0				
Digits to	o add:			
+				
Datterr	to match:*			
AAdd	1110			
-(Iult	±]]#			
Transla	tion rule:*			
+\$1				

8. In this example, we created the follow **Normalization Rules**:

Figure 3-34: Normalization Rules

Associated Normalization Rules				
🕈 New 🖹 Copy 📋 Paste 😁 Select 🥢	Show detail	sRemove 👚 🗸	ŀ	
Normalization rule	State	Pattern to match	Translation pattern	
Call to Operator	Committed	^(0)\$	+1\$1	
Call to International Operator	Committed	^(00)\$	+1\$1	
Call to Operator with Local Area Number	Committed	^(0\d{7})\$	+1\$1	
Call to Operator with Local Number	Committed	^(0\d{10})\$	+1\$1	
3 Digits	Committed	^(\d{3})\$	+1\$1	
1411	Committed	^(1411)\$	+1\$1	
In Site call	Committed	^(5\d{3})\$	+1719313\$1	
Local Area Call	Committed	^(\d{7})\$	+1719\$1	
US Call	Committed	^(\d{10})\$	+1\$1	
Prefix All	Committed	^(\d{11})\$	+\$1	
Call to Operator with International Number	Committed	^(01\d{5}\d+)\$	+1\$1	
International	Committed	^(\d{8}\d+)\$	+011\$1	

9. Example (and Translation) of Normalization Rules:

- Call to Operator: Dial exactly '0' the rule add '+1' → Lync will send to the GW '+10'.
- Call to International Operator: Dial exactly '00' the rule add '+1' → Lync will send to the GW '+100'.
- Call to Operator with Local Area Number: Dial number that start with '0' and more 7 digits ('0xxxxxx'), the rule add '+1' → Lync will send to the GW '+10xxxxxx'.
- Call to Operator with Local Number: Dial number that start with '0' and more 11 digits ('0xxxxxxxxx'), the rule add '+1' → Lync will send to the GW '+10xxxxxxxxx'.
- **3 Digits:** Dial 3 digits ('xxx') the rule add '+1' → Lync will send to the GW '+1xxx'. This rule using for emergency call (e.g., 911).
- 1411: Dial exactly '1411' the rule add '+1' → Lync will send to the GW '+11411'.
- In Site call: Dial 4 digits ('xxxx') the rule complete the number to E-164 standard number and add '+1719313' → This rule affect only in site calls and not sending to the GW.
- Local Area Call: Dial 7 digits ('xxxxxx') the rule add '+1719' → Lync will send to the GW '+1719xxxxxx'.

- US Call: Dial 10 digits ('xxxxxxxx') the rule will add '+1' → Lync will send to the GW '+1xxxxxxxxx'.
- **Prefix All:** Dial 11 digits ('xxxxxxxx') the rule will add '+' → '+1xxxxxxxx' will send to the GW.
- Call to Operator with International Number: Dial 8 or more digits that starting with '01' ('01xxxxxxxxx') the rule add '+1' → '+101xxxxxxxxxx' will send to the GW.
- International: Dial 9 or more digits ('xxxxxxxxx') the rule add '+011' → '+011xxxxxxxxx' will send to the GW.

10. Example of Normalization Rules:

Figure 3-35: Example: Normalization Rules

🧹 ок	:]	×	Cancel					
Name:	r							
Call to	Op	erato	or					
Descrip	tio	n:						
— Bui	ld a	a N	ormaliz	atio	n Rule			,
Fill in clicki	the ng	e fie Edit	lds that	you v	vant to u	IS	e, or create the rule manually l	by
Starti	ing	digi	its:					
0								
Leng	th:							
Exac	tly				•		1	•
Digit	s to) ren	nove:					
0							6	•
Digit	s to	add	1:					
+1								
Datt	orn	to r	natchi t					
-/(0)	s	101	natch.					
-								
Iran	slat	tion	rule:*					_
+13	1					_		
	Edit	it	Res	et	?			
Inte	erna	al ex	tension	?				
Dialed	nur	mbe	r to test					
								Go

- Name: The Rule name.
- **Description:** Rule description.
- Starting digits: The Digit/s Number/s that the Normalization Rule starting.
- Length: Three options: 'At Least', 'Exactly', and 'Any'. And then the numbers of dial digits (e.g., exactly 1 digit).
- **Digits to remove:** The number of Digits that need to remove from the dialed number. (e.g., Don't remove digits)
- **Digits to add:** The number of Digits that need to add to the dialed number. (e.g., add '+1' as prefix)
- Pattern to match: This field automatic filled, based on the previous fields.
- Translation rule: This field automatic filled, based on the previous fields.
- **Dialed number to test:** Use for test the rule before approve it.
- **11.** Gateway number manipulation. The number manipulation configuration of the AudioCodes device is later described in Section 4.13 on page 83.

3.5 Enabling Voice Mail in the Exchange Server

This section describes how to enable voice mail for the user on the Exchange Server.

to enable user for voice mail in the Exchange Server:

1. Open the Exchange Management Console.

Figure 3-36: Exchange Management Console

😹 Exchange Management Console						
File Action View Help						
🗢 🔿 🔰 🖬 🚺 🖬						
🔀 Microsoft Exchange	Microsoft Exchange 😥 Mailhox - Entire Forest					
🖃 📴 Microsoft Exchange On-Premises (e						
🛨 🚠 Organization Configuration	Y Create Filter					
표 🗧 Server Configuration						
🖃 🤱 Recipient Configuration	Display Name 🔶	Allas	Organizational Unit	Recipient Type Details	Primary SMTP Address	
A Mailbox	administrator 🍪	Administrator	lync.local/Users	User Mailbox	Administrator@lync.local	
🧸 Distribution Group	😡 Discovery Search Mailbox	DiscoverySearchMailbox{	lync.local/Users	Discovery Mailbox	DiscoverySearchMailbox{	
🧖 Mail Contact	🚜 Eyal Israeli	Eyal	lync.local/Users	User Mailbox	Eyal@lync.local	
a Disconnected Mailbox	🕮 Hadar Vernik	Hadar	lync.local/Users	User Mailbox	Hadar@lync.local	
🏹 Move Request	🚜 Hagai Nof	Hagai	lync.local/Users	User Mailbox	Hagai@lync.local	
💼 Toolbox	Nohn Vasquez	John	lync.local/Users	User Mailbox	John@lync.local	
	👪 Mike Erps	Mike	lync.local/Users	User Mailbox	Mike@lync.local	
	🚳 Ofer Aharonov	ofer	lync.local/Users	User Mailbox	ofer@lync.local	

- 2. In the Navigation pane, select the **'Recipient Configuration >> Mailbox'** Tree folder.
- 3. Select user for enabling voice mail from the list.
- 4. From the dropdown list, right-click on the selected user and choose **Enable Unified Messaging**.

Figure 3-37: Mailbox – Entire Forest

🚱 Mailbox - Entire Forest				
YCreate Filter				
Display Name Administrator Discovery Search Mailbox	Alias Administrator DiscoverySearchMailbox{	Organizational Unit lync.local/Users lync.local/Users		
Hadar Vernik	Enable Archive	/Users /Users		
Contraction Vasquez Contraction Contractio	Remove Enable Unified Messaging	/Users /Users /Users		
	New Local Move Request New Remote Move Request			
	Manage Send As Permission Manage Full Access Permiss	i		
	Send Mail			
	Properties			
	Help			

The Enable Unified Messaging Wizard – Introduction screen is displayed.

Figure 3-38: Enable Unified Messaging

Enable 0	Unified Messaging
 Introduction Extension Configuration Enable Unified Messaging Completion 	Introduction The selected mailbox will be enabled for Unified Messaging. Upon completion, an e-mail message will be sent to the mailbox notifying the user that they have been enabled for Unified Messaging. The message will include the PIN and the number to dial to gain access to their mailbox. By default, an extension number and PIN are automatically generated. You can also manually specify an extension number and PIN. Unified Messaging Mailbox Policy:
Help	< Back Next > Cancel

5. In the new dialog pane, click the **Browse** button; the Select UM Mailbox Policy dialog is displayed.

Figure 3-39	: Select	UM	Mailbox	Policy
-------------	----------	----	---------	--------

🍕 Sele	🍕 Select UM Mailbox Policy				
Eile	<u>V</u> iew				
<u>S</u> earch:			Find Now	Cl <u>e</u> ar	
Name	*				
🔟 ACir	nt Default Policy				

6. Select ACint Default Policy.



7. Click **Next**; the Enable Unified Messaging Wizard – Extension Configuration screen is displayed.

Figure 3-40: Enable Unified Messaging Wizard – Extension Configuration

Co Enab	le Unified Messaging
Introduction	Extension Configuration
 Extension Configuration Enable Unified Messaging Completion 	 Automatically-generated mailbox extension Manually-entered mailbox extension: SIP Resource Identifier This refers to a SIP address of a UM-enabled user when a SIP URI dial plan is used. For example, tonysmith@contoso.com. When an E.164 dial plan is used, this would refer to the E.164 address of the user. For example, +1425551234. Automatically-generated SIP resource identifier: Manually-entered SIP resource identifier: Hadar.Vernik@FE-Lync.Lync.local
Help	< <u>B</u> ack <u>N</u> ext > Cancel

8. Click **Next**; the Enable Unified Messaging Wizard Configuration Summary screen is displayed.

Figure 3-41: Enable Unified Messaging Wizard – Configuration Summary

Enable	e Unified Messaging	
Introduction	Enable Unified Messaging When you click Enable, the following recipient will be UM enabled.	
Configuration	Configuration Summary:	
Enable Unified Messaging	🛺 Hadar Vernik	*
Completion	Unified Messaging Mailbox Policy: ACint Default Policy Mailbox Extensions: 5665 SIP Resource Identifier: Hadar.Vernik@FE-Lync.Lync.local Automatically generate PIN to access Outlook Voice Access	
	To copy the contents of this page, press CTRL+C.	
Help	< Back Enable Ca	ncel

9. Click **Enable**; the Enable Unified Messaging Wizard – Completion screen is displayed.

Figure 3-42: Enable Unified Messaging Wizard – Completion

Enable	Unified Messaging
 Introduction Extension Configuration Enable Unified Messaging Completion 	Completion The wizard completed successfully. Click Finish to close this wizard. Elapsed time: 00:00:03 Summary: 1 item(s). 1 succeeded, 0 failed. Hadar Vernik Exchange Management Shell command completed: 'Lync.local/Users/Hadar Vernik' Enable-UMMailbox-PinExpired \$false -UMMailboxPolicy 'ACint Default Policy' - Extensions '5665' - SIPResourceIdentifier 'Hadar.Vernik@FE-Lync.Lync.local' Elapsed Time: 00:00:03
Help	To copy the contents of this page, press CTRL+C. < Back Finish Cancel

10. Click Finish.

_

3.6 Enabling User for Lync Voice

This section describes how to enable Lync Voice Mail for a specific Lync user.

> To enable Lync Voice Mail:

 Open the Lync Server 2010 (Start menu > Programs > Microsoft Lync Server 2010 > Lync Server Control Panel); the CSCP Home page is displayed.

Figure 3-43: CSCP Home page

	Lync Server 201	0	Administrator Sign ou 4.0.7577
~	Home		
-1	Home		
	Topology	User Information	Resources
Ģ	IM and Presence	Welcome, Administrator	Getting Started
e	Voice Routing	View your roles	First Run Checklist Using Control Panel
C	Voice Features	Top Actions	
23	Response Groups	Enable users for Lync Server	Downloadable Documentation
Ð	Conferencing	Edit or move users	Online Documentation on TechNet Library Lync Server Management Shell
6	Clients	View Monitoring Server reports	Lync Server Management Shell Script Library Lync Server Resource Kit Tools
1	External User	• view monitoring server reports	Community
	Access		Forums
	and Archiving		biogs
•	Security		
9	Network Configuration		

2. In the Navigation pane, select the 'Users' option.

Figure 3-44: Users Option

æ.	Lync Server 20	10
	Home	User Search
32	Users	
×	Topology	• Search LDAP search
Ģ	IM and Presence	Search for users by typing a user's name or clucking Add futter
ę	Voice Routing	Stable users ▼ // Edit ▼ Action ▼
S	Voice Features	Display name Enabled SIP address Registrar pool Telephony
23	Response Groups	
Ð	Conferencing	Select the type of search that you want to perform.
	Clients	IMPORTANT: Search returns users already enabled for Lync Server, but
	External User Access	does not include users enabled for previous versions. To see these users, select Legacy users in the search filter query.
	3. Ii	n the content area toolbar, click Enable users 🔻



The New Lync Server User screen is displayed.

Figure 3-45: New Lync Server User

New Lync Server User

Display name Status Add Pernove Assign users to a pool.* Image: Constraint of the provided						
Display name Status Add Pernove Pernove Assign users to a pool* Image: Convertient of the converti	or users by typing a user's n	ame or clickii	ng Add filter	Fin	d + Add filt	er ≽
Display name Status Add Renove Assign users to a pool:* Senerate user's SIP URI: ③ Use the ser principal name (UPN) ③ Use the ser principal name (UPN) ③ Use the following format: <firstname>,<lastname> @ < Status for Blowing format: <firstname>,<lastname> @ < Specify a SIP URI: precompte precompte ite URI: precompte ite URI: precompte ite URI: precompte Automatic> Automatic> . Under the Users: click Add: ; the Lync search engine opens.: Ergure 3-46: Lync Search Engine</lastname></firstname></lastname></firstname>	LDAP search					
Image: StatusAddPermove: StatusAddAsign users to a pool:*Image: StatusSecurate user's SIP URI:Image: StatusUse user's email addressImage: StatusUse the following format:Image: StatusStatus hame > Image: StatusImage: StatusUse the following format:Image: StatusStatus hame > Image: StatusImage: StatusUse the following format:Image: StatusStatus hame > Image: StatusImage: StatusSpecify a SIP URI:Image: Image: Image: StatusSpecify a SIP URI:Image: Image: Imag	om Active Directory					2 ×
Visplay name Status Add Pernove Assign users to a pool:* Visplay name Add Perto-PC only Visplay name Visplay name Perto-PC only Visplay name Visplay name Visplay name Visplay name Visplay name Automatic> Visplay name	Fig	ure 3-46: L	ync Search Engine			
Display name Status Add Remove Assign users to a pool:* Assign users to a pool:* Generate user's SIP URI: Ive the user principal name (UPN) Use the following format: <firstname>, <lastname> @ Use the following format: <samaccountname> @ Specify a SIP URI: Elephony: PC-to-PC only PC-to-PC only Conferencing policy: <automatic> View View</automatic></samaccountname></lastname></firstname>	4. In the Users: clie	Add	; the Lync search er	ngine	e opens.	
Display name Status Add Remove Assign users to a pool:* Assign users to a pool:* Generate user's SIP URI: • Use user 's email address • Use the following format: <firstname>.<lastname> @ • Use the following format: <samaccountname> @ • Specify a SIP URI: sipercomple @ Telephony: PC-to-PC only ? Line URI: tet:+123456 ?</samaccountname></lastname></firstname>	<automatic></automatic>			•	View	
Display name Status Add Assign users to a pool:* Assign users to a pool:* Semerate user's SIP URI: • Use user's email address • Use the user principal name (UPN) • Use the tollowing format: <firstname>.<lastname> @ • Use the following format: <samaccountname> @ • Specify a SIP URI: sipexample @ Telephony: PC-to-PC only PC-to-PC only Ine URI: tel:+123456</samaccountname></lastname></firstname>	Conferencing policy:					
Display name Status Add Remove Assign users to a pool:* Assign users to a pool:* Generate user's SIP URI: Use user's email address Use the ser principal name (UPN) Use the following format: Use the following format: Specify a SIP URI: Specify a SIP URI: Specify a SIP URI: Specify a SIP URI: PC-to-PC only Line URI:	tel:+123456				?	
Display name Status Add Remove Assign users to a pool:* Assign users to a pool:* Cenerate user's SIP URI: Use user's email address Use the user principal name (UPN) Use the following format: FirstName>,<lastname> @</lastname> Specify a SIP URI: Specify a SIP URI: <li< td=""><td>Line URI:</td><td></td><td></td><td></td><td></td><td></td></li<>	Line URI:					
Display name Status Add Remove Assign users to a pool:* Generate user's SIP URI: • Use user's email address • Use the user principal name (UPN) • Use the following format: <firstname>.<lastname> @ • Use the following format: <samaccountname> @ • Specify a SIP URI: sipexample @</samaccountname></lastname></firstname>	PC-to-PC only			•	?	
Display name Status Add Remove Assign users to a pool:* Senerate user's SIP URI: Ouse user's email address Use the ser principal name (UPN) Use the following format: <firstname>,<lastname> @ Specify a SIP URI: Specify a SIP URI: \$specify a SIP URI: @</lastname></firstname>	Telephony:					
Display name Status Add Remove Assign users to a pool:* Assign users to a pool:* Generate user's SIP URI: • Use user's email address • Use the user principal name (UPN) • Use the following format: <firstname>,<lastname> @ • Use the following format: <samaccountname> @ • Specify a SIP URI:</samaccountname></lastname></firstname>	sip:example	@		*		
Display name Status Add Remove Assign users to a pool:* Senerate user's SIP URI: • Use user's email address • Use the user principal name (UPN) • Use the following format: <firstname>.<lastname> @ • Use the following format: <samaccountname> @</samaccountname></lastname></firstname>	Specify a SIP URI:					
Display name Status Add Remove Assign users to a pool:* Senerate user's SIP URI: • Use user's email address • Use the user principal name (UPN) • Use the following format: <firstname>,<lastname> @</lastname></firstname>	<samaccountname></samaccountname>	@				
Display name Status Add Remove Remove Assign users to a pool:* Generate user's SIP URI: • Use user's email address • Use the user principal name (UPN) • Use the following format: <	O Use the following for	mat:				
Display name Status Add Remove Assign users to a pool:* © Generate user's SIP URI: • • Use user's email address • • Use the user principal name (UPN) • • Use the following format:	<firstname>.<lastna< td=""><td>me> @</td><td></td><td>-</td><td></td><td></td></lastna<></firstname>	me> @		-		
Display name Status Add Remove Assign users to a pool:* Senerate user's SIP URI: • Use user's email address	O Use the following for	mat:				
Display name Status Add Remove Assign users to a pool:* Senerate user's SIP URI:	Use user's email addi Use the user principal	ess Loame (LIDN)				
Display name Status Add Remove Assign users to a pool:*	Generate user's SIP URI:					
Display name Status Add Remove				•		
Display name Status Add Remove	Assign users to a pool:*				1	
Display name Status Add Remove						
Display name Status Add					Remove	
	Display name	S	tatus		Add	
Users:*	Users:*					
		Users:* Display name Assign users to a pool:* Generate user's SIP URI: Ouse user's email addr Use the user principa Use the following for <firstname>.<lastna Use the following for <samaccountname> Specify a SIP URI: sip:example Telephony: PC-to-PC only Line URI: tel:+123456 Conferencing policy: <automatic> 4. In the Users: clice Fig Display name</automatic></samaccountname></lastna </firstname>	Users:* Display name S Display name S Assign users to a pool:*	Users:* Status Display name Status Assign users to a pool:*	Display name Status Assign users to a pool:* Generate user's SIP URI: • Use user's email address • Use the ser principal name (UPN) • Use the following format: • Specify a SIP URI: • Display name • Display name • Use the following format: • Contractive Directory Telephony: Pc-to-PC only Line URI: tele: Conferencing policy: Automatic> Automatic> Conferencing policy:	Users* Display name Status Add Remove Assign users to a pool:* Generate user's SIP URI: Use user's email address Use the user principal name (UPN) Use the following format: Vise the following format: Specify a SIP URI: sipexample Specify a SIP URI: sipexample Specify a SIP URI: sipexample Specify a SIP URI:

×

5.

Click

for searching un-enable voice user in the Active Directory.

Select from Active Directory	0
 Search O LDAP search Search for users by typing a user's name or clicking Add filter X 	Find + Add filter
	Search results:
Name	Display name
Administrator	Administrator
Daniel Belenki	Daniel Belenki
DiscoverySearchMailbox {D919BA05-46A6-415f-80AD-7E09334BB852}	Discovery Search Mailbox
FederatedEmail.4c1f4d8b-8179-4148-93bf-00a95fa1e042	Microsoft Exchange Approval
Guest	
Hadar Vernik	Hadar Vernik
Itay Cohen	Itay Cohen
krbtgt	
Neitan Naamat	Neitan Naamat

6. Select one of the users from the list and click **OK**.

🔆 Enable 🗙 Cancel		
Users:*		
Display name	Status	Add
Hadar Vernik		Remove
Assign users to a pool:*		
FE-Lync.Lync.local		•
Generate user's SIP URI:		
O Use user's email address		
Use the user principal name (UPN)	
 Use the following format: 		
<firstname>.<lastname> @</lastname></firstname>	FE-Lync.Lync.local	•
Use the following format:		
<samaccountname> @</samaccountname>		Ŧ
Specify a SIP URI:		
sip:example	@	Ŧ
Telephony:		
Enterprise Voice		▼ ?
Line URI:		
tel:+17193135665		?
Dial plan policy:		
Verizon Users		▼ View
Voice policy:		
Verizon Users		View

Figure 3-48: User Properties

- 7. In the 'Assign users to a pool' field, select the Pools that you working with it.
- 8. In the 'Generate user's SIP URI' field, select the format.
- 9. In the 'Telephony' field, select Enterprise Voice.
- 10. In the 'Line URI' field, add the tel:<phone number> (e.g., tel:17193135665).
- **11.** In the 'Dial plan policy' field, select the dial plan.
- **12.** In the 'Voice policy' select the voice policy.
- 13. Click Enable.

4 Configuring AudioCodes E-SBC device

This section provides step-by-step procedures for configuring AudioCodes' E-SBC device. These procedures are based on the setup example described in the following sections.

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

- **Step 1**: Configure IP Addresses. See Section 4.1 on page 53.
- **Step 2**: Enable the SBC Capabilities. See Section 4.2 on page 59.
- **Step 3**: Configure the Number of Media Channels. See Section 4.3 on page 60.
- **Step 4**: Configure the Proxy Sets. See Section 4.4 on page 61.
- **Step 5**: Configure the IP Groups. See Section 4.5 on page 63.
- **Step 6**: Configure the Voice Coders. See Section 4.6 on page 66.
- Step 7: Define Silence Suppression and Comfort Noise. See Section 4.7 on page 68.
- **Step 8**: Configure IP Profile Settings. See Section 4.8 on page 69.
- Step 9: Configure Header Manipulation for Call Forwarding. See Section 4.9 on page 73.
- Step 10: Configure Unscreened ANI for P-Asserted Identity. See Section 4.10 on page 77.
- Step 11: Configuring Unscreened ANI using Diversion Header. See Section 4.11 on page 78.
- **Step 12:** Configure IP-to-IP Routing Setup. See Section 4.12 on page 79.
- **Step 13:** Configure Number Manipulation. See Section 4.13 on page 83.
- **Step 14:** Configuring SIP General Parameters. See Section 4.14 on page 87.
- **Step 15:** Configuring SIP Supplementary Services. See Section 4.15 on page 89.
- Step 16: Defining Reasons for Alternative Routing (see Section 4.16 on page 90).
- Step 17: Configuring Call Progress Tones file for Regional support (see Section 4.17 on page 92.

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





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

4.1 Step 1: Configuring IP Addresses

This step shows how to configure LAN IP addresses when the internal data-routing capabilities of the E-SBC device are used to connect to the Verizon SIP Trunking service. In this case, configure a separate WAN interface as described below.

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.

Figure 4-2: Physical Interface Separation



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 sub-menu > 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., 10.15.4.30).
- 4. Configure additional IP interfaces, if required.

Figure 4-3: Multiple Interface Table Page

In	lex	Application Typ	be	IP Address	Prefix Ler	ngth	Gateway	v	LAN ID	D Interface Name
0	0	OAMP + Media + Contr	rol	10.15.4.30	16		10.15.4.31		1	Voice
		-	•					_		
			WAN	Interface Name	No	ot Cor	figured 💌			

- 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	N
LAN switch VLAN 1	Connected	1 24
New Connection		4

- 3. Click the Edit 🔪 icon corresponding to the "LAN Switch VLAN 1" connection, and then click the Settings tab.
- **4.** In the 'IP Address' and 'Subnet Mask' fields, enter the required IP address (e.g., 10.15.4.31) and subnet respectively, and then click **OK**.

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

Device Name:	eth0.1
Status:	Connected
Schedule:	Always 🔽
Network:	LAN 💌
Connection Type:	Ethernet
Physical Address:	00:90:8f:36:c6:05
Underlying Connection:	LAN switch
Internet Protocol	Use the Following IP Address
IP Address:	10 .15 .4 .31
Subnet Mask:	255 .255 .0 .0

4.1.1.2 Configuring the 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.
- 3. Access the 'Settings' page (Configuration tab > Data menu > WAN Access > Settings tab).

WAN Ethernet	
Connection Type:	Manual IP Address Ethernet Connection
Name: Status: MAC Address:	WAN Ethernet Connected 00:90:8f:36:c6:06
IP Address:	195 . 189 . 192 . 138
Subnet Mask:	255 .255 .255 .240
Default Gateway:	195 .189 .192 .137
Primary DNS Server:	80 . 179 . 55 . 100
Secondary DNS Server:	80 . 179 . 52 . 100
Click here for Advanced Settings	

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., 195.189.192.138).

To assign a WAN interface for VoIP traffic:

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

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

Index	Application Type	IP Address	Prefix Length	Gateway	VLAN ID	Interface Name
0 0	OAMP + Media + Control	10.15.4.30	16	10.15.4.31	1	Voice
		•		·		
	-					
	WA	N Interface Name	WAN E	themet 🗸 🗸 🗸	2	

- 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 Configuring a 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 local button, and then clicking OK to confirm deletion.

Figure 4-9: Removing Data-Routing Connection Interface

Name	Status	Action
🗞 LAN switch	1 Ports Connected	2
S WAN Ethernet	Cable Disconnected	1
LAN switch VLAN 1	Connected	\ 🕷 🔶
New Connection		4

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

Figure 4-10: Multiple Interface Table

Index	Application Type	3P Address	Prefix Length	Gateway	VLAN ID	Interface Name	Primary DNS Server IP Address	Secondary DNS Server IP Address	Underlying Interface
0 .	OAMP + Media + Control +	10.15.4.30	36	10.15.0.1	0	Meg	0000	0.0.0	Note +

5. Click **OK** to save settings.

4.2 Step 2: Enabling the SIP SBC and IP2IP Application

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

To enable the SIP SBC and IP2IP application:

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

Figure 4-11: Application Enabling

-				
4	Enable SAS	Disable	•	·
4	Enable SBC Application	Enable		•
4	Enable IP2IP Application	Enable	~ .	·

2. From the 'Enable SBC Application' and the 'Enable IP2IP Application' drop-down lists, select **Enable**.

Reset with BURN to FLASH is required.



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

4.3 Step 3: Configuring the Number of Media Channels

In order to perform 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 E-SBC device is 120 (i.e., up to 60 IP-to-IP calls for the Mediant 1000B E-SBC, and for the Mediant 3000 E-SBC, 2016 (i.e., up to 1008 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 1		
4	Voice Streaming	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, do one of the following:
 - For Mediant 1000B E-SBC: Enter **120** to enable up to 60 IP-to-IP calls with transcoding or; click **Apply New Value**.
 - For the Mediant 3000 E-SBC: Enter **2016** to enable up to 1008 IP-to-IP calls with transcoding or; click **Apply New Value**.

4.4 **Step 4: Configuring 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 Lync Mediation server.
- Proxy Set ID #2 is assigned with the IP address of Verizon Business SIP Trunk.
- Proxy Set ID #3 is assigned with the IP address of ATA Media Gateway supporting the Fax machine.

These Proxy Sets are later assigned to IP Groups (see Section 4.5 on page 63).

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 Lync Mediation Server:
 - a. From the 'Proxy Set ID' drop-down list, select 1.
 - **b.** In the 'Proxy Address' column, enter the IP address or FQDN and the listening port of the Lync Mediation Server.
 - c. From the 'Transport Type' drop-down list, corresponding to the IP address entered above, select TCP (port 5068) or TLS (port 5067) depending on the deployed Mediation Server Transport Type.

Figure 4-13: Proxy Set ID 1 for Lync Mediation server

•				
Proxy S	et II	D 1	•	< ← 2a
		Proxy Address	Transport Type	
	1	10.15.9.11:5068 ← 2 b		-2c
	2		· · · · · · · · · · · · · · · · · · ·	
	3		· · · · · · · · · · · · · · · · · · ·	
	4		×	
	5			

AudioCodes

- 3. Configure the Proxy Set for the Verizon Business SIP Trunk:
 - a. From the 'Proxy Set ID' drop-down list, select 2.
 - b. In the 'Proxy Address' column, enter the IP address or the FQDN of the Verizon Business SIP Trunk and the listening port of the Verizon Business SIP Trunk.
 - **c.** From the 'Transport Type' drop-down list corresponding to the IP address entered above, select **UDP**.

Figure 4-14: Proxy Set ID 2 for Verizon Business SIP Trunk

•	_				
Proxy S	et II	D	2	•	∕
		Proxy Address		Transport Type]
	1	63.97.104.81:5111 ← 3 b		UDP 👻 <	3 C
	2			~	
	3			~	
	4			~	_
	5			~	

- 4. Configure the Proxy Set for the Fax supporting Media Gateway:
 - a. From the 'Proxy Set ID' drop-down list, select 3.
 - **b.** In the 'Proxy Address' column, enter the IP address or the FQDN and the listening port of the Fax Supporting Media Gateway.
 - **c.** From the 'Transport Type' drop-down list corresponding to the IP address entered above, select **UDP** Transport Type.

Figure 4-15: Proxy Set ID 3 for Fax Supporting Media Gateway

Proxy S	et I	D	3		~ ←
		Proxy Address		Transport Type]
	1	10.15.7.165:5060 ← 4b			-4c
	2			~	
	з			~	
	4			~	
	5			~	

4.5 Step 5: Configuring 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:

- Verizon Business SIP Trunk
- Lync Server 2010 Mediation Server
- ATA FAX Support via Analog Media Gateway.

These IP groups are later used by the IP2IP application for routing calls as well as the SBC application for Header Manipulation.

To configure IP Groups:

- Open the 'IP Group Table' page (Configuration tab > VoIP menu > Control Network> IP Group Table).
- 2. Define IP Group #2 for the Verizon ITSP as follows:
 - a. IP Group Index 2
 - b. Type: SERVER
 - c. Description: arbitrary name. (e.g., "Verizon")
 - **d.** Proxy Set ID: **2** (represents the IP address, configured in Section 4.4 on page 61, 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: 2: Different IP profile is used for the Verizon Business SIP Trunk and the Mediation Server. See Section 4.8 on page 69.

Figure 4-16: IP Group 2 Table

▼ Index	$2a \rightarrow 2$	~	-
Common Parameters			
Туре	2b → SERVER	×	
Description	Verizon	<mark>← 2</mark> c	
Proxy Set ID	2d → 2	~	
SIP Group Name	2e 63.97.104.81	2℃	
Contact User			
SRD	0		
Media Realm		~	
IP Profile ID	$2f \rightarrow 2$	~	
			_
Gateway Parameters			
Always Use Route Table	No	*	
Routing Mode	Not Configured	*	
SIP Re-Routing Mode	Standard	*	
Enable Survivability	Disable		



• 3a → 1 ~ Index Common Parameters 3b Туре SERVER Y 3c Description Lync 3d Proxy Set ID 1 v 195.189.192.138 SIP Group Name 3e Contact User SRD 0 Media Realm Y 3f ≻ 0 IP Profile ID ¥ Gateway Parameters Always Use Route Table No v Routing Mode Not Configured Y SIP Re-Routing Mode Standard Enable Survivability

Figure 4-17: IP Group 1 Table Page

- 3. Define IP Group #1 for Mediation Server as follows:
 - a. Select IP Group Index 1
 - b. Type: SERVER
 - c. Description: <Free Description> (e.g., "Lync Mediation Server")
 - d. Proxy Set ID: 1
 - 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: 0 (see Section 4.8 on page 69).

Figure 4-18: IP Group 3 Table Page

•			
	Index 4a	3	~
-	Common Parameters		
	Туре	SERVER	<4b
	Description 4c	ATA	
	Proxy Set ID	3	√ ←4d
	SIP Group Name		
	Contact User		
4	SRD	0	
4	Media Realm		~
	IP Profile ID 4f	3	~
•	Gateway Parameters		
	Always Use Route Table	No	v
	Routing Mode	Not Configured	v
	SIP Re-Routing Mode	Standard	×
	Enable Survivability	Disable	v

- 4. Define IP Group **#3** for supporting FAX Media Gateway as follows:
 - a. Select IP Group Index 3.
 - **b.** Type: **SERVER**
 - c. Description: <Free Description> (e.g., "ATA")
 - d. Proxy Set ID: 3
 - 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 or leave blank for default.
 - f. IP Profile ID: **3** (see Section 4.8 on page 69).

4.6 **Step 6: Configuring the Voice Coders**

This step describes how to configure the Voice Coders. Since the Mediation Server supports both the G.711A-law and G.711U-law voice coders, while the Verizon SIP trunk supports the G.729 and G.711U-law coders, you can configure a single coder table reference for both services by utilizing the G711U-law coder as a single voice coder (see Figure 4-20) or you can create a more dynamic servicing interworking based on commonality of supported vocoders via the default Coders and Coders Group tables (see Figure 4-19).

The Coder table is associated with the following IP Profiles:

- IP Profile index 1 and 3 (see Section 4.8 on page 69) references 'Coders Table' (see Figure 4-19) which is associated with the IP Groups 1 and 3 (see Section 4.5 on page 63).
- IP Profile index 2 (see Section 4.8 on page 69) references 'Coder Group 1' (see Figure 4-20), which is associated with IP Group 2 (see Section 4.5 on page 63).

> To configure Coder Table for Mediation Server and in-band FAX usage:

 Open the 'Coders Table' page (Configuration tab > VoIP menu > Coders And Profiles > Coders).

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711U-law 🗲 😽 3	20 🗸	64 💙	0	Disabled
G.711A-law	20 🗸	64 🗸	8	Disabled
~	~	~		×
×	~	~		×
×	~	~		~
~	~	~		×
×	~	~		~
×	~	~		~
×	~	~		~
×	~	~		~

Figure 4-19: Coders Table – Lync Mediation Server

- 2. From the 'Coders Table', select via the drop-down list, coder and attributes in the manner of preference with the first entry as being the preferred entry, and all others following in descending order.
- 3. Select the **G.711U-law** coder, as shown in the figure above.
- **4.** From 'Silence Suppression' drop-down list, select **Enable** or **Disabled** as shown in the figure above.
- 5. Select the **G.711 A-law** coder, as shown in the figure above.
- 6. From 'Silence Suppression' drop-down list, select **Enable** or **Disabled** as shown in the figure above.

> To configure Coder Table for Verizon SIP Trunk usage:

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

Figure 4-20: Coder Group Table 1 – Verizon SIP Trunk

▼								
Coder Group ID				1 🗸 🚽	←	2		
Coder Name		Packetiza	tion Time	Rat	e	Payload Type	Silence Supp	ression
G.729	~ 3	20	*	8	*	18	Disabled	~
G.711U-law		20	*	64	*	0	Disabled	~
	~		*		*			~
	*		*		*			*
	*		*		*			*
	*		*		~			~
	*		*		~			*
	~		*		~			~
	~		*		~			~
	~		~		*			~

- 2. From the 'Coder Group Setting' table, select via drop-down list, the index 1. This index, 1, is referenced by the datafill of parameter 'Coders Group index' of IP Profile index 2. This allows a user to list the allowed vocoders in a supported group to be referenced and utilized. This points to table 'Coder Group' for IP Profile index 2, where Coder Group 1 is explicitly referenced. As shown above, Coder Group 1 is declared to support G.729 and G.711U-law.
- 3. Select the **G.729** and **G.711U-law** coder, in the specific order as shown in the figure above. This will be the preference as advertised in the SDP for proper 'Offer/Answer' interworking.
- 4. From 'Silence Suppression' drop-down list, select **Enabled** or **Disabled** as shown in the figure above.

4.7 Step 7: Defining 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 MSBG 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 4.6 on page 66 above to enable Silence Suppression per coder.)
- Open the 'RTP/RTCP Settings' page (Configuration tab > Media menu > RTP / RTCP Settings).

•	General Settings		
	Dynamic Jitter Buffer Minimum Delay	10	
	Dynamic Jitter Buffer Optimization Factor	10]
	RTP Redundancy Depth	0]
	Packing Factor	1]
	Basic RTP Packet Interval	Default 🗸]
	RFC 2833 TX Payload Type	101]
	RFC 2833 RX Payload Type	101]
	RFC 2198 Payload Type	104]
	Fax Bypass Payload Type	102]
	Enable RFC 3389 CN Payload Type	Enable 🗸	
	Comfort Noise Generation Negotiation	Enable 🗸	← 3
	Remote RTP Base UDP Port	0]
4	RTP Multiplexing Local UDP Port	0]
4	RTP Multiplexing Remote UDP Port	0]
4	RTP Base UDP Port	6000]

Figure 4-21: 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.

4.8 Step 8: Configuring IP Profile Settings

This section describes how to configure the IP Profile Settings.

To configure IP Profile for Verizon SIP Trunk Server :

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

Figure 4-22: IP Profile Page-Verizon SIP Trunk Server

▼		
Profile ID 2	2	~
Profile Name	Verizon	
Common Parameters		
RTP IP DiffServ	46	
Signaling DiffServ	40	
Disconnect on Broken Connection	No	~
Dynamic Jitter Buffer Minimum Delay [msec](*)	10	
Dynamic Jitter Buffer Optimization Factor(*)	10	
RTP Redundancy Depth(*)	0	~
Echo Canceler(*)	Enable	~
Input Gain (-32 to 31 dB)(*)	0	
Voice Volume (-32 to 31 dB)(*)	0	
Gateway Parameters	0.544.7	_
Fax Signaling Method	G./11 Transport	~
Play Ringback Tone to IP	Don't Play	*
Enable Early Media	Enable	*
Copy Destination Number to Redirect Number	Disable	*
Media Security Behavior	Preferable - Single Media	*
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 5	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.** In the 'RTP IP DiffServ' field, enter the required value for the DSCP (see example in figure below).
- 4. In the 'Signaling DiffServ' field, enter the required value for the DSCP (see example in figure below).



Service Class Name	DSCP Name	DSCP Value	Example Apps	AudioCodes Value
Telephony	EF	101110	RTP / IP Telephony bearer	46
Signaling	<mark>C\$5</mark>	101000	RTP / SIP / IP Telephony signaling	40
Multimedia Streaming	AF31 <mark>AF32</mark> AF33	011010 011100 011110	SIP / Streaming video and audio on demand	26 28 30
Broadcast Video	CS3	011000	SIP / Broadcast TV & live events	24
Standard	DF (CSO)	000000	Undifferentiated applications	0
Low-priority data	CS1	001000	Any flow that has no BW assurance	8

Figure 4-23:DiffServ Service Classes

Source: RFC 4594. Configuration Guidelines for DiffServ Service Classes.

5. From the 'Coder Group' drop-down list, select **Coder Group 1**.

> To configure IP Profile for FAX supporting ATA:

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

Figure 4-24: IP Profile Page-FAX Supporting ATA

Profile ID		~
Profile Name	ΔΤΔ	
Comment Development		
DTR IR DiffServ	16	
	40	
Signaling Diriserv	40	
Disconnect on Broken Connection	10	×
Dynamic Jitter Buffer Minimum Delay [msec](*)	10	
Dynamic Jitter Buffer Optimization Factor(*)	10	
RTP Redundancy Depth(*)	0	~
Echo Canceler(*)	Enable	~
Input Gain (-32 to 31 dB)(*)	0	
Voice Volume (-32 to 31 dB)(*)	0	
Gateway Parameters		
Fax Signaling Method	G.711 Transport	~
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	Default Coder Group	~
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	
and the second second is the second sec		

- 2. From the 'Profile ID' drop-down list, select 3.
- 3. From the 'Fax Signaling Method' drop-down list, select **G.711 Transport**.

AudioCodes

- 4. 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.
- 5. From the 'Coder Group' drop-down list, select **Default Coder Group**. The call associated with this profile utilizes the base coders that were assigned in the 'Coders' table as opposed to coders that were assigned via an index into the Coder Group Table. The usage of G.711 U-law coder will be based on the previous datafill.
4.9 Step 9: Configuring Message Manipulation for Call Forwarding

This step describes how to configure Message Manipulation for Call Forwarding.

> To configure SBC Header Manipulation Rules:

1. Open the Manipulation Table page (Configuration tab > VoIP menu > SBC > Manipulations SBC > Message).

Figure 4-25: Message Manipulations Table

Index	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value	Row Role	
0 🔘	0	Any.Request		Header.Diversion.URL.host	Modify	'195.189.192.138'	Use Current Condition	
1 🔘	1	Any.Request	header.referred-by exists	Header.Diversion	Add	'<'+header.referred-by.URL+'>'	Use Current Condition	

- 2. Index #0 defines manipulation of calls to modify a header. For any request received modify the Header.Diversion.URL.host with '195.189.192.138'.
- 3. Index #1 defines manipulation of calls to add a header. For any Invite request received and IF a Referred-by header exists, then Add the Diversion Header with what was received within the referred-by header, '<'+header.referred-by.URL+'>'.

Each of these rules bind to a specific Manipulation Set ID as represented within this table. Indexes 0 binds to Manipulation Set ID 0 and Index 1 is bound to Manipulation Set ID 1.

Note the following:

- Index 0 was created for a specific call scenario where a PSTN user originated a call on the Verizon SIP Trunk to an Lync 2010 DID, which has its DID forwarded to the PSTN as well as have its presentation Restricted in the Source number manipulation tables. This was set to change the Diversion URL host from 'anonymous.invalid' to '195.189.192.138', which is the WAN address of the device.
- Index 1 is utilized to dynamically capture the original dialed number for Lync DIDs and present them back to Verizon when a call is forwarded back to the PSTN when creating a Diversion Header. This functionality can only be used if Lync Server 2010 is set up to send Referred-by headers. This specific index creates a new Diversion header for the call using what was presented in the Referred-by header, thus allowing the Verizon service to receive the original dialed DID extension that forwarded the call back to the PSTN SIP trunk.

These newly created message manipulation sets can now be assigned for use. There are several methods to perform this function. One method is to assign the functionality to the entire application via the *ini* file or via the AdminPage.

This functionality is configured using the parameters GWINBOUNDMANIPULATIONSET = [1] and GWOUTBOUNDMANIPULATIONSET = [-1].

When using these parameter settings, the respective Manipulation Set ID is assigned for all inbound and outbound manipulations. When using the IP to IP application, such as in this example, the manipulation for inbound messages **must be** performed at the gateway level, so therefore set **GWINBOUNDMANIPULATIONSET = 1**. The outbound message manipulation can be performed either at the device level or more dynamically, at the IP Group level.

This section describes the following procedures:

- Assign inbound manipulation to the entire device via the *ini* file or via the AdminPage. See Section 4.9.1 below.
- Assign outbound manipulation to a specific IP group index. See Section 4.9.2 below.

4.9.1 Assigning Inbound Message Manipulation Set to Entire Gateway

This step describes how to assign a message manipulation set to the entire device via the INI file.

This assignment is configured using the parameters GWINBOUNDMANIPULATIONSET = [1] and GWOUTBOUNDMANIPULATIONSET = [-1]. These parameter settings are applied for all inbound and outbound manipulations respectively. For IP2IP applications, GWINBOUNDMANIPULATIONSET must be used for header manipulation requirements on incoming messages.

To assign inbound message manipulation set to gateway:

- Open the Admin page, by appending the case-sensitive suffix 'AdminPage' to the gateway-SBC device's IP address in your Web browser's URL field (e.g., <u>http://10.15.9.118/AdminPage</u>).
- 2. On the left pane, click *ini* Parameters.

Figure 4-30: Output Window

Image	Parameter Name: Enter Value: GWINBOUNDMANIPULATIONSET	Apply New Value	
Load to Device	Output Window		•
<i>ini</i> Parameters			
Back to Main	Parameter Name: USESIPURITORDIVERSIONHEADER Parameter New Value: 1 Parameter Description:Use Tel uri or Sip uri for Diversion her	ader	l
	Parameter Name: GWINBOUNDMANIFULATIONSET Parameter New Value: 1 Parameter Description Inbound manipulation set ID for GW. If a applies for all incoming INVITE requests.	configured,	
	Parameter Name: GWOUTBOUNDMANIPULATIONSET Parameter New Value: -1 Parameter Description:Outbound manipulation set ID for GW. If manipulation set was configured in destination IP Group - this applies for all outgoing INVITE requests.	no outbound s parameter	E
			-

- 2. In the 'Parameter Name' field, enter GWINBOUNDMANIPULATIONSET or GWOUTBOUNDMANIPULATIONSET.
- **3.** In the 'Enter Value' field, enter the respective Manipulation Set ID you wish for the specific direction. e.g., **1**.
- 4. Click **Apply New Value** (GWINBOUNDMANIPULATIONSET should be set to **1** for proper interworking).

4.9.2 Assigning Outbound Message Manipulation Set to an IP Group

This step describes how to assign outbound message manipulation rules to IP groups to support outbound message manipulation. Each IP group represents a SIP entity in the device's network. You may assign manipulation rules to IP groups for the Verizon SIP Trunk entity.

When completed, the IP group is used by the IP2IP, as well as the SBC application for routing calls with advanced outbound header manipulation support.

- To assign outbound message manipulation set to IP Group Table 2 (Verizon SIP Trunk):
- Open the IP Group Table page (Configuration tab > VoIP menu > Control Network> IP Group Table).

-			
	Index	2 →2	~
•	Common Parameters		
	Туре	SERVER	~
	Description	Verizon	
	Proxy Set ID	2	~
	SIP Group Name	63.97.104.81	
	Contact User		
4	SRD	0	
4	Media Realm		*
	IP Profile ID	2	*
•	Gateway Parameters		
	Always Use Route Table	No	~
	Routing Mode	Not Configured	~
	SIP Re-Routing Mode	Standard	*
	Enable Survivability	Disable	\sim

Figure 4-26: IP Group Table 2 (Verizon SIP Trunk)

▼ SBC Parameters	
Classify By Proxy Set	Enable 👻
Max Number Of Registered Users	-1
Inbound Message Manipulation Set	-1
Outbound Message Manipulation Set 4	>0

- 2. Set Index to 2.
- 3. Set 'Inbound Message Manipulation Set' to -1. This means that the device does not perform message manipulation at this stage (for IP2IP applications, inbound message manipulation can only be performed at the gateway level).
- 4. Set 'Outbound Message Manipulation Set' to 0.

4.10 Step 10: Configuring Unscreened ANI using P-Asserted Identity

This step describes how to support proper termination of an outbound call with the correct Caller ID when the Screened Telephone Number contained in the P-Asserted Identity Header is different from the CLI contained in the From Header.

This functionality is required to support TC16 of VIT.2010.03154.TPL.001 'Test Suite for Retail VoIP Interoperability IP Trunking' for the Verizon North American retail market.

To configure P-Asserted Identity for Unscreened ANI:

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

Figure 4-27: Output Window

Image Load to Device	Parameter Name: Enter Value: PASSERTEDUSERNAME 7193135662 A	iply New Value
<i>ini</i> Parameters	Output Window	
Back to Main	Parameter Name: PASSERTEDUSERNAME The Value is invalid: Parameter Current Value: 7193135662 Parameter Description:Digit pattern used to indicate Call Forward on Do No Disturb (PBX to VoiceMail)	30

- In the 'Parameter Name' field, enter the parameter PASSERTEDUSERNAME. In the 'Enter Value' field, enter "7193135662" or any valid number which will be utilized in the P-Asserted Identity header.
- 4. Click Apply New Value.

4.11 Step 11: Configuring Unscreened ANI using Diversion Header

This step describes how to support proper termination of an outbound call with the correct Caller ID when the Screened Telephone Number contained in the Diversion Header is different from the Calling Line Identification (CLI) contained in the From Header.

This functionality is required to support TC15 of VIT.2010.03154.TPL.001 'Test Suite for Retail VoIP Interoperability IP Trunking' for the Verizon North American retail market.

To configure diversion header for unscreened ANI:

This step describes how to configure SBC Message Manipulation for creating a Diversion Header to support Unscreened ANI.

To configure SBC header manipulation rules:

Open the Manipulation Table page (Configuration tab > VoIP menu > SBC > Manipulations SBC > Message).

Figure 4-27: Message Manipulations Table

Index	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value	Row Role
0 0	0	Invite.Request		Header.Diversion	Add	<pre>'<sip:9727289417@10.15.9.11< pre=""></sip:9727289417@10.15.9.11<></pre>	Use Current Condition

Index #0 defines manipulation of calls to add a header. For any Invite request received, add a Header.Diversion with the following: <sip:9727289417@10.15.9.118;user=phone>'

3. Click Apply New Value.

4. Attaching or assigning to bind the interworking for message manipulation was discussed previously in sections 4.9, 4.9.1 and 4.9.2. Do this either at the gateway level for incoming messages or at the respective Outgoing IP Group based on routing and selection of the terminating route.

4.12 Step 12: Configuring IP-to-IP Routing Setup

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

- 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'
- 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'.

4.12.1 Configure Inbound IP Routing

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

> To configure in bound IP routing:

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

		•							
		Routing Index		1-10 💙					
		IP To Tel Routing) Mode	Route calls before ma	nipulation 👻				
	Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	- >	Trunk Group ID	IP Profile ID	Source IPGroup ID
1	$2 \rightarrow$		*		10.15.7.165		-1	3	3
2	(3)→		7193135664		63.97.104.81		-1	3	2
3	(4)	\rightarrow	*		10.15.9.11		-1	0	1
4		$5 \longrightarrow$	*		63.97.104.81		-1	2	2
5									
6									
7									
8									
9						1			
<									>

Figure 4-28: Inbound IP Routing Table Page

- Index #1 configuration identifies all IP calls received from FAX supporting ATA 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 ATA device.
 - *Trunk Group ID*: Enter -1 to indicate that these calls are IP-to-IP calls.
 - *IP Profile ID*: Enter **3** to indicate the IP Profile for interworking G.711 in-band fax support to Verizon Business SIP Trunk.
 - Source IP Group ID: Enter **3** to assign these calls to the IP Group pertaining to the ATA supporting fax services.

- 3. Index #2 configuration identifies all IP calls received from the Verizon Business SIP Trunk in the event of determining a route for a dedicated FAX line call Scenario as IP-to-IP calls. It then assigns these IP calls to the IP Group ID configured for the FAX supporting ATA and also assigns a specific IP Profile to support the FAX call.:
 - Source Host Prefix: Enter the asterisk (*) symbol to indicate all destinations.
 - Dest Phone Prefix: Enter the asterisk (*) symbol to indicate all destinations.
 - Source Phone Prefix: Enter the specific DID assigned to support FAX services.
 - Source IP Address: Enter the IP address of Verizon SIP Trunk 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 fax call for G.711 in-band service.
 - Source IP Group ID: Enter **2** to assign these originated calls to the IP Group pertaining to the Verizon SIP Trunk server.
- Index #3 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 Phone Prefix: Enter asterisk (*) symbol to indicate all sources.
 - 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 **0** indicate the IP Profile for Mediation server.
 - Source IP Group ID: Enter **1** to assign these calls to the IP Group pertaining to the Lync Mediation server.
- 5. Index #4 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 '2' indicate the IP Profile for Verizon Business SIP Trunk.
 - Source IP Group ID: Enter "2" to assign these calls to the IP Group pertaining to Verizon Business SIP Trunk.

4.12.2 Configuring Outbound IP Routing

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

> To configure outbound IP routing:

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

		•											
		Routing Inde	×			1-10	*						
		Tel To IP Rou	iting Mode			Route ca	lls before manipulation 💌						
Src. IPGroupID	Src. Host Prefix	Dest Host Prefix	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Pre	fix 5	Dest. IP Address	Port	Transport Type	Dest. IPGroup ID	Dest. SRD	IP Profile ID	Status
1 2	\rightarrow		*	*	*				Not Configured 💌	2	-1	2	n/a
2 3	(3)	\rightarrow	*	*	*				Not Configured 👻	2	-1	3	n/a
3 2	4		•	7193135664	*				Not Configured 👻	3	-1	3	n/a
4 2		5		→	*				Not Configured 👻	1	-1	0	n/a
5 -1									Not Configured 👻	-1			
6 -1									Not Configured 👻	-1			
7 -1									Not Configured 👻	-1			
8 -1									Not Configured 💌	-1			
<	£_)	1.		1							>

Figure 4-29: Outbound IP Routing Table Page

- Index #1 defines the routing of IP calls to the Verizon Business SIP Trunk. All calls received from IP Group ID 1 (i.e., Lync 2010 Mediation server) are routed to Destination IP Group ID 2 (i.e., Verizon Business SIP Trunk):
 - Source IP Group ID: Select **1** 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 **2** to indicate the destination IP Group to where the calls must be sent, i.e., to the Verizon Business SIP Trunk.
 - *IP Profile*: *Select* **2** to indicate usage of IP Profile supporting Verizon coder preference.
- 3. Index #2 defines the routing of FAX IP calls to the Verizon Business SIP Trunk. All calls received from IP Group ID 3 (i.e., ATA support FAX) are routed to Destination IP Group ID 2 (i.e., Verizon Business SIP Trunk):
 - Source IP Group ID: Select **3** to indicate received (inbound) calls identified as belonging to the IP Group configured for the ATA supporting FAX service.
 - 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 the Verizon Business SIP Trunk
 - *IP Profile*: Select **3** to indicate usage of IP Profile supporting Fax.

- 4. Index #3 defines routing of FAX IP calls to the ATA supporting FAX. All calls received from Source IP Group ID 2 (i.e., from the Verizon Business SIP Trunk) are matched against assigned FAX DIDs and then routed to Destination IP Group ID 3 (i.e., to ATA supporting FAX services):
 - Source IP Group ID: Select **2** to indicate received (inbound) calls identified as belonging to the IP Group configured for the Verizon Business SIP Trunk.
 - *Dest Phone Prefix*: Enter the assigned Verizon DID associated with the customers fax services.
 - Source Phone Prefix: Enter the asterisk (*) symbol to indicate all callers.
 - *Dest IP Group ID*: Select **3** to indicate the destination IP Group to where the calls must be sent, i.e., to ATA supporting FAX services.
 - IP Profile: Select 3 to indicate usage of IP Profile supporting Fax.
- 5. Index #4 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 **2** 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 **1** to indicate the destination IP Group to where the calls must be sent, i.e., to Lync 2010 Mediation server.
 - *IP Profile*: Select **0** to indicate no special usage of IP Profile. Instead, service will be selected from the base provisioning such as 'Coders'.

4.13 Step 13: Configuring Number Manipulation

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

- Dest Number Tel->IP. See Section 4.13.1 on page 84.
- Source Number Tel->IP. See Section 4.13.2 on page 85.
- Redirect Number Tel->IP. See Section 4.13.3 on page 86

4.13.1 Configuring Destination Phone Number Manipulation

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

- 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 sub-menu > Dest Number Tel > IP).

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

Note	: Selec	row inde	ex to modify the relevant ro	IW.					
		Add)						
Index	Sourc Trunk Group	e Source IP Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digits to Leave
1 C	-1	1	+011	•	1	0			255
2 C	-1	1	+1	•	2	0			255
зС	-1	1	+	•	1	0			255
4 C	-1	2	719	•	0	0	+1		255
4 C	-1	2	719	•	0	0	+1		255

- **Index #1** defines destination number manipulation of IP calls from Lync Server 2010. All calls received from Source IP Group 1 (i.e., from Lync Server 2010), where the destination number prefix begins with '+011', remove the '+'.
- Index #2 defines destination number manipulation of IP calls from Lync Server. All calls received from Source IP Group 1 (i.e., from Lync Server 2010), where the destination number prefix begins with '+1', remove the '+1'.
- Index #3 defines destination number manipulation of IP calls from Lync Server. All calls received from Source IP Group 1 (i.e., from Lync Server 2010), where the destination number prefix begins with '+', remove the '+'.
- Index #4 defines destination number manipulation of IP calls from Verizon Business SIP Trunk. All calls received from Source IP Group 2 (i.e., from Verizon Business SIP Trunk), where the destination number prefix is '719', add the '+1' prefix to the number.

4.13.2 Configuring Source Phone Number Manipulation

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

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 sub-menu > Source Number Tel > IP).

Figure 4-31: 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 Digits to Leave	Presentation
1	0	-1	-1	•	+17193135662	2	0			255	Restricted
2	0	-1	-1	-	+1	2	0			255	Not Configured
4	0	-1	-1	-	anonymous	20	0	7193135662		255	Restricted
5	0	-1	-1	÷	+.	1	0			255	Not Configured

- Index #1 defines Source number manipulation of IP calls from any Source IP Group, where the Source number or prefix begins with '+17193135662'. In this case, remove the '+1' prefix from the number and set the Presentation to Restricted. This enables the customer to set restricted presentation for a specific DID or range of DIDs.
- Index #2 defines Source number manipulation of IP calls from any Source IP, where the Source number prefix begins with '+1'. In this case, remove the '+1' prefix from the number.
- Index #3 defines Source number manipulation of anonymous calls from any Source IP group. All calls received a Source number 'anonymous' are replaced with a known valid Verizon DID, such as '7193135662' and the presentation should be set to 'restricted'. This is a unique interworking to support the forwarding of calls originating from the PSTN and forwarded back to the PSTN when the original calling party was restricted. In this instance, the Lync Server 2010 sends the source as 'anonymous'.
- Index #4 defines Source number manipulation of IP calls from any Source IP Group where the Source number prefix is a '+'. In this case, remove the '+' prefix from the number.

4.13.3 Configuring 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 on page 73). In this case, the E-SBC device copies and adds the referred-by number to a newly created Diversion header. For a well known number in the Diversion header (for Verizon Business SIP Trunk), a manipulation rule should be defined to remove the (+1) from the redirect number to be presented as a well known number derived from the Referred-by header received from the Lync Server 2010 environment during the forwarding scenario.

To configure redirect number Tel -> IP Table:

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

Figure 4-32: Redirect Number Tel -> IP Page

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	•	+1	2	0			255

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

The redirect number is changed to remove the "+1" prefix from the number i.e. "+17193133661" to "7193133661".

4.14 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 sub-menu > General Parameters).

▼ SIP General ▶ 195.189.192.138 AT IP Address 2 PRACK Mode Supported v Channel Select Mode Cyclic Ascending ~ Enable Early Media ~ Enable 183 Message Behavior Progress ¥ Session-Expires Time 90 Minimum Session-Expires Session Expires Method Re-INVITE Asserted Identity Mode Adding PAsserted Identity ~ Fax Signaling Method G.711 Transport Detect Fax on Answer Tone Initiate T.38 on Preamble ~ TCP SIP Transport Type ¥ SIP UDP Local Port 5060 SIP TCP Local Port 5068 SIP TLS Local Port 5067 Enable SIPS Disable Enable TCP Connection Reuse Enable v TCP Timeout 0 5068 SIP Destination Port Use user=phone in SIP URL Yes ~ Use user=phone in From Header No ~ Use Tel URI for Asserted Identity Disable Tel to IP No Answer Timeout 180 Enable Remote Party ID Disable ~ Add Number Plan and Type to RPI Header Yes ¥ Enable History-Info Header Disable ~ Use Source Number as Display Name No Use Display Name as Source Number No Enable Contact Restriction Disable v Play Ringback Tone to IP Don't Play Play Ringback Tone to Tel Play Local Until Remote Media ~ Use Tgrp information Disable Enable GRUU Disable v User-Agent Information AudiocodesGW SDP Session Owner 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 Y Enable VoiceMail URI Disable ~ Retry-After Time 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 60 9 v Enable Reason Header Enable

Figure 4-33: SIP General Parameters Page

2. In the 'NAT IP Address' 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 **TCP** in case the Mediation Server is configured to use TCP transport Type.
- 6. In the 'SIP TCP Local Port' field, enter **5068**; this port is the listening E-SBC port for TCP 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.
- 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.
- **10.** Open the 'Admin" page, by appending the case-sensitive suffix 'AdminPage' to the E-SBC device'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.

Image Load to Device	Parameter Name IGNOREALERTAFTEREARLYMEDIA	Enter Value. 1	Apply New Value
Parameters Back to Main	Outp Parameter Mane: IGHOREALERTAFTEREARLY Parameter New Value:1 Parameter Descriptioniinterwork of Al	UL WINDOW NEDIA RET from 1500 to 51P	

Figure 4-34: INI file Output Window

4.15 Step 15: Configuring SIP Supplementary Services

This section describes how to configure the SIP general parameters.

- To configure general SIP Supplementary Services parameters:
- Open the 'Supplementary Services' page (Configuration tab > VoIP menu > GW and IP to IP sub-menu > DTMF and Supplementary sub-menu > Supplementary Services).

nable Hold	2 —		~	
nable Hold to ISDN		Disable	*	
Hold Format	3—			
Held Timeout		-1		
inable Transfer	4 —		*	
Transfer Prefix				
inable Call Forward	5	Enable	*	
Enable Call Waiting	6—		*	
look-Flash Code				
nable NRT Subscription		Disable	*	
AS Subscribe IPGroupID		-1		
IRT Subscribe Retry Time		120		
Call Forward Ring Tone ID		1		

Figure 4-35: SIP Supplementary Services Page

- 2. In the 'Enable Hold' drop-down list, select Enable.
- From the 'Hold Format' drop-down list, select 0.0.0.0 to enable Hold in the no media in either direction method or select Send Only to enable Hold to support one way audio for Music on Hold type service support.
- 4. From the 'Enable Transfer' drop-down list, select **Enable**.
- 5. From the 'Enable Call Forward' drop-down list, select **Enable**.
- 6. In the 'Enable Call Waiting' drop-down list, select **Enable**.

4.16 **Step 16: Defining Reasons for Alternative Routing**

A 503 SIP response from the Mediation Server to an INVITE must cause the E-SBC device 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 sub-menu > Alternative Routing Reasons).

Reasons for Alternative Rou	ting	
	IP to Tel Reasons	
	Reason 1	
	Reason 2	
	Reason 3	
	Reason 4	
	Tel to IP Reasons	
	Reason 1 2	503
	Reason 2	
	Reason 3	
	Reason 4	

Figure 4-36: Reasons for Alternative Routing Page

- 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	Homing	~
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	~
Gateway Name	Verizon.Lync.local	
Gateway Registration Name		
DNS Query Type	A-Record	~
Proxy DNS Query Type	A-Record	*
Number of RTX Before Hot-Swap	3	
Use Gateway Name for OPTIONS	No	*
User Name		
Password	Default_Passwd	
Cnonce	Default_Cnonce	
Registration Mode	Per Gateway	~
Challenge Caching Mode	None	¥
Mutual Authentication Mode	Optional	¥

Figure 4-37: Proxy & Registration Page

4.17 Step 17: Supporting Regional Call Progress Tone

The 'Load Auxiliary Files' page allows you to load various auxiliary files to the device. For Regional Call Progress Tones support, use the "Call Progress Tones file" area to select the appropriate regional file stored on your local computer to send to the device for proper support and interworking.

> To load an auxiliary file to the device using the Web interface:

 Open the 'Load Auxiliary Files' page (Maintenance tab > Software Update menu > Load Auxiliary Files).

INI file (incremental)	Browse Load File
CAS file	Browse Load File
Voice Prompts file	Browse Load File
🗲 Call Progress Tones file	Browse Load File
Prerecorded Tones file	Browse Load File
Prerecorded Tones file	Browse Load File
Dial Plan file	Browse Load File
User Info file	Browse Load File
AMD Sensitivity file	Browse Load File

Figure 4-38:'Load Auxiliary Files Page



Note: The appearance of certain file load fields depends on the installed Software Upgrade Key.

- 2. Click the **Browse** button corresponding to the file type that you want to load, navigate to the folder in which the file is located, and then click **Open**; the name and path of the file appear in the field next to the **Browse** button.
- 3. Click the **Load File** button corresponding to the file you want to load.
- 4. Repeat steps 2 through 3 for each file you wish to load.
- 5. Save the loaded auxiliary files to flash memory, and reset the device (if you have loaded a Call Progress Tones file).



Reader's Notes

5 Troubleshooting

This section should provide some tips for troubleshooting problems, including troubleshooting commands and case reporting procedures for trouble escalation.

5.1 Debugging Procedures

This section discusses the following debugging procedures:

- Case Reporting Procedures. See Section 5.1.1 below.
- Syslog. See Section 5.1.2 on page 96.
- Wireshark Network Sniffer. See Section 5.1.3 on page 98.

5.1.1 Case Reporting Procedures

When reporting a problem to AudioCodes' Technical Support department, the following information should be provided:

- Basic information (required for all types of problems):
 - Problem description (nature of failure, symptoms, call direction, etc.)
 - Network diagram
 - *ini* configuration file (downloaded to your PC from the device using the Web interface)
 - Syslog trace (without missing messages)
 - Unfiltered IP network trace using the Wireshark application

(Note: If you are unable to collect all the network traffic, then at least collect the mandatory protocols SIP, RTP, and T38.)

- Advanced information (if required upon request):
 - PSTN message traces for PSTN problems
 - Media stream traces for problems related to voice quality, modem\fax, DTMF detection, etc.

5.1.2 Syslog

Syslog is a standard for forwarding log messages in an IP network. A syslog client, embedded in the device sends error reports/events generated by the device to a remote Syslog server using IP/UDP protocol. This information is a collection of error, warning and system messages that record every internal operation of the device. You can use the supplied AudioCodes proprietary Syslog server "ACSyslog" (shown in the figure below) or any other third-party Syslog server for receiving Syslog messages.

🎒 ACSyslog ;-)	R0.9.9		×
File View Search	Options Help		
🕞 💽 🔮	ن 👟 🤇	$\mathbf{\overline{O}}$	
🕐 Time	📃 Hast	🚫 Priority	Message
13:29:29.510	10.13.4.13	WARNING	(lgr_psbrdif)(20) !! [ERROR] #1:failed to play tor
13:29:29.510	10.13.4.13	WARNING	Invalid Tone Type (7). Channel ID:1 [Code:500c][CID:
13:29:21.949	10.13.4.13	NOTICE	(lgr_endpoint)(19) FXSEndPoint::HandleDialedStr
13:29:21.949	10.13.4.13	NOTICE	(lgr_digitmap_mngr)(18) DigitMapMngr::HandleDia
13:29:17.200	10.13.4.13	WARNING	(lgr_psbrdif)(17) !! [ERROR] #1:failed to play to
13:29:17.200	10.13.4.13	WARNING	Invalid Tone Type (7). Channel ID:1 [Code:500c][CID:
<			>
0 message(s) per min	ute (6/26)	0.9.9/Unkno	own 6 6 0 -

Figure 5-1: AudioCodes' Proprietary Syslog Server

> To activate the Syslog client on the device using the Web interface:

- Open the 'Syslog Settings' page (Configuration tab > System menu > Syslog Settings).
- 2. In the 'Syslog Server IP Address' field, enter the IP address of the Syslog server (*ini* file parameter SyslogServerIP).
- **3.** From the 'Enable Syslog' drop-down list, select 'Enable' to enable the device to send syslog messages to a Syslog server (defined in Step 2).

Figure 5-2: Enabling Syslog

Enable Syslog	Disable	~
Syslog Server IP Address		
Syslog Server Port	514	
Debug Level	0	*
Analog Ports Filter	-1	
Trunks Ports Filter	-1	

4. From the 'Debug Level' drop-down list, select '5' if debug traces are required. To enable syslog reporting, using the *ini* file, load an *ini* file to the device with the following settings:

```
[Syslog]
SyslogServerIP = 192.168.2.35
EnableSyslog = 1
SyslogServerPort = 514
GWDebugLevel = 5
```

5.1.3 Wireshark Network Sniffer

Wireshark is a freeware packet sniffer application that allows you to view the traffic that is being passed over the network. Wireshark can be used to analyze any network packets. Wireshark can also be used to analyze RTP data streams and extract the audio from the data packets (only for G.711). The audio can be saved as a *.pcm file.

> To record traffic sent to / from the device:

- 1. Install Wireshark on your PC. (You can download it from the following Web site: http://www.wireshark.org/
- 2. Connect the PC and the device to the same hub.
- **3.** If you are using a switch, use a switch with port mirroring for the port to which the Wireshark is connected.
- 4. Start Wireshark.
- Select the network interface that is currently being used by the PC on the toolbar,click Interfaces, and then in the 'Capture Interfaces' dialog box, click the Options button corresponding to the network interface:

Figure 5-3: Selecting Interface Currently used by the PC

Go Capture Analyze	Statistics Help							
s Start Stop	Restart Open	Save As	X Close	Refresh		Ard	de Bark	,
ble capture interfaces]			• Expr	ession (Jear Apply			
Wireshark: Gap	ture Inter/aces				1			
	escription	IP	Packets F	Packets/s	Stop			
Adapter for gener	ric dalup and VPN capture	unknown		4	Start Options	Details		
Broadcom NetXtre	me Gigabit Ethernet Drive	10.13.22.6	1.1+	199	Start Options	Detaik		
Help						lose		
	Start Sop Start Sop Contraction Sop Contractio	Copen Start S		Copen Sive A Copen Start Stop Restort Open Sive As Clear Expr Wiresharks: Capture Interfaces. Description IP Packets I Adapter for generic dalup and VPN capture Uniction 0 Broadcom Net/Orreme Gigabit Ethernet Driver 10.13.22.6 114 Bep	Copen Save As Case Refresh Start Stop Restart Open Save As Case Refresh Copen S	go gobure Analyze gobusts (gob Start Stop Start Stop	Start Sop Rester Open Save & Case Perfect Perfect Start Sop Rester Open Save & Case Perfect Perfect Expression Clear Apply Vireshark: Capture Interfaces Description IP Packets Packets/S Stop Adapter for generic dalup and VPN capture uninoun 0 Start Options Details Broadcom NetXtreme Gigabit Ethernet Driver 10.13.22.6 114 1 Start Options Details Help	Start Stop Restart Open Save As Cleve Refrech Per Pro Bask Start Stop Restart Open Save As Cleve Refrech Per Pro Bask Expression Clear Apply Wireshark: Capture Interfaces Description IP Packets Packets/s Stop Adapter for generic dalup and VPN capture Unicoun 0 0 Start Options Details Broadcom NetXtreme Gigabit Ethernet Driver 10.13.22.6 11 0 Start Options Details Become MetXtreme Gigabit Ethernet Driver 10.13.22.6 11 0 Start Options Details Become MetXtreme Gigabit Ethernet Driver 10.13.22.6 11 0 Start Options Details Become MetXtreme Gigabit Ethernet Driver 10.13.22.6 11 0 Start Options Details Become MetXtreme Gigabit Ethernet Driver 10.13.22.6 11 0 Start Options Details

6. In the 'Capture Options' dialog box, select the desired display options:

Figure 5-4: Configuring Wireshark Display Options

Capture		
Interface: Broadcom	NetXtreme Gigabit Ethernet Driver: (D	evice\/NPF_{20F728EA-2789-45E1-8
P address: 10.13.22.6		
Unit-layer header type:	Ethernet 🤤 Buffer size: 1	🗧 megabyte(s) Wreless Settings
Capture packets in pr	omiscuous mode	
Linit each packet to	a bytes	
Gapture Filter:		•
Capture File(s)	~	Display Options
File;	Browse	📄 🗹 Update list of packets in real time
Use multiple files		
Tiest the avery	1 negabytł(s)	Automatic smolling in live capture
Tiest fie every	1 minute(s) M	📄 🗹 Hide capture info dialog
Ring butter with	2 tits	Name Resolution
Dop capture after	1 (1)(44)	
Stop Capture		Enable MAC name resolution
🗋 after 👔	2 metat(s)	Enable getwork name resolution
after	P requbstr(s) ~	
C alter	2 merutetat	Enable transport name resolution

99



7. Click Start.

Figure 5-5: Captures Packets

a flor Day No.		shine strate	and service	Control 2 and to	CONTROL NO.						
	E start	Stop	Restart	Dom:	8	*		Pret	E Find	÷.	
et sp		-	_	Filter	Bar	ion., De	ar 6001				
Time	50	N/DR		Destrution	1	Protocol	Info				
163 09:35:22. 164 09:35:22. 209 09:35:22. 210 09:35:22.	127114 10 138134 10 362899 10 393918 1	0.33.6.1 0.33.6.1 0.33.67 0.33.67	01	Packet li	st pane	SIP SIP SIP SIP	Status Status Request status	100 Tryl 180 R1ng : PRACK 3 200 OK	ng 11ng 1p:201010	0.33.6.10	1
Internet Proto User Datagram Session Initia	col, Src Protocol tion Pro	: 10.33. , Src Po tocol	6.100 () rt: 5060	10.33.6.10 (1060), 1	0), Ost: Ost Port:	10.33.6.	101 (10. 060)	33.6.101)	Ale colore		
INVITE Sip:2 Via: SIP/2.0 Max-Forwards From: <sip:1 To: <sip:201 Call-ID: 649 Cep:1 INVI</sip:201 </sip:1 	01010.33 /UDP 10. : 70/r/m 01010.33 010.33.6 56707171	.6.101:0 33.6.100 .6.100>; .101:054 20002349	ser-pho ibranc tag-1c r-phone 25010, 3	Packet det	ir\n ails pan	r'n					
Contact: csi	p:101010	.33.6.10	n/1/40	nath rate	rea.net	etniele					
	13 64 0	3 30 1	61 49	4e 56 49 5	4 45	e	INVITE				

8. To view VoIP call flows, from the **Statistics** menu, choose **VoIP Calls**. You can view the statistics in graph format by clicking **Graph**.

Figure 5-6: Viewing VoIP Call Flows

teres Options	al. Set		*	D Ciper	E2 Sam Au		- Andread	-	8	-	4	* 3	2
-		1			• 5ar	eatr. 9	er bah			10 30 X 000			
52 08:58:4 89 08:58:4 10 08:58:4 10 08:58:8 11 08:58:8 11 08:58:8 11 08:58:8	with the	100.6			Cuel	utud (ice	cal Salachet	i Cali	15.714 (5.763 (5.767 (5.768 (5.609		El Yolng El Yolng El Yolng Have 200 K		riges 2.4 million is no 2mgm
200 0013015 395 0013015 395 0013015	jan inne -		July Zone					Pro-	12.540 12.574 12.674 12.676 12.676 17.7% 17.7%		nti Aleghine e rightskj ale rightsj Brit 2008	eri in taka Alf taka Ur falasi Ur falasi Ur falasi Ur falasi	n (M. Domont Sile on A In (M. Domont The on S
		-		Seeded Cal	Color april	peters a	05 To april 1 Corported cal	010.33.6 In (Fere					

9. To play G.711 RTP streams, click the **Player** button.

Figure 5-7: Playing G.711 RTP Streams



10. To analyze the RTP data stream and extract the audio (which can be played using programs such as CoolEdit) from the data packets (only for G.711), from the **Statistics** menu, point to **RTP**, and then choose **Stream Analysis**.



a- [tip or rtp					• (um	on. ges	r bosh							
	300	504	n.e		Deutshalor	6	Protocol	24s							
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	9.08158140.1	64885 10.	. 33. 5.	101	10.31	A CONTRACTOR	stray in the							• •	6.13
1.10	0 00130140.3	700078 201		101	10.31										
-	NO. 13. 1 17 1. 18	ALL COMPANY	_	100	20, 81			Defected 2 81P do	are. Ches	e one for form	and and revenue devi-	tion for and	lain i		
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1.14	310915814425	PO743 14.	33.4.	100	10.33	10.314.201	8000	10.35.6.107	0000	844750414	ITU1 6-711 POM	264	0 (D.P%)	20.21	
- 23	1 08158147.0	000000 10.	.33.6.	101	10,33	10.314.000	1000	10.33.6.101	6000	2025767996	TTUR G.711 PONA	258	0 (0.7%)	20.26	38
- 23	0.08138147.0	22613 10.	33.6.	100	10.33	1000100000									643
- 22	2 08:56:47.0	Contractor.		THE	101. 111						and the second second	ALC: NO			
- 55		- 10 Wi	res.hark	HITP Stream	a description							1000			C 10
23	5 08158147.0											1			177
23	6 08156147.0	Mar Parva	ed Denco	on Revense	nd Demictation								Audio	L. Owner	
23	7.08158147.0	4621		Andres	or stream how	10,123 & 101 mid	abor to 1	0.10.6.300 met 40	05 118c -	Transla		i de		- Seree	- 193
23	8 08:58:47.0	1943		an after		e prosecuti por	and of the	or so at 150 har an	on himse				-		
14.0	9.08:38:47.3	Pade Pade	et al 12	Segence	Delta (m)	381er (rei)	BV 04e	ps Marker	204		0	10 A	Seg-43	43, T388+21	825709
	0 08:59:47.1		147	44000	0.00	8.00	1.60			7.06.1		1.00	Sequered	388, T188-51	-17.5
	2 08158147.1	11	211	44083	19.59	5.01	3.20			CORS		a sata	Carriela	100 Theat	01018
1.0	3 08158147		232	4-054	20.19	6.02	+.80			[08]	-	The Party of	Same 1	47. 1100.2	124741
24	4 08158147-1	4.64	214	408	19.40	8.03	4.40			[OK]		3414.	5412-140	DO, Thes-SI	24078
24	5 08158147.1	621	236	+006	20.85	1.03	8.00			[OR]		97990.	5eq-43	48, Time-21	626757
2.4	6 08:58:47.5	643	239	+057	19.99	1.03	1.60			[Ok]		3414.	Seg-140	MI, Thursd	24218
- 24	7.98155147,1	821	240	+4000	19.99	0.02	18,259			(OR)		87990.	Seq-43	41, Time=21	626773
24	8.08:39:47,3		25	4007	19.00	0.003	12.00			100		3414.	260-140	92, T5mm-51	24398
1.64	M 08:38:47.1	100	246	44791	20.69	0.03	16.00			1001		acres.			
87.00	ma 138 / 510	The second se	240	+032	19.10	0.04	17.60			1061		_	-		-
100	Arrest IT. St		250	#4090	20.45	3.04	18.20			(OK)					
1000	strat bits	23 U	252	4-094	20.80	6.03	20.00			(Ok]					
Lun I	anitani d		274	+4295	19,79	0.07	42,40			[Ok.]					
	ander Jacori		256	+0%	20.13	0.03	24.00			108.1					
- 23	Affarant late		00	44097	19.03	A.03	25.60			100.1	-				
* 2	of all a specific			Hardet	**8.02011	sec at parlet no.	00	999 200 Day	112100	and the second					
- 23	dept 161/41 1/			54441	P patients + 2	63 Emiperhed 26(Lost RTP	pachetts = 0 (0.309	L) Selburi	a arrors = 0					
	Antes Color														
* 2.	rags: 0x00	Chair Chair	and and	ALC: No. of	(W)	Hartunda II	Anna lin	(Gast	1.4	dimente 1	- Own				
-	ins to Marti		and a second				- and - a		-						
	100 20 1168	-	_												

- a. Save the audio payload of the RTP stream to a file.
- **b.** Save the Payload as a *.pcm file.
- c. Select the 'forward' option.

5.2 Verifying Firmware

To verify the firmware load actively running on the device, log into the device and view the firmware version on the product homepage as shown in the figure below.

Figure 5-9: Viewing active firmware version



Α

Appendix: Configuring Analog Devices (ATA's) for FAX Support

This section describes how to configure the analog device entity to route its calls to the AudioCodes Media Gateway for supporting faxes. The analog device entity must be configured to send all calls to the Media Gateway without any registration process.



Note: The configuration described in this section is for ATA devices configured for AudioCodes MP-11x series.

A.1

Step 1: Configure the Endpoint Phone Number Table

The 'Endpoint Phone Number Table' page allows you to activate the MP-11x ports (endpoints) by defining telephone numbers. The configuration below uses the example of ATA1 destination phone number "7193135664" (IP address 10.15.7.165) with all routing directed to the AudioCodes E-SBC device (10.15.4.30).

To configure the Endpoint Phone Number table:

Open the 'Endpoint Phone Number Table' page (Configuration tab > VoIP menu > GW and IP to IP submenu > Hunt Group sub-menu > Endpoint Phone Number).

point	Phone Number Ta	ble			
	Chann	el(s)	Phone Number	Hunt Group ID	Tel Profile ID
1	1		7193135664		0
2					
3					
4					

Figure 5-10: Endpoint Phone Number Table Page

A.2 Step 2: Configure Tel to IP Routing Table

This step describes how to configure the Tel-to-IP routing rules to ensure that the MP-11x device sends all calls to the AudioCodes central E-SBC device.

To configure the Tel to IP Routing table:

Open the 'Tel to IP Routing' page (Configuration tab > VoIP menu > GW and IP to IP sub-menu > Routing sub-menu > Tel to IP Routing).

Tel to IP Routing								
						Basic P	arameteri	List 🔺
	▼							
	Routing Index		1-10 💙					
	Tel To IP Routing Mode		Route calls	s before manipulatio	n 🗸			
Src. Trunk Group ID Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Port	Transport Type	Dest. IPGroup ID	Dest. SRD	IP Profile ID	
1 * *	* 10	0.15.4.30 50	060	UDP 💌	-1	-1	0	
2				Not Configured 💌	-1			[
3				Not Configured 💌	-1			[
4				Not Configured 💌	-1			I

Figure 5-11: Tel to IP Routing Page

A.3 Step 3: Configure Coders Table

This step describes how to configure the coders for the MP-11x device.

To configure MP-11x coders:

Open the 'Coders' page (Configuration tab > VolP menu > Coders And Profiles sub-menu > Coders).

Coder Name		Packetization Time		Rate		Payload Type	Silence Suppression	
G.729	*	20	*	8	~	18	Disabled	*
G.711U-law	*	20	*	64	~	0	Disabled	<
	*		*		~			*
	*		~		~			*
	*		*		~			~
	*		*		~			*

Figure 5-12: Coders Table Page

A.4 Step 4: Configure SIP UDP Transport Type and Fax Signaling Method

This step describes how to configure the fax signaling method for the MP-11x device.

> To configure the fax signaling method:

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

General Parameters		
Channel Select Mode	By Dest Phone Number	
Enable Early Media	Enable	
183 Message Behavior	Progress	
Session-Expires Time	0	
Minimum Session-Expires	90	
Session Expires Method	Re-INVITE 💌	
Asserted Identity Mode	Disabled 🗸	
Fax Signaling Method	G.711 Transport	
Detect Fax on Answer Tone	Initiate T.38 on Preamble	
SIP Transport Type		
SIP UDP Local Port	5060	
SIP TCP Local Port	4 5868	
SIP TLS Local Port	5067	
Enable SIPS	Disable 🗸	
Enable TCP Connection Reuse	Enable	
TCP Timeout	0	
SIP Destination Port	5060	

Figure 5-13: SIP General Parameters Page

- 2. From the 'FAX Signaling Method' drop-down list, select G.711 Transport.
- 3. From the 'SIP Transport Type' drop-down list, select UDP.
- **4.** In the 'SIP UDP Local Port' field, enter **5060** (corresponding to the Central Gateway UDP transmitting port configuration).
- **5.** In the 'SIP Destination Port', enter **5060** (corresponding to the Central Gateway UDP listening port configuration).

Reader's Notes



Configuration Note

www.audiocodes.com