Enterprise Session Border Controllers (E-SBC)

AudioCodes[™] Mediant[™] Series

Interoperability Lab

Configuration Note

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



Microsoft Partner







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Notice

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

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1 Introduction

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

1.1 Intended Audience

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

1.2 About AudioCodes E-SBC Product Series

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

The E-SBC provides perimeter defense as a way of protecting Enterprises from malicious VoIP attacks; mediation for allowing the connection of any PBX and/or IP-PBX to any service provider; and Service Assurance for service quality and manageability.

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

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

1.3 About AudioCodes MediaPack 11x Product Series

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

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

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

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



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2 Component Information

2.1 AudioCodes E-SBC Version

Table 2-1: AudioCodes E-SBC Version

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

2.2 AudioCodes MediaPack Version

Table 2-2: AudioCodes E-SBC Version

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

2.3 SIP Trunking Version

Table 2-3: Swisscom Version

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

2.4 Microsoft Lync Server 2013 Version

Table 2-4: Microsoft Lync Server 2013 Version

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

2.5 Interoperability Test Topology

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

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



The figure below illustrates this interoperability test topology:





2.5.1 Environment Setup

The interoperability test topology includes the following environment setup:

Table 2-5: Environment Setup

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



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

2.5.2 Known Limitations

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



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3 Configuring Lync Server 2013

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



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

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

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

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



Figure 3-1: Starting the Lync Server Topology Builder



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

The following is displayed:

Figure 3-2: Topology Builder Dialog Box

26	Topology Builder	×				
Welcome to Topology Builder. Select the source of the Lync Server topology document.						
Down	nload Topology from existing deployment					
Retriev store a existin	Retrieve a copy of the current topology from the Central Management store and save it as a local file. Use this option if you are editing an existing deployment.					
O Open	Topology from a local file					
Open in proj	Open an existing Topology Builder file. Use this option if you have work in progress.					
O New T	Topology					
Create a blank topology and save it to a local file. Use this option for defining new deployments from scratch.						
Help	ОК	Cancel				

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

Figure 3-3: Save Topology Dialog Box

o o - 🕇 🖪	 Lib 	raries > Documents	v c	Search Documents	Q
Organize • New f	older			8::	• 0
🚖 Favorites	^	Name		Date modified	Туре
Desktop		Network Monitor 3		21.05.2013 10:33	File fol
Downloads		20121120.tbxml		20.11.2012 15:54	TBXM
🔢 Recent places	-	20121120-2.tbxml		20.11.2012 22:48	TBXM
	-	20121130.tbxml		30.11.2012 11:55	TBXM
Calibraries		20121218-1.tbxml		18.12.2012 16:26	TBXM
Documents		20130215.tbxml		15.02.2013 08:07	TBXM
J Music		20130402.tbxml		02.04.2013 13:30	TBXM
Pictures		20130521.tbxml		27.05.2013 15:25	TBXM
Videos		20130528.tbxml		28.05.2013 15:04	TBXM
		20130918.tbxml		18.09.2013 09:59	TBXM
: Computer	× .	< 111			>
File name:					
Save as type:	Topolo	gy Builder files (*.tbxml)			,

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

The Topology Builder screen with the downloaded Topology is displayed:

6	Lync Server 2013, To	pology B	uilder	>
File Action Help				
A Junc Server Junc Server	SIP domain			•
 Lync Server 2010 Lync Server 2013 Standard Edition Front End Servers Enterprise Edition Front End pools Director pools Mediation pools 	Default SIP domain: Additional supported SIP domains: Simple URLs	siptesti Not co	03.local nfigured	
Edge pools Trusted application servers Shared Components	Phone access URLs:	Active	Simple URL https://dialin.siptest03.local	
Branch sites	Meeting URLs:	Active	Simple URL	SIP domain
	Administrative access URL:	https://	https://meet.siptest03.local /admin.siptest03.local	siptest03.local
	Central Management Serv	ver		•
	Central Management Server:	Active	Front End	Site
		1	rint3count01 sintest03 local	Zürich

Figure 3-4: Downloaded Topology

- 4. Under the Shared Components node, right-click the PSTN gateways node, and then from the shortcut menu, choose New IP/PSTN Gateway, as shown below:
 - Figure 3-5: Choosing New IP/PSTN Gateway

16	Lync Server 2013, Topology Builder
File	Action Help Image: Server 2010 Image: Server 2013 Image: Server 2013 Image:
	PSTN gateway. 195.17 Define a new IP/PSTN gateway. 195.17 Help
	Carl Office Web Apps Servers



The following is displayed:

Figure 3-6: Define the PSTN Gateway FQDN

16	Define New IP/PSTN Gateway		×
5	Define the PSTN Gateway FQDN		
Define t	e fully qualified domain name (FQDN) for the PSTN gateway.		
sipgate	way01.siptest03.local		
Help	Back	Next	Cancel

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

Figure 3-7: Define the IP Address

16	Define New IP/PSTN Gateway	×
5	Define the IP address	
• Enable	IPv4	
🖲 Us	e all configured IP addresses.	
O Lin	nit service usage to selected IP addresses.	
PS	TN IP address:	
O Enable	IPv6	
🖲 Us	e all configured IP addresses.	
O Lin	nit service usage to selected IP addresses.	
PS	TN IP address:	
Help	Bark Navt	Cancel
Heip	Back IVext	Cancel

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

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

Notes:

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

Figure 3-8: Define the Root Trunk

d.	Define New IP/PSTN Gateway	×
5	Define the root trunk	
Trunk n	ame:*	
sipgat	eway01.siptest03.local	
Listenin	g port for IP/PSTN gateway: *	_
506d		
SIP Tran	isport Protocol:	-
TCP		•
Associa	ted Mediation Server:	
sipt3s	vlfe01.siptest03.local Zürich	•
Associa	ted Mediation Server port: *	
5068		
Help	Back Finish Cancel	
		-

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



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

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

28	Lync Server 2013, Topology Builde	×
File Action Help		
 Lync Server Zürich Lync Server 2010 Lync Server 2013 Shared Components SQL Server stores File stores File stores PSTN gateways Trunks Trunks 195.176.152.148 195.176.152.154 SipTrunk-Lync2013 SipTrunk-Lync2013 Sipgateway01.siptest03 Office Web Apps Servers Branch sites 	Incel	sipgateway01.siptest03.local sipgateway01.siptest03.local (Z0rich) 5060 TCP sipt3srvffe01.siptest03.local (Z0rich) 5068

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

18	Lync	Server 2013, T	opology B	uilder	_ _ X
File	Action Help New Central Site Edit Properties				•
	New Topology Open Topology Download Topology Save a copy of Topology As		c siptes103.local ed Not configured		
	Publish Topology				
	Merge Office Communications Server 2007 R2 Topolog Remove Deployment	9/	Active	Simple URL	
	Help	1111	~	https://dialin.siptest03.local	
	SQL Server stores Meeting	ing URLs:	Active	Simple URL	SIP domain
	P I le stores Administrative ac D Trunks URL: Office Web Apps Servers Branch sites Central Management		https://	https://meet.siptest03.local /admin.siptest03.local	siptest03.local
			erver		•
	Centra	al Management	Active	Front End	Site
	Serve	n	1	sipt3srvof01.siptest03.local	Zürich

Figure 3-10: Choosing Publish Topology

The following is displayed:

Figure 3-11: Publish the Topology

ő	Publish Topology	×
Pu	blish the topology	
In o top	order for Lync Server 2013 to correctly route messages in your deployment, you must publish your ology. 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 Persistent Chat pools and for Monitoring Servers and Archiving Servers: All SQL Server stores are installed and accessible remotely, and firewall exceptions for remote access to SQL Server are configured.	
•	For a single Standard Edition server, the "Prepare first Standard Edition server" task was completed. You are currently logged on as a SQL Server 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	
Wh	en you are ready to proceed, click Next.	×
	Help Back Next Cancel	

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

Figure 3-12: Publishing in Progress

16	Publish Topology	×
P	ublishing in progress	
PI	ease wait while Topology Builder tries to publish your topology.	
P	vominouumy voporogy	
s	Succeeded	
	Downloading global simple URL settings	
s	Succeeded	
l	Updating role-based access control (RBAC) roles	Ξ
s	Succeeded	
1	Enabling topology	v
		_
	Back Next Cance	<u> </u>



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

Figure 3-13: Publishing Wizard Complete

18	Publish Topolo	ogy	×
Put	lishing wizard complete		
Your	topology was successfully published.		
	Step	Status	
****	Publishing topology Downloading topology Downloading global simple URL settings Updating role-based access control (RBAC) roles Enabling topology	Success Success Success Success Success	⊻iew Logs
To d	lose the wizard, click Finish. Ielp	<u>B</u> ack <u>Finish</u>	Cancel

11. Click Finish.

3.2 Configuring the "Route" on Lync Server 2013

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

> To configure the "route" on Lync Server 2013:

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

Figure 3-14: Opening the Lync Server Control Panel





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



You are prompted to enter your login credentials:

Figure 3-15: Lync Server Credentials

	Windows Security X					
AdminUIH Connecting to	OST sipt3srvof01.siptest03.local.					
P	User name Password Domain: SIPTEST03 Remember my credentials					
	Connect a smart card					
	OK Cancel					

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

Figure 3-16: Microsoft Lync 3	Server 2013 Control Panel
-------------------------------	---------------------------

-	Microsoft Lync Server 201	13 Control Panel
Lync Server 2013		Administrator Sign out 5.0.8308.420 Privacy statement
👌 Home		
2 Users		
M Topology	User Information	Resources
D IM and Presence	Welcome, Administrator	Getting Started
Persistent Chat	 View your roles 	First Run Checklist Using Control Panel
😍 Voice Routing	Top Actions	Microsoft Lync Server 2013
& Voice Features	Enable users for Lync Server	Using Office 365
23 Response Groups	Edit or move users View topology status	Getting Help Downloadable Documentation
Oconferencing	 View Monitoring reports 	Online Documentation on TechNet Library Lync Server Management Shell
G Clients		Lync Server Management Shell Script Library Lync Server Resource Kit Tools
Federation and External Access		Community
Monitoring and Archiving		Forums Blogs
A Security		
Network Configuration		
the second se		

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

Figure 3-17: Voice Routing Page

2		Micros	oft Lync	Server 2013	Control Panel		- • ×
Ly	nc Server 2013						Administrator Sign out 5.0.8308.420 Privacy statement
<u>م</u>	Home	Dial Plan Voice Policy Route	PSTN Us	age Trunk Co	nfiguration Test Voice	Routing	
84 24	Topology	Create voice routing test case int	ormation				•
©	IM and Presence Persistent Chat				م		
6	Voice Routing	New V Edit V Act	Scope	State	Normalization rules	Description	0
23	Voice Features Response Groups	Global 23 All Numbers	Global User	Committed Committed	7	All Numbers	
0	Conferencing	23 E911Test 23 SIPTestAudioCodes2013	User	Committed	1		
	Federation and External Access	-					
	Monitoring and Archiving						
-	Security Network						
ŵ	Configuration						

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

Figure 3-18: Route Tab

12		Microsoft	Lync Server 2013 Control Panel	_ 0 ×
Ly	nc Server 2013			Administrator Sign out 5.0.8308.420 Privacy statement
	Home	Dial Plan Voice Policy Route P.	STN Usage Trunk Configuration Test	Voice Routing
22	Users	Create voice routing test case informa	ation	×
и	Topology	(C)		1
₽	IM and Presence		٩	
8	Persistent Chat	🔶 New 🥖 Edit 🔻 🔮 Move up	> 🐣 Move down Action 🔻 Co	nmit 🕶 😡
8	Voice Routing	Name	State PSTN usage	Pattern to match
8	Voice Features	LocalRoute	Committed	^(\+1[0-9][10])\$
22	Response Groups	All Numbers	Committed All Numbers	1
Q	Conferencing	EmergencyRoute	Committed EmergencyUsage	^\161\$
5	Clients	Lync2013	Committed Lync2013	1
ā\$	Federation and External Access			
	Monitoring and Archiving			
-	Security			
Ŷ	Network Configuration			



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

Figure 3-19: Adding New Voice Route

V K K Cancel		
Scope:		i i i
Al Numbers		
Description		
All Numbers		
Build a Pattern to Match	-	
Add the starting digits that you want to the expression manually by clicking Edi	sis route to handle, or create 1.	
Starting digits for numbers that you wa	ent to allow:	
*	A50	
	Deceptions	
	Remove	
Match this pattern: *		
1		
Frank Robert (1)		
The I was to		
Suppress caller ID		
Alternate caller ID:		

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

2		Microsoft Lync Server 2013 Control Panel	- • ×
Lyı	nc Server 2013		Administrator Sign out 60.6306-420 Privacy statement
<u>م</u>	Home	Dial Plan Voice Porky Route PSTN Usage Trunk Configuration Test Voice Routing	
22	Users	Create voice routing test case information	×.
м	Topology	(
qp.	IM and Presence	New Voice Route	
2	Persistent Chat	✓ OK X Canod	0
12	Voice Routing	tat front 🕐	
0	Voice Features		
23	Response Groups	L Suppress caller ID	
90	Conferencing		
5	Clients	Associated trunks:	
ā.	Federation and External Access	Att.	
-	Monitoring and Archiving		
	Security	Associated PSTN Usages	
÷.	Network	Mi Select Remove 🎓 👵	
	Configuration	PSTN usage record Associated voice policies	
		Translated number to test: 60	Ļ

Figure 3-20: Adding New Trunk

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

2			Microsoft Lyne	Server 2013 Control Pa	anel	= 0 X
Ly	nc Server 2013					Aproximation Sign and Sold201423 Privacy statement
-	Home	Dial Pan				
12	Users	Create vo	ice routing test case aformation			•
21						
	IM and Presence	PAGA N	Select Trunk			
9	Persistent Dist					9
					Q	
ć.	Voice Features		Service	Site		
74	Response Groups	0	PstrGateway 192,168.3.2	35 Zirio		
-	Conferencing		PutrGeteway:195.176.15	2.145 Zürich		
-		Ame	Pstr:Gateway:195.176.15	5212 Zirich		
	Federation and		PatriGateway:195.176.15	2.154 Zürich		
	External Access		PstrGateway:SigTrunk2	. Zirich		
	Monitoring and Archeving		PstrGatewaysipTrunk	Zino		
4			PstrGatewaysiggiteway	IT SIDE		
-						
*						
				04	Cancel	
			slated number to test:			
					In case of the local division of the local d	
						ł

Figure 3-21: List of Deployed Trunks

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

Figure 3-22: Selected E-SBC Trunk

9		Microsoft Lync Server 2013 Con	trol Panel	= 0 x
Lyr	nc Server 2013			Administrator Sign out
9	Home	Dia Pan Wole Policy Bank PSTN Usage Trunk Configu	ration Test wike Rowling	
22	Users	Create voice routing test case information		¥
м	Topology			
φ	IM and Presence	New Voice Route		
3	Persistent Chat	√ CK X Canot		
12	Voice Routing	tot not 🕐		
5	Voice Features			
14	Response Groups	L_ Suppress caller ID		
90	Conferencing			
5	Clients	Associated tranks:		
24	Federation and External Access	hstrifetenanspig.	Act	
-	Monitoring and Archiving		(Constant)	
	Security	Associated FSTN Usepes		
	Network	Nitela Amere 9 &		
-	Configuration	PS7N snaps record Associated write policie		
		Translated number to lost:	60	
				8



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

0		Micros	oft Lync Server 2013 Control	Panel	
Ly	nc Server 2013				Administrator Sign o 50.6306.425 Privacy stateme
4	Home	Dal Pan Voice Policy Route	PSTN Usage Trunk Configuration	Test Voke Routing	
12	Users	Create voice routing test case info	mation.		v
4	Topology				
Ð	IM and Presence	New Voice Route			
3	Persistent Chat	🖌 OK 🗰 Cancel			0
•	Voice Routing	Edt Reset	۲		•
	Voice Features				
:	Response Groups	Suppress caller ID			
D	Conferencing	Afternate caller ID:			
Ъ	Clients	Associated tranks:			
5	Federation and External Access	PstrGatewaysipg.		Add.	
	Monitoring and Archiving			- ABRICAL	
1	Security	Associated PSTN Usages			
	Network	Select. Remove	* *		
	Configuration	PSTN usage record	Associated voice policies		
		All Numbers			

Figure 3-23: Associating PSTN Usage to Route

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

Figure 3-24: Confirmation of New Voice Route

[٩	
🗣 New 🧪 Edit 🔻 👚 Mov	/e up 🛛 🦺 Move dow	n Action 🔻 Com	mit 🔻	0
Name	State	PSTN usage	Pattern to match	
SIP Trunk Route	1 Uncommit	ted Local, Internal	v/*	

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

Figure 3-25: Committing Voice Routes

[٩	
🗣 New 🧪 Edit 🔻 👚 Move up	Antice Move down	Action 🔻	Commit 🔻	0
Name	State	PSTN usa	Review uncommitted changes	
SIP Trunk Route	Uncommitted	Local, Inte	Commit all	

The Uncommitted Voice Configuration Settings page appears:



Unco	mmitted Voice Configuration	on Setting	s	😢 🗙
Re	outes			*
	Identity	Action	New value (pattern to match) Old value (pattern to match)	
	SIP Trunk Route	Added	\/*	
			6	mmit Cancel
				Curreer

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

Figure 3-27: Confirmation of Successful Voice Routing Configuration

8	Microsoft Lync Server 2013 Control Panel	
Lync Server 2013		Advisionator Sign out
A Home	Diel Plan Vese Policy 10/21 PSTN Usage Trunk Configuration Text Vace Routing	
AL Users	Create voice routing test case information	×
M Topology	6	
G IM and Presence	A	
Persistent Chat	And Anna Anna Anna Anna	
C Voice Routing	Name Sule FSIN usage Pattern to make	
C. Voice Features	Localitante Converted 1/2+1/3-5(11)(5	
Response Groups	Al Numbers Committed Al Numbers /	
Conferencing	Energecylaze Constitut Instandium Aldrid	
Clients	Lync2813 Microsoft Lync Server 2013 Control Panel	
Federation and External Access	 Successfully published voice routing configuration. 	
Monitoring and Archiving	Cose	
A Security		
Vetwork Configuration		



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

1	Conver 2012					Idministrator Sign (
Lyi	ic server 2015		_		5.0.8308	420 Privacy statem
<u>a</u>	Home	Dial Plan Voice Policy	Route PSTN Usage Tr	unk Corfiguration Test V	sice Routing	
22.	Users	Create voice routing test	t case information			~
м	Topology	(in				
P	IM and Presence			م		
3	Persistent Chat	A New / tdt *	A More un	own Action V Com	nit v	
S.	Voice Routing	Name	State	PSTN usage	Pattern to match	
6	Voice Features	LocaRoute	Committed		^(\+1[0-9][10])\$	
22	Response Groups	All Numbers	Committed	Al Numbers	1	
Q	Conferencing	EmergencyRoute	Committed	Emergency/Usage	^\1615	
5	Clients	Lync2013	Committed	Lync2013	2	
jî;	Federation and External Access					
	Monitoring and Archiving					
	Security					
¥	Network Configuration					

Figure 3-28: Voice Routing Screen Displaying Committed Routes

b.

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



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

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

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

Iv	nc Server 2013									Administrator Sig	an out
Ц		_							5.	0.8308.0 Privacy state	ement
	Home	Dial Plan	Voice Policy	Route	PSTN Usage	Trunk	Configuration	Test Voice Routing			
33	Users	Create v	pice routing tes	t case info	ormation					``	~
24	Topology										
Ģ	IM and Presence						م				
۲	Persistent Chat	- New	🔻 🧪 Edit 🤉	Actio	on 🔻 Comn	nit 🔻				(2
Ċ	Voice Routing	Na	me			Scope	State	Media bypass	PSTN usage	Ca	allin
C	Voice Features	e	Global			Global	Committed	\checkmark		0	

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

dit Trunk Configuration - Global	
✓ OK × Cancel	0
Scope: Global	
Name: *	
Global	
Description:	
Maximum early dialogs supported:	
20	
Encryption support level:	
Required	
Refer support:	
Enable sending refer to the gateway	
✓ Enable media bypass	
✓ Centralized media processing	
Enable RTP latching	
✓ Enable forward call history	
Enable forward P-Asserted-Identity data	
✓ Enable outbound routing failover timer	
^ Associated PSTN Usages	
Select Remove 🛧 🦺	
	· · ·

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

AudioCodes

- **d.** Verify de-activation of RTCP and Session timers. Since these parameters are not visible on the graphical interface, the Lync server Management Shel must be used.
- e. Use the following command on the Lync server Management Shell after reconfiguration to verify correct values.
 - Get-CsTrunkConfiguration

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

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



- EnableBypass, EnableReferSupport,RTCPActiveCalls and RTCPCallsonHold must be set to false
- SRTPMode is set to not supported
- EnableSessionTimer and ForwardCallHistory must be remain on their default value (False)
 - f. Repeat Steps 11 through 13 to commit your settings.

4 Configuring AudioCodes E-SBC

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

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

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

Notes:

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

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

- The scope of this document does **not** cover security aspects for connecting the SIP Trunk to the Microsoft Lync environment. Security measures should be implemented in accordance with your organization's security policies. For basic security guidelines, refer to the *Recommended Security Guidelines* document.
- Before you begin configuring the E-SBC, ensure that the E-SBC's Web interface Navigation tree is in Full-menu display mode. To do this, select the Full option, as shown below:

Configuration	Maintenance	Status & Diagnostics
Scenarios	Search	
🔿 Basic 🧕	Full	
System	\sim	

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



4.1 Step 1: IP Network Interfaces Configuration

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

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

Figure 4-1: Network Interfaces in Interoperability Test Topology



4.1.1 Step 1a: Configure Network Interfaces

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

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

> To configure the IP network interfaces:

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

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

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

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



4. Click Apply, and then Done.

The configured IP network interfaces are shown below:

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

12 Interfaces Table										
Add Index										
Index	Application Type	Interface Node	IP Address	Prefo	Gateway	VLAN ID	Interface Name	Primary DNS Server D ^a Address	Secondary DNS Server IP Address	Underlying Interfece
1 0	QAMP + Reda + Control	Pv4 Manual	123.3.5	24	10.3.3.1	1	KUNDE	10.3.3.10	0.0.0	OROUP_1
2 O	Illedia + Cantral	Pv4 Hanust	12.3.4.5	24	10.3.4.2	2	9W65	0.0.0.0	0.0.0	0R0VP_2

4.1.2 Step 1b: Configure the Native VLAN ID

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

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

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

Figure 4-3: Configured Port Native VLAN
4.2 Step 2: Enable the SBC Application

This step describes how to enable the SBC application.

> To enable the SBC application:

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

Figure 4-4: Enabling SBC Application

T		
🗲 SAS Application	Disable 👻]
SBC Application →	Enable 👻	
🗲 IP to IP Application	Disable 👻	

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

4.3 Step 3: Signaling Routing Domains Configuration

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

The SRD is composed of the following:

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

4.3.1 Step 3a: Configure Media Realms

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

To configure Media Realms:

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

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

Figure 4-5: Configuring Media Realm for LAN

Add Record		
Index	1	
Media Realm Name	KUNDE_MR	
IPv4 Interface Name	KUNDE	~
Pv6 Interface Name	None	~
ort Range Start	6000	
umber Of Media Session Legs	150	
ort Range End	7490	
Default Media Realm	Yes	V

3. Configure a Media Realm for WAN traffic:

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

Figure 4-6: Configuring Media Realm for WAN

Edit Record	×
Index	2
Media Realm Name	SWISSCOM_MR
IPv4 Interface Name	SWISSCOM V
IPv6 Interface Name	None 🗸
Port Range Start	7500
Number Of Media Session Legs	150
Port Range End	8990
Default Media Realm	No
	Submit × Cancel

The configured Media Realms are shown in the figure below:

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

Media Realm Table			
Add +	.)		
Index :	Media Realm Name	IPv4 Interface Name	IPv6
1	KUNDE_MR	None	None
2	SWISSCOM_MR	None	None
I < << Page 1 of 1 → ► Show 10 ∨ records per page			

Step 3b: Configure SRDs 4.3.2

This step describes how to configure the SRDs.

To configure SRDs:

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

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

Figure 4-8: Configuring LAN SRD

1 - KUNDE_SRD V
KUNDE_SRD
KUNDE_MR

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

Parameter	Value
SRD Index	2
SRD Name	SWISSCOM_SRD
Media Realm	SWISSCOM_MR

Figure 4-9: Configuring WAN SRD

•		
SRD Index	2 - SWISSCOM_SRD	
Common Parameters		
SRD Name	SWISSCOM_SRD	
Media Realm	SWISS_MR	
SBC Parameters		

4.3.3 Step 3c: Configure SIP Signaling Interfaces

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

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

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

3. Configure a SIP interface for the WAN:

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

The configured SIP Interfaces are shown in the figure below:

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

SIP I	nterface Table						
Index	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	SRD	Message Policy
1	KUNDE	SBC	0	5060	0	1	None
2	SWISS	SBC	0	5060	0	2	None
<	14	Page 1 of 1	>> Show	10 v records			View 1 - 2 of 2

4.4 Step 4: Configure Proxy Sets

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

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

- Microsoft Lync Server 2013
- Swisscom VoIP Gate

These Proxy Sets will later be associated with IP Groups.



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

> To configure Proxy Sets:

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

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

Proxy Se	et 1	ID	1		~
[Ī	Proxy Ad	Transport Type		
	1	sipt3srvlfe01.siptest	03.local:5068	TCP 🖌	
	2	sipt3srvlfe02 siptest	03.local:5068	TCP V	
	3				
	4				1
	5			V	
Enable P	ro	xy Keep Alive	Using Options		~
Proxy Ke	roxy Keep Alive Time		60		
Proxy Lo	ad	Balancing Method	Disable		×
Is Proxy	Proxy Hot Swap		Yes		
Proxy Redundancy Mode		Not Configured		¥	
Proxy Ke	SRD Index		1		
SRD Ind	ex		1		

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

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

Parameter	Value
Proxy Set ID	2
Proxy Address	nn.nn.nn:5060 (Swisscom IP address or FQDN including destination port)
Transport Type	ТСР
Enable Proxy Keep Alive	Using Options
Is Proxy Hot Swap	Νο
SRD Index	2 (enables classification by Proxy Set for SRD of IP Group belonging to Swisscom VoIP Gate)

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

•				
Proxy Set ID		2	~	
	Proxy Address		Transport Type	
1	195,176,152,148:5060		TCP ¥	٦
2				-
2				_
3				
4			v	
5			×	٦
				_
•		-		_
Enable Proxy K	eep Alive	Using Options	i 🗸	
Proxy Keep Aliv	ve Time	300		
Proxy Load Bal	ancing Method	Disable		
Is Proxy Hot Sv	vap	No	~	
Proxy Redunda	ncy Mode	Not Configure	d 🗸	
SRD Index		2		
Classification Ir	nput	IP only	~	1



This page is intentionally left blank.

4.5 Step 5: Configure IP Groups

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

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

- Lync Server 2013 (Mediation Server) located on LAN
- Swisscom VoIP Gate located on WAN
- MediaPack Analog Gateway located on LAN
- **To configure IP Groups:**
- Open the IP Group Table page (Configuration tab > VoIP menu > Control Network > IP Group Table).

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

2. Configure an IP Group for the Lync Server 2013 Mediation Server:

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

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



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

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

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

The configured IP Groups are shown in the figure below:

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

IP G	IP Group Table								
Add -	e)								
Index	Туре	Description	Proxy Set ID	SIP Group Name	Contact User	Local Host Name	SRD	Media Realm Name	IP Profile ID
1	Server	Lync	1				1	KUNDE_MR	1
2	Server	Swisscom	2	195.176.152.148			2	SWISSCOM_MR	2
3	User	Analog	-1				1	KUNDE_MR	0
				Page 1 of 1	⊳ ► Show 10 v red	ords per page			View 1 - 3

IP Group 3 will require classification rules for proper operation. The classification is based on the hostname used in the SIP signaling received from the MediaPack as well as the source username prefix applicable to all analog devices connected to the MediaPack. The configuration of the classification is explained in Step 6.

4.6 **Step 6: Configure the Condition Table**

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

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

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

The configured Condition Table is shown in the figure below:

Figure 4-14: Configured rules in Condition Table

Condition Table					
Add + Insert +					
Index :	Condition	Description			
1	header.user-agent regex MP\-1.*/v\.6\.60A\.	user-agent verification			
	ia <a< th=""><th>Page 1 of 1 🕨 🕨 Show 10 🗸 records per page</th></a<>	Page 1 of 1 🕨 🕨 Show 10 🗸 records per page			

4.7 **Step 7: Configure the Classification Table**

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

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

> To configure the Classification Table:

1. Open the IP Group Table page (Configuration tab > VoIP menu > SBC > Routing SBC > Classification Table).

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

2. Configure an Classification rule for the MediaPack analog Gateway:

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

Figure 4-15: Classification Table entry

Edit Record		×
Index	1	
Source SRD ID	1 🗸	
Source IP Address		
Source Port	0	
Source Transport Type	TCP v	
Source Username Prefix	+4161404979x#	
Source Host	sipgateway01.siptest03.l	ſ
Destination Username Prefix		
Destination Host	sipgateway01.siptest03.l	t
Message Condition	1 🗸	
Source IP Group ID	3 🗸	
Action Type	Allow 🗸	
	Submit	× Cancel

The configured Classification is shown in the figure below:

Figure 4-16: Configured rules in Classification Table

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

4.8 Step 8: Configure IP Profiles

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

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

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



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

> To configure IP Profiles:

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

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

Profile 10	
Profile ID	1 IVAIC
Prohie Name	LTNC
Common Parameters	
Gateway Parameters	
▼ SBC	
Transcoding Mode	Only if Required
Extension Coders Group ID	None
Allowed Coders Group ID	None
Allowed Coders Mode	Restriction
Diversion Mode	Don't Care
History Info Mode	Don't Care
Media Security Behavior	As Is
RFC 2833 Behavior	As Is
Alternative DTMF Method	Don't Care
P-Asserted-Identity	Don't Care
SBC Fax Coders Group ID	Coders group 1
SBC Fax Behavior	→ 1
SBC Fax Offer Mode	0
SBC Fax Answer Mode	1
SBC Session Expires Mode	Transparent
SBC Remote Early Media RTP	Delayed
SBC Remote Can Play Ringback	→ Yes
SBC Remote Supports RFC 3960	Not Supported
SBC Multiple 18x Support	supported
SBC Early Media Response Type	Transparent
SBC Remote Update Support	Supported Only After Connect
SBC Remote Re-Invite Support	Supported
SBC Remote REFER Behavior	Transparent
SBC Remote Early Media Support	supported
SBC Remote 3xx Behavior	Transparent
SBC Remote Delayed Offer Support	Not Supported
SBC PRACK Mode	Transparent
SBC Enforce MKI Size	do-not-enforce
SBC User Registration Time	0
SBC Remote Hold Format	transparent

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

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

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



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

Drofile ID	2
Profile ID	2 SWISSCOM
Pronie Name	31133COM
Common Parameters	
Gateway Parameters	
SBC	
Transcoding Mode	Only if Required
Extension Coders Group ID	None
Allowed Coders Group ID	None
Allowed Coders Mode	Restriction
Diversion Mode	Don't Care
History Info Mode	Don't Care
Media Security Behavior	RTP
RFC 2833 Behavior	As Is
Alternative DTMF Method	Don't Care
P-Asserted-Identity	Don't Care
SBC Fax Coders Group ID	None
SBC Fax Behavior	0
SBC Fax Offer Mode	0
SBC Fax Answer Mode	1
SBC Session Expires Mode	Transparent
SBC Remote Early Media RTP	Immediate
SBC Remote Can Play Ringback	
SBC Remote Supports RFC 3960	Not Supported
SBC Multiple 18x Support	supported
SBC Early Media Response Type	Transparent
SBC Remote Update Support	Supported
SBC Remote Re-Invite Support	Supported
SBC Remote REFER Behavior	Transparent
SBC Remote Early Media Support	supported
SBC Remote 3xx Behavior	Transparent
SBC Remote Delayed Offer Support	Supported
SBC PRACK Mode	Transparent
SBC Enforce MKI Size	do-not-enforce
SBC User Registration Time	0

4.9 Step 9: Configure the NTP Server Address

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

> To configure the NTP server address:

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

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

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

 NTP Settings 		
NTP Server Address (IP or FQDN)	\longrightarrow	0.ch.pool.ntp.org
NTP UTC Offset	> Hour	s: 1 Minutes: 0
NTP Updated Interval	Hour	s: 24 Minutes: 0
NTP Secondary Server IP	\longrightarrow	1.ch.pool.ntp.org
 Day Light Saving Time 		
Day Light Saving Time	Enable	~
DST Mode	Day of month	~
Start Time	Jan ∨ 01 ∨ 0	: 0
End Time	Jan v 01 v 0	: 0
Offset [min]	60	
Day of Month Start	Mar 🖌 Sunday	✓ Last ✓ 02 : 0
Day of Month End \longrightarrow	Oct v Sunday	✓ Last ✓ 03 : 0

3. Click Submit.

4.10 Step 10: Configure SRTP

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

> To configure media security:

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

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

Figure 4-20: Configuring SRTP

•	General Media Security Settings	
4	Media Security	Enable
4	Aria Protocol Support	Disable
	Media Security Behavior	Mandatory 💌
4	SRTP Tunneling Authentication for RTP	Disable
4	SRTP Tunneling Authentication for RTCP	Disable
_		
•	SRTP Setting	
	Master Key Identifier (MKI) Size	1
	Symmetric MKI Negotiation	Enable

3. Click Submit.

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

4.11 Step 11: Configure Maximum IP Media Channels

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



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

> To configure the maximum number of IP media channels:

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

Figure 4-21: Configuring Number of IP Media Channels

•		
4	Number of Media Channels \longrightarrow	30
4	Voice Streaming	Disable
	NetAnn Announcement ID	annc
	MSCML ID	ivr
	Transcoding ID	trans

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

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

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

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

- Terminate SIP OPTIONS messages on the E-SBC that are received from the LAN
- Terminate SIP REGISTER messages on the E-SBC that are received from the connected
- Calls from Lync Server 2013 are first tested against the (internal) registration database with known analog endpoints, if no match is found the calls use an alternative route to the Swisscom VoIP Gate
- Calls from Swisscom VoIP Gate are first send to the Lync Server 2013, a 404 "not found" SIP response will re-route the call towards the MediaPack Analog Gateway,
- Calls from analog endpoints are first routed to the Lync Server 2013, a 404 "not found" SIP response will re-route the call towards the Swisscom VoIP Gate
- **To configure IP-to-IP routing rules:**
- 1. Open the IP-to-IP Routing Table page (Configuration tab > VoIP menu > SBC > Routing SBC > IP-to-IP Routing Table).
- 2. Configure a rule to terminate SIP OPTIONS messages received from the LAN:

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

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

Edit Record		×
Index	0	
Source IP Group ID	-1	
Source Username Prefix	ź	
Source Host	ź	
Destination Username Prefix	±	
Destination Host	×	
Request Type	OPTIONS	-
Message Condition	None	-
ReRoute IP Group ID	-1	
Call Trigger	Any	-
Destination Type	Dest Address	
Destination IP Group ID	-1	
Destination SRD ID	None	×
Destination Address	internal	
Destination Port	0	
Destination Transport Type		-
Alternative Route Options	Route Row	v
Cost Group	None	4

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

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

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

Edit Record		>
Index	1	
Source IP Group ID	3	
Source Username Prefix	ź	
Source Host	ź	
Destination Username Prefix	ź	
Destination Host	ź	
Request Type	REGISTER	¥
Message Condition	None	~
ReRoute IP Group ID	-1	
Call Trigger	Any	~
Destination Type	IP Group	~
Destination IP Group ID	3	
Destination SRD ID	None	~
Destination Address		
Destination Port	0	
Destination Transport Type		~
Alternative Route Options	Route Row	¥
Cost Group	None	~

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

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

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

Edit Record				×
Index	2			
Source IP Group ID	1			
Source Username Prefix	ż			
Source Host	ż			
Destination Username Prefix	+4161404979	x#		
Destination Host	ż			
Request Type	All		~	
Message Condition	None		~	
ReRoute IP Group ID	-1			
Call Trigger	Any		~	
Destination Type	IP Group		~	
Destination IP Group ID	3			
Destination SRD ID	None		~	
Destination Address				
Destination Port	0			
Destination Transport Type			~	
Alternative Route Options	Route Row		~	
Cost Group	None		~	
		🛢 Submit	× Cano	el

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

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

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

Edit Record			×
Index	3		
Source IP Group ID	1		
Source Username Prefix	ż		
Source Host	ż		
Destination Username Prefix	ż		
Destination Host	ż		
Request Type	All	~	
Message Condition	None	¥	
ReRoute IP Group ID	-1		
Call Trigger	Any	¥	
Destination Type	IP Group	¥	
Destination IP Group ID	2		
Destination SRD ID	None	¥	
Destination Address			
Destination Port	0		
Destination Transport Type		~	
Alternative Route Options	Route Row	~	
Cost Group	None	¥	
	🗟 Submit) × c	ancel

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

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

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

Edit Record			×
Index	4		
Source IP Group ID	2		
Source Username Prefix	ż		
Source Host	ż		
Destination Username Prefix	ż		
Destination Host	ż		
Request Type	All	~	
Message Condition	None	¥	
ReRoute IP Group ID	-1		
Call Trigger	Any	~	
Destination Type	IP Group	~	
Destination IP Group ID	1		
Destination SRD ID	None	~	
Destination Address			
Destination Port	0		
Destination Transport Type		~	
Alternative Route Options	Route Row	~	
Cost Group	None	~	
	🖥 Submit	×	Cancel

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

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

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

Edit Record		×
Index	5	
Source IP Group ID	2	
Source Username Prefix	ż	
Source Host	ż	
Destination Username Prefix	ż	
Destination Host	ż	
Request Type	All 🗸	
Message Condition	None 🗸	
ReRoute IP Group ID	-1	
Call Trigger	Any 🗸	
Destination Type	IP Group 🗸	
Destination IP Group ID	3	
Destination SRD ID	None 🗸	
Destination Address		
Destination Port	0	
Destination Transport Type	~	
Alternative Route Options	Alt Route Ignore Inputs 🗸 🗸	
Cost Group	None 🗸	
	🗟 Submit 🗙 Car	ncel

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

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

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

Edit Record		×
Index	6	
Source IP Group ID	3	
Source Username Prefix	±	
Source Host	*	
Destination Username Prefix	ż	
Destination Host	ź	
Request Type	All	~
Message Condition	None	¥
ReRoute IP Group ID	-1	
Call Trigger	Any	¥
Destination Type	IP Group	¥
Destination IP Group ID	1	
Destination SRD ID	None	¥
Destination Address		
Destination Port	0	
Destination Transport Type		~
Alternative Route Options	Route Row	¥
Cost Group	None	¥
	🛢 Submit	× Cancel

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

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

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

Edit Record	×
Index	7
Source IP Group ID	3
Source Username Prefix	±
Source Host	±
Destination Username Prefix	±
Destination Host	±
Request Type	All 🗸
Message Condition	None 🗸
ReRoute IP Group ID	-1
Call Trigger	Any 🗸
Destination Type	IP Group 🗸
Destination IP Group ID	2
Destination SRD ID	None 🗸
Destination Address	
Destination Port	0
Destination Transport Type	~
Alternative Route Options	Alt Route Ignore Inputs 🗸
Cost Group	None v
	Submit × Cancel

The configured routing rules are shown in the figure below:

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

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

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

- To configure SBC alternative routing reasons:
- 1. Open the Alternative Routing reasons page (Configuration tab > VoIP > SBC > Routing SBC > Alternative Routing Reasons).
- 2. Configure the parameters as follows:

Parameter	Value
Reason 1	404

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

SBC Alternative Routing Reasons		
	SBC Alternative Routin	ig Reasons
	Reason 1	404 🗸
	Reason 2	
	Reason 3	
	Reason 4	
	Reason 5	· · · · · · · · · · · · · · · · · · ·

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

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



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

4.14 Step 14: Configure Message Manipulation Rules

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

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

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

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

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

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

Edit Record	×
Index	0
Manipulation Set ID	1
Message Type	invite.request
Condition	
Action Subject	header.request-uri.url.host
Action Type	Modify 🗸
Action Value	param.message.address.dst.a
Row Role	Use Current Condition V
	🗟 Submit 🗶 Cancel

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

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

Edit Record	×
Index	1
Manipulation Set ID	1
Message Type	invite.request
Condition	
Action Subject	header.to.url.host
Action Type	Modify 🗸
Action Value	param.message.address.dst.a
Row Role	Use Current Condition
	Submit × Cancel

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

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

Edit Record	×
Index	2
Manipulation Set ID	1
Message Type	invite.request
Condition	
Action Subject	header.from.url.host
Action Type	Modify 🗸
Action Value	'sipgateway01.siptest03.local'
Row Role	Use Current Condition 🗸
	Submit × Cancel

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

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

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

Edit Record	×
Index	3
Manipulation Set ID	2
Message Type	invite.request
Condition	header.referred-by exists
Action Subject	header.diversion
Action Type	Add 🗸
Action Value	'<' + header.referred-by.url + '>
Row Role	Use Current Condition 🗸
	Submit × Cancel

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

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

Edit Record	×
Index	4
Manipulation Set ID	2
Message Type	invite.request
Condition	header.referred-by exists
Action Subject	header.referred-by
Action Type	Remove v
Action Value	
Row Role	Use Current Condition
	Submit × Cancel

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

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

4.15 Step 15: Configure Call Forking Mode

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

To configure call forking:

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

Transcoding Mode	Only If Required	•
SBC No Answer Timeout	600	
SBC GRUU Mode	AsProxy	•
Minimum Session-Expires [sec]	90	
BroadWorks Survivability Feature	Disable	•
Bye Authentication	Disable	•
SBC User Registration Time	0	
SBC Proxy Registration Time	0	
SBC Survivability Registration Time	0	
SBC Forking Handling Mode	Sequential	•
Allow Unclassified Calls	Reject	•
SBC Session-Expires [sec]	180	
SBC Direct Media	Disable	-

Figure 4-38: Configuring Forking Mode

3. Click Submit.

4.16 Step 16: General Parameters

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

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

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



Note: The Gateway Name of the SBC is identical to the IP address of the SIP proxy which will be delivered by Swisscom.
~					
Use Default Proxy	No	¥			
Proxy Name					
Redundancy Mode	Parking	~			
Proxy IP List Refresh Time	60				
Enable Fallback to Routing Table	Disable				
Prefer Routing Table	No	¥			
Always Use Proxy	Disable	~			
Redundant Routing Mode	Routing Table	~			
SIP ReRouting Mode	Standard Mode	~			
Enable Registration	Disable	~			
Registrar Transport Type	Not Configured				
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	→ 195.176.152.148				
Gateway Registration Name					
DNS Query Type	A-Record	¥			
Proxy DNS Query Type	A-Record	~			
Subscription Mode	Per Endpoint	~			
Number of RTX Before Hot-Swap	3				
Use Gateway Name for OPTIONS	Yes	~			

Figure 4-39: Proxy & Registration Settings

> To configure General SIP Parameters:

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

Parameter	Value		
Retry-After Time	60		



SIP General 0.0.0.0 NAT IP Address C...... DDA CK Mada . . . Enable P-Charging Vector Disable v Enable VoiceMail URI Disable ¥ 60 Retry-After Time Enable P-Associated-URI Header Disable v Source Number Preference --- --

. .

Figure 4-40: Proxy & Registration Settings

To configure General SBC Settings:

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

Parameter	Value
Max Forwards Limit	70

Figure 4-41: SBC General Settings

•		
Transcoding Mode	Only If Required	~
SBC No Answer Timeout	600	
SBC GRUU Mode	AsProxy	¥
Minimum Session-Expires [sec]	90	
BroadWorks Survivability Feature	Disable	~
Bye Authentication	Disable	~
SBC User Registration Time	0	
SBC Proxy Registration Time	0	
SBC Survivability Registration Time	0	
SBC Forking Handling Mode	Latch On First	~
Unclassified Calls	Reject	¥
SBC Session-Expires [sec]	180	
SBC Direct Media	Disable	¥
SBC Preferences Mode	Doesn't Include Extensions	~
Max Forwards Limit	→ 70	

> To configure SBC Enforce Media Order :

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

Parameter	Value
SBCENFORCEMEDIAORDER	1

Figure 4-42: SBC General Settings

Image Load to Device	Parameter Name: SBCENFORCEMEDIAORDER Enter Value: 1 Apply New Value
<i>ini</i> Parameters	Output Window
Back to Main	Parameter Name: SBCENFORCEMEDIAORDER Parameter New Value: 1 Parameter Description:Arrange media lines according to the previous offer-answer (required by RFC 3264)

To configure the Swiss Call progress tones :

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

Configuration Maintenance Status & Diagnostics	Load Auxiliary Files
Scenarios Search	
Basic Full Maintenance Software Update Load Auxiliary Files	INI file (incremental) Browse_ No file selected. Load File
Software Upgrade Key Software Upgrade Wizard Configuration File	CAS file Browse_ No file selected. Load File
	Call Progress Tones file Browse_ switzerland.dat Load File

Figure 4-43: SBC General Settings

4.17 Step 17: Reset the E-SBC

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

- To save the configuration to flash memory:
- 1. Open the Maintenance Actions page (Maintenance tab > Maintenance menu > Maintenance Actions).

✓ Reset Configuration	
Reset Board	Reset
Burn To FLASH	Yes
Graceful Option	No
▼ LOCK / UNLOCK	
Lock	LOCK
Graceful Option	No
Gateway Operational State	UNLOCKED
✓ Save Configuration	
Burn To FLASH	BURN

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

5 **Configuring AudioCodes Analog Gateway**

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

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

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

5.1 Step 1: IP Network Interface Configuration

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

5.1.1 Step 1a: Enable the Multiple Interface Table

This step describes how to enable the Multiple Interface Table.

- 1. Open the IP Interfaces Table page (Configuration tab > VoIP menu > Network > IP Interfaces Table).
- 2. Click the 🔛 under the "Multiple Interface Settings" section

Figure 5-1: Enable the Multiple Interface Table

IP Address	10.1.10.10	۷
Subnet Mask	255.255.0.0	۷
Default Gateway Address	0.0.0.0	2
✓ VoIP DNS Settings		
🗲 DNS Primary Server IP		
🗲 DNS Secondary Server IP		
 Multiple Interface Settings 		
Multiple Interface Table		

5.1.2 Step 1b: Configure Network Interface

This step describes how to configure the IP network interface.

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

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

Figure 5-2: Configure Network Interface

Index	Application Type	IP Ad	Idress Prefix Length	Gateway	VLAN 10	Interface Name	Primary DNS Server IP Address	Secondary DNS Server IP Address
0 0	OAMP + Media + Control	10.3.3.50	24	10.3.3.1	1	0+M+C	10.3.3.10	0.0.0.0
			and the second second					
			🗲 VLAN Mode	D	sable	×		
			Native VLAN ID	1				
			IP Interface Statu	us Table		[inde		

5.2 Step 2: FXS Endpoints Configuration

This step describes how to configure the analog endpoints.

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

Parameter	Value
Channels	1 (the port number, 124)
Phone Number	+41614049799 (the applicable phone number)
Trunk Group ID	1
Tel Profile ID	1 (for analog phones)2 (for modem or Fax devices)



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

Figure 5-3: Configure FXS endpoints

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

5.3 Step 3: Hunt Group Settings Configuration

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

- Channel select method by which IP-to-Tel calls are assigned to the Hunt Group's channels.
- Registration method for registering Hunt Groups to selected Serving IP Group IDs.
- > To configure the Hunt Group Settings:
- Open the IP Interfaces Table page (Configuration tab > VoIP menu > GW and IP to IP submenu > Hunt Group submenu > Hunt Group Settings).
- 2. Change the view of the table to advanced view by clicking on the "Advanced Parameter List" button in the upper left corner



3. Configure the Hunt group settings Table as follows:

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



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

Figure 5-4: Configure Hunt Group Settings

	Trunk Group ID	Channel Select M	Channel Select Mode Registration Mode		Serving IP Group ID	Gateway Name
1	1	By Dest Phone Number	By Dest Phone Number 🗸 🗸		1 🗸	sipgateway01.siptest03
2			~	V	¥	

5.4 Step 4: IP to Hunt Group Routing Configuration

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

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

> To configure the IP to Hunt Group Routing Table:

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

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

Figure 5-5: Configure IP to Hunt Group routing

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

5.5 Step 5: Proxy & Registration Configuration

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

> To configure the Proxy and Registration settings:

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

Parameter	Value			
Enable Registration	Enable			
Registration time	3600			

Figure 5-6: Configure Proxy & Registration

systration		
Lise Default Proxy	No	
Provy Name		۲
Redundancy Mode	Parking	7
Provy IP List Refresh Time	60	۲
Enable Fallback to Routing Table	Disable	7
Prefer Routing Table	No	1
Always Use Proxy	Disable	1
Redundant Routing Mode	Routing Table	7
SIP ReRouting Mode	Standard Mode	1
Enable Registration	Enable	ī
Registrar Name		٦
Registrar IP Address		٦
Registrar Transport Type	Not Configured	1
Registration Time	3600	٦
Re-registration Timing [%]	₁ 50	1
Registration Retry Time	30	٦
Registration Time Threshold	0	1
Re-register On INVITE Failure	Disable	Ī
ReRegister On Connection Failure	Disable	7
Gateway Name		٦
Gateway Registration Name		1

5.6 Step 6: Proxy Set Configuration

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

> To configure the Proxy Sets Table:

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

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

Figure 5-7: Configure the Proxy Set

•				_			
Proxy Set ID				1		¥	
			Proxy Address			Transport Ty	ре
t	1		ipgateway01.siptest03.local:50	60		TCP 🗸	
2	2					~	
3	3					~	
4	4					~	
5	5					~	
•							
Enable Proxy K	(eep Aliv	ve	\longrightarrow	Us	sing Option	s v	
Proxy Keep Aliv	ve Time	e		60)		
Proxy Load Bal	Proxy Load Balancing Method		Disable		~		
Is Proxy Hot Sv	Is Proxy Hot Swap		No		¥		
Proxy Redunda	ancy Mo	de		Not Configured		ed 🗸 🗸	
SRD Index				0			
Classification In	nput			IP	only	~	

5.7 Step 7: Coders Configuration

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

To configure the default Coders:

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

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

Figure 5-8: Configure the Coders

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

> To configure the Coder Groups:

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

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

Figure 5-9: Configure the Coders Group 1

-				
Coder Group ID		1 🗸		
Coder Name Packetization Time		Rate	Payload Type	Silence Suppression
G.711A-law v	20 🗸	64 🗸	8	Disabled v

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

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

Figure 5-10: Configure the Coders Group 2

-								
Coder Group ID				2 🗸				
Coder Name		Packetiza	tion Time	Rat	te	Payload Type	Silence Suppre	ession
G.711A-law	~	20	~	64	~	8	Disabled	~
T.38	~	N/A	~	N/A	~	N/A	N/A	¥
	~		*		V			¥

5.8 **Step 8: Tel Profile Configuration**

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

> To configure the Tel Profiles:

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

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

Figure 5-11: Configure the Tel Profile 1

•		
Profile ID	1	~
Profile Name	> Telefon	
Profile Parameters		
Profile Preference	1	~
Fax Signaling Method	No Fax	~
Enable Polarity Reversal	Disable	~
Enable Current Disconnect	Disable	v
MWI Analog Lamp	Disable	v
MWI Display	Disable	~
Echo Canceler	Enable	~
Flash Hook Period	700	
Enable Early Media	Disable	~
Progress Indicator to IP	Not Configured	~
Disconnect Call on Detection of Busy Tone	Enable	~
Time For Reorder Tone [sec]	255	
Enable 911 PSAP	Disable	~
Call Priority Mode	None	~
Swap Tel To IP Phone Numbers	Disable	~
- Cada Cau		
	Cada Caur 1	
Coder Group	Coder Group T	~

3. Configure the Second Tel Profile as follows:

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

Figure 5-12: Configure the Tel Profile 2

2	
> Fax and Modem	
1	
No Fax	
Disable	
Disable	
Disable	
Disable	
Enable	
700	
Disable	
Not Configured	
Enable	
255	
Disable	
None	
Disable	
	2 Fax and Modem 1 No Fax Disable Disable Disable Disable Enable 700 Disable Not Configured Enable 255 Disable None None Disable None None Disable Non

5.9 Step 9 IP Profile Configuration

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

To configure the IP Profiles:

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

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

Figure 5-13: Configure the IP Profile

Profile ID	1	
Profile Name	SBC	
Common Parameters		
RTP IP DiffServ	46	
Signaling DiffServ	40	
Disconnect on Broken Connection	Yes	
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	
Symmetric MKI Negotiation	Enable	
MKI Size	→ 1	
Reset SRTP State Upon Re-key	Disable	
Gateway Parameters		
Fax Signaling Method	No Fax	
Play Ringback Tone to IP	Play	
Early Media	Disable	
Copy Destination Number to Redirect Number	Disable	

5.10 Step 10 IP Group Configuration

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

To configure the IP Group:

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

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

Figure 5-14: Configure the IP Group

Common Gateway	
Index	1
Description	SBC
Proxy Set ID	1
SIP Group Name	sipgateway01.siptest03.k
Contact User	
Local Host Name	
Media Realm Name	
IP Profile ID	1
	Submit × Cancel

5.11 Step 11: Number Manipulations

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

- Tel-to-IP calls:
 - Destination Phone Number Manipulation Table for Tel > IP Calls
 - Convert a 00 prefix into + for international dialing
 - Destination Prefix: 00
 - Strip digits from Left: 2
 - Prefix to add: +
 - Convert a 0 prefix into +41 for national dialing
 - Destination Prefix:
 - ✓ Strip digits from Left: 1
 - Prefix to add: +41
 - Convert emergency and service numbers to include the +41 prefix

0

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

> To configure the number Manipulations:

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

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

Index	Destination Prefix	Source Prefix	Source Trunk Group	Destination IP Group	Prefix to Add	Suffix to Add
0	00	*	-1	-1	+	
1	0	*	-1	-1	+41	
2	[112,117,118,143,144,145,147,161,162,163,164,187]#	*	-1	-1	+41	
3	[1811,1818,1850,1414,1415]#	*	-1	-1	+41	
4	xxxx#	*	-1	-1	+4161404	
5	*	*	-1	-1	+41	
	Page 1 of 1 >> > Show 10 v records per page View 1					

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

Rule Action			
Index	0		
Destination Prefix	00		
Source Prefix	*		
Source Trunk Group	-1		
Destination IP Group	-1		
		🖥 Submit	× Cancel

Rule Action		
Index	þ	
Stripped Digits From Left	2	
Stripped Digits From Right	0	
Number of Digits to Leave	255	
Prefix to Add	+	
Suffix to Add		
TON	Not Configured	~
NPI	Not Configured	¥
Presentation	Not Configured	~
		Submit × Cance

Rule Action			
Index	1		
Destination Prefix	0		
Source Prefix	±		
Source Trunk Group	-1		
Destination IP Group	-1		
		B Submit	× Cancel

Rule Action		
Index	þ	
Stripped Digits From Left	1	
Stripped Digits From Right	0	
Number of Digits to Leave	255	
Prefix to Add	+41	
Suffix to Add		
TON	Not Configured	~
NPI	Not Configured	¥
Presentation	Not Configured	¥
		Submit × Cancel

Rule Action	
Index	2
Destination Prefix	[112,117,118,143,144,145,147,161,162,1]
Source Prefix	*
Source Trunk Group	-1
Destination IP Group	-1
	Submit & Canc

Note: the full entry in Destination Prefix is:

 $[112,\!117,\!118,\!143,\!144,\!145,\!147,\!161,\!162,\!163,\!164,\!187]\#$

Rule Action			
Index	2		
Stripped Digits From Left	0		
Stripped Digits From Right	0		
Number of Digits to Leave	255		
Prefix to Add	+41		
Suffix to Add			
TON	Not Configured		¥
NPI	Not Configured		¥
Presentation	Not Configured		¥
		🗃 Submit	× Cancel

Rule Action	
Index	3
Destination Prefix	[1811,1818,1850,1414,1415]#
Source Prefix	ż
Source Trunk Group	-1
Destination IP Group	-1
	🗟 Submit 🗙 Cancel

Rule Action			
Index	β		
Stripped Digits From Left	0		
Stripped Digits From Right	0		
Number of Digits to Leave	255		
Prefix to Add	+41		
Suffix to Add			
TON	Not Configured		~
NPI	Not Configured		¥
Presentation	Not Configured		¥
		🖶 Submit	* Cancel

Rule Action	
Index	4
Destination Prefix	xxxx#
Source Prefix	*
Source Trunk Group	-1
Destination IP Group	-1
	🖶 Submit 🗙 Cancel

Rule Action		
index	4	
Stripped Digits From Left	0	
Stripped Digits From Right	0	
Number of Digits to Leave	255	
Prefix to Add	+4161404	
Suffix to Add		
FON	Not Configured	~
NPI	Not Configured	Ý
Presentation	Not Configured	~

Rule Action		
Index	5	
Destination Prefix	ż	
Source Prefix	ż	
Source Trunk Group	-1	
Destination IP Group	-1	
	🗟 Submit	× Cancel

Rule Action			
Index	Ξ		
Stripped Digits From Left	0		
Stripped Digits From Right	0		
Number of Digits to Leave	255		
Prefix to Add	+41		
Suffix to Add			
TON	Not Configured	~	
NPI	Not Configured	~	
Presentation	Not Configured	~	
		🖥 Submit 🗙 Can	cel

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

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

Index :	Source Prefix	Source IP Address	Source Host Prefix	Destination Prefix	Destination Host Prefix	Prefix to Add	Suffix to Add
0	+41	*	*	*	*	0	
1	+	*	*	*	*	00	
	Page 1 of 1 Page 10 of 1 Page 10 of 1 Page 10 of 10 Page 10 of 10 Page						

Rule Action	
Index	0
Source Prefix	+41
Source IP Address	ź
Source Host Prefix	±.
Destination Prefix	ź
Destination Host Prefix	ź
	B Submit × Cancel

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

Rule Action	
Index	þ
Stripped Digits From Left	3
Stripped Digits From Right	0
Number of Digits to Leave	255
Prefix to Add	0
Suffix to Add	
Presentation	Not Configured 🗸
	Submit × Cancel
Rule Action	
Index	1
Stripped Digits From Left	1
Stripped Digits From Right	0
Number of Digits to Leave	255
Prefix to Add	00
Suffix to Add	
Presentation	Not Configured 🗸
	Submit × Cancel

Rule Action	
Index	1
Source Prefix	+
Source IP Address	ź
Source Host Prefix	*
Destination Prefix	ź
Destination Host Prefix	ź
	Submit × Cancel

5.12 Step 12: General Configuration

Various general configurations complete the configuration of the MediaPack Analog Gateway

- > To configure NTP and Daylight Saving rules:
- 1. Open the Message Manipulations page (Configuration tab > System menu > Application Settings).
- 2. Configure the following Parameters:

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

Figure 5-19: Application Settings

▼ NTP Settings					
NTP Server Address (IP or FQDN)	→ 0.	.ch.pool.ntp.	org		
NTP UTC Offset		1	Minutes:	0	
NTP Updated Interval	Hours:	24	Minutes:	0	
NTP Secondary Server IP	$\longrightarrow 1$.ch.pool.ntp	org		

Day Light Saving Time	\longrightarrow	Enable	~
DST Mode	\longrightarrow	Day of month	~
Start Time		Jan v 01 v 0 :	0
End Time		Jan v 01 v 0 :	0
Offset [min]		60	
Day of Month Start	\longrightarrow	Mar 🗸 Sunday 🗸	Last v 02 : 0
Day of Month End	\longrightarrow	Oct v Sunday v	Last v 03 : 0

> To configure the Media Analog Settings:

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

Parameter	Value
FXS Coefficient Type	Europe

•	FXS_FXO Settings		
4	Analog Metering Type	12 kHz sinusoidal bursts	/
4	Min. Hook-Flash Detection Period [msec]	300	
	Max. Hook-Flash Detection Period [msec]	700	
4	FXS Coefficient Type	Europe	/
4	FXO Coefficient Type	Europe	/

Figure 5-20: SBC General Settings

To configure the FAX/Modem Settings:

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

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

Figure 5-21: FAX/Modem Settings

•	General Settings		
	Fax Transport Mode	Bypass	~
	Caller ID Transport Type	Mute	¥
	Caller ID Type	Standard ETSI	¥
	V.21 Modem Transport Type	Enable Bypass	¥
	V.22 Modem Transport Type	Enable Bypass	\checkmark
	V.23 Modem Transport Type	Enable Bypass	\checkmark
	V.32 Modem Transport Type	Enable Bypass	¥
	V.34 Modem Transport Type	Enable Bypass	¥
	Fax CNG Mode	Doesn't send T.38 re-INVITE	¥
	CNG Detector Mode	Disable	¥

✓ Fax Relay Settings	
Fax Relay Redundancy Depth	0
Fax Relay Enhanced Redundancy Depth	4
Fax Relay ECM Enable	Enable 🗸
Fax Relay Max Rate (bps)	14400bps 🗸

Bypass Settings		
Fax/Modem Bypass Coder Type	→ G711Alaw_64 ✓]
Fax/Modem Bypass Packing Factor	1	
Fax Bypass Output Gain	0	
Modem Bypass Output Gain	0	

To configure the Media Security Settings:

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

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

Figure 5-22: FAX/Modem Settings

•	General Media Security Settings		
4	Media Security	Enable	¥
	Media Security Behavior	Preferable - Single media	¥
	Authentication On Transmitted RTP Packets	Active	¥
	Encryption On Transmitted RTP Packets	Active	¥
	Encryption On Transmitted RTCP Packets	Active	¥
4	SRTP Tunneling Authentication for RTP	Disable	¥
4	SRTP Tunneling Authentication for RTCP	Disable	¥

 SRTP Setting 		
Master Key Identifier (MKI) Size	→ 1	
Symmetric MKI Negotiation	> Enable	×

SRTP Offered Suites

> To configure the Supplementary Services:

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

Parameter	Value
Enable Call Waiting	Disable
Enable Caller ID	Enable

▼	
Enable Hold	Enable 🗸
Hold Format	Send Only 🗸
Held Timeout	-1
Call Hold Reminder Ring Timeout	30
Enable Transfer	Enable 🗸
Transfer Prefix	
Enable Call Forward	Enable 🗸
Enable Call Waiting	Disable 🗸
Number of Call Waiting Indications	2
Time Between Call Waiting Indications	10
Time Before Waiting Indications	0
Waiting Beep Duration	300
Enable Caller ID	Enable 🗸
Hook-Flash Code	
Flash Keys Sequence Style	Flash hook 🗸

Figure 5-23: Supplementary Services

> To configure the DTMF and Dialing Settings:

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

Parameter	Value
Max Digits in Phone Numb	30

Figure 5-24: Supplementary Services

\mathbf{A}	
Max Digits In Phone Num	30
Inter Digit Timeout [sec]	4
Declare RFC 2833 in SDP	Yes 🗸
1st Tx DTMF Option	RFC 2833 🗸
2nd Tx DTMF Option	~ ·
RFC 2833 Payload Type	96
Default Destination Number	1000

> To configure the SIP General Parameters:

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

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

Figure 5-25: Supplementary Services

 SIP Gener 	al			
🗲 NAT IP Add	ress	0.	0.0.0	
PRACK Mod	e	S	upported	¥
Enable Con	tact Restriction	D	lisable	¥
Play Ringba	ck Tone to IP	$\longrightarrow P$	lay	¥
Play Ringba	ck Tone to Tel	$\longrightarrow P$	lay Local Until Remote Media Ar	¥
Use Tgrp inf	formation	D	lisable	¥
Enable GRU	U	D	lisable	¥
User-Agent	Information			
SDP Sessio	n Owner	A	udiocodesGW	
Subject				
Multiple Pac	ketization Time Format	N	lone	¥
Enable Sem	i-Attended Transfer	D	lisable	¥
3xx Behavi	or	F	orward	¥
Enable P-C	narging Vector	D)isable	¥
Enable Voic	eMail URI	D	lisable	¥
Retry-After	Time		0	
Enable P-As	sociated-URI Header	D)isable	¥
Source Nur	her Preference			

> To configure the Routing General Parameters:

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

Parameter	Value
Source IP Address Input	Layer 3 Source IP

✓ General Parameters		
Add Hunt Group ID as Prefix	No	1
Add Trunk ID as Prefix	No	,
Replace Empty Destination with B-channel Phone Number	No	,
Add NPI and TON to Called Number	No	1
Add NPI and TON to Calling Number	No	1
IP to Tel Remove Routing Table Prefix	No	,
Source IP Address Input	Layer 3 Source IP	1
Enable Alt Routing Tel to IP	Disable	1
Alt Routing Tel to IP Mode	Both	

Figure 5-26: Supplementary Services

To configure Enable Early 183 :

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

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

Figure 5-27: configuring EnableEarly183

Image Load to Device <i>ini</i> Parameters	Parameter Name: ENABLEEARLY183 Enter Value: 1 Output Window	Apply New Value
Back to Main	Parameter Name: ENABLEEARLY183 Parameter New Value: 1 Parameter Description:Enable Early 183	

AudioCodes

	Figure 5-28: configuring BellModemTransportType		
	Parameter Name: BELLMODEMTRANSPORTTYPE Enter Value: 2		
Image Load	Apply New Value		
to Device			
ini			
Parameters	Output Window		
Back to			
Main	Parameter New Value. 0		
	Parameter Description:Sets the Bell modem transport over the network method.		
	Parameter Name: BELLMODEMTRANSPORTTYPE		
	Parameter New Value: 2		
	Parameter Description:Sets the Bell modem transport over the network method.		

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

Configuration Maintenance Status & Diagnostics	Load Auxiliary Files
Scenarios Search	
Basic Full Maintenance Software Update Load Auxiliary Files	INI file (incremental) Browse_ No file selected. Load File
Software Upgrade Key Software Upgrade Wizard Configuration File	CAS file Browse_ No file selected. Load File
	Call Progress Tones file Browse_ switzerland.dat Load File

Α

AudioCodes MediaPack INI File

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



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

```
; * * * * * * * * * * * * * *
;** Ini File **
; * * * * * * * * * * * * * *
;Board: MP-118 FXS
;Board Type: 56
;Serial Number: 3555183
;Slot Number: 1
;Software Version: 6.60A.245
;DSP Software Version: 204IM=> 660.10
;Board IP Address: 10.3.3.50
;Board Subnet Mask: 255.255.255.0
;Board Default Gateway: 10.3.3.1
;Ram size: 32M Flash size: 8M
;Num of DSP Cores: 2 Num DSP Channels: 6
; Profile: NONE
;License Key limits aren't active full features capabilities are
available !;
;______
[SYSTEM Params]
SyslogServerIP = 10.3.3.98
NTPServerUTCOffset = 3600
DayLightSavingTimeStart = '03:SUN/05:02:00'
DayLightSavingTimeEnd = '10:SUN/05:03:00'
DayLightSavingTimeEnable = 1
NTPServerIP = '0.ch.pool.ntp.org'
NTPSecondaryServerIP = '1.ch.pool.ntp.org'
[Analog Params]
FXSCountryCoefficients = 66
[Voice Engine Params]
CallerIDType = 1
FaxTransportMode = 2
V21ModemTransportType = 2
FaxBypassPayloadType = 8
```

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```
BellModemTransportType = 2
ENABLEMEDIASECURITY = 1
SRTPTxPacketMKISize = 1
[SIP Params]
ENABLECALLERID = 1
MAXDTGTTS = 30
ENABLECALLWAITING = 0
PLAYRBTONE2IP = 1
REGISTRATIONTIME = 3600
ISREGISTERNEEDED = 1
PLAYRBTONE2TEL = 3
GWDEBUGLEVEL = 5
MEDIASECURITYBEHAVIOUR = 3
SOURCEIPADDRESSINPUT = 1
ENABLEEARLY183 = 1
FAKERETRYAFTER = 60
ENABLESYMMETRICMKI = 1
[ InterfaceTable ]
FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_VlanID, InterfaceTable_InterfaceName,
InterfaceTable PrimaryDNSServerIPAddress,
InterfaceTable_SecondaryDNSServerIPAddress;
InterfaceTable 0 = 6, 10, 10.3.3.50, 24, 10.3.3.1, 1, "O+M+C",
10.3.3.10, 0.0.0.0;
[ \InterfaceTable ]
[ TrunkGroup ]
FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum,
TrunkGroup_FirstTrunkId, TrunkGroup_FirstBChannel,
TrunkGroup_LastBChannel, TrunkGroup_FirstPhoneNumber,
TrunkGroup_ProfileId, TrunkGroup_LastTrunkId, TrunkGroup_Module;
TrunkGroup 0 = 1, 255, 1, 1, "+41614049794", 1, 255, 255;
TrunkGroup 1 = 1, 255, 2, 2, "+41614049798", 2, 255, 255;
TrunkGroup 2 = 1, 255, 3, 3, "+41614049799", 2, 255, 255;
[ \TrunkGroup ]
[ NumberMapTel2Ip ]
FORMAT NumberMapTel2Ip_Index = NumberMapTel2Ip_DestinationPrefix,
NumberMapTel2Ip SourcePrefix, NumberMapTel2Ip NumberType,
NumberMapTel2Ip NumberPlan, NumberMapTel2Ip RemoveFromLeft,
NumberMapTel2Ip RemoveFromRight, NumberMapTel2Ip LeaveFromRight,
NumberMapTel2Ip Prefix2Add, NumberMapTel2Ip Suffix2Add,
NumberMapTel2Ip_IsPresentationRestricted,
```

```
NumberMapTel2Ip_SrcTrunkGroupID, NumberMapTel2Ip_SrcIPGroupID,
NumberMapTel2Ip_DestIPGroupID;
NumberMapTel2Ip 0 = "00", "*", 255, 255, 2, 0, 255, "+", "", 255,
-1, -1, -1;
NumberMapTel2Ip 1 = "0", "*", 255, 255, 1, 0, 255, "+41", "", 255,
-1, -1, -1;
NumberMapTel2Ip 2 =
"[112,117,118,143,144,145,147,161,162,163,164,187]#", "*", 255,
255, 0, 0, 255, "+41", "", 255, -1, -1, -1;
NumberMapTel2Ip 3 = "[1811,1818,1850,1414,1415]#", "*", 255, 255,
0, 0, 255, "+41", "", 255, -1, -1, -1;
NumberMapTel2Ip 4 = "xxxx#", "*", 255, 255, 0, 0, 255, "+4161404",
"", 255, -1, -1, -1;
NumberMapTel2Ip 5 = "*", "*", 255, 255, 0, 0, 255, "+41", "", 255,
-1, -1, -1;
[ \NumberMapTel2Ip ]
[ SourceNumberMapIp2Tel ]
FORMAT SourceNumberMapIp2Tel_Index =
SourceNumberMapIp2Tel_DestinationPrefix,
SourceNumberMapIp2Tel_SourcePrefix,
SourceNumberMapIp2Tel_SourceAddress,
SourceNumberMapIp2Tel_SrcHost, SourceNumberMapIp2Tel_DestHost,
SourceNumberMapIp2Tel_NumberType,
SourceNumberMapIp2Tel_NumberPlan,
SourceNumberMapIp2Tel_RemoveFromLeft,
SourceNumberMapIp2Tel_RemoveFromRight,
SourceNumberMapIp2Tel_LeaveFromRight,
SourceNumberMapIp2Tel_Prefix2Add,
SourceNumberMapIp2Tel_Suffix2Add,
SourceNumberMapIp2Tel_IsPresentationRestricted;
SourceNumberMapIp2Tel 0 = "*", "+41", "*", "*", "*", 255, 255, 3,
0, 255, "0", "", 255;
SourceNumberMapIp2Tel 1 = "*", "+", "*", "*", "*", 255, 255, 1, 0,
255, "00", "", 255;
[ \SourceNumberMapIp2Tel ]
[ PstnPrefix ]
FORMAT PstnPrefix Index = PstnPrefix DestPrefix,
PstnPrefix_TrunkGroupId, PstnPrefix_SourcePrefix,
PstnPrefix_SourceAddress, PstnPrefix_ProfileId,
PstnPrefix_SrcIPGroupID, PstnPrefix_DestHostPrefix,
PstnPrefix_SrcHostPrefix, PstnPrefix_SrcSRDID, PstnPrefix_TrunkId,
PstnPrefix_CallSetupRulesSetId;
PstnPrefix 0 = "*", 1, "", "10.3.3.5", 0, -1, "", "", , -1, -1;
[ \PstnPrefix ]
[ ProxyIp ]
```

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```
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = "sipgateway01.siptest03.local:5060", 1, 1;
[ \ProxyIp ]
[ TxDtmfOption ]
FORMAT TxDtmfOption_Index = TxDtmfOption_Type;
TxDtmfOption 0 = 4;
[ \TxDtmfOption ]
[ TrunkGroupSettings ]
FORMAT TrunkGroupSettings Index = TrunkGroupSettings TrunkGroupId,
TrunkGroupSettings ChannelSelectMode,
TrunkGroupSettings_RegistrationMode,
TrunkGroupSettings_GatewayName, TrunkGroupSettings_ContactUser,
TrunkGroupSettings_ServingIPGroup,
TrunkGroupSettings_MWIInterrogationType,
TrunkGroupSettings_TrunkGroupName;
TrunkGroupSettings 0 = 1, 0, 0, "sipgateway01.siptest03.local",
"", 1, 255, "";
[ \TrunkGroupSettings ]
[ TelProfile ]
FORMAT TelProfile_Index = TelProfile_ProfileName,
TelProfile_TelPreference, TelProfile_CodersGroupID,
TelProfile_IsFaxUsed, TelProfile_JitterBufMinDelay,
TelProfile_JitterBufOptFactor, TelProfile_IPDiffServ,
TelProfile_SigIPDiffServ, TelProfile_DtmfVolume,
TelProfile_InputGain, TelProfile_VoiceVolume,
TelProfile_EnableReversePolarity,
TelProfile_EnableCurrentDisconnect,
TelProfile_EnableDigitDelivery, TelProfile_EnableEC,
TelProfile_MWIAnalog, TelProfile_MWIDisplay,
TelProfile_FlashHookPeriod, TelProfile_EnableEarlyMedia,
TelProfile_ProgressIndicator2IP, TelProfile_TimeForReorderTone,
TelProfile_EnableDIDWink, TelProfile_IsTwoStageDial,
TelProfile_DisconnectOnBusyTone, TelProfile_EnableVoiceMailDelay,
TelProfile_DialPlanIndex, TelProfile_Enable911PSAP,
TelProfile_SwapTelToIpPhoneNumbers, TelProfile_EnableAGC,
TelProfile_ECNlpMode, TelProfile_DigitalCutThrough,
TelProfile_EnableFXODoubleAnswer, TelProfile_CallPriorityMode,
TelProfile_FXORingTimeout;
TelProfile 1 = "Telefon", 1, 1, 0, 10, 10, 46, 40, -11, 0, 0, 0,
0, 0, 1, 0, 0, 700, 0, -1, 255, 0, 1, 1, 1, -1, 0, 0, 0, 0, 0, 0,
0, 0;
```

TelProfile 2 = "FAX & MODEM", 1, 2, 0, 10, 10, 46, 40, -11, 0, 0, 0, 0, 0, 1, 0, 0, 700, 0, -1, 255, 0, 1, 1, 1, -1, 0, 0, 0, 0, 0, 0, 0, 0; [\TelProfile] [IpProfile] FORMAT IpProfile Index = IpProfile ProfileName, IpProfile IpPreference, IpProfile CodersGroupID, IpProfile_IsFaxUsed, IpProfile_JitterBufMinDelay, IpProfile_JitterBufOptFactor, IpProfile_IPDiffServ, IpProfile_SigIPDiffServ, IpProfile_SCE, IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort, IpProfile_CNGmode, IpProfile_VxxTransportType, IpProfile_NSEMode, IpProfile IsDTMFUsed, IpProfile PlayRBTone2IP, IpProfile EnableEarlyMedia, IpProfile ProgressIndicator2IP, IpProfile EnableEchoCanceller, IpProfile CopyDest2RedirectNumber, IpProfile MediaSecurityBehaviour, IpProfile CallLimit, IpProfile DisconnectOnBrokenConnection, IpProfile FirstTxDtmfOption, IpProfile SecondTxDtmfOption, IpProfile_RxDTMFOption, IpProfile_EnableHold, IpProfile_InputGain, IpProfile_VoiceVolume, IpProfile_AddIEInSetup, IpProfile_SBCExtensionCodersGroupID, IpProfile MediaIPVersionPreference, IpProfile TranscodingMode, IpProfile_SBCAllowedCodersGroupID, IpProfile_SBCAllowedCodersMode, IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior, IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCAssertIdentity, IpProfile_AMDSensitivityParameterSuit, IpProfile_AMDSensitivityLevel, IpProfile_AMDMaxGreetingTime, IpProfile_AMDMaxPostSilenceGreetingTime, IpProfile_SBCDiversionMode, IpProfile_SBCHistoryInfoMode, IpProfile_EnableQSIGTunneling, IpProfile_SBCFaxCodersGroupID, IpProfile_SBCFaxBehavior, IpProfile_SBCFaxOfferMode, IpProfile_SBCFaxAnswerMode, IpProfile_SbcPrackMode, IpProfile SBCSessionExpiresMode, IpProfile SBCRemoteUpdateSupport, IpProfile SBCRemoteReinviteSupport, IpProfile SBCRemoteDelayedOfferSupport, IpProfile_SBCRemoteReferBehavior, IpProfile_SBCRemote3xxBehavior, IpProfile_SBCRemoteMultiple18xSupport, IpProfile_SBCRemoteEarlyMediaResponseType, IpProfile_SBCRemoteEarlyMediaSupport, IpProfile_EnableSymmetricMKI, IpProfile_MKISize, IpProfile SBCEnforceMKISize, IpProfile SBCRemoteEarlyMediaRTP, IpProfile SBCRemoteSupportsRFC3960, IpProfile SBCRemoteCanPlayRingback, IpProfile EnableEarly183, IpProfile_EarlyAnswerTimeout, IpProfile_SBC2833DTMFPayloadType, IpProfile_SBCUserRegistrationTime, IpProfile_ResetSRTPStateUponRekey, IpProfile_AmdMode, IpProfile_SBCReliableHeldToneSource, IpProfile_SBCPlayHeldTone, IpProfile_SBCRemoteHoldFormat, IpProfile_GenerateSRTPKeys; IpProfile 1 = "SBC", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 1, 0, -1, 1, 0, 2, -1, 1, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, -1, 0, 0, 0, 0, 0, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 2, 2, 1, 0, 0, 1, 0, 1, 1, 1, 0, 0, 0, 1, 1, 0, 0, 0, 0, 0, 1, 0, 0, 0; [\IpProfile]

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```
[ ProxySet ]
FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive,
ProxySet ProxyKeepAliveTime, ProxySet ProxyLoadBalancingMethod,
ProxySet_IsProxyHotSwap, ProxySet_SRD,
ProxySet_ClassificationInput, ProxySet_ProxyRedundancyMode;
ProxySet 0 = 0, 60, 0, 0, 0, 0, -1;
ProxySet 1 = 1, 60, 0, 0, 0, 0, -1;
[ \ProxySet ]
[ IPGroup ]
FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Description,
IPGroup_ProxySetId, IPGroup_SIPGroupName, IPGroup_ContactUser,
IPGroup_EnableSurvivability, IPGroup_ServingIPGroup,
IPGroup_SipReRoutingMode, IPGroup_AlwaysUseRouteTable,
IPGroup_RoutingMode, IPGroup_SRD, IPGroup_MediaRealm,
IPGroup_ClassifyByProxySet, IPGroup_ProfileId,
IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet,
IPGroup_OutboundManSet, IPGroup_RegistrationMode,
IPGroup_AuthenticationMode, IPGroup_MethodList,
IPGroup_EnableSBCClientForking, IPGroup_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName;
IPGroup 1 = 0, "SBC", 1, "sipgateway01.siptest03.local", "", 0, -
1, -1, 0, -1, 0, "", 1, 1, -1, -1, 0, 0, "", 0, -1, -1, "";
[ \IPGroup ]
[ CodersGroup0 ]
FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce;
CodersGroup0 0 = "g711Alaw64k", 20, 0, -1, 0;
CodersGroup0 1 = "g711Ulaw64k", 20, 0, -1, 0;
CodersGroup0 2 = "t38fax", 255, 255, -1, 255;
[ \CodersGroup0 ]
[ CodersGroup1 ]
FORMAT CodersGroup1_Index = CodersGroup1_Name, CodersGroup1_pTime,
CodersGroup1_rate, CodersGroup1_PayloadType, CodersGroup1_Sce;
CodersGroup1 0 = "g711Alaw64k", 20, 0, -1, 0;
CodersGroup1 1 = "q711Ulaw64k", 20, 0, -1, 0;
[ \CodersGroup1 ]
[ CodersGroup2 ]
```

```
FORMAT CodersGroup2_Index = CodersGroup2_Name, CodersGroup2_pTime,
CodersGroup2_rate, CodersGroup2_PayloadType, CodersGroup2_Sce;
CodersGroup2 0 = "g711Alaw64k", 20, 0, -1, 0;
CodersGroup2 1 = "g711Ulaw64k", 20, 0, -1, 0;
CodersGroup2 2 = "t38fax", 255, 255, -1, 255;
```

[\CodersGroup2]



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B AudioCodes E-SBC INI File

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



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

```
; * * * * * * * * * * * * * *
;** Ini File **
; * * * * * * * * * * * * * *
;Board: Mediant 1000 - MSBG
;HW Board Type: 47 FK Board Type: 67
;Serial Number: 3273006
;Slot Number: 1
;Software Version: 6.60A.257.004
;DSP Software Version: 204IM=> 660.10
;Second DSP Software Version: 624AE3=> 660.10
;Board IP Address: 10.3.3.5
;Board Subnet Mask: 255.255.255.0
;Board Default Gateway: 10.3.3.1
;Ram size: 495M Flash size: 64M
;Num of DSP Cores: 9 Num DSP Channels: 52
;Num of physical LAN ports: 7
;Profile: NONE
;Key features:;Board Type: Mediant 1000 - MSBG ;Channel Type: RTP
DspCh=240 IPMediaDspCh=240 ;PSTN Protocols: ISDN IUA=4 CAS
;E1Trunks=4 ;T1Trunks=4 ;Security: IPSEC MediaEncryption
StrongEncryption EncryptControlProtocol ;Coders: G723 G729 GSM-FR
G727 ILBC ;DSP Voice features: IpmDetector ;IP Media: Conf VXML
VoicePromptAnnounc(H248.9) ExtVoicePrompt=1MB ;Control Protocols:
MSFT MGCP MEGACO SIP SBC=30 FEU=30; Default features:; Coders: G711
G726;
;----- Mediant 1000 - MSBG HW components ------
; Slot # : Module type : # of ports : # of DSPs
1 : Empty
;
      2 : FXS
                     :
                                4 :
;
                                            1
      3 : IPMedia
                    :
                               0:
                                            2
;
      4 : Empty
;
      5 : Empty
;
      6 : Empty
          _____
[SYSTEM Params]
```

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```
NTPServerUTCOffset = 3600
DayLightSavingTimeStart = '03:SUN/05:02:00'
DayLightSavingTimeEnd = '10:SUN/05:03:00'
DayLightSavingTimeEnable = 1
NTPServerIP = '0.ch.pool.ntp.org'
NTPSecondaryServerIP = '1.ch.pool.ntp.org'
[Voice Engine Params]
ENABLEMEDIASECURITY = 1
NoOpEnable = 1
SRTPTxPacketMKISize = 1
CallProgressTonesFilename = 'switzerland.dat'
[SIP Params]
MEDIACHANNELS = 144
SIPGATEWAYNAME = `nn.nn.nn'
USEGATEWAYNAMEFOROPTIONS = 1
MEDIASECURITYBEHAVIOUR = 1
ENABLESBCAPPLICATION = 1
FAKERETRYAFTER = 60
SBCMAXFORWARDSLIMIT = 70
ENABLESYMMETRICMKI = 1
SBCFORKINGHANDLINGMODE = 1
SBCENFORCEMEDIAORDER = 1
[ PhysicalPortsTable ]
FORMAT PhysicalPortsTable Index = PhysicalPortsTable Port,
PhysicalPortsTable Mode, PhysicalPortsTable NativeVlan,
PhysicalPortsTable_SpeedDuplex,
PhysicalPortsTable_PortDescription,
PhysicalPortsTable_GroupMember, PhysicalPortsTable_GroupStatus;
PhysicalPortsTable 0 = "GE 0 1", 1, 1, 4, "KUNDE Port #0",
"GROUP_1", "Redundant";
PhysicalPortsTable 1 = "GE_0_2", 1, 1, 4, "KUNDE Port #1",
"GROUP_1", "Active";
PhysicalPortsTable 2 = "GE_7_1", 1, 2, 4, "SWISS Port #2",
"GROUP_2", "Redundant";
PhysicalPortsTable 3 = "GE_7_2", 1, 2, 4, "SWISS Port #3",
"GROUP_2", "Active";
PhysicalPortsTable 4 = "GE_7_3", 1, 3, 4, "OAMP Port #4",
"GROUP_3", "Active";
PhysicalPortsTable 5 = "GE_7_4", 1, 3, 4, "OAMP Port #5",
"GROUP_3", "Redundant";
[ \PhysicalPortsTable ]
[ EtherGroupTable ]
```

```
FORMAT EtherGroupTable_Index = EtherGroupTable_Group,
EtherGroupTable_Mode, EtherGroupTable_Member1,
EtherGroupTable_Member2;
EtherGroupTable 0 = "GROUP_1", 2, GE_0_1, GE_0_2;
EtherGroupTable 1 = "GROUP_2", 2, GE_7_1, GE_7_2;
EtherGroupTable 2 = "GROUP 3", 2, GE 7 3, GE 7 4;
[ \EtherGroupTable ]
[ InterfaceTable ]
FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_VlanID, InterfaceTable_InterfaceName,
InterfaceTable PrimaryDNSServerIPAddress,
InterfaceTable SecondaryDNSServerIPAddress,
InterfaceTable_UnderlyingInterface;
InterfaceTable 1 = 6, 10, 10.3.3.5, 24, 10.3.3.1, 1, "KUNDE",
10.3.3.10, 0.0.0.0, GROUP 1;
InterfaceTable 2 = 5, 10, 10.3.4.5, 24, 10.3.4.3, 2, "SWISSCOM",
0.0.0.0, 0.0.0.0, GROUP_2;
[ \InterfaceTable ]
[ DspTemplates ]
;
  *** TABLE DspTemplates ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts.
[ \DspTemplates ]
[ IPMediaChannels ]
FORMAT IPMediaChannels_Index = IPMediaChannels_ModuleID,
IPMediaChannels_DSPChannelsReserved;
IPMediaChannels 0 = 3, 48;
[ \IPMediaChannels ]
[ CpMediaRealm ]
FORMAT CpMediaRealm_Index = CpMediaRealm_MediaRealmName,
CpMediaRealm_IPv4IF, CpMediaRealm_IPv6IF,
CpMediaRealm_PortRangeStart, CpMediaRealm_MediaSessionLeg,
CpMediaRealm_PortRangeEnd, CpMediaRealm_IsDefault;
CpMediaRealm 1 = "KUNDE_MR", KUNDE, , 6000, 150, 7490, 1;
```



```
CpMediaRealm 2 = "SWISSCOM_MR", SWISSCOM, , 7500, 150, 8990, 0;
[ \CpMediaRealm ]
[ SRD ]
FORMAT SRD_Index = SRD_Name, SRD_MediaRealm,
SRD_IntraSRDMediaAnchoring, SRD_BlockUnRegUsers,
SRD_MaxNumOfRegUsers, SRD_EnableUnAuthenticatedRegistrations;
SRD 1 = "KUNDE_SRD", "KUNDE_MR", 0, 0, -1, 1;
SRD 2 = "SWISSCOM_SRD", "SWISSCOM_MR", 0, 0, -1, 1;
[\SRD]
[ SBCAlternativeRoutingReasons ]
FORMAT SBCAlternativeRoutingReasons Index =
SBCAlternativeRoutingReasons_ReleaseCause;
SBCAlternativeRoutingReasons 0 = 404;
[ \SBCAlternativeRoutingReasons ]
[ ProxyIp ]
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = "sipt3srvlfe01.siptest03.local:5068", 1, 1;
ProxyIp 2 = "nn.nn.nn.si5060", 1, 2;
[ \ProxyIp ]
[ IpProfile ]
FORMAT IpProfile_Index = IpProfile_ProfileName,
IpProfile_IpPreference, IpProfile_CodersGroupID,
IpProfile_IsFaxUsed, IpProfile_JitterBufMinDelay,
IpProfile_JitterBufOptFactor, IpProfile_IPDiffServ,
IpProfile_SigIPDiffServ, IpProfile_SCE,
IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort,
IpProfile_CNGmode, IpProfile_VxxTransportType, IpProfile_NSEMode,
IpProfile_IsDTMFUsed, IpProfile_PlayRBTone2IP,
IpProfile_EnableEarlyMedia, IpProfile_ProgressIndicator2IP,
IpProfile_EnableEchoCanceller, IpProfile_CopyDest2RedirectNumber,
IpProfile_MediaSecurityBehaviour, IpProfile_CallLimit,
IpProfile_DisconnectOnBrokenConnection,
IpProfile_FirstTxDtmfOption, IpProfile_SecondTxDtmfOption,
IpProfile_RxDTMFOption, IpProfile_EnableHold, IpProfile_InputGain,
IpProfile_VoiceVolume, IpProfile_AddIEInSetup,
IpProfile_SBCExtensionCodersGroupID,
IpProfile_MediaIPVersionPreference, IpProfile_TranscodingMode,
IpProfile_SBCAllowedCodersGroupID, IpProfile_SBCAllowedCodersMode,
```

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IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior,
IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCAssertIdentity,
IpProfile_AMDSensitivityParameterSuit,
IpProfile_AMDSensitivityLevel, IpProfile_AMDMaxGreetingTime,
IpProfile_AMDMaxPostSilenceGreetingTime,
IpProfile_SBCDiversionMode, IpProfile_SBCHistoryInfoMode,
IpProfile_EnableQSIGTunneling, IpProfile_SBCFaxCodersGroupID,
IpProfile_SBCFaxBehavior, IpProfile_SBCFaxOfferMode,
IpProfile_SBCFaxAnswerMode, IpProfile_SbcPrackMode,
IpProfile_SBCSessionExpiresMode, IpProfile_SBCRemoteUpdateSupport,
IpProfile_SBCRemoteReinviteSupport,
IpProfile_SBCRemoteDelayedOfferSupport,
IpProfile_SBCRemoteReferBehavior, IpProfile_SBCRemote3xxBehavior,
IpProfile_SBCRemoteMultiple18xSupport,
IpProfile_SBCRemoteEarlyMediaResponseType,
IpProfile_SBCRemoteEarlyMediaSupport,
IpProfile_EnableSymmetricMKI, IpProfile_MKISize,
IpProfile SBCEnforceMKISize, IpProfile SBCRemoteEarlyMediaRTP,
IpProfile SBCRemoteSupportsRFC3960,
IpProfile SBCRemoteCanPlayRingback, IpProfile EnableEarly183,
IpProfile EarlyAnswerTimeout, IpProfile SBC2833DTMFPayloadType,
IpProfile SBCUserRegistrationTime,
IpProfile_ResetSRTPStateUponRekey, IpProfile_AmdMode,
IpProfile_SBCReliableHeldToneSource, IpProfile_SBCPlayHeldTone,
IpProfile_SBCRemoteHoldFormat, IpProfile_GenerateSRTPKeys;
IpProfile 1 = "LYNC", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 0, 0,
0, 0, 0, -1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, -1,
0, 0, 0, 0, 0, 0, 8, 300, 400, 0, 0, 0, 1, 1, 0, 1, 3, 0, 1, 2, 0,
0, 0, 1, 0, 1, 0, 0, 0, 1, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 0, 0;
IpProfile 2 = "SWISSCOM", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 0,
0, 0, 0, 0, -1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, -
1, 0, 2, 0, 0, 0, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 2,
2, 1, 0, 0, 1, 0, 1, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 1, 0, 0,
0;
[ \IpProfile ]
[ ProxySet ]
FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive,
ProxySet_ProxyKeepAliveTime, ProxySet_ProxyLoadBalancingMethod,
ProxySet_IsProxyHotSwap, ProxySet_SRD,
ProxySet_ClassificationInput, ProxySet_ProxyRedundancyMode;
ProxySet 0 = 0, 60, 0, 0, 0, 0, -1;
ProxySet 1 = 1, 300, 0, 1, 1, 0, -1;
ProxySet 2 = 1, 300, 0, 0, 2, 0, -1;
[ \ProxySet ]
[ IPGroup ]
FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Description,
IPGroup_ProxySetId, IPGroup_SIPGroupName, IPGroup_ContactUser,
IPGroup_EnableSurvivability, IPGroup_ServingIPGroup,
IPGroup_SipReRoutingMode, IPGroup_AlwaysUseRouteTable,
```

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IPGroup_RoutingMode, IPGroup_SRD, IPGroup_MediaRealm,
IPGroup_ClassifyByProxySet, IPGroup_ProfileId,
IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet,
IPGroup_OutboundManSet, IPGroup_RegistrationMode,
IPGroup_AuthenticationMode, IPGroup_MethodList,
IPGroup_EnableSBCClientForking, IPGroup_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName;
IPGroup 1 = 0, "LYNC", 1, "", "", 0, -1, -1, 0, -1, 1, "KUNDE_MR",
1, 1, -1, -1, 1, 0, 0, "", 0, -1, -1, "";
IPGroup 2 = 0, "SWISSCOM", 2, "nn.nn.nn", "", 0, -1, -1, 0, -1,
2, "SWISSCOM_MR", 1, 2, -1, 4, 2, 0, 0, "", 0, -1, -1, "";
IPGroup 3 = 1, "ANALOG", -1, "", "", 0, -1, -1, 0, -1, 1,
"KUNDE MR", 0, 0, -1, -1, -1, 0, 0, "", 0, -1, -1,
[ \IPGroup ]
[ ConditionTable ]
FORMAT ConditionTable Index = ConditionTable Condition,
ConditionTable_Description;
ConditionTable 1 = "header.user-agent regex MP-1.*/v.6.60A.",
"user-agent prüfen";
[ \ConditionTable ]
[ IP2IPRouting ]
FORMAT IP2IPRouting_Index = IP2IPRouting_SrcIPGroupID,
IP2IPRouting_SrcUsernamePrefix, IP2IPRouting_SrcHost,
IP2IPRouting_DestUsernamePrefix, IP2IPRouting_DestHost,
IP2IPRouting_RequestType, IP2IPRouting_MessageCondition,
IP2IPRouting_ReRouteIPGroupID, IP2IPRouting_Trigger,
IP2IPRouting_DestType, IP2IPRouting_DestIPGroupID,
IP2IPRouting_DestSRDID, IP2IPRouting_DestAddress,
IP2IPRouting_DestPort, IP2IPRouting_DestTransportType,
IP2IPRouting_AltRouteOptions, IP2IPRouting_CostGroup;
IP2IPRouting 0 = -1, "*", "*", "*", 6, , -1, 0, 1, -1, ,
"internal", 0, -1, 0, ;
IP2IPRouting 1 = 3, "*", "*", "*", "*", 2, , -1, 0, 0, 3, , "", 0,
-1, 0, ;
IP2IPRouting 2 = 1, "*", "*", "*", 0, , -1, 0, 0, 3, , "", 0,
-1, 0, ;
IP2IPRouting 3 = 1, "*", "*", "*", "*", 0, , -1, 0, 0, 2, , "", 0,
-1, 1, ;
IP2IPRouting 4 = 2, "*", "*", "*", 0, , -1, 0, 0, 1, , "", 0,
-1, 0, ;
IP2IPRouting 5 = 2, "*", "*", "*", 0, , -1, 0, 0, 3, , "", 0,
-1, 1, ;
IP2IPRouting 6 = 3, "*", "*", "*", "*", 0, , -1, 0, 0, 1, , "", 0,
-1, 0, ;
IP2IPRouting 7 = 3, "*", "*", "*", "*", 0, , -1, 0, 0, 2, , "", 0,
-1, 1, ;
[ \IP2IPRouting ]
```

```
[ Classification ]
FORMAT Classification_Index = Classification_MessageCondition,
Classification_SrcSRDID, Classification_SrcAddress,
Classification_SrcPort, Classification_SrcTransportType,
Classification_SrcUsernamePrefix, Classification_SrcHost,
Classification_DestUsernamePrefix, Classification_DestHost,
Classification_ActionType, Classification_SrcIPGroupID;
Classification 1 = 1, 1, "", 0, 1, "+4161404979x#",
"sipgateway01.siptest03.local", "",
"sipgateway01.siptest03.local", 1, 3;
[ \Classification ]
[ SIPInterface ]
FORMAT SIPInterface_Index = SIPInterface_NetworkInterface,
SIPInterface_ApplicationType, SIPInterface_UDPPort,
SIPInterface_TCPPort, SIPInterface_TLSPort, SIPInterface_SRD,
SIPInterface_MessagePolicy, SIPInterface_TLSMutualAuthentication,
SIPInterface_TCPKeepaliveEnable,
SIPInterface_ClassificationFailureResponseType;
SIPInterface 1 = "KUNDE", 2, 0, 5060, 0, 1, , -1, 0, 500;
SIPInterface 2 = "SWISS", 2, 0, 5060, 0, 2, , -1, 0, 500;
[ \SIPInterface ]
[ MessageManipulations ]
FORMAT MessageManipulations_Index = MessageManipulations_ManSetID,
MessageManipulations_MessageType, MessageManipulations_Condition,
MessageManipulations ActionSubject,
MessageManipulations ActionType, MessageManipulations ActionValue,
MessageManipulations RowRole;
MessageManipulations 0 = 1, "invite.request", "", "header.request-
uri.url.host", 2, "param.message.address.dst.address", 0;
MessageManipulations 1 = 1, "invite.request", "",
"header.to.url.host", 2, "param.message.address.dst.address", 0;
MessageManipulations 2 = 1, "invite.request", "",
"header.from.url.host", 2, "'sipgateway01.siptest03.local'", 0;
MessageManipulations 3 = 2, "invite.request", "header.referred-by
exists", "header.diversion", 0, "'<' + header.referred-by.url +
'>'", 0;
MessageManipulations 4 = 2, "invite.request", "header.referred-by
exists", "header.referred-by", 1, "", 0;
[ \MessageManipulations ]
```



Configuration Note



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