Ribbon SBC Edge SWe Lite R9.0 Interop with NICE Engage : Interoperability Guide



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Document Overview

This document outlines the best practices for configuring Ribbon SBC SWe Lite for SIPREC feature verification.

About Ribbon SBC SWe Lite

The Ribbon Session Border Controller Software Edition Lite (SBC SWe Lite) provides best-in-class communications security. Microservices design to optimize resource allocation, dynamic scaling, automated lifecycle management are all attributes of the SBC SWe Lite, delivering edge SBC capabilities, such as robust network security, overload controls, SIP normalization, SIP Recording, IPv4-IPv6 interworking, and audio transcoding.

About NICE Engage platform

The NICE Engage Platform provides comprehensive omnichannel interaction recording. Omnichannel recording helps organizations provide customers a coherent experience by providing a single place to define and implement compliance and quality practices across all channels.

About SIP Recording

SIP Recording (SIPREC) is a recording capability that helps users to comply with regulations, to monitor the quality of service of representatives, to store call information for quality analysis, and so on. The Ribbon SBC SWe Lite supports SIPREC towards multiple recorders based on the Internet Engineering Task Force (IETF) standard.

The Ribbon SBC SWe Lite SIPREC supports the RFC standard for a SIP recording interface. To support SIPREC, the SBC SWe Lite acts as a Session Recording Client (SRC) initiating a Recording Session (RS) towards a Session Recording Server (SRS). The SBC SWe Lite initiates a recording session for all the Communication Sessions (CS) to record over SIP towards the SRS. The CS output is based on the SBC SWe Lite's Web UI configuration for enabling recording.

SIP Recording is supported on the SBC SWe Lite for the following purposes:

- · Storing call information for quality analysis.
- Recording call and media sessions on a third-party recording server.
- Checking the call detail records and determining if a call is recording or not.
- Providing call detail records for recorded calls.

Table 1 : Terminology

Term	Definition
CS	Communication Session
RS	Recording Session
SRC	Session Recording Client
SRS	Session Recording Server

Scope

This document provides configuration best practices for deploying Ribbon's SBC SWe Lite for SIPREC interop. Note that these are configuration best practices, and each customer may have unique needs and networks. Ribbon recommends that customers work with network design and deployment engineers to establish the network design that best meets their requirements.

Non-Goals

It is not the goal of this guide to provide detailed configurations that meets the requirements of every customer. Use this guide as a starting point and build the SBC configurations in consultation with network design and deployment engineers.

Audience

This is a technical document intended for telecommunications engineers to configure both the Ribbon SBC and the third-party product. Navigating the third-party product as well as the Ribbon SBC SWe Lite GUI is required. Understanding the basic concepts of TLS/TCP/UDP, IP/Routing, and SIP /SRTP is also necessary to complete the configuration and for any required troubleshooting.

Prerequisites

Before proceeding with the interop, make sure you have the following:

- Ribbon SBC SWe Lite
- SBC SWe Lite License
 - This interop requires the acquisition and application of SIP sessions. Refer to Working with Licenses for more information.
- NICE Engage Platform
 - NICE Engage platform SIPREC server running on 4.1 version or above.
 - Licenses for the required number of recording sessions.
 - NICE Dispatch Integration Services running on the SIPREC server.

Product and Device Details

The configuration uses the following equipment and software:

Table 2 : Requirements

Product	Equipment	Software Version
Ribbon Networks	Ribbon SBC SWe Lite	9.0.1
Third-party Equipment	NICE Engage platform	4.1
Administration and Debugging Tools	Wireshark	3.2.7
	LX Tool	2.1.0.6

Network Topology and E2E Flow Diagrams

SBC SWe Lite - NICE Engage Deployment Topology



Interoperability Test Lab Topology and Call Flow Diagram



Installing the SBC SWe Lite

To install and run the SBC SWe Lite on the VMware ESXi, refer to Installing SBC SWe Lite on VMware ESXi.

SBC SWe Lite Configuration

Accessing the SBC SWe Lite

Open any browser and enter the SBC SWe Lite IP address.



Click Enter and log in with a valid User ID and Password.



License and TLS Certificates

View License

Q Search	Current Licenses								
Expand All Collapse All Reload	Historical Usage Download License File								
Call Routing Signaling Groups Stetworking Interfaces	License Format Version 3								
V System		Feature Licenses							
Vode-Level Settings	Total 6 Feature License Rows								
Install New License	Feature	Licensed	Total Licenses	Available Licenses	Feature Expiration				
🕨 🭺 Software Management	SIP Signaling Sessions		300	300	August 09, 2021 23:59:59				
Auth and Directory Services	Enhanced Media Sessions with Transcoding	₩⁄	100	100	August 09, 2021 23:59:59				
Protocols SIP	Enhanced Media Sessions without Transcoding	₩⁄	600	600	August 09, 2021 23:59:59				
Security	SIP Registrations		300	300	August 09, 2021 23:59:59				
Media Interpretation	AMR-WB	6	Not Licensed	Not Licensed	Not Applicable				
Telephony Mapping Tables	SIP Recording	₩⁄	300	300	August 09, 2021 23:59:59				

For more details on Licenses, refer to Working with Licenses.

Import Trusted Root CA Certificates

A trusted certificate authority issues a Trusted CA Certificate. Trusted CA Certificates are imported to the SBC SWe Lite to establish its authenticity on the network.

From the Settings tab, navigate to Security > SBC Certificates > Trusted CA Certificates.

The following procedure shows you how to import Trusted Root CA Certificates, using either the File Upload or Copy and Paste method.

- 1. To import a Trusted CA Certificate, click the Import Trusted CA Certificate (
- 2. Select either Copy and Paste or File Upload from the Mode menu.

- 3. If you choose File Upload, use the Select File button to find the file.
- 4. Click OK.

Note Follow the above steps to import the SIPREC App Server certificates.

View Networking Interfaces

This section describes how to manage the way Ribbon SBC SWe Lite interfaces with the network. The SBC SWe Lite supports five system-created logical interfaces known as **Administrative IP**, **Ethernet 1 IP**, **Ethernet 2 IP**, **Ethernet 3 IP**, and **Ethernet 4 IP**. In addition to the system-created logical interfaces, the SBC SWe Lite supports user-created VLAN logical sub-interfaces.

From the Settings tab, navigate to Networking Interfaces > Logical Interfaces.

Administrative IP

The SBC SWe Lite system supports a logical interface called the Admin IP (Administrative IP, also known as the Management IP). A Static IP or DHCP is used for running the initial setup of the SBC SWe Lite system.

Q Search	Logical I	agical Interfaces								
Expand All Collapse All Reload	VIØI	I 🖉 Create VLAN I/F 🗙 Total 5 LogicalInterface Rows								
🕨 💋 Call Routing		Interface Name	IPv4 Address	IPv6 Address	Description	Admin State	Display	Primary Key		
Signaling Groups	Þ 🗀 🗆	Admin IP	10.34.3.132			Enabled	Counters	35		
 Logical Interfaces 	Þ 🗀 🗆	Ethernet 1 IP	10.10.10.10			Enabled	Counters	36		
Admin IP	Þ 🗎 🗆	Ethernet 1.370 IP	10.34.5.132	fd00:10:6b50:5040::184		Enabled	Counters	1		
Ethernet 1 IP Ethernet 1.370 IP	۱ 🗈 🗆	Ethernet 2 IP	20.20.20.20			Enabled	Counters	37		
Ethernet 2 IP	Þ 🔲 🗆	Ethernet 2.371 IP	10.34.7.132	fd00:10:6b50:5060::184		Enabled	Counters	2		
Ethernet 2.371 IP										

Ethernet 1 IP

Q. Sauch	v 📋 🗌 Ethernet 1.370 IP 10.34.	Enabled	Counters 1
C Search			
Expand All Collapse All Reload	Identification/Status		
 Call Routing Signaling Groups Networking Interfaces Admin IP Ethernet 1 IP Ethernet 1 270 IP Ethernet 2.371 IP System Auth and Directory Services Protocols Scurity 	Interface Name Ethernet 1.370 IP I/F Index 8 Alias Description Admin State Enabled Networking MAC Address 00:00: VIAN tag 370 Ib Address 10:00:		
🕨 🃁 Media	IP Addressing Mode Both 🗸		
Tone Tables			
SNMP/Alarms	IPv4 Information		
 Jegging Configuration Emergency Services 	IP Assign Method Static Primary Address 10.34 Primary Netmask 255.255.254.0 * XXXXX Media Next Hop IP 10.34 * XXXXX		
	IPv6 Information		
	Link Local Address fe80::20c Link Local Prefix 64 Primary Address fd00:10:**xccccc Primary Address Prefix 60 * (1127) Media Next Hop IP fd00:10:**xccccc		

Ethernet 2 IP



Configure Static Routes

Static routes communicate with remote networks. In a production environment, static routes are mainly configured for routing from a specific network to another network and accessed through one point or one interface (that is, single path access or default route).

- Configure static routes for smaller networks with just one or two routes, so that a link is not wasted for exchanging dynamic routing information.
- You do not require static routes for networks that have a LAN-side gateway on Voice VLAN or Multi-Switch Edge Devices (MSEs) with voice VLAN towards the SBC Edge,

Destination IP

Specifies the destination IP address.

Mask

Specifies the network mask of the destination host or subnet. If the 'Destination IP Address' field and 'Mask' field are both 0.0.0.0, the static route is called the 'default static route'.

Gateway

Specifies the IP address of the next-hop router to use for this static route.

Metric

Specifies the cost of this route, hence indirectly specifying the preference of the route. Lower values indicate more preferred routes. The typical value is 1 for most static routes, indicating that static routes are preferred over dynamic routes.

Q Search	St	itic IP Route Table							
Expand All Collapse All Reload	+	- 🗙 Total 1 IP Route Row							
▶ 🥬 Call Routing		Row ID	Destination IP	Mask	Gateway	Administrative Distance	Primary Key		
Signaling Groups	o	1	0.0.0.0	0.0.0.0	10.34.	1	1		
Networking Interfaces	E								
System Auth and Directory Services									
Protocols									
🕨 🥟 DNS									
Routing Table									
Static ARP									
Access Control Lists									
🕨 📁 NAT									
▶ 🍺 IPv6									

Configure SIP Profile

SIP Profiles control how the SBC Edge communicates with SIP devices. They control important characteristics, such as session timers, SIP header customization, SIP timers, MIME payloads, and option tags.

From the **Settings** tab, navigate to**SIP > SIP Profiles**.



SIP Server Tables

SIP Server

SIP Server Tables contain information about the SIP devices connected to the SBC Edge. The entries in the tables provide information about the IP Addresses, ports, and protocols used for communicating with each server.

From the Settings tab, navigate to SIP > SIP Server Tables.

- 1. Click the **Create** (+) icon.
- 2. In the **Description** field, enter a descriptive name for the SIP Server Table.
- 3. From the Type drop-down box, select SIP Server.
- 4. Click OK.

Create SIP Se	erver Table	April 21, 2021 05:21:30	0
Row ID Description Type	11 UAC SIP Server		
	ОК		

SIP Server Table Entry

From the Settings tab, navigate to SIP > SIP Server Tables.

1. Click on the desired SIP Server Table.

2. From the Create SIP Server drop-down list, select IP/FQDN.

UAC - IPv4

Q Search	UAC						
Expand All Collapse All Reload	Create SIP Server 🔻 🗶 /	Total 1 S	SIP Server Row				
🕨 🥖 Call Routing	Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
Signaling Groups	▼ □ 10.54.	IP/FQDN	35070	UDP	Counters	1	1
Networking Interfaces							
System	Se	rver Host		Tra	nsport		
Autin and Directory Services							
	Server Lookup IP/FQDN			Monitor Non	e 🗸		
/ Local Registrars	Priority 1	~					
📁 Local / Pass-thru Auth Tables	Host FODN/IP 10.54						
SIP Profiles							
V SIP Server Tables	Port 35070	* [165535]					
Default SIP Server	Protocol UDP	*					
	Remote Autho	rization and Contacts	3				
(0AST							
UAS3	Remote Authorization Table	None	× +				
UACV6	Contact Registrant Table	None	V				
UASV6	Contact Registrant Table	None					
E SIP_RECORDER1	Session URI Validation	Liberal 🗸					
SIP_RECORDER2							
SIP_RECORDERV6							
📁 Trunk Groups					Apply		
📁 NAT Qualified Prefix Tables							



Q Search	UACV6						
Expand All Collapse All Reload	Create SIP Server 🔻 🗶 🖉	Total 1 SIP Server	Row				
▶ 💋 Call Routing	Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
Signaling Groups	🔻 📄 🗌 fd00:10:	IP/FQDN	35110	UDP	Counters	1	1
Metworking Interfaces							
System	Server	lost		Transport			
Auth and Directory Services				manoport	_		
Protocols	Server Lookup IP/FQDN		Monito	None	~		
V SIP	Priority 1	•		None			
Local Registrars							
SID Profiles	Host FQDN/IP fd00:10:	*					
SIP Fromes	Port 35110	* [165535]					
Default SIP Server	Protocol UDD	•]*					
UAC							
UAS			_				
UAS1	Remote Authorization	on and Contacts					
UAS2							
UAS3	Remote Authorization Table No	one 🗸 🕇					
UACV6	Contact Registrant Table	one 🗸 🕇					
UASV6	Contine URI Validation						
SIP_RECORDER1	Session URI Validation	oeral V					
SIP_RECORDER2			_				
SIP_RECORDERV6							
📁 Trunk Groups					Apply		
📁 NAT Qualified Prefix Tables							

UAS - IPv4

Q Search	UAS1							
Expand All Collapse All Reload	Create SIP Serv	er 🔻 🗶	/12 Tota	l 1 SIP Server i	Row			
Call Routing Signaling Groups Networking Interfaces Auth and Directory Services Call Registrars Cocal Registrars Cocal / Pass-thru Auth Tables SIP Profiles	v 10. Server Look Prior Host FQDN	t / Domain 54. Secup IP/FQDN rity 1 /IP 10.54. pt 25000	Server Lookup IP/FQDN erver Host	Port 35090	Protocol UDP Monitor	Display Counters Counters Transport	Priority 1	Primary Key 1
VIP Server Tables	Proto	col UDP Remote Author horization Table Registrant Table	* None None Liberal	acts			Apply	



Q Search	UASV6						
Expand All Collapse All Reload	Create SIP Server 🔻 🗶 🥂	Total 1 SIP Serve	Row				
🕨 🥖 Call Routing	Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
 Signaling Groups Networking Interfaces System 	▼ 📄 🗋 fd00:10:	IP/FQDN	35130	UDP	Counters	1	1
Auth and Directory Services	Server Hos	st		Transpo	ort		
 Protocols Cocal Registrars Local / Pass-thru Auth Tables SIP Profiles SIP Profiles SIP Server Tables Default SIP Server UAC 	Server Lookup IP/FQDN Priority 1 ~ Host FQDN/IP fd00:10 Port 35130 Protocol UDP ~	* [165535]	м	onitor None	~		
UAS	Remote Authorization	and Contacts					
UAS2 UAS3 UACV6 SIP_RECORDER1 SIP_RECORDER2 SIP_RECORDER2 SIP_RECORDER2	Remote Authorization Table None Contact Registrant Table None Session URI Validation Liber	▼ ▼ ■ ▼					
📁 Trunk Groups 📁 NAT Qualified Prefix Tables					Apply		

Note

The number of Sip Server Table Entries depends on the requirements.

SIPREC configuration on Ribbon SBC SWe Lite

SIP Server Table for SIPREC

SIP Recorder

A SIP Recorder Table contains information about the Network interface through which the SRC initiates recording requests toward the SRS. To support the SIPREC, the SBC Edge acts as a Session Recording Client (SRC), initiating a Recording Session (RS) toward a Session Recording Server (SRS). The SBC Edge initiates a recording session for all the Communication Sessions (CS). The CS output is based on the SBC Edge's Web UI configuration for enabling recording. The SIP Server (SIP Recorder) supports either an FQDN or IP Address (V4 or V6). Each SIPREC Server entry enables configuring multiple servers. When the standby SRS is configured, the SBC Edge sends the recording first to the active SRS. If the request fails (due to server reachability fault), the SBC SWe Lite sends the recording session to the standby SRS.

From the Settings tab, navigate to SIP > SIP Server Tables.

1. Click the **Create** (+) icon.

- 2. In the **Description** field, enter a descriptive name for the SIP Server Table.
- 3. From the Type drop-down box, select SIP Recorder.
- 4. Click OK.

Create SIP Se	rver Table	April 21, 2021 05:17:46 🕜
Row ID Description Type	11 SIP_RECORDER1 SIP Recorder 🗸	
	ОК	

SIP Server Table entry for SIPREC

From the Settings tab, navigate to SIP > SIP Server Tables

- 1. Click on the desired SIP Server Table.
- 2. From the Create SIP Server drop-down list, select IP/FQDN.

SIPREC - IPv4

Q Search	SIP_RECORDER	1						
Expand All Collapse All Reload	Create SIP Recorde	I 🗙 I	/1 Total 1 SI	P Server R	w			
🕨 🏓 Call Routing	Host /	Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
Signaling Groups	▼ □ 10.34.	-	IP/FQDN	5060	UDP	Counters	1	1
Metworking Interfaces								
System Auth and Directory Services		Server H	ost		Transport	t		
Protocols								
	Server Lookup	IP/FQDN		Mo	nitor None	~		
/ Local Registrars	Priority	1	~					
💋 Local / Pass-thru Auth Tables	Host FODN/IP	10.34	*					
SIP Profiles								
V SIP Server Tables	Port	5060	* [165535]					
Default SIP Server	Protocol	UDP	*					
UAC								
UAS								
UAS1								
						Apply		
UASV6								
SIP_RECORDER1								
SIP_RECORDER2								
SIP_RECORDERV6								

SIPREC - IPv6

Q Search	SIP_RECORDER	V6						
Expand All Collapse All Reload	Create SIP Recorde	r 🔻 l 🗙 l 🥂	Total 1 SIP Se	rver Row				
🕨 🧯 Call Routing	Host / [Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
Signaling Groups	▼ 📄 🗌 fd00:10	D:	IP/FQDN	35140	UDP	Counters	1	1
Metworking Interfaces								
▶ 📁 System		Server Host		Tran	sport			
Generation Auth and Directory Services					•			
	Server Lookup	IP/FQDN		Monitor None	~			
Local Registrars	Priority	1 ~		Contractor Sector				
💋 Local / Pass-thru Auth Tables	Host FODN/IP	Fd00-10-						
SIP Profiles	100011000111	1000.10.						
V SIP Server Tables	Port	35140 *	[165535]					
Default SIP Server	Protocol	UDP 🗸 *						
UAC 📑								
UAS 👘								
UAS1	I					_		
UAS2					Apply			
UAS3								
UACV6	:							
UASV6								
SIP_RECORDER1								
UE SIP_RECORDERV6								

SIP Recording Table entry

A SIP Recorder Table contains information about how a Signaling Group starts a SIP Recording (SIPREC) session. Through the configuration options, you select the Recording Server (SIP Server), SIP channels, routing, and media, which directs the SBC to communicate with the Session Recording Server (SRS).

From the **Settings** tab, navigate to **SIP > SIP Recording**.

SIP Recording Table 1

Q Search	SIP Recording Table				
Expand All Collapse All Reload	🧹 l 🥥 l 🕂 🗙 🛛 Total 2 SIP	Recording Rows			
🕨 🏓 Call Routing	Description	Admin State	Service Status	Display	Primary Key
🕨 📁 Signaling Groups	v SIP_Recorder_SG1	U	Up	Counters Channels Sessions	50001
Metworking Interfaces		•			
🕨 🣁 System					
Auth and Directory Services	Description SIP Recorder SG1				
Protocols					
💌 🥔 SIP	Admin State Enabled V				
📁 Local Registrars					_
💋 Local / Pass-thru Auth Tables	SIP Channels	and Routing		SIP IP Details	
SIP Profiles					
🕨 🍺 SIP Server Tables	No. of Channels 100	* (1.050)		Signaling/Media	
📁 Trunk Groups		[1900]		Source IP Ethernet 1.370 IP (10.34.	<u> </u>
💋 NAT Qualified Prefix Tables	SIP Profile Default SIP P	rofile 🗸 🕇		Signaling DSCP 40 * [063]	
🥖 Remote Authorization Tables	Recording Server Table SIP_RECORD	ER1 🗸 🕈			
🥖 Contact Registrant Table	Load Balancing Round Pohin	×			
🕨 🥖 Message Manipulation	Round Robin	•			
Node-Level SIP Settings	Channel Hunting Most Idle	~			
V SIP Recording					_
SIP_Recorder_SG1	Listen	Ports		Federated IP/FQDN	
SIP_Recorder_SG2					

SIP Recording Table 2

Q Search	SIP Recording Table					
Expand All Collapse All Reload	√ I ⊘ I + I × T	otal 2 SIP Recording Rows				
🕨 🥖 Call Routing	Description	Admin State	Service Status		Display	Primary Key
Signaling Groups	SIP_Recorder_SG1	V	Up		Counters Channels Sessions	50001
// Networking Interfaces	v 📄 SIP_Recorder_SG2	2 11/2	Up		Counters Channels Sessions	50002
Auth and Directory Services						
Protocols						
V 💋 SIP	Description SIP_Recorder	_SG2				_
📁 Local Registrars	Admin State Enabled	~				_
📁 Local / Pass-thru Auth Tables						
🕨 📁 SIP Profiles		Ohannala and Dautian				
🕨 🥩 SIP Server Tables	SIP	Channels and Routing			SIP IP Details	
📁 Trunk Groups				Cian	aliaa (Madia	
💋 NAT Qualified Prefix Tables	No. of Channels 10	* [1960]		Sign	Source IP Ethernet 2.371 IP (10.34.) 🗸
📁 Remote Authorization Tables	SIP Profile De	efault SIP Profile	~ +	Sign	aling DSCP 40 * /0.6	27
💋 Contact Registrant Table	Recording Server Table SI			Ung.		<i>'</i>
🕨 📁 Message Manipulation						
Node-Level SIP Settings	Load Balancing Ro	ound Robin	<u>~</u>			
🔻 💋 SIP Recording	Channel Hunting M	ost Idle	~			
SIP_Recorder_SG1						
SIP_Recorder_SG2		Listen Ports			Federated IP/FQDN	

🕑 Tip

For details on the SIP Recording (SIPREC) feature, refer to Working with SIP Recording - SIPREC.

Configure Media Profiles

From the Settings tab, navigate to Media > Media Profiles.

- 1. From the Create Media Profile drop-down box at the top of the Media Profiles page, select Voice Codec Profile.
- 2. From the Codec drop-down box, select a codec.

Codec - G.729

Q Search	Media Profiles		
Expand All Collapse All Reload	Create Media Profile 🔻 🇙	Total 4 Med	i a Profile Rows
🕨 🥖 Call Routing	Codec	Description	Primary Key
Signaling Groups	▶ 💼 🗌 G.711 A-Law	Default G711A	1
 Metworking Interfaces System 	▶ 📄 🗌 G.711 µ-Law	Default G711u	2
Auth and Directory Services	🔻 📋 🗌 G.729	G729A	5
Protocols			
🕨 🍎 SIP	Voice Codec C	onfiguration	
Security			_
🔻 💋 Media	Description G729A		
Media System Configuration	Codes 0770		
🔻 🥟 Media Profiles	Codec G./29	~	
Default G711A	Payload Size 20	✓ ms	
Default G711u			
G729A			
G726			
DES-SRTP Profiles		Ар	ply



Q Search	Media Profiles			
Expand All Collapse All Reload	Create Media Profi	ile 🔻 🗙	Total 4 M	ledia Profile Rows
🕨 🥖 Call Routing	Codec		Description	Primary Key
Signaling Groups	▶ 📄 🗌 G.711	A-Law	Default G711A	1
 jew Networking Interfaces jew System 	▶ 📄 🗌 G.711	µ- <mark>L</mark> aw	Default G711u	2
Auth and Directory Services	▶ 📴 🗌 G.729		G729A	5
Protocols Ø SIP Image:	🔻 📄 🗌 G.726		G726	6
 Security Media Media System Configuration Media Profiles Default G711A Default G711u G729A G726 SDES-SRTP Profiles 	Vo Description Codec Rate Payload Size Payload Type	ice Codec C G726 G.726 32000 b/s 20 2	onfiguration w ms	
 Media List Tone Tables Telephony Mapping Tables SNMP/Alarms 				Арріу

Attach Media Profiles to Media lists

From the **Settings** tab, navigate to **Media > Media Lists**.

- Click the Create Media List (⁺) icon at the top of the Media List View page.
 Provide the desired description.
 Click Add/Edit and select the Media Profiles List.

- Click OK.

Q Search	Media List View			
Expand All I Collapse All I Beload	🕂 🗙 Total	4 Media List Rows		
Call Routing	Description			Primary Key
🕨 📁 Signaling Groups	🕨 💼 📄 Default Media	List		1
Networking Interfaces	ThoressMedi	iaProfile		2
V System				-
Auth and Directory Services				
Protocols	Description	IngressMediaProfile		
Socurity			, 1	
Media		Default G711A	qU	
Media System Configuration		G729A	Down	
🕨 🍺 Media Profiles	Media Profiles List	G726		
SDES-SRTP Profiles			Add/Edit	
🔻 🥟 Media List		-	Remove	
Default Media List				
IngressMediaProfile	SDES-SRTP Profile	None	Associated SIP SG Listen Ports should be TI	S only. 💠
EgressMediaProfile	Media DSCP	46	* [063]	
SiprecMediaProfile	Dead Call Detection	Disabled 🗸		
🕨 📁 Tone Tables	Silence Suppression	Enabled 🗸	1	
🕨 💋 Telephony Mapping Tables			J	
O Search	Media List View			
Searcha	i i v Tota	4 Media List Rows		
Expand All Collapse All Reload	- · ·			
🕨 📁 Call Routing	Description			
🕨 🍺 Signaling Groups	🕨 📄 🗌 Default Media	a List		
Metworking Interfaces	IngressMedia	Profile		
System		Drofile		
Protocols	EgressMedia	arrone		
▶ d SIP				
Security	Description	ForessMediaProfile		
V Media	2 coulption	Parsenticements		
Media System Configuration		Default G711A	Un	

🕨 🃁 Media Profiles		G729A		D
💋 SDES-SRTP Profiles	Media Profiles List	G726		bown *
💌 🥟 Media List		2010		Add/Edit
Default Media List				Remove
IngressMediaProfile				
EgressMediaProfile	SDES-SRTP Profile	None	~ ,	Associated SIP SG Listen Ports should be TLS only. 💠
SiprecMediaProfile	Media DSCP	46	,	* [063]
🕨 🃁 Tone Tables	Dead Call Detection			
🕨 🍺 Telephony Mapping Tables	Deau Can Detection	Disabled	•	
SNMP/Alarms	Silence Suppression	Enabled	~	

Default G711u

Configure Signaling Groups

Signaling groups allow grouping telephony channels for routing and shared configuration. They are used for routing calls and selecting Call Routes.

Up

From the Settings tab, navigate to Signaling Groups (Add SIP SG).

INGRESS_SG

🕨 💋 Media Profiles

- Attach the Sip Profile you created earlier.
 - Specifies the SIP Profile for this Signaling Group.
- Select the Agent Type as Back-to-Back User Agent.
 - The SBC maintains the state and participates in all SIP signaling between both endpoints.
- Attach the SIP Server Table "UAC".
 - Specifies the SIP Server Table for this Signaling Group.
- Select the appropriate Media List.
 - Specifies the Media List this Signaling Group uses.

ons
ld/Edit
* move
ld/Edit
move

- Enable SIP Recording and attach the Recording Server Table created for SIP recording.
 When the SIPREC is enabled for the specific Signaling Group, this field allows you to select an entry from a SIP Recording Table, using the drop-down list. This list is populated from the entries configured in the SIP Recording table. For details, refer to Creating and Modifying Entries in the SIP Recorder Table.
- Select Ethernet 1(IPv4) as the Signaling/Media Source IP
 - Specifies the Logical IP address for receiving SIP messages. Use this address as the source IP for all SIP messages, leaving the SBC SWe Lite or SBC 1000/2000 through this Signaling Group.

Q Search	Signaling Group Table										
Expand All Collapse All Reload	🧹 🔖 🧭 Add :	ILI I 🖉 I Add SIP SG I 🗙 Total 6 Signaling Group Rows									
🕨 🃁 Call Routing	🗌 Туре	Description	Admin State	Service Status	D	isplay					
🔻 🤣 Signaling Groups	🔻 📋 🗌 SIP	INGRESS_SG	V	Up	2	Counters Channels Session	1				
(SIP) INGRESS_SG (SIP) EGRESS_SG		SIP Recording			Марр	bing Tables					
(SIP) TRANSFER_SG2 (SIP) TRANSFER_SG2 (SIP) INGRESS_SGV6 (SIP) EGRESS_SGV6	SIP Recordin	g Status Enabled Recorder SIP_Recorder_SG1	v v	SIP To (2.850 Override Table	Default (RFC4497) Default (RFC4497)	~ ~				
 Metworking Interfaces System Auth and Directory Services 				Pass-thr	u Peer SIP Response Code	+ Enable	~				
 Protocols SIP 					SIP	IP Details					
 Security Media Tone Tables 				Teams Loca Signa	l Media Optimization ling/Media Source IP	Disable Ethernet 1.370 IP (10.34.	~				
 Telephony Mapping Tables SNMP/Alarms 					Signaling DSCP	40 * [0_63]					

- Configure the UAC IP Address in the Federated IP/FQDN field.
 - The Federated IP/FQDN feature acts as an access control, defining from which server a SIP Signaling Group will accept messages.

Q Search	Signaling Group Table								
Expand All Collapse All Reload	🗸 🔖 🧭 Add S	IP SG 🗙	Total 6 Signaling Grou	p Rows					
▶ 💋 Call Routing	🗌 Туре	Description		Admin State	Service Status	Display			
V Signaling Groups	V 📋 🗌 SIP	INGRESS_SG		₽⁄	Up	Counters Channels Sessions			
(SIP) INGRESS_SG	5								
SIP) TRANSFER_SG1			Listen Ports			Federated IP/FQDN			
SIP) TRANSFER_SG2		Total 2 SID Liston	Part Powe			Total 1 CID Enderstad ID Dow	- 1		
(SIP) INGRESS_SGV6		Total 3 317 Listen	FOIL ROWS		- ×				
(SIF) EGRESS_SOVO	Port	Protocol	TLS Profile ID		IP/FQDN	Netmask/Prefix			
Getworking interfaces	/ 🗌 5060	UDP	N/A		/ 🗌 10.54.	255.255.255.255			
Auth and Directory Services	/ 5060	TCP	N/A						
Protocols	/ 5061	TLS	Default TLS Profi	e					
SIP									
Security Media									
🕨 🍺 Tone Tables	Message Manipula	ion Disabled 🗸							
🕨 🃁 Telephony Mapping Tables							_		
SNMP/Alarms									
Dogging Configuration							Apply		
Emergency Services									

EGRESS_SG

- Attach the Sip Profile you created earlier.
- Select the Agent Type as Back-to-Back User Agent.
- Attach the SIP Server Table "UAS".
- Select the appropriate Media List.

Q Search	Signaling Group Table				
Expand All Collapse All Reload	🧹 📙 ⊘ Add SIP SG 🗙	Total 6 Signaling Group	Rows		
▶ 💋 Call Routing	Type Descri	ption	Admin State	Service Status	Display
V Signaling Groups	SIP INGRE	SS_SG	V	Up	Counters Channels Sessions
(SIP) INGRESS_SG	v 📄 SIP EGRES	S_SG	₽ v	Up	Counters Channels Sessions
SIP) TRANSFER_SG1	Description EGRESS_SG				
(SIP) TRANSFER_SG2 (SIP) INGRESS_SGV6	Admin State Enabled Service Status Up	~			
(or) ECRESS_SOVE		SIP Channels and Routing			
Auth and Directory Services				M	edia Information
Protocols	Action Set Tab	None	✓ +		
 ▶ 10 SIP ▶ 10 Security ▶ 10 Media ▶ 10 Tone Tables 	Call Routing Tab No. of Channe SIP Profi	le EGRESS Is 10 * [1960] le Sip Profile	 ✓ + ✓ + 	Supported Audio Proxy Modes Direct Proxy wi	th Local SRTP
Telephony Mapping Tables	SIP Mod	le Basic Call	~	Supported Proxy	Add/Edit
F SNMP/Alarms Jogging Configuration	Agent Typ	Back-to-Back User Agent	✓✓	Video/Application Modes	* Remove *
Emergency Services	Load Balancir	g Round Robin	~	Media List ID EgressM Proxy Local SRTP	ediaProfile 🗸 +

• Select Ethernet 2 (IPv4) as the Signaling/Media Source IP.

Q Search	Signaling Group Table									
Expand All Collapse All Reload	🧹 🔖 🧭 Add SI	U II. 10 I Add SIP SG I X Total 6 Signaling Group Rows								
🕨 🃁 Call Routing	🗌 Туре	Description	Admin State	Serv	ice Status	Display				
Signaling Groups	▶ 📄 🗌 SIP	INGRESS_SG	₩/	Up		Counters Channels Sessions				
(SIP) INGRESS_SG	V 📄 SIP	EGRESS_SG	∎⁄	Up		Counters Channels Sessions				
SIP) TRANSFER_SG1		SIP Recording			Maj	oping Tables				
SIP) INGRESS_SGV6	SIP Recording	Status Disabled 🗸			SIP To Q.850 Override Table	Default (RFC4497)]			
 Metworking Interfaces System 					Q.850 To SIP Override Table	Default (RFC4497)	J			
 Auth and Directory Services Protocols 					Pass-thru Peer SIP Response Code	Enable 🗸]			
 ▶ ≠ SIP ▶ ≠ Security 					SI	P IP Details	=			
🕨 🥟 Media										
Tone Tables				_	Teams Local Media Optimization	Disable 🗸				
Interprint wapping lables SNMP/Alarms					Signaling/Media Source IF	Ethernet 2.371 IP (10.34.)				
Logging Configuration					Signaling DSC	* [063]				

• Configure the UAS IP Address in the Federated IP/FQDN field.

Signaling Group Table							
🗸 I 🛄 I 🌘	👌 Add SIP S	ig 🗙 Tol	al 6 Signaling Grou	Rows			
	Туре	Description		Admin State	-	Service Status	Display
۱ 🗋 🕨	SIP	INGRESS_SG		₽⁄		Up	Counters Channels Sessions
v 🗋 🗌	SIP	EGRESS_SG		₽ ∕		Up	Counters Channels Sessions
		Liste	en Ports			Fed	lerated IP/FQDN
	,	Total 3 SIP Listen Port I	Pows			Total 1 SIP Fe	ederated TP Pow
	Port	Protocol	TLS Profile ID			IP/FQDN	Netmask/Prefix
/ 🗆	5060	UDP	N/A			/ 🗌 10.54.	255.255.255
/ 0	5060	тср	N/A				
/ 0	5061	TLS	Default TLS Profile				
Messag	e Manipulation	Disabled 🗸					
_	_				-		
							Apply
	Signaling	Signaling Group Tab Add StP S Type SIP SIP Port 5060 5060 5060 Soci Message Manipulation	Signaling Group Table				

INGRESS_SGV6

- Attach the Sip Profile you created earlier.
 Select the Agent Type as Back-to-Back User Agent.
 Attach the SIP Server Table "UACV6".
 Select the appropriate Media List.

Q Search	Signaling Group Table							
Expand All Collapse All Reload	🧹 🛄 🥝 Add SIP SG 🗶	Total 6 Signaling Group Rows						
🕨 🍺 Call Routing	Type Description	Admin State	Service Status	Display				
Signaling Groups	▶ □ SIP INGRESS_SG	₽⁄	Up	Counters Channels Sessions				
SIP) INGRESS_SG	F SIP EGRESS_SG	₹.	Up	Counters Channels Sessions				
(SIP) TRANSFER_SG1	F SIP TRANSFER_SG	1 🔍	Up	Counters Channels Sessions				
(SIP) TRANSFER_SG2	Find SIP TRANSFER_SG	12 🖏	Up	Counters Channels Sessions				
SIP) EGRESS_SGV6	v 📄 SIP INGRESS_SG	V6 🔍	Up	Counters Channels Sessions				
System Auth and Directory Services P Protocols SiP SiP Security	Admin State Enabled V Service Status Up	Channels and Routing	_					
🕨 🍎 Media			Me	dia Information				
Cone Tables Tone Tables Telepown Mapping Tables SNMP/Alarms SNMP/Alarms Jogning Configuration Emergency Services	Action Set Table Ne Call Routing Table 1Ni No. of Channels 10 SIP Profile Sir SIP Mode Ba Agent Type Ba SIP Server Table U. Load Balancino Bon	nne GRESSV6 (1.566) Profile ck-to-Back User Agent CV6 udf Abbin	Supported Audio Modes Direct Proxy Video/Application Media List ID IngressM	h Local SRIP				

- Enable SIP Recording and attach the Recording Server Table created for the SIP recording.
- Select Ethernet 1(IPv6) as the Signaling/Media Source IP.

Q Search	Signalin	Signaling Group Table							
Expand All Collapse All Reload	🛷 l 🛄 l	⊘ Add SIP	SG 🗙 Total 6 Signalir	ig Group Rows					
Call Routing		Туре	Description	Admin State		Service Status	Display		
Signaling Groups	• 🖬 🗆	SIP	INGRESS_SG	ŧ√.		Up	Counters Channels Sessions		
(SIP) INGRESS_SG	• DC	SIP	EGRESS_SG	∎⁄		Up	Counters Channels Sessions		
(SIP) TRANSFER_SG1	۱ 🗈 🗆	SIP	TRANSFER_SG1	₩/		Up	Counters Channels Sessions		
(SIP) TRANSFER_SG2	• 🖬 🗆	SIP	TRANSFER_SG2	₩/		Up	Counters Channels Sessions		
(SIP) EGRESS_SGV6	v	SIP	INGRESS_SGV6	∎ v		Up	Counters Channels Sessions		
Metworking Interfaces			SIP Recording			Мар	pping Tables		
System System Auth and Directory Services Protocols Security Security Media Tone Tables Tone Tables		IP Recording S	status Enabled V SIP_Recorder_SG1 V			SIP To Q.850 Override Table Q.850 To SIP Override Table Pass-thru Peer SIP Response Code	Default (RFC4497) * Default (RFC4497) * Enable		
SNMP/Alarms						SIF	P IP Details		
Emergency Services						Teams Local Media Optimization	Disable 🗸		
						Signaling/Media Source IP	Ethernet 1.370 IP (fd00:10:		
						Signaling DSCP	40 * [0_63]		

• Configure the UACV6 IP Address in the Federated IP/FQDN field.

Q Search	Signaling Group Table						
Expand All Collapse All Reload	🛷 📙 ⊘ Add S	IP SG 🗙	Total 6 Signaling Group R	ows			
🕨 🃁 Call Routing	Туре	Description		Admin State		Service Status	Display
Signaling Groups	Image: SIP	INGRESS_SG		₩		Up	Counters Channels Sessions
(SIP) INGRESS_SG	Image: SIP	EGRESS_SG		₩/		Up	Counters Channels Sessions
(SIP) TRANSFER_SG1	▶ 📄 🗆 SIP	TRANSFER_SG1		₽⁄		Up	Counters Channels Sessions
(SIP) TRANSFER_SG2	▶ 📄 🗆 SIP	TRANSFER_SG2		₩		Up	Counters Channels Sessions
(SIP) EGRESS_SGV6	V 📋 SIP	INGRESS_SGV6		₩.	_	Up	Counters Channels Sessions
Metworking Interfaces							
🕨 🧀 System		Li	sten Ports			Fed	lerated IP/FQDN
Auth and Directory Services		T-t-1 2 CTD List-s D-			a II.	The The LACENCE	- dameter data para
	- ×	Total 3 SIP Listen Po	In Rows				Sociated IP Row
Security	Port	Protocol	TLS Profile ID			IP/FQDN	Netmask/Prefix
🕨 🏓 Media	/ 5060	UDP	N/A		a 11	/fd00:10:	128
🕨 🥩 Tone Tables	. 5060	TCP	N/A			-	
🕨 🃁 Telephony Mapping Tables					- II.		
SNMP/Alarms	/ 5061	ILS	Default TLS Profile				
Logging Configuration							
Emergency Services							
	Message Manipula	Disabled V					
							Apply

EGRESS_SGV6

- Attach the Sip Profile you created earlier.
 Select the Agent Type as Back-to-Back User Agent.
 Attach the SIP Server Table "UASV6".
- Select the appropriate Media List.

Q Search	Signaling Group Table							
Expand All Collapse All Reload	🗸 🔖 🧭 Add SIP	SG 🗙 Total 6 Signaling Gro	up Rows					
▶ 🥬 Call Routing	Туре	Description	Admin State	Service Status	Display			
Signaling Groups	Image: SIP	INGRESS_SG		Up	Counters Channels Sessions			
(SIP) INGRESS_SG	▶ 📄 🗆 SIP	EGRESS_SG	₩⁄	Up	Counters Channels Sessions			
SIP) TRANSFER_SG1	🕨 📄 🗌 SIP	TRANSFER_SG1	₩⁄	Up	Counters Channels Sessions			
(SIP) TRANSFER_SG2	Image: SIP	TRANSFER_SG2	∎⁄	Up	Counters Channels Sessions			
(SIP) EGRESS_SGV6	Image: SIP	INGRESS_SGV6	₩⁄	Up	Counters Channels Sessions			
🕨 🥖 Networking Interfaces	V SIP	EGRESS_SGV6	₽⁄	Up	Counters Channels Sessions			
▶ 📁 Protocols ▶ 📁 SIP ▶ 💋 Security ▶ 💋 Media	Admin State Er Service Status Up	sip Channels and Dauging						
🕨 💋 Tone Tables		SIP Channels and Routing		Me	adia Information			
Galaction (Mapping Tables SNMP/Alams SNMP/Alams Goging Configuration Goging Configuration Genergency Services	Actic Call Ru No SIP 5 Log	n Set Table None EGRESSV6 GEGRESSV6 GEGRESSV6 GEGRESSV6 GEGRESSV6 GEGRESSV6 (1.960) SIP Profile SIP Mode Basic Call Agent Type Back-to-Back User Agent ierver Table UASV6 Gelandnon Gelan	 → →	Supported Audio Modes Direct Video/Application Modes Media List D EgressMr	th Local SRTP			

• Select Ethernet 2 (IPv6) as the Signaling/Media Source IP.

Q Search	Signaling Group Table									
Expand All Collapse All Reload	🧹 I 📙 I	VIL 10 I Add SIP SG X Total 6 Signaling Group Rows								
🕨 🥖 Call Routing	Type Description Admin State					vice Status	Display			
Signaling Groups	• 🗈 C	SIP	INGRESS_SG	₩/	Up		Counters Channels Sessions			
(SIP) INGRESS_SG	F 🗖 🗆	SIP	EGRESS_SG	₩⁄	Up		Counters Channels Sessions			
(SIP) TRANSFER_SG1	• DC	SIP	TRANSFER_SG1	₩/	Up		Counters Channels Sessions			
(SIP) TRANSFER_SG2	• 🗈 C	SIP	TRANSFER_SG2	₩⁄	Up		Counters Channels Sessions			
(SIP) EGRESS_SGV6	• 🗈 C	SIP	INGRESS_SGV6	₩/	Up		Counters Channels Sessions			
Metworking Interfaces	- D	SIP	EGRESS_SGV6	₩/	Up		Counters Channels Sessions			
System			SIP Recording			Maj	ping Tables			
Auth and Directory Services										
▶ Ø SIP	5	IP Recording St	atus Disabled 🗸			SIP To Q.850 Override Table	Default (RFC4497)	~		
🕨 🥖 Security						-	+	_		
🕨 🥖 Media						Q.850 To SIP Override Table	Default (RFC4497)	~		
🕨 🥩 Tone Tables						Pace thru Poor SID Pospons		_		
Telephony Mapping Tables						Code	Enable	~		
SNMP/Alarms						L				
Logging Configuration							P IP Details			
Emergency Services						31	- IF Details			
						Teams Local Media Optimization	Disable	~		
						Signaling/Media Source IF	Ethernet 2.371 IP (fd00:10:	~		
						Signaling DSC	40 * /0631	-		
						bighting book				

• Configure the UASV6 IP Address in the Federated IP/FQDN field.

Q Search	Signaling Group Table						
Expand All Collapse All Reload	🧹 🔖 🧭 Add SI	PSG 🗙 T	otal 6 Signaling Group Rows				
Call Routing	Туре	Description	Admin State	Service Status	Display		
V Signaling Groups	▶ 📄 🗋 SIP	INGRESS_SG	۹.	Up	Counters Channels Sessions		
(SIP) INGRESS_SG	▶ 📄 🗆 SIP	EGRESS_SG	₩ /	Up	Counters Channels Sessions		
SIP) TRANSFER_SG1	Image: SIP	TRANSFER_SG1	₽⁄	Up	Counters Channels Sessions		
(SIP) TRANSFER_SG2	▶ 📄 🗆 SIP	TRANSFER_SG2	₽∕	Up	Counters Channels Sessions		
(SIP) EGRESS_SGV6	▶ 📄 🗆 SIP	INGRESS_SGV6	₽⁄	Up	Counters Channels Sessions		
Metworking Interfaces	🔻 📋 🗌 SIP	EGRESS_SGV6	I [√	Up	Counters Channels Sessions		
Protocoli Protocoli Scurty Scurty Media Toisphony Mapping Tables SubNP/Nams Loging Configuration Security Emergency Services	Port Port 5060 5060 5060 Message Manipulati	List Total 3 SIP Listen Port Protocol UDP TCP TLS on Disabled V	ILS Profile ID N/A N/A Default TLS Profile	Image: Total 1 St Image	Federated IP/FODN PF Federated IP Rov Retmask/Prefix 128		

Configure Transformation Tables

Transformation Tables facilitate the conversion of names, numbers, and other fields when routing a call. They can, for example, convert a public PSTN number into a private extension number or a SIP address (URI). Every entry in a Call Routing Table requires a Transformation Table and is available for user selection. Transformation tables are also configurable as a reusable pool that Action Sets can reference.

To Create a Transformation Table

Each Transformation Table contains a list of entries for creating routing rules. Create each rule in the available order until you reach the end of the table, or when a mandatory entry fails to execute.

From the Settings tab, navigate to Call Routing > Transformation.

- 1. Click the Create (+) icon.
- 2. Enter a descriptive name in the Description text field.
- 3. Click OK.

Create Transf	ormation Table	April 19, 2021 10:17:02	9
Row ID Description	9 ROUTING		
		ОК	

Creating an Entry in a Message Transformation Table

- 1. Click the **Create**(+) icon next to the table you created in the previous step.
- 2. Set the following fields:

Admin State:

Enabled - The default state is Enabled.

Match Type:

Optional: Optional entries must match at least one of the Input Field types.

When a call arrives at a Transformation Table, the incoming message contains several Informational Elements (IEs). These IEs include important call information, such as Called Address/Number, Called Extension, Calling Name, Redirecting Number, and others. Each IE is processed row-by-row in the Transformation Table.

Value (Input/Output):

Specifies the value to match against the selected type. Depending upon the type selected, values are free-form or selected from a menu.

3. Click Apply.

Q Search	ROUTING						
Expand All Collapse All Reload	√1 ⊘ 1+1×1∦	Total 1 Transformation Entr	y Row				_
Call Routing	Admin State Input Field	Type Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key
Transformation	🔻 🔲 🗌 🍢 🛛 Called Add	ress/Number (.*)	Called Address/Number	\1	Mandatory (Must Match)	MY_NUM	1
7777_ROUTE							
8867_ROUTE	Description MY_NUM						
9535_ROUTE	Admin State Enabled	~					
9916_ROUTE	Match Type Mandatory (N	lust Match) 🗸					
Passthrough Untouched					_		
Day Table	Input	Field	Output Field	4			
Call Routing Table	input		ouputrien				
Call Actions	Type Called Address/f	Number 🗸	Type Called Address/Numb	per 🗸			
Signaling Groups	Value (.*)		Value \1				
System							
Auth and Directory Services				_			
Protocols							
▶ D SIP				1	Apply		
Security							

Note

For details on Transformation Table Entry configuration, refer to Creating and Modifying Entries to Transformation Tables. For call digit matching and manipulation through the use of regular expressions, refer to Creating Call Routing Logic with Regular Expressions.

Configure Call Routing Tables

Call Routing tables allow you to configure flexible routes for transferring calls between Signaling Groups and translating the calls.

From the Settings tab, navigate to Call Routing > Call Routing Table.

Modifying an Entry in a Call Routing Table

- 1. Click the **expand** () icon next to the entry you wish to modify.
- 2. Edit the entry properties as required.

Creating an Entry in a Call Routing Table

Call Routing Tables are one of the central connection points of the system, linking Transformation Tables, Message Translations, Cause Code Reroute Tables, Media Lists, and the three types of Signaling Groups (ISDN, SIP, and CAS).

In the SBC Edge, call routing occurs between Signaling Groups.

To route any call to or from a call system connected to the SBC, you must first configure a Signaling Group to represent that device or system. The following list illustrates the hierarchical relationships of the various Telephony routing components of an SBC call system:

- Signaling Group describes the source call and points to a routing definition known as a Call Route Table.
- Call Route Table contains one or more Call Route Entries.
- Call Route Entries points to the destination Signaling Group(s).

Each call routing entry describes how to route the call and also points to a Transformation Table, which defines the conversion of names, numbers, and other fields when routing a call.

To create an entry:

- 1. Click the Create Routing Entry (+) icon.
- 2. Set the following fields:

Admin State:

Enabled - Enables the call route entry for routing the call, displaying it in the configuration header as

Route Priority:

Shows the priority of the route from 1 (highest) to 10 (lowest). Regardless of the order of the routes in the table, higher priority routes are matched first, before matching the lower priority routes.

Number/Name Transformation Table:

Specifies the Transformation Table for this routing entry. The drop-down list is populated from the entries in the Transformation Table.

Destination Signaling Groups:

Specifies the Signaling Groups used as the destination of calls. The first operational Signaling Group from the list is chosen to place the call. Click the Add/Edit button to select the destination signaling group.

Audio Stream Mode:

DSP (default entry): The SBC uses DSP resources for media handling (transcoding); however, it does not support the capabilities and features between endpoints if they are not supported within the SBC (codec/capability mismatch). When you configure DSPs, the Signaling Groups support the DSP in a specific order.

Media Transcoding:

Enabled: Enable Transcoding on SIP-to-SIP calls.

3. Click Apply.





SBC SWe Lite Configuration with TLS

This section describes the steps to configure the SBC SWe Lite with TLS for the SIPREC.

Modify SIP Recording Table entry

A SIP Recorder Table contains information that a Signaling Group uses to start a SIP Recording (SIPREC) session. Use the configuration options to select the Recording Server (SIP Server), SIP channels, routing, and media, which direct the SBC to communicate with the Session Recording Server (SRS).

From the Settings tab, navigate to SIP > SIP Recording.

• Configure the TLS Listen Port and attach the TLS Profile.

Q Search	SIP Recording Table			
Expand All Collapse All Reload	🧹 l 🥥 l 🕂 l 🗙 🛛 Total 1 SI	P Recording Row		
Call Routing	Description Tescription SIP_Recorder_SG1	Admin State	Service Status	Display Counters Channels Sessions
 Metworking Interfaces System Auth and Directory Services 	Liste	n Ports		Federated IP/FQDN
 Protocols SIP Local Registrars Local / Pass-thru Auth Tables SIP Profiles SIP Server Tables Trunk Groups Trunk Groups 	I Total 3 SIP Liste Port Protocol Image: Solid Decision UDP Image: Solid Decision TCP Image: Solid Decision TLS	n Port Rows TLS Profile ID N/A N/A Default TLS Profile		IP/FQDN Netmask/Prefix Table is empty
Remote Authorization Tables Contact Registrant Table Message Manipulation Node-Level SIP Settings SIP Recording SIP Recording SIP Recording	Message Manipulation Disabled V			Арріу

Update SIP Server Table

SIP Server Tables contain information about the SIP devices connected to the SBC Edge. The entries in the tables provide information about the IP Addresses, ports, and protocols used for communicating with each server. The table entries also contain links to counters that are useful for troubleshooting.

From the Settings tab, navigate to SIP > SIP Server Tables > SIPRECORDER_1. Click the expand () icon next to the entry.

- 1. Select the TLS protocol with Port Number 5061.
- 2. Attach the TLS Profile.

Search	SIPRECORDER_1				
Expand All Collapse All Reload	Create SIP Recorder	\bullet \mathbf{X} \mathbf{Z}_2^1	Total 1 SI	IP Server Row	
 Call Routing Signaling Groups Networking Interfaces 	 Host / Do ▼ □ 10.34. 	omain	Server Lookup IP/FQDN	Port 5061	Protocol TLS
 System Auth and Directory Services 		Server Host			Transport
Protocols SIP Local Registrars Local / Pass-thru Auth Tables SIP Profiles SIP Server Tables Default SIP Server UAC UAS	Server Lookup Priority Host FQDN/IP Port Protocol TLS Profile	IP/FQDN 1 10.34. 5061 TLS Default TLS Profil]]* [165535]]* le ~	Monitor	None
SKS SIPRECORDER_1					Apply

NICE Engage Configuration

Make sure you meet the following configuration requirements:

- Ensure that the Nice Integration Dispatch service is running on the NICE server.
- Note that the SBC SWe Lite specific configurations are not required on the NICE server since it accepts SIP traffic from all the IP Addresses.
- Import the SBC Primary Certificate to the Application Server of the NICE Engage Platform (Version: 6.15).

(i) Info

Make sure that the NICE Engage Platform is running on 6.15 or later versions for TLS support.

Trusted Certificate installation on the Application server

You must export and copy the certificate installed on each server in .cer format to all the other servers. For example, if you exported the certificate from the appserver, then be sure to copy it to AIR1/AIR2 and Interact servers.

- 1. Copy AIR1, AIR2, Interaction Server, and SWe Lite root certificates to the Application Server.
- 2. Open the Run command box in Windows, type certmgr.msc, and press OK.

	Run	x
	Type the name of a program, folder, document, or Internet resource, and Windows will open it for you.	
Open:	certmgr.msc 🗸 🗸]
	This task will be created with administrative privileges.	
-		
	OK Cancel Browse	

3. From the certmgr window, navigate to Trusted Root Certification Authorities > Certificates. Right-click and select All Tasks > Import.



4. In the Import Certificate window, select the .cer certificate that you copied in step1 and import it to Trusted Root Certification Authorities >

Certificates. Click Finish.

- 5. Repeat step 4 for other .cer certificates that you copied earlier.
- 6. Repeat the above steps to import each certificate for other servers.

TLS Configuration on NICE Desktop Application

1. Open the NICE Desktop Application in a web browser of the Application Server.

2. Provide a valid User Name and Password and log in to the Application.

Welcome to NICE Engage S	Solutions NICE
User name:	
Password:	Show password
	Forgot your password?
Copyright © 20	

3. Navigate to Administration > System Administrator.



4. Navigate to Settings and select Technician Mode.

NICE®					He	llo NICE, Superus	er Help Setting	s Logout	System Adminis	strator
My Universe Business Analyzer	Reporter	Monitor	Insight Manager	ClearSight	PBO Requests	Tools		Change Password Technician Mode	V Insight Amplifier	•
								Label IT		
Actions								Label IT		

5. From the left Navigation Pane, go to Master Site > CTI Integrations > Media Provider Controllers > Select VRSP server.

Actions V		6
	General	
Active Directory		
Agent Center	Media Provider Controller General Information	
🗄 🖑 Customer Center		
Distributed Cache	Media Provider Controller Type	
Import/Export	General Details	
E License Manager		
E Storage	Name: SIPRECINTERACI MPC 1	
□ Master Site	* Location	
Applications	TP/HostName- APPSERVER	
CIT Integrations		
Henger Connection Manager		
Drivers	Attach Connection Manager	
Key Managers	Additional Media Provider Controller Parameters	
🖃 🏠 Media Provider Conti	Media Provider Controller Reporting Level	
b secondary vrsp		
SIPREC INTERAC		
🗉 🏠 Data Marts		
Database Servers		
insight to Impact		
Interactions Centers		
E Logger Channel Mapping		
Logger Servers		
Media Interconnect		
Playback		
Becorders		
Resiliency		
Security		
Sustem Manning	×	
<		

6. Expand the additional Media Provider Controller parameters.

7. Set the value of the Sip Stack TIs Enabled parameter to Yes.

Actions 🔻 🔀					n H
Organization Active Directory Agent Center Source Center	General Media Prov	ider Controller Gener	ral Information		
🖶 🗣 Distributed Cache	Media Provide	r Controller Type			
🗉 🎡 Import/Export	General Detail	s			
🕀 📴 License Manager	Attach Connec	ction Manager		8	
🖅 🗒 Storage	Additional Med	lia Provider Controller	Parameters		
Master Site	🗌 Display Read	d Only Information Manda	atory fields are marked in bold	Add	
🖻 🧑 CTI Integrations	Parameter Name	•	Parameter Value	^	
Connection Manager	MemoryPageSize	1	1024		
CTI Interfaces	SipStack TIsPort	led	5061		
⊡ 🛱 Drivers 🗉	SipStackTlsIpAdd	dress			
Key Managers	SipStack TisCertil	ficateSerialNumber	00 9d 70 8e 65 ca 22 82 8f		
E Media Provider Conti	SipStackTlsCrlRe	evocationMode	Offline		
secondary vrsp	RedundancvisEn	abled	Yes	~	
Data Marta	Description:	Enable TLS connection.		A	
Data Marts				V	
Insight to Impact					
Interactions Centers	Media Provide	r Controller Reporting	Level		
Logger Channel Mapping					
+ T Logger Servers					
Media Interconnect					
🔄 🛃 Media Library Servers					
Resiliency					
🗄 🔒 Security					
III >					

8. Select Sip Stack TIsCertificateSerialNumber and then enter the serial number of the APPSERVER certificate. Refer to How to get

certificate serial number to obtain the certificate serial number.

Actions 💌 🔀	M					n 🖺
Organization		General				
Active Directory						
Agent Center		Media Provid	er Controller Gen	eral Information		
🗉 💑 Customer Center						
Distributed Cache		Media Provider (Controller Type		8	
Import/Export		General Details			8	
🖃 🚰 License Manager		Attach Connecti	on Manager		8	
E Storage		Additional Media	Provider Controlle	er Parameters	(A)	
🖻 🔝 Master Site		Display Read (only Information Mar	datory fields are marked in hold		
🖶 🛃 Applications		j Display Read o	any mornadon na		Add Add	
😑 🧑 CTI Integrations		Parameter Name		Parameter Value	^	
Connection Manage	n	MemoryPageSize		1024		
CTI Interfaces		SipStack HsEnable	d	Yes 5061		
i γ 🚰 Drivers	=	SipStackTlsIpAddre	ess	5501		
		SipStack TIsCertific	ateSerialNumber	00 9d 70 8e 65 ca 22 82 8f		
🖻 🍈 Media Provider Con	ti	SipStackTIsCertific	ateStoreLocation	Local Machine		
- Secondary vrsp		RedundancvisEnab	led	Yes	×	
SIPREC INTERA	9	Description: s	erial Number of the X5	109 (for Client/Server).		
Data Marts						
Database Servers					· · · · · · · · · · · · · · · · · · ·	
Insight to Impact		Media Provider O	Controller Reportin	g Level	8	
Interactions Centers						
中國 Logger Channel Mappin	15					
B Madia Intersegnedt						
Media Library Servers						
Playback	н					
Becorders						
Resiliency						
Security						
System Manning	~					
<						

9. Click the Save icon displayed on the right-hand corner of the Application.

10. Navigate to Master Site > CTI Integrations > Recorders and select AIR1.

Actions 💌 🔀				5
	Filter: All 💌 Add 😥 Export	General Advanced NICs		
🗈 🎒 Active Directory	Recorders	Recorder Details		
🗈 🗔 Agent Center	AIR 2		AIR1	
🖻 🚭 Customer Center	AIR1	Name:		
🗉 🎯 Distributed Cache		Recorder ID:	65	
Import/Export		Location:	NICEAIR1:51333	
E 📴 License Manager				
E Storage		Recorder Storage		
Master Site		Recorder Storage Partitions		
H EI Applications		E:\	Edit	
E Data Marte				
Database Servers				
Insight to Impact				
Interactions Centers				
E Stanger Channel Mapping				
Logger Servers		Logging		
🗈 💽 Media Interconnect				
Media Library Servers		Edit logging details		
🗊 🔄 Playback		Apply to:	TP Capture	
E Recorders				
- 🖓 Maintenance Mode		Reset to default:	Reset	
Resiliency				
🗄 🛗 Security		Logging details		
System Mapping		Landing launds	Debug	
Iext Recording Servers		Logging level:		
United Logger Servers		Number of log files:	50 🛨	
I		Max, log file size:	20 🕂	
		<	III	>

- 11. After selecting AIR1, click on the Advanced tab and expand IP Capture.
- 12. Select the Certificate Serial parameter, enter the serial number of the AIR1 certificate, and save your changes.

Actions V				n E
Organization	Filter: Al 💌 Add 🔀 Export	General Advanced NICs		
Active Directory	Recorders	IP Capture		(S)
Agent Center	AIR 2		Edit	
🗉 🥵 Customer Center	AIR1			
🕫 😨 Distributed Cache		Parameter Name	Value	
Import/Export		AAC LATM dynamic payload types		
License Manager		AAC-LD dynamic payload types		
T Storage		Audio file cache size	4096	
Master Site		Certificate serial	00 bb 8c e8 a2 42 b1 70 8e	
Applications		Default Target Compression	G729	
CTI Integrations		Dialer Session Duration - Total Recording (. 300	•
A Data Marts				
Database Servers		Description:		
Insight to Impact		Serial number of the certificate for ILS com	munication.	
Interactions Centers				
In the Logger Channel Manning		1		
El Logger Servers		Screen Capture		8
A Media Interconnect		Global		8
Media Library Servers		Recording Session Manager		8
Playback		Retriever		No. 1
Recorders		Archiving Manager		8
🔄 🤪 Maintenance Mode		Recorder Administrator		8
⊕		Text Recording		8
E Security		Real-Time Voice Buffering		8
System Mapping				
Text Recording Servers				
Video Logger Servers				
		<	ш	>

13. Select the SIP transport mode parameter and update it to the TLS.

Actions 🔻 🔣				n 🗄
Organization	Filter: All 💌 Add 🔀 Export	General Advanced NICs		
Active Directory	Recorders	IP Capture		(in the second s
Agent Center	AIR 2			Edit
😨 😴 Customer Center	AIR1			
Distributed Cache		Parameter Name	Δ Value	_
Import/Export		SIP timer mode	1	
💼 🔤 License Manager		SIP transport mode	TLS	
🗉 🛅 Storage		SRTP enabled	True	
Master Site		Summation wait time (milliseconds)	1000	
Applications		Support Late Packet Arrival	False	
CTI Integrations		TLS port	5061	•
🗈 🏠 Data Marts		Description		
Database Servers		Indicates whether or not SRTP is enabled	1.	
Insight to Impact				
Interactions Centers				
🖬 🌆 Logger Channel Mapping				
Logger Servers		Screen Capture		
Media Interconnect		Global		8
Media Library Servers		Recording Session Manager		8
Playback		Retriever		8
Recorders		Archiving Manager		
Maintenance Mode		Recorder Administrator		
A Security		Text Recording		
System Mapping		Real-Time Voice Buffering		
Text Recording Servers				
T Video Logger Servers				
VoIP Recording Gateway				
		<	Ш	>

- 14. Repeat the procedure from step10 to step13 for AIR2 using the certificate serial number of the AIR2 certificate.
- 15. Click **Save** displayed on the right-hand corner of the Application.
- 16. In the CTI Integrations branch, click Apply.
- 17. Restart the following services on the Application server and AIR servers:
 - NICE Dispatch Service
 - NICE Recorder Administrator
 - Media Provider Controller manager
 - NICE IP capture

How to get the Certificate Serial number

Double click on the .cer Certificate and go to the Details tab. Select the Serial Number field and copy the serial number of the certificate.

R	Certificate	x
General Details Certification	n Path	
Show: <all></all>	~	
Field	Value	-
Serial number	V3	
Signature algorithm	sha 1RSA	=
Signature hash algorithm	sha1 bshwetha@rbbn.com, *, SVT,	
Valid to	Wednesday, July 12, 2020 3:00:4 Wednesday, July 10, 2030 3:	
Subject	bshwetha@rbbn.com, *, SVT,	≤
00 9d 70 8e 65 ca	22 82 8f	
	Edit Properties Copy to File	
	OK	

Supplementary Services and Features Coverage

The following checklist depicts the set of services/features covered through the configuration defined in this Interop Guide.

Table	3	:	Feature	Coverage
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Sr. No	Supplementary Services/Features	Coverage
1	SIPREC License Test	✓
2	Call Setup and Termination	✓
3	IP-Interworking	✓
4	Direct Media	✓
5	Call Transfer (Blind/Unattended)	✓
6	Call Transfer (Attended)	✓
7	Multiple Call Transfer - Blind	✓
8	Multiple Call Transfer - Attended	X
9	Call with Audio and Video	X
10	DTMF - Inband and RFC2833	✓

11	Round Robin Option for SRS recorder	\checkmark
12	Channel Hunting	✓
13	Metadata Validation	\checkmark
14	Transcode Calls	\checkmark
15	4xx/5xx Response Handling	\checkmark
16	Call Hold and Resume	\checkmark
17	Call Forward - Unconditional, Busy, and No Answer	\checkmark
18	SIP Keepalive	✓
19	MESSAGE/INFO/REFER/SUBSCRIBE/NOTIFY Handling	✓
20	Long Duration Calls	✓
21	Upgrade and Downgrade Test	✓
22	FAX	✓
23	Basic Calls - Skype Clients	✓
24	Transfer Calls (Blind and Attended) - Skype Clients	✓
25	Conference Calls - Skype Clients	✓
26	Call Park and Retrieve	✓
27	Basic Calls - Teams Clients	✓
28	Call Hold and Resume - Teams Clients	✓
29	Call Transfer before and after connect - Teams Clients	✓
30	Call Queue	✓
31	Music On Hold	✓
32	Call Conference	✓
33	Voice Mail	✓
34	Simultaneous Ringing	✓
35	E911 Calls	✓
36	Group Call Pickup	✓
37	SRTP on Recording Session	X

<u>Legend</u>



Caveats

Note the following limitations about this Interop:

- SBC SWe Lite does not honor Request/Response from the SIPREC with a=inactive/sendonly/sendrecv
- SBC SWe Lite does not send encrypted media to the SIPREC.
- SBC SWe Lite supports recording of incoming calls only.

Support

For any support related queries about this guide, please contact your local Ribbon representative or the following numbers and website:

- Sales and Support: 1-833-742-2661
- Other Queries: 1-877-412-8867
- Website: https://ribboncommunications.com/about-us

References

For detailed information about Ribbon products and solutions, visit: https://ribboncommunications.com/products

For detailed information about the NICE Engage platform, visit: https://www.nice.com/

Conclusion

This Interoperability Guide describes how to successfully configure the NICE interop for Ribbon SBC SWe Lite and NICE Engage platform SIPREC server.

The guide provides information about all tested features and capabilities. It records all limitations, notes, and observations to provide you with an accurate understanding of what this guide covers and what it does not.

Configuration guidance in this document enables you to replicate the same base setup; however, you may require additional configuration changes to suit the exact deployment environment.

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