

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

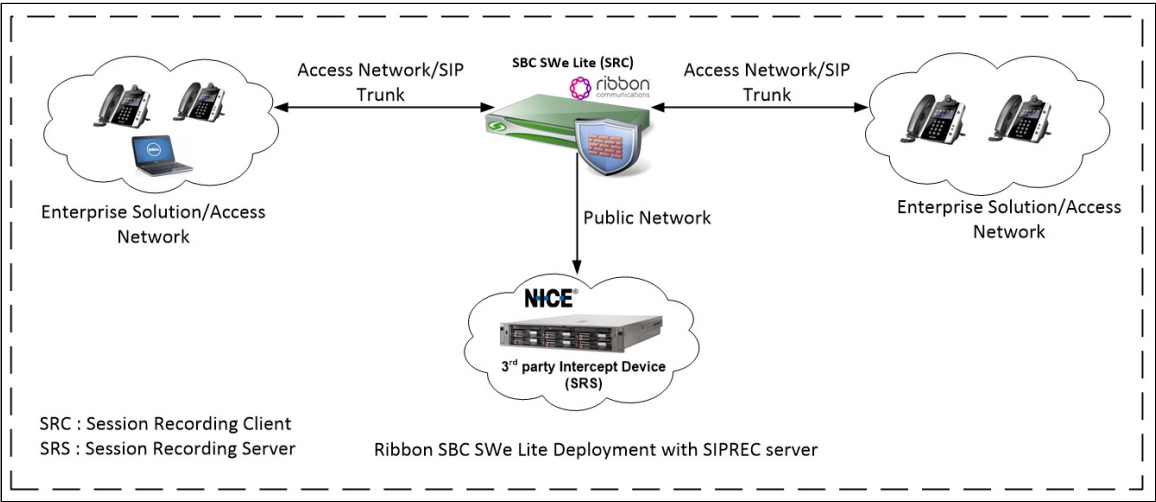
The configuration uses the following equipment and software:

Table 2 : Requirements

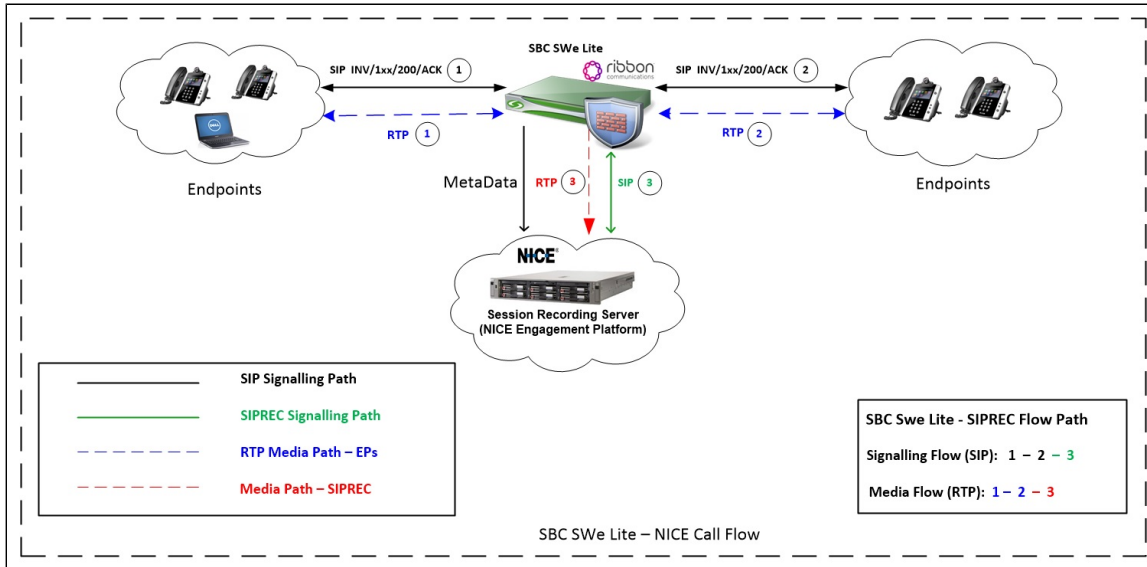
Product	Equipment	Software Version
Ribbon Networks	Ribbon SBC SWe Lite	9.0.1
Third-party Equipment	NICE Engage platform	4.1
Administration and Debugging Tools	Wireshark	3.2.7
	LX Tool	2.1.0.6

Network Topology and E2E Flow Diagrams

SBC SWe Lite - NICE Engage Deployment Topology



Interoperability Test Lab Topology and Call Flow Diagram



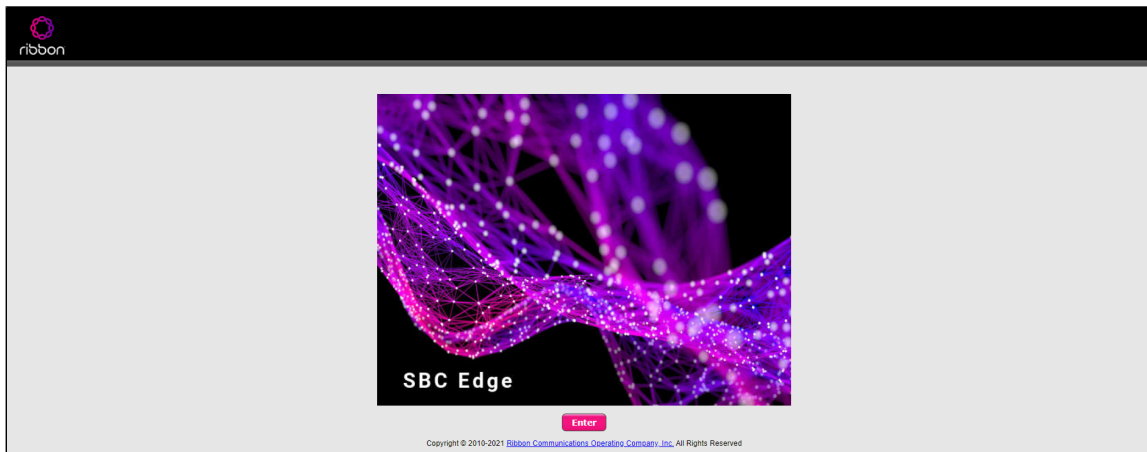
Installing the SBC SWe Lite

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.



Welcome to Ribbon SBC SWe Lite

Users (authorized or unauthorized) have no explicit or implicit expectation of privacy. Any or all uses of this system and all files on this system may be intercepted, monitored, recorded, copied, audited, inspected, and disclosed to authorized site, customer administrative, and law enforcement personnel, as well as authorized officials of government agencies, both domestic and foreign. By using this system, the user consents to such interception, monitoring, recording, copying, auditing, inspection, and disclosure at the discretion of authorized personnel.

Unauthorized or improper use of this system may result in administrative disciplinary action and civil and criminal penalties. By continuing to use this system you indicate your awareness of and consent to these terms and conditions of use. CANCEL YOUR LOGIN IMMEDIATELY if you do not agree to the conditions stated in this warning.

User Name
admin

Password
.....

Login
Cancel

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License and TLS Certificates

View License

Search...

Expand All | Collapse All | Reload

- Call Routing
- Signaling Groups
- Networking Interfaces
- System
 - Node-Level Settings
 - Licensing
 - Current Licenses
 - Install New License
 - Software Management
- Auth and Directory Services
- Protocols
- SIP
- Security
- Media
- Tone Tables
- Telephony Mapping Tables

Current Licenses

Historical Usage | Download License File

License Format Version 3

Feature Licenses

Total 6 Feature License Rows

Feature	Licensed	Total Licenses	Available Licenses	Feature Expiration
SIP Signaling Sessions		300	300	August 09, 2021 23:59:59
Enhanced Media Sessions with Transcoding		100	100	August 09, 2021 23:59:59
Enhanced Media Sessions without Transcoding		600	600	August 09, 2021 23:59:59
SIP Registrations		300	300	August 09, 2021 23:59:59
AMR-WB		Not Licensed	Not Licensed	Not Applicable
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.

- To import a Trusted CA Certificate, click the Import Trusted CA Certificate () Icon.
- 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.

Logical Interfaces							
Total 5 LogicalInterface Rows							
Interface Name	IPv4 Address	IPv6 Address	Description	Admin State	Display	Primary Key	
Admin IP	10.34.3.132			Enabled	Counters	35	
Ethernet 1 IP	10.10.10.10			Enabled	Counters	36	
Ethernet 1.370 IP	10.34.5.132	fd00:10:6b50:5040::184		Enabled	Counters	1	
Ethernet 2 IP	20.20.20.20			Enabled	Counters	37	
Ethernet 2.371 IP	10.34.7.132	fd00:10:6b50:5060::184		Enabled	Counters	2	

Ethernet 1 IP

Search...	Ethernet 1.370 IP	10.34.5.132	fd00:10:6b50:5040::184	Enabled	Counters	1
Identification/Status						
Interface Name Ethernet 1.370 IP						
I/F Index 8						
Alias						
Description						
Admin State Enabled						
Networking						
MAC Address 00:0c:29:00:00:00						
VLAN tag 370						
IP Addressing Mode Both						
IPv4 Information						
IP Assign Method Static						
Primary Address 10.34.5.132						
Primary Netmask 255.255.254.0						
Media Next Hop IP 10.34.7.132						
IPv6 Information						
Link Local Address fe80::20c:29ff:fe00:0000						
Link Local Prefix 64						
Primary Address fd00:10:6b50:5040::184						
Primary Address Prefix 60						
Media Next Hop IP fd00:10:6b50:5060::184						

Ethernet 2 IP

Search...

Expand All | Collapse All | Reload

- Call Routing
- Signaling Groups
- Networking Interfaces
 - Logical Interfaces
 - Admin IP
 - Ethernet 1 IP
 - Ethernet 1.370 IP
 - Ethernet 2 IP
 - Ethernet 2.371 IP**
- System
- Auth and Directory Services
- Protocols
- SIP
- Security
- Media
- Tone Tables
- Telephony Mapping Tables
- SNMP/Alarms
- Logging Configuration
- Emergency Services

Ethernet 2.371 IP

10.34. [redacted]

fd00:10: [redacted]

Enabled

Counters 2

Identification/Status

Interface Name Ethernet 2.371 IP

I/F Index 9

Alias

Description

Admin State Enabled

Networking

MAC Address 00:0c: [redacted]

VLAN tag 371

IP Addressing Mode Both

IPv4 Information

IP Assign Method Static

Primary Address 10.34. [redacted] * [redacted]

Primary Netmask 255.255.254.0 * [redacted]

Media Next Hop IP 10.34. [redacted] * [redacted]

IPv6 Information

Link Local Address fe80::20c: [redacted]

Link Local Prefix 64

Primary Address fd00:10: [redacted] * [redacted]

Primary Address Prefix 60 * [redacted]

Media Next Hop IP fd00:10: [redacted] * [redacted]

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.

Search...

Expand All | Collapse All | Reload

- Call Routing
- Signaling Groups
- Networking Interfaces
- System
- Auth and Directory Services
- Protocols**
 - DNS
 - IP**
 - Static Routes**
 - Routing Table
 - Static ARP
 - Access Control Lists
 - NAT
 - IPv6

Static IP Route Table

+

-

Total 1 IP Route Row

Row ID	Destination IP	Mask	Gateway	Administrative Distance	Primary Key
1	0.0.0.0	0.0.0.0	10.34. [redacted]	1	1

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**.

SIP Server Table Entry

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

1. Click on the desired SIP Server Table.

UAC - IPv4

Search...

Expand All | Collapse All | Reload

- Call Routing
- Signaling Groups
- Networking Interfaces
- System
- Auth and Directory Services
- Protocols
- SIP**
 - Local Registrars
 - Local / Pass-thru Auth Tables
 - SIP Profiles
 - SIP Server Tables**
 - Default SIP Server
 - UAC**
 - UAS
 - UAS1
 - UAS2
 - UAS3
 - UACV6
 - UASV6
 - SIP_RECORDER1
 - SIP_RECORDER2
 - SIP_RECORDERV6
 - Trunk Groups
 - NAT Qualified Prefix Tables

UAC

Create SIP Server ✖ !

Total 1 SIP Server Row

	Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
	10.54. [redacted]	IP/FQDN	35070	UDP	Counters	1	1

Server Host

Server Lookup IP/FQDN

Priority 1

Host FQDN/IP 10.54. [redacted]

Port 35070

Protocol UDP

Transport

Monitor None

Remote Authorization and Contacts

Remote Authorization Table None

Contact Registrant Table None

Session URI Validation Liberal

Apply

Search...

Expand All | Collapse All | Reload

- Call Routing
- Signaling Groups
- Networking Interfaces
- System
- Auth and Directory Services
- Protocols
 - SIP**
 - Local Registrars
 - Local / Pass-thru Auth Tables
 - SIP Profiles
 - SIP Server Tables**
 - Default SIP Server
 - UAC
 - UAS
 - UAS1
 - UAS2
 - UAS3
 - UACV6**
 - UASV6
 - SIP_RECORDER1
 - SIP_RECORDER2
 - SIP_RECORDERV6
 - Trunk Groups
 - NAT Qualified Prefix Tables

UACV6

Create SIP Server ✖ 🔍

Total 1 SIP Server Row

	Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
<input type="checkbox"/>	fd00:10::	IP/FQDN	35110	UDP	Counters	1	1

Server Host

Server Lookup IP/FQDN

Priority 1

Host FQDN/IP fd00:10::

Port 35110 * [1..65535]

Protocol UDP *

Remote Authorization and Contacts

Remote Authorization Table None +

Contact Registrant Table None +

Session URI Validation Liberal

Transport

Monitor None

Apply

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Search...

Expand All | Collapse All | Reload

- Call Routing
- Signaling Groups
- Networking Interfaces
- System
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 - UAS3
 - UACV6
 - UASV6
 - SIP_RECORDER1
 - SIP_RECORDER2
 - SIP_RECORDERV6
 - Trunk Groups
 - NAT Qualified Prefix Tables

UAS1

Create SIP Server ✕ ! ? Total 1 SIP Server Row

	Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
<input checked="" type="checkbox"/>	10.54. [REDACTED]	IP/FQDN	35090	UDP	Counters	1	1

Server Host

Server Lookup IP/FQDN

Priority 1

Host FQDN/IP 10.54. [REDACTED] *

Port 35090 * [1..65535]

Protocol UDP *

Transport

Monitor None

Remote Authorization and Contacts

Remote Authorization Table None +

Contact Registrant Table None +

Session URI Validation Liberal

Apply

UAS - IPv6

Search...

Expand All | Collapse All | Reload

- Call Routing
- Signaling Groups
- Networking Interfaces
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- Protocols
 - SIP**
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 - UAS3
 - UACV6
 - UASV6**
 - SIP_RECORDER1
 - SIP_RECORDER2
 - SIP_RECORDERV6
 - Trunk Groups
 - NAT Qualified Prefix Tables

UASV6

Create SIP Server ✕ ! ? Total 1 SIP Server Row

	Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
<input checked="" type="checkbox"/>	fd00:10: [REDACTED]	IP/FQDN	35130	UDP	Counters	1	1

Server Host

Server Lookup IP/FQDN

Priority 1

Host FQDN/IP fd00:10: [REDACTED] *

Port 35130 * [1..65535]

Protocol UDP *

Transport

Monitor None

Remote Authorization and Contacts

Remote Authorization Table None +

Contact Registrant Table None +

Session URI Validation Liberal

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 Server Table

April 21, 2021 05:17:46

Row ID

11

Description

SIP_RECORDER1

Type

SIP Recorder

OK

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

Search...

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SIP Profiles

SIP Server Tables

Default SIP Server

UAC

UAS

UAS1

UAS2

UAS3

UACV6

UASV6

SIP_RECORDER1

SIP_RECORDER2

SIP_RECORDERV6

SIP_RECORDER1

Create SIP Recorder

Total 1 SIP Server Row

Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
10.34.	IP/FQDN	5060	UDP	Counters	1	1

Server Host

Server Lookup

IP/FQDN

Priority

1

Host FQDN/IP

10.34.

Port

5060

Protocol

UDP

Transport

Monitor

None

Apply

SIPREC - IPv6

Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

Networking Interfaces

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Auth and Directory Services

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SIP Profiles

SIP Server Tables

Default SIP Server

UAC

UAS

UAS1

UAS2

UAS3

UACV6

UASV6

SIP_RECORDER1

SIP_RECORDER2

SIP_RECORDERV6

SIP_RECORDERV6

Create SIP Recorder

Total 1 SIP Server Row

Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
fd00:10:	IP/FQDN	35140	UDP	Counters	1	1

Server Host

Server Lookup

IP/FQDN

Priority

1

Host FQDN/IP

fd00:10:

Port

35140

Protocol

UDP

Transport

Monitor

None

Apply

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

The screenshot shows the 'SIP Recording Table' configuration page for 'SIP_Recorder_SG1'. The left sidebar shows the navigation tree with 'SIP' and 'SIP Recording' highlighted. The main panel displays the configuration for 'SIP_Recorder_SG1' with the following details:

- Description:** SIP_Recorder_SG1
- Admin State:** Enabled
- SIP Channels and Routing:**
 - No. of Channels: 100 (range [1..960])
 - SIP Profile: Default SIP Profile
 - Recording Server Table: SIP_RECORDER1
 - Load Balancing: Round Robin
 - Channel Hunting: Most Idle
- SIP IP Details:**
 - Signaling/Media Source IP: Ethernet 1.370 IP (10.34. [redacted])
 - Signaling DSCP: 40 (range [0..63])

SIP Recording Table 2

The screenshot shows the 'SIP Recording Table' configuration page for 'SIP_Recorder_SG2'. The left sidebar shows the navigation tree with 'SIP' and 'SIP Recording' highlighted. The main panel displays the configuration for 'SIP_Recorder_SG2' with the following details:

- Description:** SIP_Recorder_SG2
- Admin State:** Enabled
- SIP Channels and Routing:**
 - No. of Channels: 100 (range [1..960])
 - SIP Profile: Default SIP Profile
 - Recording Server Table: SIP_RECORDER2
 - Load Balancing: Round Robin
 - Channel Hunting: Most Idle
- SIP IP Details:**
 - Signaling/Media Source IP: Ethernet 2.371 IP (10.34. [redacted])
 - Signaling DSCP: 40 (range [0..63])



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

[Expand All](#) | [Collapse All](#) | [Reload](#)

- Call Routing
- Signaling Groups
- Networking Interfaces
- System
- Auth and Directory Services
- Protocols
- SIP
- Security
- Media
 - Media System Configuration
 - Media Profiles
 - Default G711A
 - Default G711u
 - G729A**
 - G726
 - SDES-SRTP Profiles

Media Profiles

Create Media Profile
|
✖

Total 4 Media Profile Rows

	Codec	Description	Primary Key
▶	<input type="checkbox"/> G.711 A-Law	Default G711A	1
▶	<input type="checkbox"/> G.711 μ-Law	Default G711u	2
▼	<input type="checkbox"/> G.729	G729A	5

Voice Codec Configuration

Description

Codec

Payload Size
ms

Apply

Codec - G.726

[Expand All](#) | [Collapse All](#) | [Reload](#)

- Call Routing
- Signaling Groups
- Networking Interfaces
- System
- Auth and Directory Services
- Protocols
- SIP
- Security
- Media
 - Media System Configuration
 - Media Profiles
 - Default G711A
 - Default G711u
 - G729A
 - G726**
 - SDES-SRTP Profiles
 - Media List
- Tone Tables
- Telephony Mapping Tables
- SNMP/Alarms

Media Profiles

Create Media Profile
|
✖

Total 4 Media Profile Rows

	Codec	Description	Primary Key
▶	<input type="checkbox"/> G.711 A-Law	Default G711A	1
▶	<input type="checkbox"/> G.711 μ-Law	Default G711u	2
▶	<input type="checkbox"/> G.729	G729A	5
▼	<input type="checkbox"/> G.726	G726	6

Voice Codec Configuration

Description

Codec

Rate
32000 b/s

Payload Size
ms

Payload Type
2

Apply

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**.

[Expand All](#) | [Collapse All](#) | [Reload](#)

- Call Routing
- Signaling Groups
- Networking Interfaces
- System
- Auth and Directory Services
- Protocols
- SIP
- Security
- Media**
 - Media System Configuration
 - Media Profiles
 - SDES-SRTP Profiles
 - Media List**
 - Default Media List
 - IngressMediaProfile**
 - EgressMediaProfile
 - SiprecMediaProfile
 - Tone Tables
 - Telephony Mapping Tables

Media List View

+

×

Total 4 Media List Rows

Description	Primary Key
Default Media List	1
IngressMediaProfile	2

Description

IngressMediaProfile

Media Profiles List

Default G711A

Default G711u

G729A

G726

Up

Down

Add/Edit

Remove

SDES-SRTP Profile

None

Associated SIP SG Listen Ports should be TLS only. +

Media DSCP

46

* [0..63]

Dead Call Detection

Disabled

Silence Suppression

Enabled

[Expand All](#) | [Collapse All](#) | [Reload](#)

- Call Routing
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- Media**
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 - Media List**
 - Default Media List
 - IngressMediaProfile
 - EgressMediaProfile**
 - SiprecMediaProfile
 - Tone Tables
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Media List View

+

×

Total 4 Media List Rows

Description	Primary Key
Default Media List	1
IngressMediaProfile	2
EgressMediaProfile	3

Description

EgressMediaProfile

Media Profiles List

Default G711A

Default G711u

G729A

G726

Up

Down

Add/Edit

Remove

SDES-SRTP Profile

None

Associated SIP SG Listen Ports should be TLS only. +

Media DSCP

46

* [0..63]

Dead Call Detection

Disabled

Silence Suppression

Enabled

Configure Signaling Groups

Signaling groups allow grouping telephony channels for routing and shared configuration. They are used for routing calls and selecting [Call Routes](#).

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

INGRESS_SG

- 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.

Signaling Group Table

Total 6 Signaling Group Rows

Type	Description	Admin State	Service Status	Display
<input checked="" type="checkbox"/> SIP	INGRESS_SG	Enabled	Up	Counters Channels Sessions

Description: INGRESS_SG
Admin State: Enabled
Service Status: Up

SIP Channels and Routing

Action Set Table: None
Call Routing Table: INGRESS
No. of Channels: 10 (1..960)
SIP Profile: Sip Profile
SIP Mode: Basic Call
Agent Type: Back-to-Back User Agent
SIP Server Table: UAC
Load Balancing: Round Robin

Media Information

Supported Audio Modes: DSP, Proxy, Direct, Proxy with Local SRTP
Supported Video/Application Modes: Proxy, Direct
Media List ID: IngressMediaProfile

- 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. 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.

Signaling Group Table

Total 6 Signaling Group Rows

Type	Description	Admin State	Service Status	Display
<input checked="" type="checkbox"/> SIP	INGRESS_SG	Enabled	Up	Counters Channels Sessions

SIP Recording

SIP Recording Status: Enabled
SIP Recorder: SIP_Recorder_SG1

Mapping Tables

SIP To Q.850 Override Table: Default (RFC4497)
Q.850 To SIP Override Table: Default (RFC4497)
Pass-thru Peer SIP Response Code: Enable

SIP IP Details

Teams Local Media Optimization: Disable
Signaling/Media Source IP: Ethernet 1.370 IP (10.34.10.34)
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.

Signaling Group Table

Total 6 Signaling Group Rows

Type	Description	Admin State	Service Status	Display
<input checked="" type="checkbox"/> SIP	INGRESS_SG	Enabled	Up	Counters Channels Sessions

Listen Ports

Total 3 SIP Listen Port Rows

Port	Protocol	TLS Profile ID
5060	UDP	N/A
5060	TCP	N/A
5061	TLS	Default TLS Profile

Federated IP/FQDN

Total 1 SIP Federated IP Row

IP/FQDN	Netmask/Prefix
10.54.10.34	255.255.255.255

Message Manipulation: Disabled

Apply

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.

Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

(SIP) INGRESS_SG

(SIP) EGRESS_SG

(SIP) TRANSFER_SG1

(SIP) TRANSFER_SG2

(SIP) INGRESS_SGV6

(SIP) EGRESS_SGV6

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Emergency Services

Signaling Group Table

Total 6 Signaling Group Rows

Type	Description	Admin State	Service Status	Display
SIP	INGRESS_SG	Up	Up	Counters Channels Sessions
SIP	EGRESS_SG	Up	Up	Counters Channels Sessions

Description: EGRESS_SG

Admin State: Enabled

Service Status: Up

SIP Channels and Routing

Action Set Table: None

Call Routing Table: EGRESS

No. of Channels: 10

SIP Profile: Sip Profile

SIP Mode: Basic Call

Agent Type: Back-to-Back User Agent

SIP Server Table: UAS

Load Balancing: Round Robin

Media Information

Supported Audio Modes: DSP, Proxy, Direct, Proxy with Local SRTP

Supported Video/Application Modes: Proxy, Direct

Media List ID: EgressMediaProfile

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

Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

(SIP) INGRESS_SG

(SIP) EGRESS_SG

(SIP) TRANSFER_SG1

(SIP) TRANSFER_SG2

(SIP) INGRESS_SGV6

(SIP) EGRESS_SGV6

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Signaling Group Table

Total 6 Signaling Group Rows

Type	Description	Admin State	Service Status	Display
SIP	INGRESS_SG	Up	Up	Counters Channels Sessions
SIP	EGRESS_SG	Up	Up	Counters Channels Sessions

SIP Recording

SIP Recording Status: Disabled

Mapping Tables

SIP To Q.850 Override Table: Default (RFC4497)

Q.850 To SIP Override Table: Default (RFC4497)

Pass-thru Peer SIP Response Code: Enable

SIP IP Details

Teams Local Media Optimization: Disable

Signaling/Media Source IP: Ethernet 2.371 IP (10.34)

Signaling DSCP: 40

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

Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

(SIP) INGRESS_SG

(SIP) EGRESS_SG

(SIP) TRANSFER_SG1

(SIP) TRANSFER_SG2

(SIP) INGRESS_SGV6

(SIP) EGRESS_SGV6

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Signaling Group Table

Total 6 Signaling Group Rows

Type	Description	Admin State	Service Status	Display
SIP	INGRESS_SG	Up	Up	Counters Channels Sessions
SIP	EGRESS_SG	Up	Up	Counters Channels Sessions

Listen Ports

Total 3 SIP Listen Port Rows

Port	Protocol	TLS Profile ID
5060	UDP	N/A
5060	TCP	N/A
5061	TLS	Default TLS Profile

Federated IP/FQDN

Total 1 SIP Federated IP Row

IP/FQDN	Netmask/Prefix
10.54	255.255.255.255

Message Manipulation: Disabled

Apply

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.

Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

(SIP) INGRESS_SG

(SIP) EGRESS_SG

(SIP) TRANSFER_SG1

(SIP) TRANSFER_SG2

(SIP) INGRESS_SGV6

(SIP) EGRESS_SGV6

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Signaling Group Table

Total 6 Signaling Group Rows

Type	Description	Admin State	Service Status	Display
SIP	INGRESS_SG	Up	Up	Counters Channels Sessions
SIP	EGRESS_SG	Up	Up	Counters Channels Sessions
SIP	TRANSFER_SG1	Up	Up	Counters Channels Sessions
SIP	TRANSFER_SG2	Up	Up	Counters Channels Sessions
SIP	INGRESS_SGV6	Up	Up	Counters Channels Sessions

Description: INGRESS_SGV6

Admin State: Enabled

Service Status: Up

SIP Channels and Routing

Action Set Table: None

Call Routing Table: INGRESS_SG

No. of Channels: 10

SIP Profile: SIP Profile

SIP Mode: Basic Call

Agent Type: Back-to-Back User Agent

SIP Server Table: UACV6

Load Balancing: Round Robin

Media Information

Supported Audio Modes: DSP, Proxy, Direct, Proxy with Local SRTP

Supported Video/Application Modes: Proxy, Direct

Media List ID: IngressMediaProfile

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

Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

(SIP) INGRESS_SG

(SIP) EGRESS_SG

(SIP) TRANSFER_SG1

(SIP) TRANSFER_SG2

(SIP) INGRESS_SGV6

(SIP) EGRESS_SGV6

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Signaling Group Table

Total 6 Signaling Group Rows

Type	Description	Admin State	Service Status	Display
SIP	INGRESS_SG	Up	Up	Counters Channels Sessions
SIP	EGRESS_SG	Up	Up	Counters Channels Sessions
SIP	TRANSFER_SG1	Up	Up	Counters Channels Sessions
SIP	TRANSFER_SG2	Up	Up	Counters Channels Sessions
SIP	INGRESS_SGV6	Up	Up	Counters Channels Sessions

SIP Recording

SIP Recording Status: Enabled

SIP Recorder: SIP_Recorder_SG1

Mapping Tables

SIP To Q.850 Override Table: Default (RFC4497)

Q.850 To SIP Override Table: Default (RFC4497)

Pass-thru Peer SIP Response Code: Enable

SIP IP Details

Teams Local Media Optimization: Disable

Signaling/Media Source IP: Ethernet 1.370 IP (fd00:10::)

Signaling DSCP: 40

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

Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

(SIP) INGRESS_SG

(SIP) EGRESS_SG

(SIP) TRANSFER_SG1

(SIP) TRANSFER_SG2

(SIP) INGRESS_SGV6

(SIP) EGRESS_SGV6

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Signaling Group Table

Total 6 Signaling Group Rows

Type	Description	Admin State	Service Status	Display
SIP	INGRESS_SG	Up	Up	Counters Channels Sessions
SIP	EGRESS_SG	Up	Up	Counters Channels Sessions
SIP	TRANSFER_SG1	Up	Up	Counters Channels Sessions
SIP	TRANSFER_SG2	Up	Up	Counters Channels Sessions
SIP	INGRESS_SGV6	Up	Up	Counters Channels Sessions

Listen Ports

Total 3 SIP Listen Port Rows

Port	Protocol	TLS Profile ID
5060	UDP	N/A
5060	TCP	N/A
5061	TLS	Default TLS Profile

Federated IP/FQDN

Total 1 SIP Federated IP Row

IP/FQDN	Netmask/Prefix
fd00:10::	128

Message Manipulation: Disabled

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.

Signaling Group Table

Total 6 Signaling Group Rows

Type	Description	Admin State	Service Status	Display
<input type="checkbox"/> SIP	INGRESS_SG		Up	Counters Channels Sessions
<input type="checkbox"/> SIP	EGRESS_SG		Up	Counters Channels Sessions
<input type="checkbox"/> SIP	TRANSFER_SG1		Up	Counters Channels Sessions
<input type="checkbox"/> SIP	TRANSFER_SG2		Up	Counters Channels Sessions
<input type="checkbox"/> SIP	INGRESS_SGV6		Up	Counters Channels Sessions
<input checked="" type="checkbox"/> SIP	EGRESS_SGV6		Up	Counters Channels Sessions

Description: EGRESS_SGV6

Admin State: Enabled

Service Status: Up

SIP Channels and Routing

Action Set Table: None

Call Routing Table: EGRESSV6

No. of Channels: 10

SIP Profile: Sip Profile

SIP Mode: Basic Call

Agent Type: Back-to-Back User Agent

SIP Server Table: UASV6

Load Balancing: Round Robin

Media Information

Supported Audio Modes: DSP, Proxy, Direct, Proxy with Local SRTP

Supported Video/Application Modes: Proxy, Direct

Media List ID: EgressMediaProfile

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

Signaling Group Table

Total 6 Signaling Group Rows

Type	Description	Admin State	Service Status	Display
<input type="checkbox"/> SIP	INGRESS_SG		Up	Counters Channels Sessions
<input type="checkbox"/> SIP	EGRESS_SG		Up	Counters Channels Sessions
<input type="checkbox"/> SIP	TRANSFER_SG1		Up	Counters Channels Sessions
<input type="checkbox"/> SIP	TRANSFER_SG2		Up	Counters Channels Sessions
<input type="checkbox"/> SIP	INGRESS_SGV6		Up	Counters Channels Sessions
<input checked="" type="checkbox"/> SIP	EGRESS_SGV6		Up	Counters Channels Sessions

SIP Recording

SIP Recording Status: Disabled

Mapping Tables

SIP To Q.850 Override Table: Default (RFC4497)

Q.850 To SIP Override Table: Default (RFC4497)

Pass-thru Peer SIP Response Code: Enable

SIP IP Details

Teams Local Media Optimization: Disable

Signaling/Media Source IP: Ethernet 2/371 IP (fd00:10::...)

Signaling DSCP: 40

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

Signaling Group Table

Total 6 Signaling Group Rows

Type	Description	Admin State	Service Status	Display
<input type="checkbox"/> SIP	INGRESS_SG		Up	Counters Channels Sessions
<input type="checkbox"/> SIP	EGRESS_SG		Up	Counters Channels Sessions
<input type="checkbox"/> SIP	TRANSFER_SG1		Up	Counters Channels Sessions
<input type="checkbox"/> SIP	TRANSFER_SG2		Up	Counters Channels Sessions
<input type="checkbox"/> SIP	INGRESS_SGV6		Up	Counters Channels Sessions
<input checked="" type="checkbox"/> SIP	EGRESS_SGV6		Up	Counters Channels Sessions

Listen Ports

Total 3 SIP Listen Port Rows

Port	Protocol	TLS Profile ID
5060	UDP	N/A
5060	TCP	N/A
5061	TLS	Default TLS Profile

Federated IP/FQDN

Total 1 SIP Federated IP Row

IP/FQDN	Netmask/Prefix
fd00:10::...	128

Message Manipulation: Disabled

Apply

Configure Transformation Tables

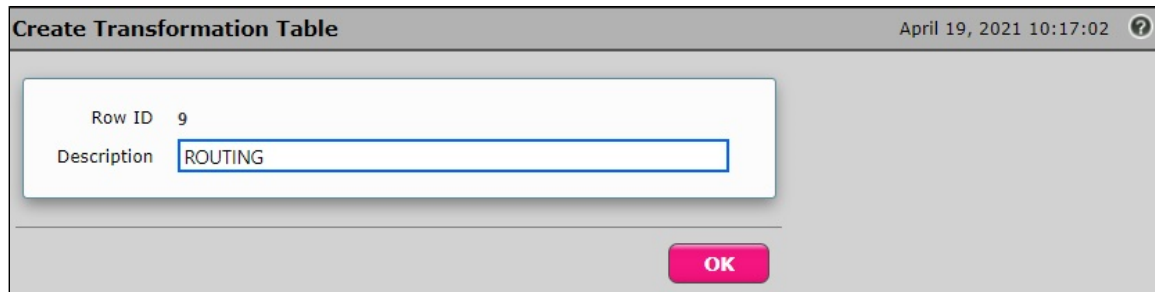
Transformation Tables facilitate the conversion of names, numbers, and other fields when routing a call. They can, for example, convert a public PSTN number into a private extension number or a SIP address (URI). Every entry in a [Call Routing Table](#) requires a Transformation Table and is available 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**.



The dialog box titled "Create Transformation Table" has a date and time stamp "April 19, 2021 10:17:02" and a help icon. It contains a form with "Row ID" set to 9 and a "Description" field containing the text "ROUTING". An "OK" button is at the bottom right.

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:

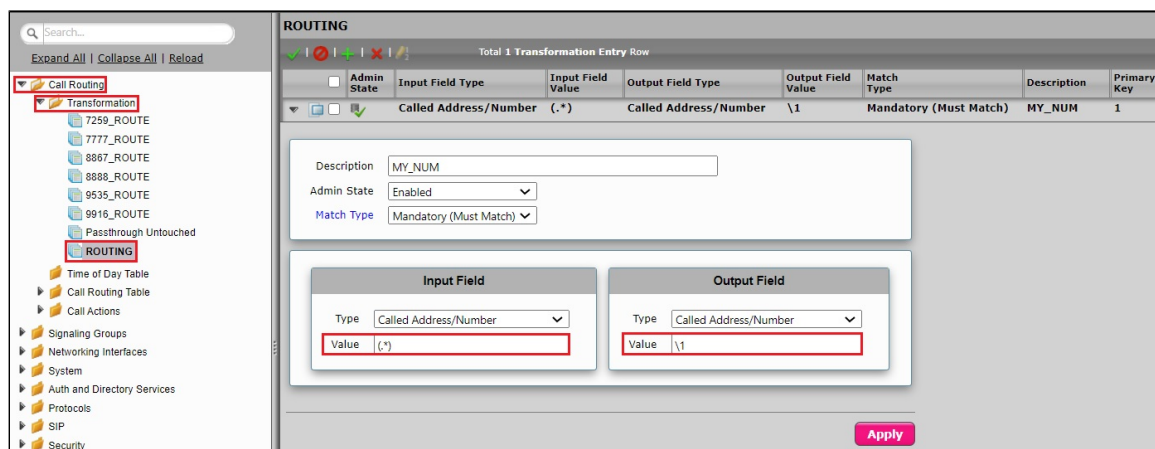
Optional: Optional entries must match at least one of the Input Field types.

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**.



The interface shows the "ROUTING" table configuration. On the left is a tree view with "Call Routing" expanded, showing "Transformation" and "ROUTING" (highlighted). The main area shows a table with one entry. Below the table is a configuration form for the entry.

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key
<input checked="" type="checkbox"/>	Called Address/Number	(*)	Called Address/Number	\1	Mandatory (Must Match)	MY_NUM	1

Below the table, the entry configuration is shown:

- Description: MY_NUM
- Admin State: Enabled
- Match Type: Mandatory (Must Match)
- Input Field: Type: Called Address/Number, Value: (*)
- Output Field: Type: Called Address/Number, Value: \1

An "Apply" button is at the bottom right.



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**.

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.

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).



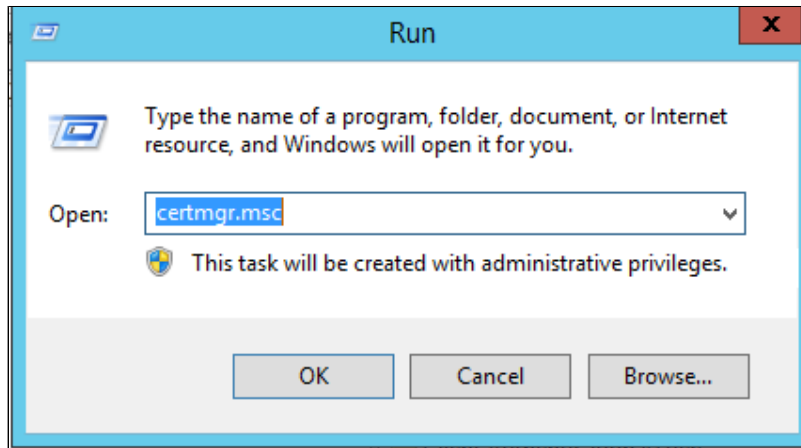
Info

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

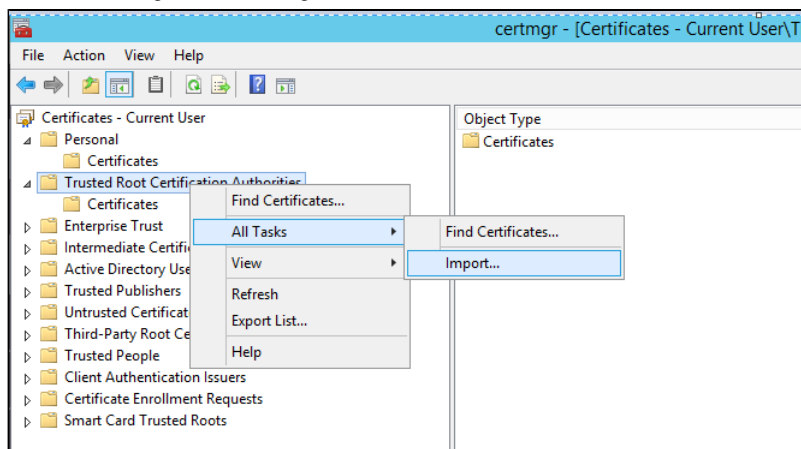
Trusted Certificate installation on the Application server

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**.



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

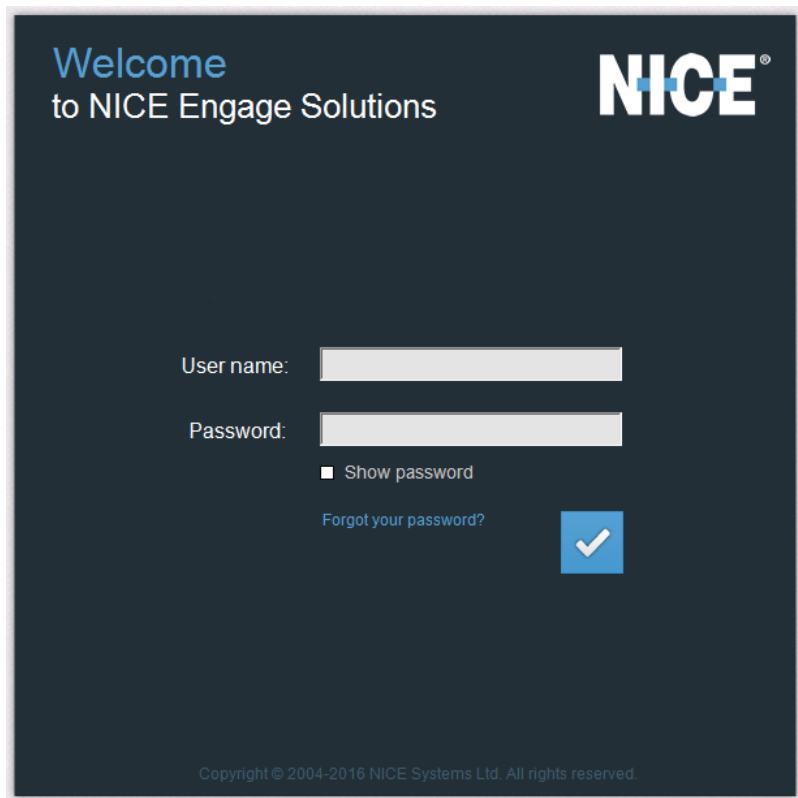


4. In the Import Certificate window, select the .cer certificate that you copied in [step 1](#) and import it to **Trusted Root Certification Authorities > Certificates**. Click **Finish**.
5. Repeat [step 4](#) for other .cer certificates that you copied earlier.
6. Repeat the above steps to import 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 Solutions

NICE®

User name:

Password:

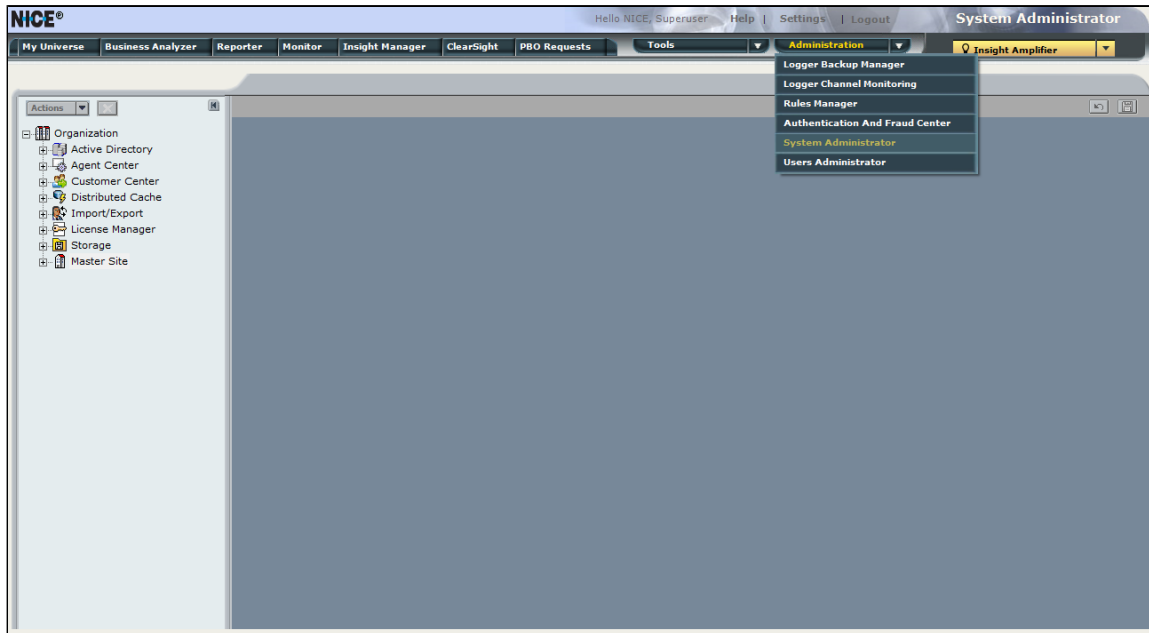
☐ Show password

[Forgot your password?](#)

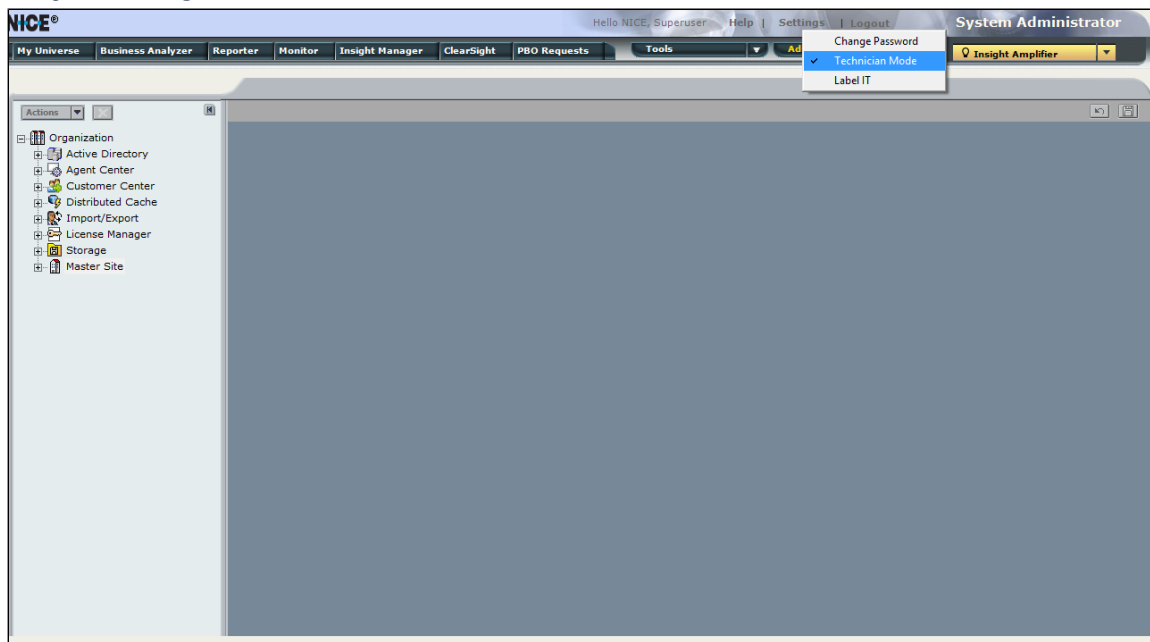
☒

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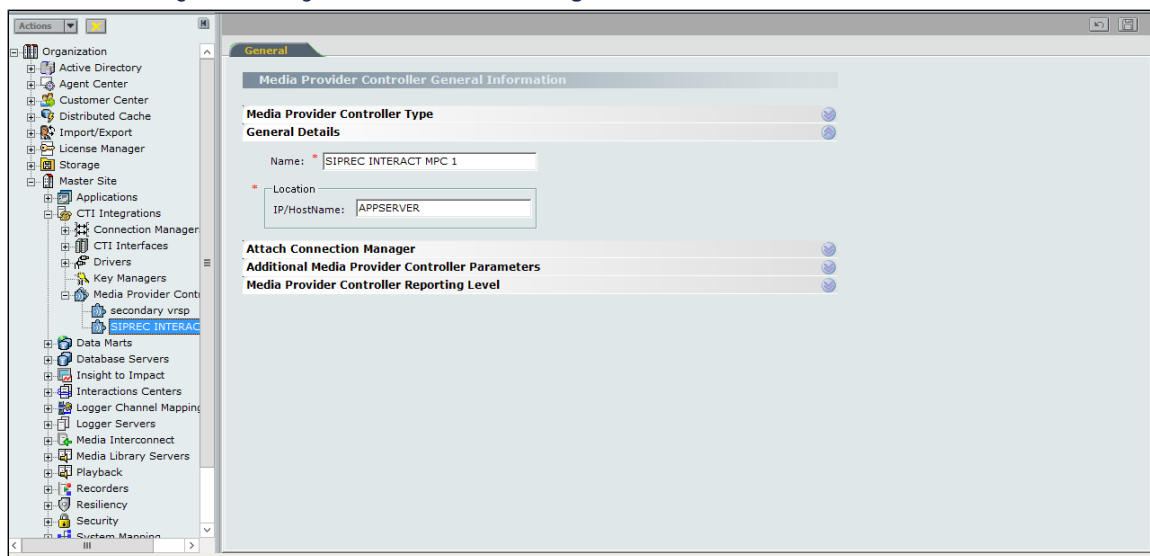
3. Navigate to **Administration > System Administrator**.



4. Navigate to **Settings** and select **Technician Mode**.



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



6. Expand the additional **Media Provider Controller** parameters.

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

The screenshot shows the 'Media Provider Controller General Information' window. The left sidebar contains a tree view with categories like Organization, Applications, and Data Marts. The main pane is titled 'General' and contains sections for 'Media Provider Controller Type', 'General Details', 'Attach Connection Manager', and 'Additional Media Provider Controller Parameters'. Under 'Additional Media Provider Controller Parameters', there is a table with two columns: 'Parameter Name' and 'Parameter Value'. The table contains the following entries:

Parameter Name	Parameter Value
MemoryPageSize	1024
SipStackTlsEnabled	Yes
SipStackTlsPort	5061
SipStackTlsIpAddress	
SipStackTlsCertificateSerialNumber	00 9d 70 8e 65 ca 22 82 8
SipStackTlsCertificateStoreLocation	LocalMachine
SipStackTlsCertificateRevocationMode	Offline
RedundancyIsEnabled	Yes

Below the table, there is a 'Description' field with the text 'Enable TLS connection.' and a 'Media Provider Controller Reporting Level' section.

8. Select **Sip Stack TlsCertificateSerialNumber** and then enter the serial number of the APPSERVER certificate. Refer to [How to get certificate serial number](#) to obtain the certificate serial number.

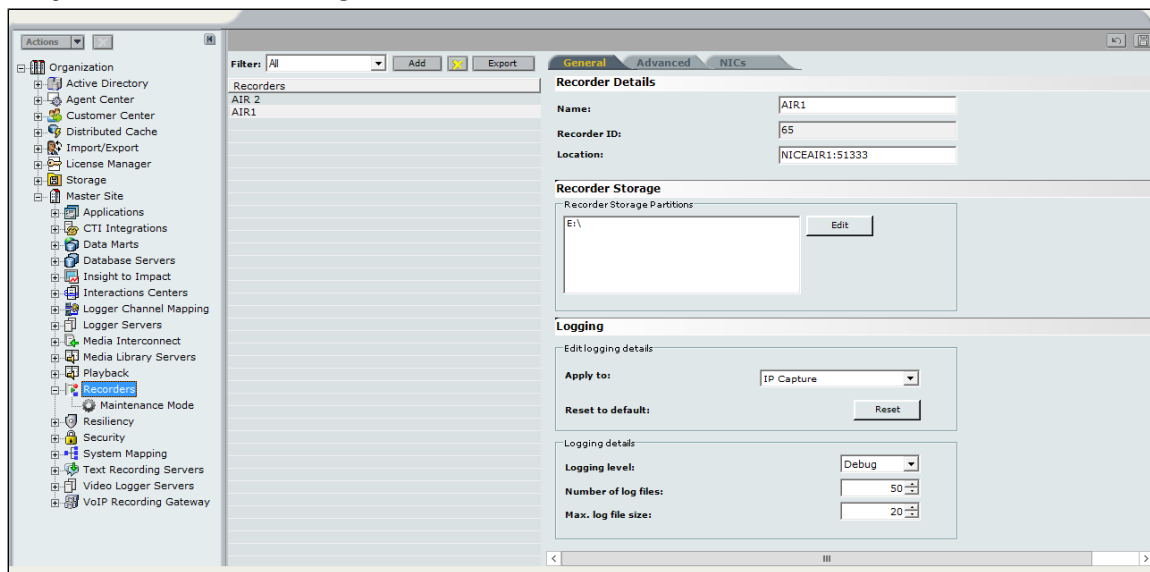
The screenshot shows the same 'Media Provider Controller General Information' window. In this view, the 'SipStackTlsCertificateSerialNumber' parameter is highlighted with a blue selection bar. The table now contains the following entries:

Parameter Name	Parameter Value
MemoryPageSize	1024
SipStackTlsEnabled	Yes
SipStackTlsPort	5061
SipStackTlsIpAddress	
SipStackTlsCertificateSerialNumber	00 9d 70 8e 65 ca 22 82 8
SipStackTlsCertificateStoreLocation	LocalMachine
SipStackTlsCertificateRevocationMode	Offline
RedundancyIsEnabled	Yes

The 'Description' field now shows 'Serial Number of the X509 (for Client/Server).' and the 'Media Provider Controller Reporting Level' section remains at the bottom.

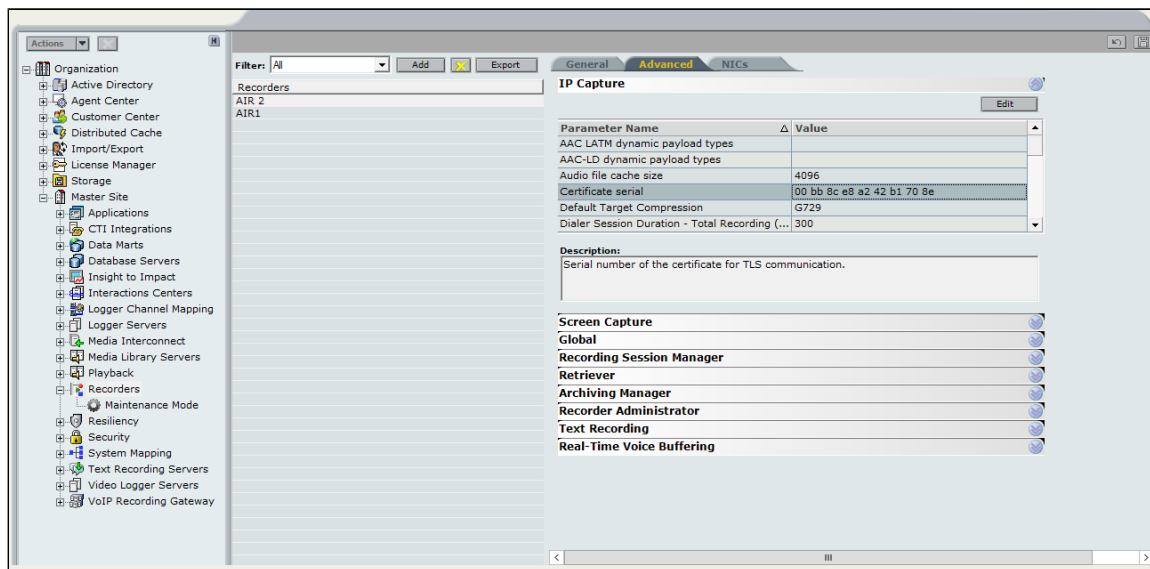
9. Click the **Save** icon displayed on the right-hand corner of the Application.

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

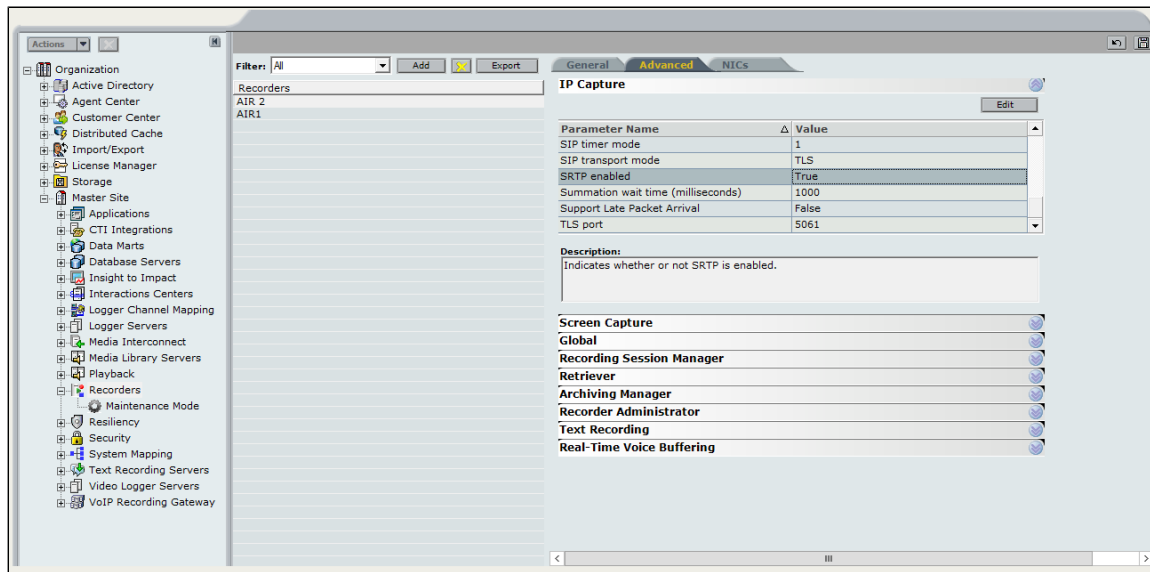


11. After selecting **AIR1**, click on the **Advanced** tab and expand **IP Capture**.

12. Select the **Certificate Serial** parameter, enter the serial number of the AIR1 certificate, and save your changes.



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



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.

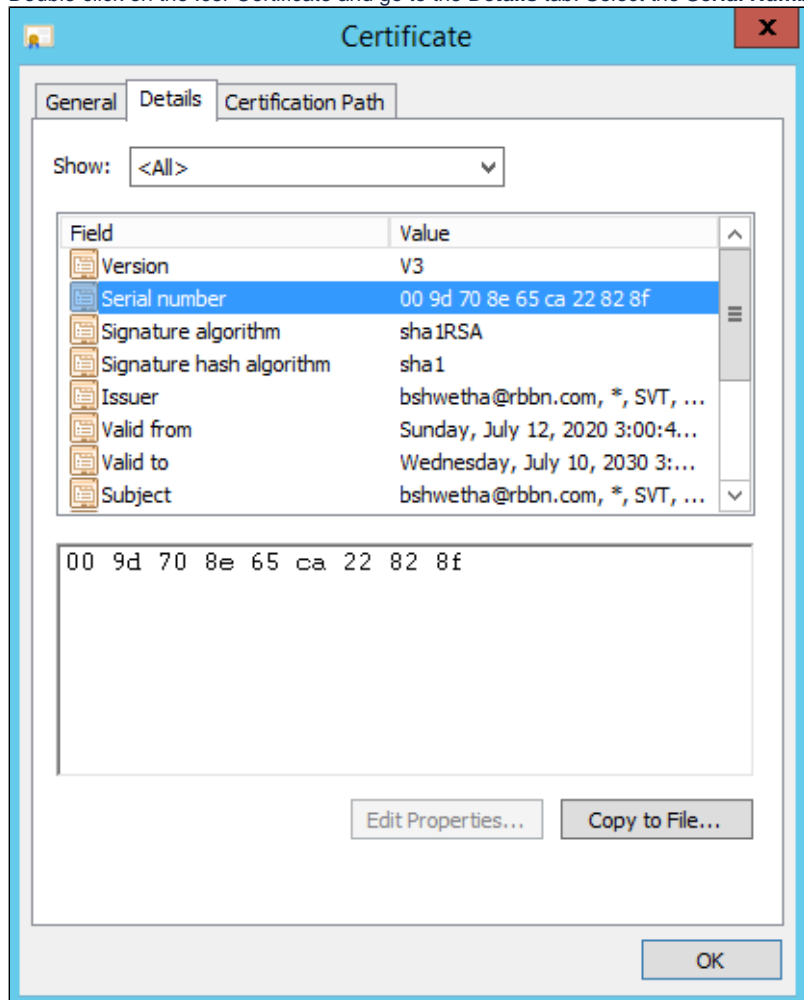
16. In the CTI Integrations branch, click **Apply**.

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.



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

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	✗
9	Call with Audio and Video	✗
10	DTMF - Inband and RFC2833	✓

11	Round Robin Option for SRS recorder	✓
12	Channel Hunting	✓
13	Metadata Validation	✓
14	Transcode Calls	✓
15	4xx/5xx Response Handling	✓
16	Call Hold and Resume	✓
17	Call Forward - Unconditional, Busy, and No Answer	✓
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	✗

Legend

Supported	✓
Not Supported	✗

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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