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# Ribbon SBC Core SWe R10.1 Interop with NICE Engage 6.15 : Interoperability Guide

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# Interoperable Vendors

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## Document Overview

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This document outlines the configuration best practices for the Ribbon SBC SWe Core & PSX when deployed with NICE Recording Server.

### About Ribbon SBC SWe Core :

The SBC SWe Core addresses the next-generation needs of SIP communications by delivering embedded media transcoding, robust security and advanced call routing in a high-performance, small form-factor device enabling service providers and enterprises to quickly and securely enhance their network by implementing services like SIP Trunking, secure Unified Communications and Voice over IP (VoIP).

The SBC SWe Core provides a reliable, scalable platform for IP interconnect to deliver security, session control, bandwidth management, advanced media services and integrated billing/reporting tools in an SBC appliance. This versatile series of SBCs can be deployed as peering SBCs, access SBCs or enterprise SBCs (eSBCs). The SBC product family is tested for interoperability and performance against a variety of third-party products and call flow configurations in the customer networks.

### About Ribbon PSX :

The Ribbon PSX provides centralized policy and call routing engine for both Ribbon distributed Call Processing Node (CPN) such as GSX/SBC and also third-party call processing nodes. When deployed in Service Provider network or Enterprises network, it interfaces with these call processing nodes while processing either TDM (SS7, PRA) or SIP calls.

### About NICE SIP Recorder :

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

## Scope/Non-Goals

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This document provides configuration best practices for deploying Ribbon's SBC SWe Core for NICE SIP recording 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 which best meets their requirements.

It is not the goal of this guide to provide detailed configurations that meet 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

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This is a technical document intended for telecommunications engineers with the purpose of configuring the Ribbon SBC SWe Core & PSX .

To perform this interop, you need to:

- use the graphical user interface (GUI) or command line interface (CLI) of the Ribbon product,
- understand the basic concepts of TCP/UDP/TLS and IP/Routing, and
- have understanding of SIP/RTP/RTCP to complete the configuration and for troubleshooting.

**Note**

This configuration guide is offered as a convenience to Ribbon customers. The specifications and information regarding the product in this guide are subject to change without notice. All statements, information, and recommendations in this guide are believed to be accurate but are presented without warranty of any kind, express or implied, and are provided AS IS. Users must take full responsibility for the application of the specifications and information in this guide.

## Prerequisites

The following aspects are required before proceeding with the interop:

- Ribbon SBC SWe Core
- Ribbon SBC SWe Core license
  - A valid license from Ribbon is required to enable functionality on Ribbon SBCs. Each SBC license provides a base set of capabilities to allow enabling and adding of additional features and capacity, as required.
- TLS certificates for SBC SWe Core
  - Please refer to [Managing Certificates](#)
- Ribbon PSX
- NICE Engage setup



NICE VRSP server functions as a SIP Proxy to set up SIP sessions between the SBC and the NICE.VRSP internally communicates to NICE AIR server which acts as a recording server. In active standby mode we have two VRSP servers with one as Active and one as Standby. Throughout this document from SBC perspective, we will be mentioning VRSP server as SRS[Session Recording Server].

## Product and Device Details

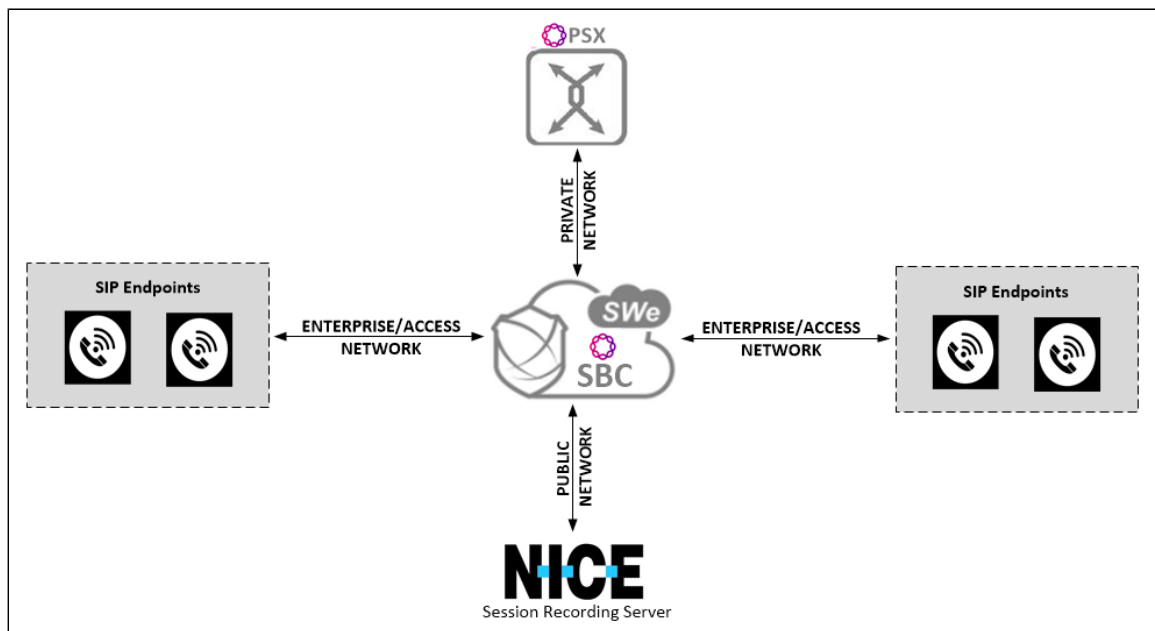
The configuration uses the following equipment and software:

Product	Equipment/Service	Software Version
Ribbon Communications	SBC SWe Core	V10.01.00-R000
	PSX	V14.1
Third-Party Equipment	NICE Recording Server	V6.15
Endpoints	PhonerLite	V2.96
	Zoiper5	V5.5.8
Administration and Debugging Tools	Wireshark	V3.0.1

## Network Topology and E2E Flow Diagrams

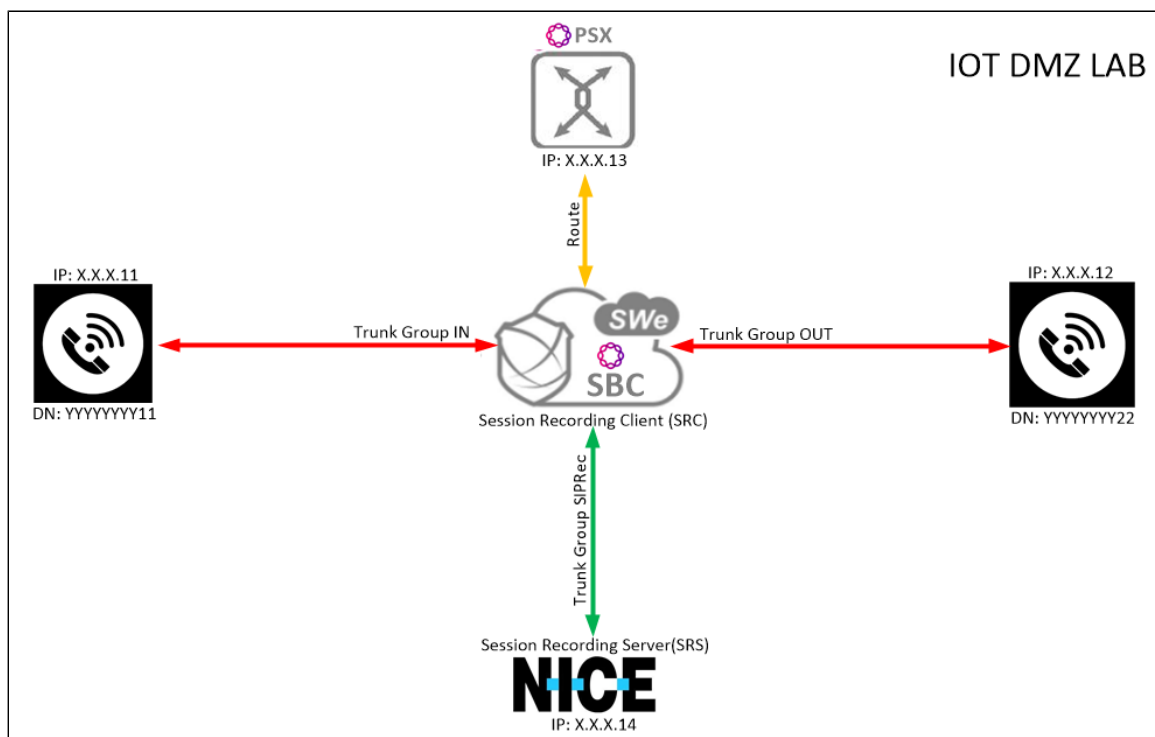
### Deployment Topology

Figure 1:



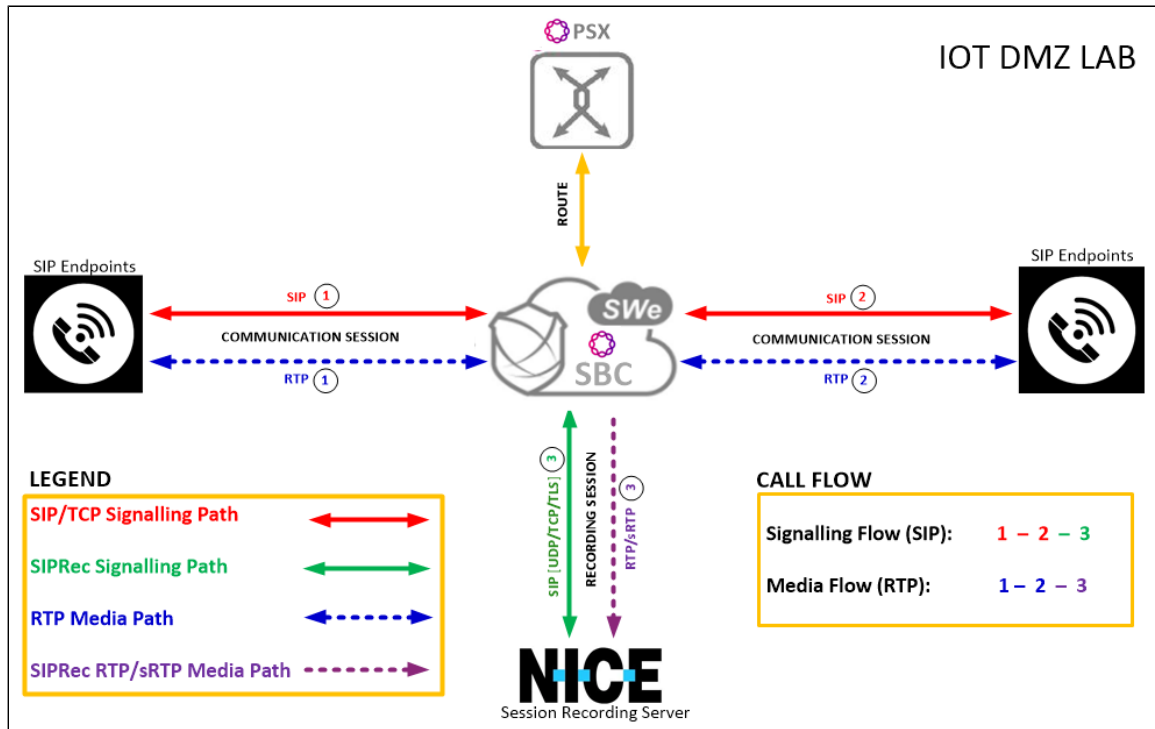
## Interoperability Test Lab Topology

Figure 2:



## Call Flow Diagram

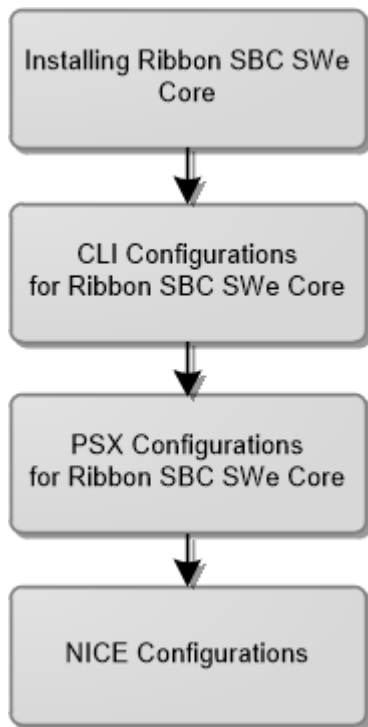
Figure 3:



## Document Workflow

The sections in this document follow the sequence below. The reader is advised to complete each section for the successful configuration.

Figure 4:



## Installing Ribbon SBC SWe Core

### Ribbon SBC Standalone

To deploy Ribbon SBC SWe Core StandAlone instance, refer to [SBC Core 10.1.x Documentation](#)

## Ribbon SBC High Availability

To deploy Ribbon SBC SWe Core in HA mode on different platforms, refer to [SBC Core Software Installation and Upgrade Guide](#)



During this interop, SBC SWe Core HA was installed on VMware platform by following the procedure described in [Installing SBC Application in High Availability Mode](#).



- After successful installation, ensure the time on both Active and Standby SBCs is in sync.
- NTP Sync verification:
  - Run the command 'timedatectl' to check if NTP is synchronized.
  - File /etc/ntp.conf should contain the IP of the NTP server that you have configured during installation

## CLI Configurations for RibbonSBC SWe Core

### Global Configuration

#### 1. Configure IP Interface Group

An IP Interface Group is a named object containing one or more IP interfaces (IP addresses). The IP Interface Group is Address Context-specific (e.g. permanently bound to a particular Address Context), and is the primary tool to manage disjointed networks (separate networks that are not designed to communicate directly). An IP Interface Group is the local manifestation of a segregated network domain. The service section of an IP trunk group and a Signaling Port typically reference an IP Interface Group in order to restrict signaling and/or media activity to that IP Interface Group.

```
set addressContext default ipInterfaceGroup IG1 ipInterface IP1 ceName SBCSIIPREC
set addressContext default ipInterfaceGroup IG1 ipInterface IP1 portName pkt0
set addressContext default ipInterfaceGroup IG1 ipInterface IP1 ipAddress <The primary IP address of the interface>
set addressContext default ipInterfaceGroup IG1 ipInterface IP1 prefix <The IP subnet prefix of this Interface>
set addressContext default ipInterfaceGroup IG1 ipInterface IP1 mode inService
set addressContext default ipInterfaceGroup IG1 ipInterface IP1 state enabled
set addressContext default ipInterfaceGroup IG2 ipInterface IP2 ceName SBCSIIPREC
set addressContext default ipInterfaceGroup IG2 ipInterface IP2 portName pkt1
set addressContext default ipInterfaceGroup IG2 ipInterface IP2 ipAddress <The primary IP address of the interface>
set addressContext default ipInterfaceGroup IG2 ipInterface IP2 prefix <The IP subnet prefix of this Interface>
set addressContext default ipInterfaceGroup IG2 ipInterface IP2 mode inService
set addressContext default ipInterfaceGroup IG2 ipInterface IP2 state enabled
commit
```

#### 2. Configure Static Route

IP Static Route object specifies the gateway to which you wish to direct traffic from your Packet, Management, or Link Interface. In effect, this object allows you to add, change, and delete gateways (next Hops) to these interfaces. Interface and static routes combine to form the IP routing table for your network.

An IP Static Route provides a route to each potential call destination IP address. The static route is used to add static IP routes for the IP interfaces. A static route indicates the next Hop gateway and IP interface to use for a particular peer network IP prefix.

```
set addressContext default staticRoute <destinationIpAddress> 0 <nextHopIpAddress> IG1 IP1 preference 100
set addressContext default staticRoute <destinationIpAddress> 0 <nextHopIpAddress> IG2 IP2 preference 100
commit
```

## SBC Configuration for Endpoints

#### 1. Create new Zone and configure sipSigPort

A Zone is used to group a set of objects unique to a particular customer environment.

A SIP Signaling Port is a logical address permanently bound to a specific zone, and is used to send and receive SIP call signaling packets. A SIP Signaling Port is capable of multiple transports such as UDP, TCP, and TLS/TCP.

```

set addressContext default zone zone1 id 111
set addressContext default zone zone1 sipSigPort 1 ipInterfaceGroupName IG1
set addressContext default zone zone1 sipSigPort 1 ipAddressV4 <IPv4 address>
set addressContext default zone zone1 sipSigPort 1 portNumber <1-65535>
set addressContext default zone zone1 sipSigPort 1 mode inService
set addressContext default zone zone1 sipSigPort 1 state enabled
set addressContext default zone zone1 sipSigPort 1 transportProtocolsAllowed sip-udp,sip-tcp,sip-tls-tcp
set addressContext default zone zone2 id 222
set addressContext default zone zone2 sipSigPort 2 ipInterfaceGroupName IG2
set addressContext default zone zone2 sipSigPort 2 ipAddressV4 <IPv4 address>
set addressContext default zone zone2 sipSigPort 2 portNumber <1-65535>
set addressContext default zone zone2 sipSigPort 2 mode inService
set addressContext default zone zone2 sipSigPort 2 state enabled
set addressContext default zone zone2 sipSigPort 2 transportProtocolsAllowed sip-udp,sip-tcp,sip-tls-tcp
commit

```

## 2. Create basic Trunk Group Configurations

SIP Trunk Groups are used to apply a wide-ranging set of call management functions to a group of peer devices (endpoints) within the network. SIP Trunk Groups are created within a specific address context and zone.

All SBC signaling and routing (both Trunking and Access) are based upon Trunk Group configurations defined within zones. A zone can contain multiple Trunk Groups.



Please ensure to configure similar transport preferences in CLI and PSX Trunk Group configurations

```

set addressContext default zone zone1 sipTrunkGroup SIPREC_TG1 signaling transportPreference preference1 tcp
set addressContext default zone zone1 sipTrunkGroup SIPREC_TG1 media mediaIpInterfaceGroupName IG1
set addressContext default zone zone1 sipTrunkGroup SIPREC_TG1 ingressIpPrefix <IP address> <prefix>
set addressContext default zone zone1 sipTrunkGroup SIPREC_TG1 state enabled
set addressContext default zone zone1 sipTrunkGroup SIPREC_TG1 mode inService
set addressContext default zone zone2 sipTrunkGroup SIPREC_TG2 signaling transportPreference preference1 tcp
set addressContext default zone zone2 sipTrunkGroup SIPREC_TG2 media mediaIpInterfaceGroupName IG2
set addressContext default zone zone2 sipTrunkGroup SIPREC_TG2 ingressIpPrefix <IP address> <prefix>
set addressContext default zone zone2 sipTrunkGroup SIPREC_TG2 state enabled
set addressContext default zone zone2 sipTrunkGroup SIPREC_TG2 mode inService
commit

```

## SBC Configurations for SIPRec

We must make a separate TG with separate zone and sipSigPort and attach that to egress IP interface group. This sip trunk is toward NICE recorder.

### 1. Create new Zone and Configure Sip Sigport for SIPRec Zone.

```

set addressContext default zone zone4 id 444
set addressContext default zone zone4 sipSigPort 4 ipInterfaceGroupName IG2
set addressContext default zone zone4 sipSigPort 4 ipAddressV4 <IPv4 address>
set addressContext default zone zone4 sipSigPort 4 portNumber <1-65535>
set addressContext default zone zone4 sipSigPort 4 transportProtocolsAllowed sip-udp,sip-tcp,sip-tls-tcp
set addressContext default zone zone4 sipSigPort 4 siprec enabled
set addressContext default zone zone4 sipSigPort 4 mode inService
set addressContext default zone zone4 sipSigPort 4 state enabled
commit

```

### 2. Configure Trunk group for SIPRec zone.



Please ensure to configure similar transport preferences in CLI and PSX Trunk Group configurations

Also, Transport preference mentioned in SRS Group profile should match transport preferences in Trunk Group towards SIPRec zone.



```

set addressContext default zone zone4 sipTrunkGroup SIPREC_TG4 media mediaIpInterfaceGroupName IG2
set addressContext default zone zone4 sipTrunkGroup SIPREC_TG4 ingressIpPrefix <IP address> <prefix>
set addressContext default zone zone4 sipTrunkGroup SIPREC_TG4 signaling transportPreference preferencel tls-tcp
set addressContext default zone zone4 sipTrunkGroup SIPREC_TG4 state enabled
set addressContext default zone zone4 sipTrunkGroup SIPREC_TG4 mode inService
commit

```

3. The Path Check Profile specifies the conditions that constitute a connectivity failure, and in the event of such a failure, the conditions that constitute a connectivity recovery. This profile specifies the configuration for OPTIONS PING.

```

set profiles services pathCheckProfile sip_recording1 protocol sipOptions
set profiles services pathCheckProfile sip_recording1 sendInterval 10
set profiles services pathCheckProfile sip_recording1 replyTimeoutCount 3
set profiles services pathCheckProfile sip_recording1 recoveryCount 1
set profiles services pathCheckProfile sip_recording1 failureResponseCodes [ all5xx ]
set profiles services pathCheckProfile sip_recording1 transportPreference preferencel tls-tcp

```

4. Configure the SRS IP as an ipPeer in the SIPREC zone (the zone containing the Trunk Group configured for the SRS) and attach the pathcheck profile to it.

```

set addressContext default zone zone4 ipPeer SIPREC_VRSP1 ipAddress <The IPv4 or IPv6 address of the Peer>
set addressContext default zone zone4 ipPeer SIPREC_VRSP1 ipPort <0-65535>
set addressContext default zone zone4 ipPeer SIPREC_VRSP1 pathCheck profile sip_recording1
set addressContext default zone zone4 ipPeer SIPREC_VRSP1 pathCheck state enabled
set addressContext default zone zone4 ipPeer SIPREC_VRSP2 ipAddress <The IPv4 or IPv6 address of the Peer>
set addressContext default zone zone4 ipPeer SIPREC_VRSP2 ipPort <0-65535>
set addressContext default zone zone4 ipPeer SIPREC_VRSP2 pathCheck profile sip_recording1
set addressContext default zone zone4 ipPeer SIPREC_VRSP2 pathCheck state enabled
commit

```

5. NICE does not support SIP INFO method towards SIPRec . So, disable SIP INFO method towards SIPRec Trunk Group.

```

set addressContext default zone zone4 sipTrunkGroup SIPREC_TG4 signaling methods info reject
commit

```

5. Create sipRecMetadataProfile with version 1 as per RFC 7865 and associate the profile to SIPRec Trunk Group.



When sipRecMetadataProfile is not configured, by default SBC supports backward compatibility and pre-defined metadata for passing proprietary call specific information from the SRC to the SRS.

Refer to [MetadataSupport](#) for additional NICE configurations.

```

set profiles services sipRecMetadataProfile t1 state enabled
set profiles services sipRecMetadataProfile t1 version 1
comm
set addressContext default zone zone4 sipTrunkGroup SIPREC_TG4 services sipRecMetadataProfile t1
comm

```

## TLS Certificates

The Public Key Infrastructure (PKI) provides a common set of infrastructure features supporting public key and certificate-based authentication based on the RSA public/private key pairs and X.509 digital certificates.

Import all the required certificated to SBC under /opt/sonus/external and execute the following commands.

```

#### SRS1 Application Server Certificate Import ####
set system security pki certificate NICE_REMOTE1 state enabled
set system security pki certificate NICE_REMOTE1 fileName <SRS1 Certificate filename imported in SBC>
set system security pki certificate NICE_REMOTE1 type remote
comm

#### SRS2 Interaction Server Certificate Import ####
set system security pki certificate NICE_REMOTE2 state enabled
set system security pki certificate NICE_REMOTE2 fileName <SRS2 Certificate filename imported in SBC>
set system security pki certificate NICE_REMOTE2 type remote
comm

#### SBC Certificate Import ####
set system security pki certificate SBC_LOCAL state enabled
set system security pki certificate SBC_LOCAL fileName <SBC local Certificate filename imported in SBC>
set system security pki certificate SBC_LOCAL passphrase xxxx
set system security pki certificate SBC_LOCAL type local
comm

```

## TLS Profile

This object creates and configures a profile for implementing the Transport Layer Security (TLS) protocol to use with SIP over TLS. TLS is an IETF protocol for securing communications across an untrusted network. Normally, SIP packets travel in plain text over TCP or UDP connections. Secure SIP is a security measure that uses TLS, the successor to the Secure Sockets Layer (SSL) protocol.

To add a TLS protection-level policy, create a TLS PROFILE and configure each of the parameters.

```

#### TLS Profile for SIP Endpoint ####
set profiles security tlsProfile TLS_SIPREC1 appAuthTimer 5
set profiles security tlsProfile TLS_SIPREC1 handshakeTimer 5
set profiles security tlsProfile TLS_SIPREC1 sessionResumpTimer 3600
set profiles security tlsProfile TLS_SIPREC1 cipherSuite1 rsa-with-aes-128-cbc-sha
set profiles security tlsProfile TLS_SIPREC1 cipherSuite2 rsa-with-aes-256-cbc-sha
set profiles security tlsProfile TLS_SIPREC1 cipherSuite3 tls_rsa_with_aes_256_gcm_sha384
set profiles security tlsProfile TLS_SIPREC1 allowedRoles clientandserver
set profiles security tlsProfile TLS_SIPREC1 authClient true
set profiles security tlsProfile TLS_SIPREC1 clientCertName SBC_LOCAL
set profiles security tlsProfile TLS_SIPREC1 serverCertName SBC_LOCAL
set profiles security tlsProfile TLS_SIPREC1 acceptableCertValidationErrors none
set profiles security tlsProfile TLS_SIPREC1 v1_0 enabled
set profiles security tlsProfile TLS_SIPREC1 v1_1 enabled
set profiles security tlsProfile TLS_SIPREC1 v1_2 enabled
set profiles security tlsProfile TLS_SIPREC1 suppressEmptyFragments disabled
set profiles security tlsProfile TLS_SIPREC1 peerNameVerify disabled
commit

#### TLS Profile for NICE SIP Recording Trunk ####
set profiles security tlsProfile testsiprectlsroot appAuthTimer 5
set profiles security tlsProfile testsiprectlsroot handshakeTimer 5
set profiles security tlsProfile testsiprectlsroot sessionResumpTimer 3600
set profiles security tlsProfile testsiprectlsroot cipherSuite1 rsa-with-aes-128-cbc-sha
set profiles security tlsProfile testsiprectlsroot cipherSuite2 rsa-with-aes-256-cbc-sha
set profiles security tlsProfile testsiprectlsroot cipherSuite3 tls_rsa_with_aes_256_gcm_sha384
set profiles security tlsProfile testsiprectlsroot allowedRoles clientandserver
set profiles security tlsProfile testsiprectlsroot authClient true
set profiles security tlsProfile testsiprectlsroot clientCertName SBC_LOCAL
set profiles security tlsProfile testsiprectlsroot serverCertName SBC_LOCAL
set profiles security tlsProfile testsiprectlsroot acceptableCertValidationErrors none
set profiles security tlsProfile testsiprectlsroot v1_0 enabled
set profiles security tlsProfile testsiprectlsroot v1_1 enabled
set profiles security tlsProfile testsiprectlsroot v1_2 enabled
set profiles security tlsProfile testsiprectlsroot suppressEmptyFragments disabled
set profiles security tlsProfile testsiprectlsroot peerNameVerify disabled
commit

```

The TLS profile is specified on the SIP Signaling Port and controls behavior of all TLS connections established on that signaling port.

```
##### Attach TLS profile to SIPrec zone #####
set addressContext default zone zone4 sipSigPort 4 tlsProfileName testsiprectlsroot
comm

##### Attach TLS profile to SIPrec zone (If TLS transport is enabled)#####
set addressContext default zone zone1 sipSigPort 1 tlsProfileName TLS_SIPREC1
set addressContext default zone zone2 sipSigPort 2 tlsProfileName TLS_SIPREC1
comm
```

## SBC Configuration to enable PSX

We need to disable local PolicyServer and configure remote PSX details in SBC SWe Core.

```
set system policyServer localServer PSX_LOCAL_SERVER state disabled
set system policyServer localServer PSX_LOCAL_SERVER mode outOfService
set system policyServer remoteServer IOTPSX ipAddress 172.16.100.216
set system policyServer remoteServer IOTPSX state enabled
set system policyServer remoteServer IOTPSX mode active
set system policyServer remoteServer IOTPSX action force
commit
```

## PSX Configurations for RibbonSBC SWe Core

### Configuring Class of Service

Please note that we have used default Class Of Service 'DEFAULT\_IP' for our testing.

Figure 5:

Class Of Service: **DEFAULT\_IP**

Description:

<p><b>Service Flags</b></p> <ul style="list-style-type: none"> <li><input type="checkbox"/> Authcode</li> <li><input type="checkbox"/> Blocking</li> <li><input type="checkbox"/> Business Group Blocking</li> <li><input type="checkbox"/> Business Group Origination Blocking</li> <li><input type="checkbox"/> Calling Forced Routing</li> <li><input type="checkbox"/> Destination Forced Routing</li> <li><input type="checkbox"/> DTMF</li> <li><input type="checkbox"/> Hifraud Countries</li> <li><input type="checkbox"/> Infodigit Screening</li> <li><input type="checkbox"/> Ingress CPC Screening</li> <li><input type="checkbox"/> Message Waiting Indicator</li> <li><input type="checkbox"/> Message Waiting Indicator Update</li> <li><input type="checkbox"/> SAC/Non-SAC Routing</li> <li><input type="checkbox"/> Services Standard Routing</li> <li><input type="checkbox"/> Short Key Translation</li> </ul>	<p><b>Non-Subscriber Call Routing</b></p> <ul style="list-style-type: none"> <li>0+</li> <li>0+ IDDD</li> <li>0-</li> <li>00</li> <li>1+</li> <li>Carrier Cut Through</li> <li>IDDD</li> <li>Private</li> <li>Switch ID Trunk Group ID</li> <li>User Name</li> </ul>	<p><b>Casual Calling Routing</b></p> <ul style="list-style-type: none"> <li>0+</li> <li>0+ IDDD</li> <li>0-</li> <li>00</li> <li>1+</li> <li>Carrier Cut Through</li> <li>IDDD</li> </ul>
--	--	---

Figure6:

Services

Authcode Script: <None> [Runtime Variables](#)

International Number Blocking: <None> [Runtime Variables](#)

Screening: <None> [Runtime Variables](#)

DTMF Profile: <None>

Short Key Profile: <None>

Message Waiting Indicator

Script: <None>

Service Number:

Callino Forced Routes	Infodigit Screening	Ingress CPC Screening	Non-SAC Routing	SAC Routing	Services Standard Routes
Blocking	Business Group Blocking	Business Group Origination Blocking	Destination Forced Routes		Hifraud Country Blocking
Blocking Profile	Script	Sequence	Time Range		

Service Exception Profile: <None>

New Open Delete [Runtime Variables](#)

## Configuring Gateway

1. Configure a gateway with the SBC name and the management IP address.

Figure 7:

PSX Manager V14.01.00R000  
User: - North America

Menu

<Configure>

<Admin>

Gateway

Gateway

SQL Search Criteria (8 entries)

Gateway: \*

Search More...

Gateway

DEFAULT

IOTCHANDANCE

NATSW

SBCPOOJA

SBCSYAM1

STISBC

TESTGW

ZOOM

2. From the Gateway configuration UI, enter the name of gateway that is configured in the SBC.



Gateway name should be same as systemname in SBC conf file and should be capitalized.

**Figure8:**

GATEWAY: SBCSYAM1 LRNS

Switch: SOFTSWITCH

Gateway Group: DEFAULT

Cluster Profile: <None>

Default Trunk Group: SIP

Charge Band Profile: <None>

Traffic Control Escape Profile: <None>

Mobile Switch ID: 1 ☒ None

Signaling Gateway Group: <None>

Enum Authority Profile: <None>

Address Reachability Service Profile: <None>

SMM Profile Group: <None>

Peer Throttling Profile: <None>

P-Origination-ID:  ☐ Autogenerate

Context Info

Flags

☐ CAMEL Services Supported ☐ Route CAMEL Subscription Calls

☐ CDP Gateway ☒ Traffic Management

☐ MTRR Supported ☐ Logical SBC

Display

☐ Allow Mixed Characters in Gateway Name

H.323 Control

☐ Prune Routes

Configure SBC management IP in IPv4 Address and default port number 2569.

**Figure 9:**

IPv4 Address: 8 . 8 . 8 . 8 Port Number: 2569

IPv6 Address: 0 : 0 : 0 : 0 : 0 : 0 : 0 : 0

☒ Prefer IPv4 ☐ Prefer IPv6

☐ H.323 IP Address: 0 . 0 . 0 . 0 H.323 Port Number: 1720

☐ H323 IPv6 Address: 0 : 0 : 0 : 0 : 0 : 0 : 0 : 0

☐ Set As Default H.323 Gateway For This IP Address

☒ Prefer IPv4 ☐ Prefer IPv6

☐ SIP IP Address: 0 . 0 . 0 . 0 SIP Port Number: 5060

☐ SIP IPv6 Address: 0 : 0 : 0 : 0 : 0 : 0 : 0 : 0

☐ Set As Default SIP Gateway For This IP Address

☒ Prefer IPv4 ☐ Prefer IPv6

☐ Server FQDN 0 FQDN Port Number: 0

☒ Prefer IPv4 ☐ Prefer IPv6

☐ Perform DNS query for SIP Server Selection

Local Routes Filtering/Prioritization

☐ Apply At Ingress Gateway ☐ Apply At Ingress Cluster

Services

☒ Not Screened ☐ Screened - Normal ☐ Screened - Fraud

Class Of Service: DEFAULT\_IP

Service Exception Profile: <None>

## Configuring Globalization Profile

This object is used to define numbers that are to be globalized for egress SIP message headers. Specify a profile entry for each number type that needs to be globalized. The profile includes a digit type, a source for the country code, and a flag to enable the globalization. Assign Globalize Profiles to egress trunk groups by selecting them on the IP Signaling Profile for each trunk group.

**Figure10:**

**Globalize Profile:** SIPREC\_Profile

Description: Doing globalization plus populating country code

Globalize Profile Data

Number Type: Calling Number

☐ Use Digit Type For URI

Digit Type

<All>

900 Premium Toll

950 Carrier Access

Carrier Access

Directory Assistance

Easily Identifiable Number

Emergency

Government Emergency

Country Code Source: Origination

Country Code Fallback: <None>

☒ Globalize Flag

Add/Modify

Number Type	Country Code	Globalize Flag	Country Code Fallback	...
Calling Number	Origination	Enabled	<None>	...
Called Number	Origination	Enabled	<None>	...
LNP Routing Number	Destination	Enabled	<None>	...
Redirection	Destination	Enabled	<None>	...
Original Called Number	Origination	Enabled	<None>	...
Location	Origination	Enabled	<None>	...
Redirecting	Origination	Enabled	<None>	...
Billing Number	Origination	Enabled	<None>	...
GN: Dialed Digits	Destination	Enabled	<None>	...
GN: Destination	Destination	Enabled	<None>	...

Delete

Figure11:

Number Type	Country Code	Globalize Flag	Country Code Fallback	...
GN: User Calling, Not Screened	Origination	Enabled	<None>	...
GN: Redirecting Terminating	Destination	Enabled	<None>	...
GN: Ported Dialed	Destination	Enabled	<None>	...
GN: Called CES Id	Destination	Enabled	<None>	...
GN: Additional Called	Destination	Enabled	<None>	...
GN: Additional Connected	Destination	Enabled	<None>	...
GN: Additional Calling	Origination	Enabled	<None>	...
GN: Additional Original Called	Destination	Enabled	<None>	...
GN: Additional Redirecting	Origination	Enabled	<None>	...
GN: Additional Redirection	Destination	Enabled	<None>	...
Contractor Number	Origination	Enabled	<None>	...

Figure12:

Number Type	Country Code	Globalize Flag	Country Code Fallback	...
GN: Additional Calling	Origination	Enabled	<None>	...
GN: Additional Original Called	Destination	Enabled	<None>	...
GN: Additional Redirecting	Origination	Enabled	<None>	...
GN: Additional Redirection	Destination	Enabled	<None>	...
Contractor Number	Origination	Enabled	<None>	...
GN: Network Provided Number	Destination	Enabled	<None>	...
Dialed Number	Origination	Enabled	<None>	...
GN: Third Party Number	Destination	Enabled	<None>	...
GN: Collect Call Number	Destination	Enabled	<None>	...
GN: Local ANI	Origination	Enabled	<None>	...
To URI User	Origination	Enabled	<None>	...

## Configuring IP Signaling Profile

This object specifies parameters associated with H.323, SIP, SIP-I communication that are sent as part of the outgoing signaling message after standard protocol rules have been applied.

You can associate IP signaling profiles with IP trunk groups and virtual trunk groups.

Figure13:

IP SIGNALING PROFILE: SIPREC\_IPSP\_TCP

Common IP Attributes - Communicating With The Peer Regardless Of Call Direction

<input type="checkbox"/> Accept Alert Info	<input type="checkbox"/> No Content Disposition
<input type="checkbox"/> Add P-Charging Function Addr	<input type="checkbox"/> No Port Number 5060
<input type="checkbox"/> Add Path/Service Route Per TG	<input type="checkbox"/> No Userinfo In Contact Header
<input type="checkbox"/> Audio Codec Change through Empty TCS	<input type="checkbox"/> Only Selected Codec In Session Refresh
<input type="checkbox"/> Call Hold Interworking	<input type="checkbox"/> Override Relay For Non SIP Egress Leg
<input type="checkbox"/> Calling Party Type Number If Present	<input type="checkbox"/> P-Called-Party-Id-Support
<input type="checkbox"/> Clearmode For Data Calls	<input type="checkbox"/> P-ChgMsg-Info
<input type="checkbox"/> Create P-Charging-Vector	<input type="checkbox"/> Relay Data Path Mode Changes To The Other Leg
<input type="checkbox"/> Create P-Visited-Network Id	<input type="checkbox"/> Reject REFER
<input type="checkbox"/> Create Path Header	<input type="checkbox"/> Replace Host On Via Header
<input type="checkbox"/> Create Service-Route Header	<input type="checkbox"/> Reject REFER With IP
<input type="checkbox"/> Customized Session Timer Behavior	<input type="checkbox"/> Reject REFER With TN
<input type="checkbox"/> Disable Also Header	<input type="checkbox"/> ReQuery PSX on REGISTER Refresh
<input type="checkbox"/> Disable Constrained Capacities	<input type="checkbox"/> Restrict History Info Header
<input type="checkbox"/> Disable Host Translation	<input type="checkbox"/> Route Using Received FQDN
<input type="checkbox"/> Disable Media Lock Down	<input type="checkbox"/> SDP O-line Only Compares
<input type="checkbox"/> Disable Refer-To URI Parameters	<input type="checkbox"/> Send All Allowed Codecs For Late Media Invite Or Re-Invite
<input type="checkbox"/> Discard Received Reason Header	<input type="checkbox"/> Send Direct Media Info In SDP Attribute
<input type="checkbox"/> Do Not Include SS Attribute In Re-INVITE	<input type="checkbox"/> Send Empty TCS

Figure14:

<input type="checkbox"/> Don't Send REFER With IP	<input type="checkbox"/> Send Only Preferred Codec
<input type="checkbox"/> Don't Send REFER With TN	<input type="checkbox"/> Send PTIME In SDP
<input type="checkbox"/> End To End BYE	<input type="checkbox"/> Send RTCP Port In SDP
<input type="checkbox"/> End To End RE-INVITE	<input type="checkbox"/> Session Timer Refresh Update
<input type="checkbox"/> End To End UPDATE	<input type="checkbox"/> Set Accept Header To Application SDP Only
<input type="checkbox"/> Suppress End To End Session Refresh	<input type="checkbox"/> Set Oline Dash
<input type="checkbox"/> End To End PRACK	<input type="checkbox"/> Set Session Version Zero
<input type="checkbox"/> Enable Default PUI Procedures	<input type="checkbox"/> Set Sline Dash
<input type="checkbox"/> Enable Dial String Handling	<input type="checkbox"/> Store P-Charging Function Addr
<input type="checkbox"/> Include G729 with G729A when offer PSP has G729A	<input checked="" type="checkbox"/> Store P-Charging Vector
<input type="checkbox"/> Include IP Ports In FROM And TO Headers	<input type="checkbox"/> Store Path Header
<input type="checkbox"/> Include Reason Header (Q.850)	<input type="checkbox"/> Store Service-Route Header
<input type="checkbox"/> Include SS Attribute In Initial Invite	<input type="checkbox"/> Suppress Min-SE if not received
<input checked="" type="checkbox"/> Include Transport Type In Contact Header	<input type="checkbox"/> Terminal Portability Interworking
<input type="checkbox"/> Insert Peer Address As Top Route Header	<input type="checkbox"/> Send RTCP BandWidth Info
<input type="checkbox"/> Lockdown Preferred Codec	<input type="checkbox"/> Validate Access Nw Info Header
<input type="checkbox"/> Map Cause Location	<input type="checkbox"/> Use Psx Route for Registered Invite
<input type="checkbox"/> Map SGD In P-Sig-Info Header	<input type="checkbox"/> From Header Anonymisation
<input type="checkbox"/> Map Suspend/Resume Event In P-Svc-Info Header	<input type="checkbox"/> Create ISUP Message Body

Figure15:

<input type="checkbox"/> Map UUI In P-Sig-Info Header	<input type="checkbox"/> Disable Transparently Passing ISUP Message Body
<input type="checkbox"/> MIME Cause Precede Reason Header Cause	<input type="checkbox"/> aiToPcmInterworking
<input type="checkbox"/> Minimize Relaying Of Media Changes From Other Call Leg	<input type="checkbox"/> Send SBC Supported Codecs For Late Media Re-Invite
<input type="checkbox"/> No Service Route Hdr For Emergency Registration	<input type="checkbox"/> Select Core Stream For Multi Stream Audio Or Image Call
<input type="checkbox"/> Publish IP In Hold SDP	<input type="checkbox"/> Disable Non Core Audio And Image Streams
<input type="checkbox"/> Insert PAccess Network Info	<input type="checkbox"/> Map DPM to Send and Receive for Initial Dialog
<input type="checkbox"/> Contact Transparency For Isfocus Media Tag	<input type="checkbox"/> Suppress Refer Relay From Other Leg
<input type="checkbox"/> Support S-CSCF Restoration Procedures	<input type="checkbox"/> Support Call Info With SIP Cause 608 RFC 8688
<input type="checkbox"/> Insert UE Flow Info	
<input type="checkbox"/> Include SIP Reason Header	
<b>Call Preservation Flags</b> <input type="checkbox"/> Call Preservation Call Preservation Time Out: <input type="text" value="5"/>	
<b>Call Transfer Flags</b> Handle IP Addresses Not Present In Network Selector Table (NST): <input type="text" value="Route Via Transferring IPTG"/>	
<input type="checkbox"/> Force Re-Route Via PSX Query <input type="checkbox"/> Skip Re-Route Via PSX Query	
<b>Local Media Control Flags</b> <input type="checkbox"/> Enable HOLD on REFER	

**Figure16:**

<b>Option Tag In Require Header</b> <input type="checkbox"/> Suppress Replace Tag																	
<b>Option Tag In Supported Header</b> <input type="checkbox"/> Suppress Replace Tag																	
<b>PreConditions Profile</b> <input type="checkbox"/> State <input type="checkbox"/> Support If Egress IPTG <input type="checkbox"/> Strength Mandatory Policy <input type="checkbox"/> Strength Optional Policy <input type="checkbox"/> UPDATE Preconditions Policy Strength Mandatory Priority: <input type="text" value="1"/> Strength Optional Priority: <input type="text" value="1"/> UPDATE Preconditions Priority: <input type="text" value="1"/>																	
<b>Relay Flags</b> <table border="0"> <tr> <td><input type="checkbox"/> Conference Event Package</td> <td><input type="checkbox"/> PUBLISH</td> </tr> <tr> <td><input type="checkbox"/> Dialog Event Package</td> <td><input type="checkbox"/> REFER</td> </tr> <tr> <td><input type="checkbox"/> DTMF Body</td> <td><input type="checkbox"/> Reg Event Package</td> </tr> <tr> <td><input type="checkbox"/> Force 503 To 500 Relay</td> <td><input type="checkbox"/> Ribbon Media Body</td> </tr> <tr> <td><input type="checkbox"/> Info</td> <td><input type="checkbox"/> Status Code 3XX</td> </tr> <tr> <td><input type="checkbox"/> Message</td> <td><input type="checkbox"/> Status Code 4XX-6XX</td> </tr> <tr> <td><input type="checkbox"/> Notify</td> <td><input type="checkbox"/> Third Party Bodies</td> </tr> <tr> <td><input type="checkbox"/> Options</td> <td><input type="checkbox"/> Update without SDP</td> </tr> </table>		<input type="checkbox"/> Conference Event Package	<input type="checkbox"/> PUBLISH	<input type="checkbox"/> Dialog Event Package	<input type="checkbox"/> REFER	<input type="checkbox"/> DTMF Body	<input type="checkbox"/> Reg Event Package	<input type="checkbox"/> Force 503 To 500 Relay	<input type="checkbox"/> Ribbon Media Body	<input type="checkbox"/> Info	<input type="checkbox"/> Status Code 3XX	<input type="checkbox"/> Message	<input type="checkbox"/> Status Code 4XX-6XX	<input type="checkbox"/> Notify	<input type="checkbox"/> Third Party Bodies	<input type="checkbox"/> Options	<input type="checkbox"/> Update without SDP
<input type="checkbox"/> Conference Event Package	<input type="checkbox"/> PUBLISH																
<input type="checkbox"/> Dialog Event Package	<input type="checkbox"/> REFER																
<input type="checkbox"/> DTMF Body	<input type="checkbox"/> Reg Event Package																
<input type="checkbox"/> Force 503 To 500 Relay	<input type="checkbox"/> Ribbon Media Body																
<input type="checkbox"/> Info	<input type="checkbox"/> Status Code 3XX																
<input type="checkbox"/> Message	<input type="checkbox"/> Status Code 4XX-6XX																
<input type="checkbox"/> Notify	<input type="checkbox"/> Third Party Bodies																
<input type="checkbox"/> Options	<input type="checkbox"/> Update without SDP																

**Figure17:**



☐ Reason Phrase 4XX 6XX

Refer To Header Relay
 

☒ Reject the REFER request if no match is found
 ☐ relay the REFER request if no match is found
 ☐ relay the REFER request without matching

Transparency Flags
 

☐ Accept-Contact Header
 ☐ Reason Header

☐ Accept-Language Header
 ☐ Referred-By Header

☐ Accept Header
 ☐ Resource Priority Option Tag

☐ Alert Information Header
 ☐ Request-URI

☐ Allow Header
 ☐ Resource-Lists Body

☐ Authcode Headers
 ☐ RLMI Body

☐ Call-Info Header
 ☐ Route Header

☐ Contact Header\*
 ☐ Server Header

☐ Error Info
 ☐ Service-Route Header

☐ Event Header
 ☐ Simple-Filter Body

☐ External Body
 ☐ SIP Body

☐ From Header
 ☐ SIPFRAG Body

☐ Geo Location Error
 ☐ Target-Dialog Header

☐ Geo Location Header
 ☐ To Header

☐ Geo Location Route
 ☐ Tone Body

☐ History Info
 ☐ Unknown Body

Figure18:

☐ Image Body
 ☐ Unknown Header

☐ Max\_forwards Header
 ☐ User-Agent Header

☐ MWI Body
 ☐ User-To-User Header

☐ Pass Complete Contact Header
 ☐ Via Header

☐ P-Access-Network-Info Header
 ☐ Warning Header

☐ P-Called-Party-Id
 ☐ Watcherinfo Body

☐ P-Charging-Vector Header
 ☐ X-ATP

☐ P-Early-Media

☐ P-Visited-Network ID Header

☐ Path Header

☐ Pidf Body

☐ Pidf-Diff Body

☐ QSIG Body

Transparency Profile
 SBC Transparency Profile:

Flags
 

☐ Apply Setting to SBC TG
 ☐ Apply Setting to "Use SIP In Core" Egress TG if Applicable

PDCS-Billing Info Header
 ☐ Transparency

From the drop down, select Globalization Profile created above.

Figure19:

<input type="checkbox"/> Include Privacy	
Sip In Core <input type="checkbox"/> Use SIP In Core	
Header Encryption Flags <input type="checkbox"/> Path Header <input type="checkbox"/> Service Route Header	
Subscription Package Support <input type="checkbox"/> Support Reg Event <input type="checkbox"/> Use PSX Route For SBC Initiated Subscribe	
Registrar Recovery <input type="checkbox"/> Register to Alternate on Primary Down <input type="checkbox"/> Override Internal Expires Timer <input type="checkbox"/> Revert to Primary On Recovery <input type="checkbox"/> Deregister Alternate on Primary Recovery	
Egress IP Attributes - Sending A Call In The Forward Direction To The Peer	
IP Protocol Type: <input checked="" type="radio"/> SIP Only <input type="radio"/> SIP-I <input type="radio"/> H.323 <input type="radio"/> Wireless	
IP Signaling MIME Content Type:	ISUP
IP Signaling Treatment:	Interwork
MIME Content Type Version:	0
Globalize Number Profile:	SIPREC_Profile
Localize Profile:	<None>
Phone-Context Parameter Length:	0
Media Qos Kpi Profile:	<None>
Signaling Qos Kpi Profile:	<None>

Figure20:

Flags	
<input type="checkbox"/> Accept 3XX With RN	<input type="checkbox"/> Qos Based Routing
<input type="checkbox"/> BGCF Target Scheme Transparency	<input type="checkbox"/> Prefix RN to Dialed Digits
<input type="checkbox"/> Convert Inactive To Sendrecv	<input type="checkbox"/> Reject 3XX With IP
<input type="checkbox"/> Delay Cut Through	<input type="checkbox"/> Reject 3XX With TN
<input type="checkbox"/> Disable 2806 Compliance	<input type="checkbox"/> Same CallId For Required Authorization
<input type="checkbox"/> Disable Optional Register Parameters	<input type="checkbox"/> Transit PAI From Unregistered Peer
<input checked="" type="checkbox"/> Disposition Handling Required	<input type="checkbox"/> Suppress UNREGISTER
<input type="checkbox"/> Don't Send Fast Start Proposal	<input type="checkbox"/> TTC-ISUP Mapping
<input type="checkbox"/> Enable 3261 Cancel Handling	<input type="checkbox"/> Use Called Party In Request URI
<input type="checkbox"/> Include ENUM Parameters	<input type="checkbox"/> Use Colon In SDP Media Type Parameter
<input type="checkbox"/> Insert In Band Indication	<input type="checkbox"/> Use JIP from 3XX Response in PDCS-Billing-Info-Header
<input type="checkbox"/> Add Loop Back Route Header	<input type="checkbox"/> Validate ISUB Address
<input type="checkbox"/> Map 181 Or 182 Message To 183	<input type="checkbox"/> Wait Till Connect Before Abandon FastStart
<input type="checkbox"/> Map 3xx Contact URL To Route Header	<input type="checkbox"/> Restrict User Equal To Phone
<input type="checkbox"/> Map Contractor Number In P-Sig-Info Header	<input type="checkbox"/> Ignore SDP After Offer Answer Completed
<input type="checkbox"/> Use Network Provided Screening Indicator For Calling Number	<input type="checkbox"/> Map Diversion Header To Charge Number
<input type="checkbox"/> MonitorRtpOnEgressUpdate	<input type="checkbox"/> Map RN, OCN, RDI To Diversion Header
<input type="checkbox"/> Honor Subsequent SDP Answer	<input type="checkbox"/> Enable Globalization of Numbers starting with Alphabet

Figure21:

<input type="checkbox"/> Ignore Unmodified Called Userpart If Truncated	<input type="checkbox"/> Ignore Unmodified Calling Userpart If Truncated
<b>BCI</b> <input type="checkbox"/> BCI Interwork Encountered <span style="float: right;"><input type="checkbox"/> BCI ISDN Access</span>	
<b>Carrier Information</b> <input type="checkbox"/> Disconnect If Neither Terminating CA Nor CIC Received <span style="float: right;"><input type="checkbox"/> Use Terminating CIC From SIP</span> <input type="checkbox"/> Use Terminating CA From SIP	
<b>Domain Name</b> <div style="display: flex; justify-content: space-between;"> <div style="width: 48%;"> <input type="checkbox"/> Preserve Ingress FROM Domain Name  <input type="checkbox"/> Preserve Ingress R-URI Domain Name  <input checked="" type="checkbox"/> Use IP Signaling Peer Domain In R-URI  <input type="checkbox"/> Use DM/PM Manipulated Host Name In R-URI  <input type="checkbox"/> Use Zone Level Domain Name in Path Header  <input type="checkbox"/> Use SIP Domain Name In PAI Header  <input type="checkbox"/> Do not use PSX Unmodified From URI Host Part         </div> <div style="width: 48%;"> <input type="checkbox"/> Use Lower Case Domain Names  <input type="checkbox"/> Use SIP Domain Name In FROM Field  <input type="checkbox"/> Use Zone Level Domain Name In Contact  <input type="checkbox"/> Use SIP Domain Name In Request URI  <input type="checkbox"/> Use Called URI As R-URI  <input type="checkbox"/> Use PSX Modified To URI Host Part  <input type="checkbox"/> Do not use PSX Unmodified PAI URI Host Part         </div> </div>	
<b>ISUB</b> <input type="checkbox"/> Allow NSAP ISUB <span style="float: right;"><input type="checkbox"/> Include Called Party ISUB</span> <input type="checkbox"/> Allow User Specified ISUB <span style="float: right;"><input type="checkbox"/> Include Calling Party ISUB</span>	
<b>Number Portability Attributes</b> NPDI Options: <span style="margin-left: 20px;"><input checked="" type="radio"/> Include npdi</span> <span style="margin-left: 20px;"><input type="radio"/> Include npdi=yes</span> <span style="margin-left: 20px;"><input type="radio"/> Do Not Include npdi</span>	

**Figure22:**

<b>Flags</b> <input type="checkbox"/> Disable rn
<b>Privacy</b> <input type="checkbox"/> Transparency <input type="checkbox"/> AnonymizeHostIpAddress Privacy Information: <span style="margin-left: 20px;"><input checked="" type="radio"/> P-Preferred-ID</span> <span style="margin-left: 20px;"><input type="radio"/> P-Asserted-ID</span> <span style="margin-left: 20px;"><input type="radio"/> Remote-Party-ID</span>
<b>Flags</b> <input checked="" type="checkbox"/> Include Privacy <span style="float: right;"><input type="checkbox"/> Privacy Required by Proxy</span> <input type="checkbox"/> MS Lync Privacy Support <span style="float: right;"><input type="checkbox"/> Include Embedded PAI Header in Redirected INVITE</span> <input type="checkbox"/> Do Not Include Tel URI In PAI Header
<b>Redirect</b> Mode: <span style="border: 1px solid black; padding: 2px;">Accept Redirection</span>
Contact Handling: <span style="margin-left: 20px;"><input checked="" type="radio"/> Merge Received Contacts with Existing Contacts</span> <span style="margin-left: 20px;"><input type="radio"/> Purge Existing Contacts</span>
<b>Flags</b> <div style="display: flex; justify-content: space-between;"> <div style="width: 48%;"> <input type="checkbox"/> Skip Crankback Profile And Always Crankback  <input type="checkbox"/> Force Re-query for Redirection  <input type="checkbox"/> Skip DTG Lookup For 3XX Contact         </div> <div style="width: 48%;"> <input type="checkbox"/> Honor Embedded Headers in 3xx  <input type="checkbox"/> Enhanced Local Redirection         </div> </div>
<b>SIP Cause Mapping</b> Internal To SIP Cause Mapping: <span style="border: 1px solid black; padding: 2px;">1 - DEFAULT</span> SIP To Internal Cause Mapping: <span style="border: 1px solid black; padding: 2px;">1 - DEFAULT</span>

**Figure23:**

Internal to SIP Cause Mapping Profile Name

SIP to Internal Cause Mapping Profile Name

**SIP Headers And Parameters**

Include Charge Information: ☒ Include None ☐ Include P-Charge-Info

Session-Expires Refresher: ☒ Not Send ☐ UAC ☐ UAS

☐ None ☒ Original Called Number (OCN)

SIP TO Header Mapping: ☐ Called Number ☐ GAP Dialed Number

☐ Fallback to called number if OCN is not present ☐ Fallback to called number if GAP Dialed number is not present

☐ PI Allowed Send CPC In: ☒ DEFAULT ☐ FROM ☐ PAI ☐ BOTH

Destination Trunk Group Options:

Originating Trunk Group Options:

Generate Call-ID Using:

**Flags**

☐ Include CIC ☐ Include PSTN Parameters

☐ Include CPC Information ☐ Include Qvalue

☐ Include NPI ☐ Skip CSeq Check In Early Dialog

☐ Include OLIP ☒ Transparency For Destination Trunk Group Parameter

☐ Include P-K-Adn ☐ End To End Ack

☐ No CDR Change In End To End Ack

**Figure24:**

**Call Forwarding**

☐ Diversion-History Info Interworking (RFC 6044 compliance)

**Redirection Information**

☒ Diversion ☐ Diversion With Transparency

☐ PK Header

**History Information**

☐ Include History-Info ☐ Cause Parameter In RFC 4458 ☐ Reason With Cause Value As Per RFC 4244

**CPC Mapping Flags**

☐ Map CPC when Presentation Indicator is Restricted

☒ Any CPC ☐ CPC=Priority

**Send CPC Param In**

☒ Default ☐ PAI ☐ From ☐ Both (PAI and From)

**P Charge Info**

☐ Transparency

P-Charge-Info Information: ☒ URI Parameter ☐ User Parameter ☐ Header Parameter

**Flags**

☒ Include NPI ☐ Include NOA

**SIP Jurisdiction Support**

Jurisdiction Support: ☐ Enable ☒ Disable

SBC JIP Profile:

Use Transport Type object to configure the preferred transport.

**Figure 25:**

<b>Flags</b> <input type="checkbox"/> Apply Settings to SBC TG <input type="checkbox"/> Apply Settings to "Use SIP In Core" Egress TG if Applicable	
<b>SIP RPH ETS</b> Action For ETS 400 Response With 417 Reason Code: <span>Retry Without ETS</span>	
ETS Default Priority Value: <span>0</span>	
<b>Flags</b> <input type="checkbox"/> Add/Modify ETS Resource Priority Header <input type="checkbox"/> Use Incoming ETS Resource Value <input type="checkbox"/> Do Not Include Require RPH	
<b>SIP Variant Type</b> SIP Variant Type: <span>sonus</span>	
<b>Flags</b> <input type="checkbox"/> Apply Setting to SBC TG <input type="checkbox"/> Apply Setting to "Use SIP In Core" Egress TG if Applicable	
<b>Transport Type</b> Transport Type 1: <span>TCP</span> Transport Type 2: <span>&lt;None&gt;</span> Transport Type 3: <span>&lt;None&gt;</span> Transport Type 4: <span>&lt;None&gt;</span> <input type="checkbox"/> Use configured transport for egress leg	

**Figure26:**

<b>Ingress IP Attributes - Signaling Back A Message To The Peer That We Receive A Call From</b>	
<b>Flags</b>	
<input type="checkbox"/> 181 Supported <input type="checkbox"/> 182 Supported <input type="checkbox"/> Convert Progress To Alert <input type="checkbox"/> Don't Send Facility Message <input type="checkbox"/> Don't Send 3XX With IP <input type="checkbox"/> Don't Send 3XX With TN <input type="checkbox"/> Map Called Party Category In P-Sig-Info Header <input type="checkbox"/> No SDP In 180 Supported <input type="checkbox"/> Refuse Fast Start Proposal <input type="checkbox"/> Registration Expires in Expires Header <input type="checkbox"/> Map Subsequent 180 to 183 <input type="checkbox"/> Early Media Authorization <input type="checkbox"/> Report Early Media Auth	<input type="checkbox"/> Registration Support 3xx <input type="checkbox"/> Send 183 On Initiating Disconnect Treatment <input type="checkbox"/> Send Fast Start Response In CP <input type="checkbox"/> Send SDP In 200 OK If 18x Reliable <input type="checkbox"/> Send Updated SDP In 200 OK <input type="checkbox"/> Send SDP In Subsequent 18x <input type="checkbox"/> Send TLS Connection Failure Response <input type="checkbox"/> Suppress 183 For 3xx Redirect Response <input type="checkbox"/> Suppress 183 Without SDP <input type="checkbox"/> Override 3xx Relay <input type="checkbox"/> Send BIT-H Of BCI In Outgoing Invite <input type="checkbox"/> Convert Alert To Progress <input type="checkbox"/> Process Qtype and Attach DPC/SSN info in 3xx

**Figure27:**

<b>Carrier Information</b> <input type="checkbox"/> Generate Terminating CA <input type="checkbox"/> Generate Terminating CIC	
<b>History Information</b> <input type="checkbox"/> Include History-Info <input type="checkbox"/> Cause Parameter In RFC 4458 <input type="checkbox"/> Reason With Cause Value As Per RFC 4244	
Access Transfer Profile: <span>&lt;None&gt;</span>	
<b>Trf Parameters</b> Preferred Trf Uri: <input type="text"/> Preferred Mrb Uri: <input type="text"/>	
<b>Enume Parameters</b> TTL: <span>0</span>	

## Configuring Codec Entry Profile

Codecs define the audio encoding methods and their associated attributes. You can add custom codec entries which are then available to include when configuring codecs in a Packet Service Profile. When you add a codec entry, the parameters available change, depending on the base codec you select. You can also configure options for a selected Codec Entry that specify how to handle DTMF digits in the media stream.

Define the codec entry priorities and codec names.

### DTMF Types Configuration

Use the DTMF relay window under Codec Entry configured in Packet Service Profiles to specify how to handle DTMF digits in the media stream.

**Figure28:**

Codec Entry:	G711-DEFAULT
Audio Encoding:	G.711
Coding Rate (kbts/s):	6.3
Fax Tone Treatment:	<None>
Packet Size (ms):	10
Preferred RTP Payload Type:	128
Max Interleave Depth:	0
Fax Treatment Failure Handling	
<input type="radio"/> Disconnect <input checked="" type="radio"/> Continue	
G.711 Law	
<input type="radio"/> Law From Other Leg <input type="radio"/> A Law <input checked="" type="radio"/> U Law <input type="checkbox"/> G.711 Send SID	
Modem Tone Treatment	
<input checked="" type="radio"/> None <input type="radio"/> Notify Peer <input type="radio"/> Disconnect <input type="radio"/> Fallback To G.711 <input type="radio"/> Apply Fax Treatment	
Modem Treatment Failure Handling	
<input type="radio"/> Disconnect <input checked="" type="radio"/> Continue	
Honor Tone Detection	
<input type="checkbox"/> Fax <input type="checkbox"/> Modem	
DTMF Relay	
<input checked="" type="radio"/> None <input type="radio"/> Out-Of-Band <input type="radio"/> RFC 2833 <input type="radio"/> Either OOB Or 2833 <input type="radio"/> Both OOB And 2833 <input checked="" type="checkbox"/> DTMF Remove Digits <input type="checkbox"/> enable DTMF Duration	
DTMF Duration(ms): 300	

**Figure29:**

AMR & AMR-WB Options		
<input type="checkbox"/> AMRWB lu-UP Mode	<input type="checkbox"/> Mode Change Neighbor	
<input type="checkbox"/> RTCP APP CMR	<input type="checkbox"/> Initial Codec Mode as per 3GPP 26.114	
FEC Redundancy		
<input checked="" type="radio"/> 0 <input type="radio"/> 1 <input type="radio"/> 2		
AMR-WB Mode Set (Kbps)		
<input type="checkbox"/> 6.6	<input type="checkbox"/> 14.25	<input type="checkbox"/> 19.85
<input type="checkbox"/> 8.85	<input type="checkbox"/> 15.85	<input type="checkbox"/> 23.05
<input type="checkbox"/> 12.65	<input type="checkbox"/> 18.25	<input type="checkbox"/> 23.85
Silence Suppression		
<input type="checkbox"/> Silence Suppression <input type="radio"/> vad1 <input checked="" type="radio"/> vad2		
OPUS Options		
<input type="checkbox"/> UseCBR <input type="checkbox"/> UseFEC <input type="checkbox"/> UseDTX		
Max Average Bit Rate (bits/sec): 20000		

**Figure30:**

EVS Options		
<input type="checkbox"/> UseCompatHeader	<input type="checkbox"/> Support EVS-AMR-WB-IO Mode	<input type="checkbox"/> Support Asymmetric Bit Rate
Partial Redundancy		
<input checked="" type="radio"/> -1 <input type="radio"/> 0 <input type="radio"/> 2 <input type="radio"/> 3 <input type="radio"/> 5 <input type="radio"/> 7		
Max Channels		
<input checked="" type="radio"/> 1 <input type="radio"/> 2 <input type="radio"/> 6 <input type="radio"/> 3 <input type="radio"/> 4 <input type="radio"/> 5		
BR Set		
Min Bit Rate: <input type="text"/>		
Max Bit Rate: <input type="text"/>		
SILK Options		
<input type="checkbox"/> UseSilkDTX		
Max Average Bit Rate (bits/sec): 0		

## Video Call Configuration

Configure Maximum Video Bandwidth and Video Bandwidth Reduction Factor in packet service profile to enable video calls.

Figure 44:

Video Calls

Maximum Video Bandwidth (kbps):

2000

Video Bandwidth Reduction Factor (%):

1

☒ Audio Only If Video Is Prevented

IPv4 TOS:

0

IPv6 Traffic Class:

0

IEEE 802.1Q VLAN COS:

0

Codec List Profile:

<None>

Configuring Packet Service Profile

Each Packet Service Profile is configured for a pair of gateways, and includes entries for up to four audio/video encoding methods. The pair of gateways can be originating for destination gateways in the same gateway group, or can be originating for destination gateways in an inter-gateway group.

Packet Service Profile IN

From the Drop Down, select the codec Entry profiles created during initial steps,

Figure 31:

Packet Service Profile:

SIPREC\_PSP\_INGRESS

Silence Factor:

40

Voice Initial Playout Buffer Delay (ms):

10

Type Of Service:

0

AAL1 Payload Size:

47

Preferred RTP Payload Type For DTMF Relay:

<None>

Media Packet COS:

0

Monitoring Profile:

<None>

Media Peer Inactivity Timeout (s):

0

Codec Entry

Codec Entry:

G729A-DEFAULT

Add

Update

Codec Entry	Value
1	G711-DEFAULT

Figure32:

T.38			
Number of Redundant Packets			
<input checked="" type="radio"/> 0		<input type="radio"/> 1	
Low Speed Number of Redundant Packets			
<input checked="" type="radio"/> 0		<input type="radio"/> 1	
T.38v0 Maximum Bit Rate			
<input checked="" type="radio"/> 2.4 kbits/s		<input type="radio"/> 4.8 kbits/s	<input type="radio"/> 9.6 kbits/s
Data Rate Management Type			
<input checked="" type="radio"/> Type 1 - Local Generation of TCF		<input type="radio"/> Type 2 - Transfer of TCF	
Use Max Bit Rate Only			
<input checked="" type="radio"/> Disabled		<input type="radio"/> Enabled	
ECM			
<input type="checkbox"/> ECM Preferred			
T.38FaxMaxDatagram Size without Redundancy			
<input checked="" type="radio"/> Disabled		<input type="radio"/> Enabled	
T.38FaxProtocolVersion: T.38(v0)			
Honor Remote Precedence			
<input checked="" type="radio"/> Disabled		<input type="radio"/> Enabled	
Send Route PSP Precedence			
<input checked="" type="radio"/> Disabled		<input type="radio"/> Enabled	
Packet-To-Packet Control			
Transcode			
<input type="radio"/> Only	<input checked="" type="radio"/> Conditional	<input type="radio"/> Determined By PSP For Other Leg	<input type="radio"/> Trans

### Transcoding:

Use the Codecs Allowed For Transcoding window to specify, for a Packet Service profile (PSP), between which codecs you want the SBC to allow transcoding. Checking options on this window specifies that the codecs selected in the "This Leg" row can be transcoded to those selected in the "Other Leg" row, and vice versa.

PSPs are assigned to both legs of a call. Therefore the Codecs Allowed For Transcoding values applied to a particular call reflect the contributions of both profiles, with the ingress and egress call legs being viewed as "This Leg" by one profile and as the "Other Leg" by the other profile.

This control specifies the transcoding method used for the associated packet flow.

The SBC performs transcoding for media streams carried between two IP devices by translating the streams from the ingress audio encoding format to the egress audio encoding format when the devices do not share a common codec. In some environments, transcoding may be preferred over negotiating the attributes of a common codec.

- **Conditional** The SBC performs transcoding when any of the conditions specified in the Conditions In Addition To "No Common Codec" section are met.
- **Determined By PSP For Other Leg** The SBC performs transcoding based on the transcoding options specified in the packet service profile assigned to the other leg of the call. When selected, PSX Manager disables the check boxes in the Codecs Allowed for Transcoding section.
- **Only** The SBC performs transcoding for the codecs selected in the Codec Allowed For Transcoding section (see definition below). None of the conditions specified in the Conditions In Addition To "No Common Codec" section are used in determining when to perform transcoding.
- **Transcoder Free Transparency** When selected, the SBC transparently passes the PSP from the ingress call-leg to the egress call-leg, bypassing transcoding.

**Figure33:**



Packet-To-Packet Control	
Transcode	
<input type="radio"/> Only <input checked="" type="radio"/> Conditional <input type="radio"/> Determined By PSP For Other Leg <input type="radio"/> Transcode	
Conditions In Addition To "No Common Codec"	
<input type="checkbox"/> Apply Fax Tone Treatment <input type="checkbox"/> Different DTMF Relay <input type="checkbox"/> Different Packet Size	<input type="checkbox"/> Different Silence Suppression <input type="checkbox"/> Honor Answer Preference <input type="checkbox"/> Honor Offer Preference <input type="checkbox"/> Different 2833 Payload Type
Codes Allowed For Transcoding	
This Leg: <input checked="" type="checkbox"/> G.711 A <input checked="" type="checkbox"/> G.711 U <input type="checkbox"/> G.722 <input type="checkbox"/> G.722.2 <input type="checkbox"/> G.723.1 <input type="checkbox"/> G.726 <input type="checkbox"/> G.729 <input type="checkbox"/> OPUS <input type="checkbox"/> EVS <input type="checkbox"/> SILK <input type="checkbox"/> T.38 <input type="checkbox"/> iLBC <input type="checkbox"/> AMR Other Leg: <input checked="" type="checkbox"/> G.711 A <input checked="" type="checkbox"/> G.711 U <input type="checkbox"/> G.722 <input type="checkbox"/> G.722.2 <input type="checkbox"/> G.723.1 <input type="checkbox"/> G.726 <input type="checkbox"/> G.729 <input type="checkbox"/> OPUS <input type="checkbox"/> EVS <input type="checkbox"/> SILK <input type="checkbox"/> T.38 <input type="checkbox"/> iLBC <input type="checkbox"/> AMR	
RTCP	
<input checked="" type="checkbox"/> RTCP         Packet Loss Threshold (Packets Lost/100,000 Packets): 0	
RR Bandwidth: 250	
RS Bandwidth: 250	
Packet Loss Action	
<input checked="" type="radio"/> None <input type="radio"/> Trap <input type="radio"/> Trap And Disconnect	
<input type="checkbox"/> Enable RTCP Only For HELD Calls <input type="checkbox"/> Termination For Pass-Through Calls	
<input type="checkbox"/> RTCP-MUX <input type="checkbox"/> Generate RTCP for T140 if not received from other leg	
RTCP-XR	
<input type="checkbox"/> Relay <input type="checkbox"/> Relay Or Terminate	

### RTCP configuration:

Use this object to specify Real Time Control Protocol (RTCP) options for the call. RTCP is used to report network traffic congestion data.

When set to **Enable**, Use RTCP for the call leg.

**Figure34:**

<b>Packet Service Profile</b> SQL Search Criteria (19 entries) Packet Service Profile: * <input type="text"/> Search <input type="button" value="More..."/> <input type="button" value="Reset"/>	<input type="checkbox"/> Different Packet Size <input type="checkbox"/> Honor Offer Preference <input type="checkbox"/> Different 2833 Payload Type
Packet Service Profile ATOS_IN ATOS_OUT DEFAULT ENT_ACM_DEFAULT_PSP ENT_AWAYASM_DEFAULT_PSP ENT_CUCM_DEFAULT_PSP ENT_LYNXMS_PSP_TCP ENT_LYNXMS_PSP_TLS PSTN_PSP SIPREC_PSP SIPREC_PSP_EGRESS SIPREC_PSP_EGRESS1 SIPREC_PSP_EGRESS2 SIPREC_PSP_EGRESS_pob SIPREC_PSP_INGRESS SIPREC_PSP_INGRESS_G729 SRTP_SIPREC_PSP_EGRESS	Codes Allowed For Transcoding This Leg: <input checked="" type="checkbox"/> G.711 A <input checked="" type="checkbox"/> G.711 U <input type="checkbox"/> G.722 <input type="checkbox"/> G.722.2 <input type="checkbox"/> G.723.1 <input type="checkbox"/> G.726 <input checked="" type="checkbox"/> G.729 <input type="checkbox"/> OPUS <input type="checkbox"/> EVS <input type="checkbox"/> SILK <input type="checkbox"/> T.38 <input type="checkbox"/> iLBC <input type="checkbox"/> AMR Other Leg: <input checked="" type="checkbox"/> G.711 A <input checked="" type="checkbox"/> G.711 U <input type="checkbox"/> G.722 <input type="checkbox"/> G.722.2 <input type="checkbox"/> G.723.1 <input type="checkbox"/> G.726 <input checked="" type="checkbox"/> G.729 <input type="checkbox"/> OPUS <input type="checkbox"/> EVS <input type="checkbox"/> SILK <input type="checkbox"/> T.38 <input type="checkbox"/> iLBC <input type="checkbox"/> AMR
	RTCP <input checked="" type="checkbox"/> RTCP         Packet Loss Threshold (Packets Lost/100,000 Packets): 0
	RR Bandwidth: 250
	RS Bandwidth: 250
	Packet Loss Action
	<input checked="" type="radio"/> None <input type="radio"/> Trap <input type="radio"/> Trap And Disconnect
	<input type="checkbox"/> Enable RTCP Only For HELD Calls <input type="checkbox"/> Termination For Pass-Through Calls
	<input type="checkbox"/> RTCP-MUX <input type="checkbox"/> Generate RTCP for T140 if not received from other leg
	RTCP-XR
	<input type="checkbox"/> Relay <input type="checkbox"/> Relay Or Terminate

**Figure35:**

Peer Absence Action	
<input checked="" type="radio"/> None	<input type="radio"/> Trap
<input type="radio"/> Trap And C	
Silence Insertion Descriptor	
G.711 Silence Insertion Descriptor RTP Payload Type: 13	
<input checked="" type="checkbox"/> Silence Insertion Descriptor Heartbeat	
Data Calls	
Initial Playout Buffer Delay (ms):	50
Packet Size:	20
Preferred RTP Payload Type:	56
Video Calls	
Maximum Video Bandwidth (kbps):	0
Video Bandwidth Reduction Factor (%):	0
<input checked="" type="checkbox"/> Audio Only If Video Is Prevented	
IPv4 TOS:	0
IPv6 Traffic Class:	0
IEEE 802.1Q VLAN COS:	0
Codec List Profile:	<None>
Qos Values	
MSRP DSCP:	0
DTLS SCTP DSCP:	0
T140 DSCP:	0
Application Dscp:	0

Secure RTP/RTCP > Crypto Suite Profile is used for srtp configurations. Please refer [Media Encryption](#) for more details

**Figure36:**

Non RTP Stream	
Max Non Rtp Bandwidth(kbps):	0
Non RTP TLS Profile Name:	defaultTlsProfile
Audio Transparency	
Unknown Codec Packet Size(ms)	10
Unknown Codec Bit Rate(kbps)	124
Secure RTP/RTCP	
Crypto Suite Profile:	<None>
Flags	
<input type="checkbox"/> Allow Fallback	<input type="checkbox"/> Enable SRTP
<input type="checkbox"/> Reset ROC On Session Key Change	<input type="checkbox"/> Reset Enc/Dec/ROC on Decryption Key Change
<input type="checkbox"/> Update Crypto On Modify	<input type="checkbox"/> Allow Pass Through
DTLS/SRTP	
Crypto Suite Profile:	<None>
Flags	
<input type="checkbox"/> Allow Fallback	<input type="checkbox"/> Enable DTLS
<input type="checkbox"/> Relay DTLS SRTP	<input type="checkbox"/> Relay DTLS SCTP

**Figure37:**

Flags	
<input type="checkbox"/> DSCP Passthrough	<input type="checkbox"/> Interwork DTMF OOB-2833 Without Transcoding
<input type="checkbox"/> Digit Detect Send Enabled	<input type="checkbox"/> Use Direct Media
<input type="checkbox"/> Disallow Data Calls	<input type="checkbox"/> Validate Peer Support for DTMF Events
<input type="checkbox"/> SSRC Randomize	<input type="checkbox"/> HD Codec Preferred
<input type="checkbox"/> Reserve BW for Preferred Audio Common Codec	<input type="checkbox"/> Prefer NB PassThru Over HDTranscode
<input type="checkbox"/> Police on Heaviest Audio Codec	<input type="checkbox"/> Match Offered Codec Group If Nb Only
<input type="checkbox"/> t140 Call	<input type="checkbox"/> Force Route PSP Order
<input type="checkbox"/> Allow Audio Transcode For MultiStream Call	<input type="checkbox"/> SSRC Randomize For Srtp
<input type="checkbox"/> Generate and Signal SSRC and CName	<input type="checkbox"/> Vtp Support
<input type="checkbox"/> Allow Mid Call SSRC Modification	<input type="checkbox"/> Always Send Timestamp

## Packet Service Profile OUT

Figure38:

Packet Service Profile:	SIPREC_PSP_EGRESS
Silence Factor:	40
Voice Initial Playout Buffer Delay (ms):	10
Type Of Service:	0
AAL1 Payload Size:	47
Preferred RTP Payload Type For DTMF Relay:	<None>
Media Packet COS:	0
Monitoring Profile:	<None>
Media Peer Inactivity Timeout (s):	0

Codec Entry

Codec Entry: G729A-DEFAULT

AddUpdate

Codec Entry	Value
1	G711-DEFAULT

Delete

Figure39:

T.38

Number of Redundant Packets

☒ 0☐ 1

Low Speed Number of Redundant Packets

☒ 0☐ 1

T.38v0 Maximum Bit Rate

☒ 2.4 kbits/s☐ 4.8 kbits/s☐ 9.6 kbits/s

Data Rate Management Type

☒ Type 1 - Local Generation of TCF☐ Type 2 - Transfer of T

Use Max Bit Rate Only

☒ Disabled☐ Enabled

ECM

☐ ECM Preferred

T38FaxMaxDatagram Size without Redundancy

☒ Disabled☐ Enabled

T.38FaxProtocolVersion: T.38(v0)

Honor Remote Precedence

☐ Disabled☒ Enabled

Send Route PSP Precedence

☒ Disabled☐ Enabled

Packet-To-Packet Control

Transcode

☐ Only☒ Conditional☐ Determined By PSP For Other Leg☐ Trans

Figure40:

Packet-To-Packet Control	
Transcode	
<input type="radio"/> Only <input checked="" type="radio"/> Conditional <input type="radio"/> Determined By PSP For Other Leg <input type="radio"/> Transcode	
Conditions In Addition To "No Common Codec"	
<input type="checkbox"/> Apply Fax Tone Treatment <input type="checkbox"/> Different DTMF Relay <input type="checkbox"/> Different Packet Size	<input type="checkbox"/> Different Silence Suppression <input type="checkbox"/> Honor Answer Preference <input type="checkbox"/> Honor Offer Preference <input type="checkbox"/> Different 2833 Payload Type
Codecs Allowed For Transcoding	
This Leg: <input checked="" type="checkbox"/> G.711 A <input checked="" type="checkbox"/> G.711 U <input type="checkbox"/> G.722 <input type="checkbox"/> G.722.2 <input type="checkbox"/> G.723.1 <input type="checkbox"/> G.726 <input type="checkbox"/> G.729 <input type="checkbox"/> OPUS <input type="checkbox"/> EVS <input type="checkbox"/> SILK <input type="checkbox"/> T.38 <input type="checkbox"/> iLBC <input type="checkbox"/> AMR	
Other Leg: <input checked="" type="checkbox"/> G.711 A <input checked="" type="checkbox"/> G.711 U <input type="checkbox"/> G.722 <input type="checkbox"/> G.722.2 <input type="checkbox"/> G.723.1 <input type="checkbox"/> G.726 <input type="checkbox"/> G.729 <input type="checkbox"/> OPUS <input type="checkbox"/> EVS <input type="checkbox"/> SILK <input type="checkbox"/> T.38 <input type="checkbox"/> iLBC <input type="checkbox"/> AMR	
RTCP	
<input checked="" type="checkbox"/> RTCP         Packet Loss Threshold (Packets Lost/100,000 Packets): 0	
RR Bandwidth: 250	
RS Bandwidth: 250	
Packet Loss Action	
<input checked="" type="radio"/> None <input type="radio"/> Trap <input type="radio"/> Trap And Disconnect	
<input type="checkbox"/> Enable RTCP Only For HELD Calls <input type="checkbox"/> Termination For Pass-Through Calls	
<input type="checkbox"/> RTCP-MUX <input type="checkbox"/> Generate RTCP for T140 if not received from other leg	
RTCP-XR	
<input type="checkbox"/> Relay <input type="checkbox"/> Relay Or Terminate	

Figure41:

Peer Absence Action	
<input checked="" type="radio"/> None <input type="radio"/> Trap <input type="radio"/> Trap And Disconnect	
Silence Insertion Descriptor	
G.711 Silence Insertion Descriptor RTP Payload Type: 13	
<input checked="" type="checkbox"/> Silence Insertion Descriptor Heartbeat	
Data Calls	
Initial Playback Buffer Delay (ms): 50	
Packet Size: 20	
Preferred RTP Payload Type: 56	
Video Calls	
Maximum Video Bandwidth (kbps): 0	
Video Bandwidth Reduction Factor (%): 0	
<input checked="" type="checkbox"/> Audio Only If Video Is Prevented	
IPv4 TOS: 0	
IPv6 Traffic Class: 0	
IEEE 802.1Q VLAN COS: 0	
Codec List Profile: <None>	
Qos Values	
MSRP DSCP: 0	
DTLS SCTP DSCP: 0	
T140 DSCP: 0	
Application DSCP: 0	

Figure 42:

Non RTP Stream	
Max Non Rtp Bandwidth(kbps):	0
Non RTP TLS Profile Name: defaultTlsProfile	
Audio Transparency	
Unknown Codec Packet Size(ms)	10
Unknown Codec Bit Rate(kbps)	124
Secure RTP/RTCP	
Crypto Suite Profile: <None>	
Flags	
<input type="checkbox"/> Allow Fallback	<input type="checkbox"/> Enable SRTP
<input type="checkbox"/> Reset ROC On Session Key Change	<input type="checkbox"/> Reset Enc/Dec/ROC on Decryption Key Change
<input type="checkbox"/> Update Crypto On Modify	<input type="checkbox"/> Allow Pass Through
DTLS/SRTP	
Crypto Suite Profile: <None>	
Flags	
<input type="checkbox"/> Allow Fallback	<input type="checkbox"/> Enable DTLS
<input type="checkbox"/> Relay DTLS SRTP	<input type="checkbox"/> Relay DTLS SCTP

Figure 43:

Flags	
<input type="checkbox"/> DSCP Passthrough	<input type="checkbox"/> Interwork DTMF OOB-2833 Without Transcoding
<input type="checkbox"/> Digit Detect Send Enabled	<input type="checkbox"/> Use Direct Media
<input type="checkbox"/> Disallow Data Calls	<input type="checkbox"/> Validate Peer Support for DTMF Events
<input type="checkbox"/> SSRC Randomize	<input type="checkbox"/> HD Codec Preferred
<input type="checkbox"/> Reserve BW for Preferred Audio Common Codec	<input type="checkbox"/> Prefer NB PassThru Over HDTranscode
<input type="checkbox"/> Police on Heaviest Audio Codec	<input type="checkbox"/> Match Offered Codec Group If Nb Only
<input type="checkbox"/> t140 Call	<input type="checkbox"/> Force Route PSP Order
<input type="checkbox"/> Allow Audio Transcode For MultiStream Call	<input type="checkbox"/> SSRC Randomize For Srtsp
<input type="checkbox"/> Generate and Signal SSRC and CName	<input type="checkbox"/> Vtp Support
<input type="checkbox"/> Allow Mid Call SSRC Modification	<input type="checkbox"/> Always Send Timestamp

## Configuring IP Signaling Peer Group

IP Peer is an entity of the Session Border Controller, which is configured inside the Zone. It acts as a destination endpoint for the call to be routed towards. An IP Peer constitutes an IPv4/IPv6 address or a Fully Qualified Domain Name (FQDN) with a port number.

Figure 45:

IP Signaling Peer Group: SIPREC_PEER1							
Description:							
Policy Profile Group: <None>							
Flags							
<input type="checkbox"/> Send All Peer IP Addresses/FQDNs							
<input type="checkbox"/> Number of Routes to Try: 1 <input type="checkbox"/> All							
Route Prioritization: <input type="radio"/> Sequence <input checked="" type="radio"/> Round Robin <input type="radio"/> All Proportion							
Peer Group Data							
Sequence Number: 0							
<input checked="" type="radio"/> IPv4 Address: 8 . 8 . 8 . 8 Port Number: 8							
<input type="radio"/> IPv6 Address: 0 : 0 : 0 : 0 : 0 : 0 : 0 : 0 Port Number: 0							
<input type="radio"/> Server FQDN: Port Number: 0							
Proportion: 0							
<input checked="" type="checkbox"/> In Service							
<a href="#">Add/Update</a>							
Sequence Number	IP Address	Port Number	Server FQDN	Port Number	Send	Service Status	Proportion
0	8.8.8.8	8		0	IP Address	In Service	0

Figure 46:

IP Signaling Peer Group: SIPREC\_PEER2

Description:

Policy Profile Group: <None>

Flags

☐ Send All Peer IP Addresses/FQDNs

☐ Number of Routes to Try: 1 ☐ All

Route Prioritization: ☐ Sequence ☒ Round Robin ☐ All Proportion

Peer Group Data

Sequence Number: 0

☒ IPv4 Address: 8 . 8 . 8 . 8 Port Number: 8

☐ IPv6 Address: 0 : 0 : 0 : 0 : 0 : 0 : 0 : 0 Port Number: 0

☐ Server FQDN: Port Number: 0

Proportion: 0

☒ In Service

Add/Update

Sequence Number	IP Address	Port Number	Server FQDN	Port Number	Send	Service Status	Proportion
0	8.8.8.8	8		0	IP Address	In Service	0

Configuring Carrier

Please note that we have used default Carrier '0000' for our testing.

Figure 47:

Carrier: 0000

Partition: DEFAULT

Preferred Packet Service Profile ID Group: <None>

Signalling Profile: <None>

SIP Domain: <None>

Service Provider Id (Hex): 0

Context Info

Ingress CVT Rule: <None>

Egress CVT Rule: <None>

Flags

☐ Escaped

☐ Ignore Tandem Script On Redirection

Scripts

Casual Routing: <None> Runtime Variables

Nonsubscriber: <None> Runtime Variables

Tandem: <None> Runtime Variables

Services

☐ Not Screened ☒ Screened - Normal ☐ Screened - Fraud

Class Of Service: <None>

Service Exception Profile: <None>

Configuring Element Routing Priority Profile

Please note that we have cloned and used default Element Routing Priority for our testing.

Figure 48:

Element Routing Priority: SIPREC

---

Call Property

Call Type: Private

Priority: 1

Network: All

Toll Indication: <All>

---

Entity Type: <None>

Priority: 1

Add Update

---

Call Type	Call Priority	Network	Toll Indication	Entity Type	Priority
Private	1	All	<All>	<None>	1
0+	1	All	<All>	<None>	1
0-	1	All	<All>	<None>	1
1+	1	All	<All>	Trunk Group	1
1+	2	All	<All>	<None>	2
IDDD	1	All	<All>	<None>	1
0+ IDDD	1	All	<All>	<None>	1
00	1	All	<All>	<None>	1
IP VPN Service	1	All	<All>	<None>	1
Test	1	All	<All>	<None>	1
Transit	1	All	<All>	<None>	1
Other Carrier Chosen	1	All	<All>	<None>	1
Carrier Cut Through	1	All	<All>	<None>	1
User Name	1	All	<All>	<None>	1
Mobile	1	All	<All>	<None>	1

## Configuring SignalingProfile

Please note that we have used default Signaling Profile 'DEFAULT\_IP\_PROFILE' for our testing.

Figure 49:

SIGNALING PROFILE: DEFAULT\_IP\_PROFILE

---

Transit Carrier Indicator Profile: <None>

Generic Digit Type: <Unknown>

---

Ingress

CFT

☐ Send CFT Information

CFT Information For Early Backward Message: ☐ Off Net ☒ On Net

---

Ingress Flags

<input type="checkbox"/> Disallow Missing Calling Number	<input type="checkbox"/> Generate Charge Message
<input type="checkbox"/> Disallow Without Billing Number	<input type="checkbox"/> Generate CPG for Call Forward Notify
<input type="checkbox"/> Disallow Without OLIP	<input type="checkbox"/> Inbound TNS Allowed
<input type="checkbox"/> Don't Generate Exit Message	<input type="checkbox"/> Normalize Carrier Code
<input type="checkbox"/> Don't Send Restricted Connected Line Identity	<input type="checkbox"/> Propagate Egress Channel Information
<input type="checkbox"/> Don't Send Connected Number	<input type="checkbox"/> Propagate FE Parameter
<input checked="" type="checkbox"/> Don't Send Unrequested Connected Line Identity	<input type="checkbox"/> Treat CIC 0000 As No CIC
<input type="checkbox"/> Enable Redirection Capability	<input type="checkbox"/> Use ISUP Immediate REL On SUS Timer
<input type="checkbox"/> Enable Transfer Connect	<input type="checkbox"/> Validate GAP Type Ported Number
<input type="checkbox"/> FE Parameter In Short Form	

Figure 50:

Egress			
<b>TNS Flags</b>			
Inter LATA Local:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
Intra LATA Local:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
Inter LATA Toll:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
Intra LATA Toll:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
0:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
0+ Inter LATA:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
0+ Intra LATA:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
00:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
IDDD:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
0+IDDD:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
Calling Name:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
Calling Number:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
Charge Number:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
CIP:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
CSP:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
JIP:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
OLIP:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
Original Called Number:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send
Redirecting Number:	<input checked="" type="radio"/> No Input	<input type="radio"/> Send	<input type="radio"/> Don't Send

Figure 51:

Redirect Capability:	<input checked="" type="radio"/> Yes	<input type="radio"/> No
Redirect Count:	<input checked="" type="radio"/> Yes	<input type="radio"/> No
Redirect Information:	<input checked="" type="radio"/> Yes	<input type="radio"/> No
Calling Party/Billing Number: <None>		
<b>Egress Flags</b>		
<input type="checkbox"/> Add Prefix 011 For International Calls	<input type="checkbox"/> Propagate Charge Message	
<input type="checkbox"/> Add Prefix 1 For Inter LATA Calls	<input type="checkbox"/> Propagate GD Parameter	
<input type="checkbox"/> Add Prefix 1 For Intra LATA Calls	<input type="checkbox"/> Propagate Ingress Channel Information	
<input type="checkbox"/> Annex E Support	<input type="checkbox"/> Reorder Trunk as Low Priority Based On ISUP Preference	
<input type="checkbox"/> Apply Switch Type CPC Profile	<input type="checkbox"/> Reroute On Signaling Congestion	
<input type="checkbox"/> Called Number 7 Digits	<input checked="" type="checkbox"/> Reset OLIP For Toll Free Calls	
<input type="checkbox"/> Calling Number 7 Digits	<input type="checkbox"/> Restore Calling Number If Derived From Billing Number	
<input type="checkbox"/> Change Bearer Cap From 3.1KHz To Speech	<input type="checkbox"/> Restore Calling Number If Derived From OCN	
<input type="checkbox"/> Convert Numbers To E164 Format	<input type="checkbox"/> Restore Calling Number If Derived From Redirecting Number	
<input type="checkbox"/> CPC Mapping	<input type="checkbox"/> Restore Calling Number If Derived From Trunk Group	
<input type="checkbox"/> Dialed Number As Called Number	<input type="checkbox"/> Restore FCI International Bit	
<input type="checkbox"/> Discard GAP Additional Calling If Same As Calling Number And Ingress SIP	<input type="checkbox"/> Send Billing Number As Calling Number	
<input type="checkbox"/> Don't Strip Calling Number For Restricted Presentation	<input type="checkbox"/> Send Billing Number As Calling Number If Calling Number Not Present	
<input type="checkbox"/> Forced Override OLIP Value	<input type="checkbox"/> Send Contract Number If Allowed By Ingress SIP	
<input type="checkbox"/> Generate FE Parameter	<input type="checkbox"/> Send DM/PM Manipulated Billing Number	
<input type="checkbox"/> OLI Mapping	<input type="checkbox"/> Send Toll Free Number In GAP Parameter	

Figure 52:



<input type="checkbox"/> Prefix RN to Dialed Digits	<input checked="" type="checkbox"/> Send Toll Free Number In OCN Parameter <input type="checkbox"/> Suppress ONI <input type="checkbox"/> Undo LNP <input type="checkbox"/> Use Output ANI For CDNIS
<b>CFT</b> Egress CFT Information: <span style="margin-left: 100px;"><input checked="" type="radio"/> Off Net</span> <span style="margin-left: 100px;"><input type="radio"/> On Net</span>	
<b>Generate PartitionID + NetID In NetworkData In IAM</b> <input type="checkbox"/> Generate PartitionId + NetId In NetworkData In IAM <input type="checkbox"/> Propagate PartitionId + NetId In NetworkData In IAM <input type="checkbox"/> Override PartitionId + NetId In NetworkData In IAM	
<b>IP Double Dip Control Flags</b> <div style="display: flex; justify-content: space-between;"> <div> <input type="checkbox"/> Called Number From Alternate Called Number  <input type="checkbox"/> Restore Ingress Numbers Except Translated Numbers         </div> <div> <input type="checkbox"/> Restore Translated Numbers  <input type="checkbox"/> Skip Egress Trunk Group Processing         </div> </div>	
<b>Mobile Call Delivery</b> Original Called Number: <span style="margin-left: 50px;"><input checked="" type="radio"/> No Input</span> <span style="margin-left: 50px;"><input type="radio"/> Send</span> <span style="margin-left: 50px;"><input type="radio"/> Don't Send</span> Redirection Information: <span style="margin-left: 50px;"><input checked="" type="radio"/> No Input</span> <span style="margin-left: 50px;"><input type="radio"/> Send</span> <span style="margin-left: 50px;"><input type="radio"/> Don't Send</span>	

**Figure 53:**

<b>Redirection Capability Flags</b> <input type="checkbox"/> Enable Redirection Capability Number Control Profile: <None> <span style="float: right;">▼</span> Redirect Information Profile: <None> <span style="float: right;">▼</span> <div style="border: 1px solid black; padding: 5px; margin-top: 5px;"> <b>Flags</b>  <div style="display: flex; justify-content: space-between;"> <input type="checkbox"/> Check Ingress Trunk Group Redirection Capability           <input type="checkbox"/> Check Redirection Capability Of Number Used For Routing In Number Control Profile         </div> <div style="display: flex; justify-content: space-between;"> <input type="checkbox"/> Check Number Control Profile For Received Called Number           <input type="checkbox"/> Check SIP Indirect DIP And Username Translation Source Number         </div> <input type="checkbox"/> Check Received Redirection Parameters       </div>
<b>Common</b> <b>Common Flags</b> <input checked="" type="checkbox"/> Trusted For COL <span style="margin-left: 100px;"><input type="checkbox"/> COLP/COLR IGW Support</span>
<b>Access Transport</b> <input checked="" type="radio"/> Yes <span style="margin-left: 100px;"><input type="radio"/> No</span>
<b>International Gateway Support</b> <input type="checkbox"/> Don't Convert Called Number <span style="margin-left: 100px;"><input type="checkbox"/> Don't Convert Calling Number</span>

## Configuring Feature Control Profile

Please note that we have used default Feature Control Profile 'DEFAULT\_IP' for our testing.

**Figure 54:**

Feature Control Profile:	DEFAULT_IP
Features (Set 1)	
<input type="checkbox"/>	Always Apply Default Calling Party Number From Trunk Group
<input type="checkbox"/>	Always Apply Default Presentation Indicator From Trunk Group
<input checked="" type="checkbox"/>	Apply Business Group Services
<input checked="" type="checkbox"/>	Apply Calling Party Services
<input type="checkbox"/>	Apply Default If Calling Party Number Not Present
<input type="checkbox"/>	Apply Default If Calling Party Number Not Subscriber
<input checked="" type="checkbox"/>	Apply Destination Services
<input checked="" type="checkbox"/>	Apply Dial Plan
<input checked="" type="checkbox"/>	Apply Digit Length Enforcement
<input type="checkbox"/>	Apply OLIP Services
<input type="checkbox"/>	Determine JIP
<input checked="" type="checkbox"/>	Determine LATA, Region and MTA
<input type="checkbox"/>	Determine MTA For LRN in Ported Calls
<input type="checkbox"/>	Determine LATA, Region and MTA for LRN in Ported Calls
<input type="checkbox"/>	Exclude LATA Sub-Zone Id For Determining Toll Indication
<input type="checkbox"/>	Filter Routes Before Prioritization
<input type="checkbox"/>	Normalize Digits
<input checked="" type="checkbox"/>	Process Called Number
<input checked="" type="checkbox"/>	Process Calling Number
<input checked="" type="checkbox"/>	Process Generic Digits

**Figure 55:**

<input type="checkbox"/>	Process Presentation Setting
<input type="checkbox"/>	Process Screening Setting
<input type="checkbox"/>	Use Billing Number For Normalization
<input type="checkbox"/>	Use Billing Number For Subscriber
<input type="checkbox"/>	Use Trunk Group Country
<input type="checkbox"/>	Use Trunk Group Country For Blocking Profile
Features (Set 2)	
<input type="checkbox"/>	Always Use Billing Number For Calling Party Number
<input type="checkbox"/>	Always Use Ingress CSP
<input type="checkbox"/>	Always Use Redirecting Number For Calling Number
<input type="checkbox"/>	Always Use Trunk Group JIP
<input checked="" type="checkbox"/>	Apply All Countries Routing
<input type="checkbox"/>	Apply CPC Services
<input type="checkbox"/>	Determine Charge Band
<input type="checkbox"/>	Do Not Replace Calling Number For Emergency Calls
<input type="checkbox"/>	Error On Misrouted LRN
<input type="checkbox"/>	No Local Calls
<input checked="" type="checkbox"/>	Process Called Party NOA
<input checked="" type="checkbox"/>	Process Called Party NPI
<input checked="" type="checkbox"/>	Process Calling Party NOA
<input checked="" type="checkbox"/>	Process Calling Party NPI

**Figure 56:**

<input type="checkbox"/> Skip Called Party Services For Misrouted LRN <input type="checkbox"/> Skip LRN Validation And Unporting From LNP <input type="checkbox"/> Skip LNP For Toll Calls <input type="checkbox"/> Treat Not Presubscribed Input Carrier Input As Not A Casual Call <input type="checkbox"/> Treat Presubscribed Input Carrier Input As Not A Casual Call <input type="checkbox"/> Trigger LNP For 0+ Dialed Calls <input type="checkbox"/> Use Billing Number For Calling Party Number If Calling Party Number Not Present <input type="checkbox"/> Use OCN For Calling Party Number If Redirecting Number Not Present <input type="checkbox"/> Use Redirecting Number For Subscriber
<b>Features (Set 3)</b> <input type="checkbox"/> Add Number of Prefix Digits Stripped To Overlap Dialing Parameters <input type="checkbox"/> All Provisioned Calling and Called Digits Matched for Local Calling Area Determination <input type="checkbox"/> Allow CMT Call <input type="checkbox"/> Apply Network Traffic Management On Indirect Dip <input type="checkbox"/> Always Process Called Number If NOA Unknown <input type="checkbox"/> Always Process Calling Number If NOA Unknown <input type="checkbox"/> Don't Apply Called Party Services During LNP Transition <input type="checkbox"/> Fetch Subscriber With Country Code Prefixed <input type="checkbox"/> Generate ECI <input checked="" type="checkbox"/> Process Redirection Number <input type="checkbox"/> SSG Calling Party Use Signal-In Number

**Figure 57:**

<input type="checkbox"/> Translated Emergency Number <input type="checkbox"/> Try Alternate Address For SIPE <input type="checkbox"/> Use Redirecting BG <input type="checkbox"/> Use Redirecting Number For Called Number Normalization <input type="checkbox"/> Use Redirecting Number Instead Of CLI For DDI Screening <input type="checkbox"/> Apply LATA from Trunk Group If Calling Number Not Present <input checked="" type="checkbox"/> Perform Route Header Based Routing <input type="checkbox"/> Use Destination IP address in Standard Routing <input type="checkbox"/> Disable Fallback To 7 Digits Hosted LNP Lookup <input type="checkbox"/> Determine Charge Band Profile from TG <input type="checkbox"/> Don't Send \1 In Enum Response
<b>Features (Set 4)</b> <input type="checkbox"/> Accept Calls With RPH If Dialed Number Is Non ETS <input type="checkbox"/> Enable RPH ETS <div style="border: 1px solid black; padding: 5px;"> <b>Process Destination Trunk Group And Trunk-Context</b>  <input checked="" type="checkbox"/> Process TGRP  <input type="checkbox"/> Process Trunk-context </div> <input type="checkbox"/> Process Enumdi Parameter <input checked="" type="checkbox"/> Process Originating Trunk Group And Trunk-Context Over OTG <input type="checkbox"/> SIP Cause Code Mapping

**Figure 58:**

☐ Skip Number Translations For Valid Service Routes
 ☐ Include Retry After For 503 Responses
 ☐ Process Swid And Tgid From Sip Invite
 ☐ Don't Restart Timer C on 1xx
 ☐ Override Trunkgroup With Subscriber End Point Profile
 ☐ Fetch State For ENUM SIP AoR
 ☐ Enable Per Route Routing Label
 ☐ Do Not Validate GAP
 ☐ Process ISUP MIME From SIP Message Body
 ☐ Use Flex Variable for Origination Jurisdiction Determination
 ☐ Use Flex Variable for Destination Jurisdiction Determination
 ☐ Process Screening For Call Origination

**Figure 59:**

URI Processing
 

☐ Process TO URI User
 ☐ Process FROM URI User
 ☐ Process PAI URI User
 ☐ Process Diversion URI User
 ☐ Process Called URI User
 ☐ Process Calling URI User
 ☐ Process History-Info URI User
 ☐ Start Using Processed URI User Data

IP Protocol Flags
 

☐ Use IP Protocol Flags
 

Flags
 

☐ Default Called User As A User Name
 ☐ Default Calling User As A User Name
 ☐ Disable Egress Check And Don't Send Contract Number
 ☐ Prefer BICC instead of ISUP routes for FCI preferred value
 ☐ Proxy/Redirector Force Route Calls With Non-Local IP Address
 ☐ Reject Calls To Non-Local Domains
 ☐ Reject Calls To Non-Local IP Addresses
 ☐ Support Domain Name In 300 Contact
 ☐ Support PAI Header in CONTACT
 ☐ Honor Phone-Context Parameter
 ☐ Enable Stir Shaken

PSX Processing Mode
 

☐ Proxy
 ☒ Redirector

## Configuring Trunk Groups

Create two Trunk Groups for Ingress and Egress and associate the Trunk Groups to the gateway created in Step-1.



### Warning

Mandatory! You must capitalize SIP Trunk Group names.

## Trunk Group IN

Follow the instructions below for Ingress Trunk Group.

**Figure 60:**

Trunk Group:	SIPREC_TG1	<input type="checkbox"/> Unrestricted
Gateway:	SBCSYAM1	
Description:		
Auto Recall Profile:	<None>	
Call Processing Localization Variant:	North America	
Calling Area:	<None>	
Carrier:	0000	
Carrier Selection Priority:	<None>	
Country:	1 - USA, Canada and Caribbean	
DDI Range Profile:	<None>	
Destination Switch Type:	Access	
Direction:	Two Way	
Element Routing Priority Profile:	SIPREC	
Feature Control Profile:	DEFAULT_IP	
IP Signaling Profile:	SIPREC_IPSP_TCP	
LATA:	<None>	
Local Recursion Profile:	<None>	
Maximum Satellite Hops:	Three or More Satellite Hops	
Network Data Partition:	0	
Network Data Net:	0	
Next Hop Domain:	<None>	
Number Analysis Profile:	<None>	
Number Length Enforcement:	<None>	

Figure 61:

Originating Carrier:	<None>
PPR Profile:	<None>
Pseudo Carrier:	<None>
Remote Sip Peer Type:	None
Region:	<None>
Routing Criteria Profile:	<None>
SCP Business Service Group:	0
Signaling Profile:	DEFAULT_IP_PROFILE
Signaling Flag:	GR394 ISUP
SIP Domain:	<None>
SIP Response Code Profile:	<None>
TDM Type:	Other
Tone And Announcement Profile:	<None>
Trunk Group COS:	
Trunk Group COS Profile:	<None>
Trunk Group Domain:	<None>
Trunk Number:	
Zone Index Profile:	<None>
ZZ Profile:	<None>
Charge Band Profile:	<None>

Figure 62:

Enum Domain Profile:	<None>	
Flexible Variable Rule:	<None>	
STI Profile:	<None>	
P-Originator-ID:	<input type="text"/>	<input type="checkbox"/> Autogenerate <input type="button" value="Clear"/>
RPH Signaling Profile:	<None>	
Beep Tone Profile:	<None>	
STI Generic Profile:	<None>	
IPSP Generic Profiles:	<None>	
Context Info	<input type="text"/>	

**Ingress**

Charge Indicator:	None
Default CPC:	<None>
Default OLIP:	<None>
Dial Plan Profile:	<None>
Forced OLIP Value:	<None>
In DM/PM Rule:	<None>
Info Transfer Capability Profile:	<None>
IP Version Preference:	IPv4 Only
ONI:	<input type="text"/>
JIP:	<input type="text"/>

**Figure 63:**

NPA:	<input type="text"/>
Numbering Plan:	NANP_ACCESS
In Policy Profile Group:	<None>
CVT Rule:	<None>
Service Detect Policy Profile Group:	<None>

**Flags**

<input type="checkbox"/> Allow Hex Digits In Cdpn	<input type="checkbox"/> Non-Zero Video Bandwidth Based Routing for H.323
<input type="checkbox"/> Discard NPDI	<input type="checkbox"/> Non-Zero Video Bandwidth Based Routing for SIP
<input type="checkbox"/> Discard RN	<input type="checkbox"/> Overlap Dialing
<input type="checkbox"/> HD Preferred Routing	<input type="checkbox"/> TNS Circuit Code Based Routing
<input type="checkbox"/> HD Supported Routing	<input type="checkbox"/> Use IPTG Routing (Hop By Hop Routing) For Ingress

**Egress**

Charge Indicator:	None
Out DM/PM Rule:	<None>
Out Policy Profile Group:	<None>
CVT Rule:	<None>
Trunk Context:	<input type="text"/>
R-URI Host:	<input type="text"/>
R-URI Host Port:	0

**Figure 64:**

<b>Flags</b> <input type="checkbox"/> Disable Crankback <input type="checkbox"/> Enable JIP Interwork <input type="checkbox"/> Use Preferred Identity <input type="checkbox"/> Send STI Verified Display Name	
<b>Billing</b> Billing Plan: <None> Billing Information: <None> Default Billing Number: <input type="text"/> Nature Of Address: <None> Numbering Plan Indicator: <None>	
<b>Calling Party Number</b> Calling Party: <input type="text"/> Nature Of Address: <None> Numbering Plan Indicator: <None> Presentation: <None> Screening: <None> Default Presentation: <None>	

**Figure 65:**

<b>Flags</b> <input checked="" type="checkbox"/> Do Not Use For Fallback Bearer Capability <input type="checkbox"/> Escaped <input type="checkbox"/> Out Of Service <input type="checkbox"/> Satellite Trunk <input type="checkbox"/> Use Sac NonSac Call Types For ZZ Profile	
<b>IP TG</b> IP Signaling Peer Group: SIPREC_PEER1 <input checked="" type="checkbox"/> IP Peer Supported Packet Service Profile ID Group: SIPREC_INGRESS <input checked="" type="checkbox"/> Egress IP Signaling Profile: SIPREC_IPSP_TCP	
<b>Packet Service Profile</b> Preferred Packet Service Profile ID Group: <None> <input type="checkbox"/> Destination Override	
<b>Traffic Management Options</b> Trunk Group Reservation Level 1: 10 Trunk Group Reservation Level 2: 5	
<b>VPN Information</b> Business Group: <None> Business Location: <None> <input type="checkbox"/> Business Group From CLI	

**Figure 66:**

<b>Services</b> <input checked="" type="radio"/> Not Screened <input type="radio"/> Screened - Normal <input type="radio"/> Screened - Fraud Class Of Service: <None> Service Exception Profile: <None>	
<b>Use SIP in Core</b> Inter Gateway IP Signaling Profile: <None> Egress IP Signaling Profile: <None>	
<b>SIP Used in Core</b> Inter Gateway IP Signaling Profile: <None> Egress IP Signaling Profile: <None>	

## Trunk Group OUT

Follow the instructions below for Egress Trunk Group.

**Figure 67:**

Trunk Group:	SIPREC_TG2	<input type="checkbox"/> Unrestricted
Gateway:	SBSCSYAM1	
Description:		
Auto Recall Profile:	<None>	
Call Processing Localization Variant:	North America	
Calling Area:	<None>	
Carrier:	0000	
Carrier Selection Priority:	<None>	
Country:	1 - USA, Canada and Caribbean	
DDI Range Profile:	<None>	
Destination Switch Type:	Access	
Direction:	Two Way	
Element Routing Priority Profile:	SIPREC	
Feature Control Profile:	DEFAULT_IP	
IP Signaling Profile:	SIPREC_IPSP_TCP	
LATA:	<None>	
Local Recursion Profile:	<None>	
Maximum Satellite Hops:	Three or More Satellite Hops	
Network Data Partition:	0	
Network Data Net:	0	
Next Hop Domain:	<None>	
Number Analysis Profile:	<None>	
Number Length Enforcement:	<None>	

Figure 68:

Originating Carrier:	<None>
PPR Profile:	<None>
Pseudo Carrier:	<None>
Remote Sip Peer Type:	None
Region:	<None>
Routing Criteria Profile:	<None>
SCP Business Service Group:	0
Signaling Profile:	DEFAULT_IP_PROFILE
Signaling Flag:	GR394 ISUP
SIP Domain:	<None>
SIP Response Code Profile:	<None>
TDM Type:	Other
Tone And Announcement Profile:	<None>
Trunk Group COS:	
Trunk Group COS Profile:	<None>
Trunk Group Domain:	<None>
Trunk Number:	
Zone Index Profile:	<None>
ZZ Profile:	<None>
Charge Band Profile:	<None>

Figure 69:



Enum Domain Profile:	<None>
Flexible Variable Rule:	<None>
STI Profile:	<None>
P-Originator-ID:	<input type="text"/> <input type="checkbox"/> Autogenerate <input type="button" value="Clear"/>
RPH Signaling Profile:	<None>
Beep Tone Profile:	<None>
STI Generic Profile:	<None>
IPSP Generic Profiles:	<None>
Context Info	<input type="text"/>

**Ingress**

Charge Indicator:	None
Default CPC:	<None>
Default OLIP:	<None>
Dial Plan Profile:	<None>
Forced OLIP Value:	<None>
In DM/PM Rule:	<None>
Info Transfer Capability Profile:	<None>
IP Version Preference:	IPv4 Only
ONI:	<input type="text"/>
JIP:	<input type="text"/>

**Figure 70:**

NPA:	<input type="text"/>
Numbering Plan:	NANP_ACCESS
In Policy Profile Group:	<None>
CVT Rule:	<None>
Service Detect Policy Profile Group:	<None>

**Flags**

<input type="checkbox"/> Allow Hex Digits In Cdpn	<input type="checkbox"/> Non-Zero Video Bandwidth Based Routing for H.323
<input type="checkbox"/> Discard NPDI	<input type="checkbox"/> Non-Zero Video Bandwidth Based Routing for SIP
<input type="checkbox"/> Discard RN	<input type="checkbox"/> Overlap Dialing
<input type="checkbox"/> HD Preferred Routing	<input type="checkbox"/> TNS Circuit Code Based Routing
<input type="checkbox"/> HD Supported Routing	<input type="checkbox"/> Use IPTG Routing (Hop By Hop Routing) For Ingress

**Egress**

Charge Indicator:	None
Out DM/PM Rule:	<None>
Out Policy Profile Group:	<None>
CVT Rule:	<None>
Trunk Context:	<input type="text"/>
R-URI Host:	<input type="text"/> R-URI Host Port: <input type="text" value="0"/>

**Figure 71:**

**Flags**

- ☐ Disable Crankback
- ☐ Enable JIP Interwork
- ☐ Use Preferred Identity
- ☐ Send STI Verified Display Name

**Billing**

Billing Plan: <None>

Billing Information: <None>

Default Billing Number:

Nature Of Address: <None>

Numbering Plan Indicator: <None>

**Calling Party Number**

Calling Party:

Nature Of Address: <None>

Numbering Plan Indicator: <None>

Presentation: <None>

Screening: <None>

Default Presentation: <None>

**Figure 72:**

**Flags**

- ☒ Do Not Use For Fallback Bearer Capability
- ☐ Escaped
- ☐ Out Of Service
- ☐ Satellite Trunk
- ☐ Use Sac NonSac Call Types For ZZ Profile

**IPTG**

IP Signaling Peer Group: SIPREC\_PEER2

☒ IP Peer Supported

Packet Service Profile ID Group: SIPREC\_EGRESS

☒ Egress IP Signaling Profile: SIPREC\_IPSP\_TCP

**Packet Service Profile**

Preferred Packet Service Profile ID Group: <None>

☐ Destination Override

**Traffic Management Options**

Trunk Group Reservation Level 1: 10

Trunk Group Reservation Level 2: 5

**VPN Information**

Business Group: <None>

Business Location: <None>

☐ Business Group From CLI

**Figure 73:**

**Services**

☒ Not Screened      ☐ Screened - Normal      ☐ Screened - Fraud

Class Of Service: <None>

Service Exception Profile: <None>

**Use SIP in Core**

Inter Gateway IP Signaling Profile: <None>

Egress IP Signaling Profile: <None>

**SIP Used in Core**

Inter Gateway IP Signaling Profile: <None>

Egress IP Signaling Profile: <None>

## Configuring Routes

Routing allows you to send calls to the correct destination. You can use routing options based on your requirements. Configure the standard and specific routes (with usernames) to ensure that no matter how the called party is addressed (a number or username), the SBC routes the message to the Core. Create Route entries for standard Trunk Group routing with Matching Criteria and a Routing Label destination.

## Routing Label

A routing label is associated with a route. Each route includes a gateway/trunk group pair. Routing labels provide the link between an entry in the Standard Route table and the set of routes associated with that Standard Route table entry.

### Routing Label 1

Figure 74:

Routing Label: SIPREC\_RL1

Action

☒ Routes ☐ Script ☐ Route Hopping ☐ LCR

Number Of Routes Requested: 10 ☒ All

Number Of Routes Per Call: 1

Script: <None> [Runtime Variables](#)

Partition: <None>

DMPM Rule: <None> ☐ Apply Later

CPC Screening: <None>

Overflow Number:

Overflow Nature Of Address: <None>

Overflow Numbering Plan Indicator: <None>

Call Parameter Filter Group: <None>

Call Parameter Filter Profile Script: <None>

Call Parameter Filter Criteria Cluster: <None>

Routing Criteria

☐ Use Entity Type <None>

Partition

☒ Ignore ☐ Do not Use ☐ Use

Destination

☒ Ignore ☐ Do not Use ☐ Use

Route Prioritization Type

☒ Sequence ☐ Proportion ☐ Round Robin ☐ All Proportion ☐ Least Cost Routing

Figure 75:

☐ Use TAR Routes

TAR Route Prioritization Type

☒ Sequence ☐ Proportion ☐ Round Robin ☐ All Proportion ☐ Least Cost Routing

Route Prioritization Type For Equal Cost Routes: Sequence

Local Routes

☐ Pass Only Local Routes ☐ Prioritize Local Routes ☒ Do Nothing

Filter Criteria Routes

☐ Pass Only Filter Criteria Routes ☐ Prioritize Filter Criteria Routes ☒ Do Not Change Route Order

Flags

☐ Continue Number Translation ☐ Continue CNAM Translation ☐ No Connect Signal To Be Sent ☐ Use Configured NAPTR Order and Preference Value

Routes

Type	Endpoint 1	Endpoint 2	IP Peer	Sequence	Proport...	Status	TAR Ac...	TAR Lo...	DM/P...	Apply ...	Testing	Cost	Skip LR	STI T...	N...	N...
GSX Gateway	SIPREC_TG1	SBCSYAM1		1	0	In Service	Normal	0		Do Not...	Normal	1000000	Disab...	0	65...	65...

Figure 76:

Route

Type: GSX Gateway

Gateway: SBCSYAM1

Trunk Group: SIPREC\_TG1

IP Peer: <None>

Sequence: 1

Proportion: 0

Cost: 1000000

TAR Action: Normal

TAR Location: 0

NAPTR Order: 65536

NAPTR Preference: 65536

DM/PM Rule: <None> ☐ Apply Later

Testing: ☒ Normal ☐ Test ☐ Non-Test

☒ In Service ☐ Skip Local Recursion

☐ Signing ☐ Local Tagging ☐ Verification

OK

Cancel

## Routing Label 2

Figure 77:

Routing Label: SIPREC\_RL2

Action

☒ Routes
 ☐ Script
 ☐ Route Hopping
 ☐ LCR

Number Of Routes Requested: 10 ☒ All

Number Of Routes Per Call: 1

Script: <None> [Runtime Variables](#)

Partition: <None>

DM/PM Rule: <None> ☐ Apply Later

CPC Screening: <None>

Overflow Number:

Overflow Nature Of Address: <None>

Overflow Numbering Plan Indicator: <None>

Call Parameter Filter Group: <None>

Call Parameter Filter Profile Script: <None>

Call Parameter Filter Criteria Cluster: <None>

Routing Criteria

☐ Use Entity Type <None>
 

Partition

☒ Ignore
 ☐ Do not Use
 ☐ Use

Destination

☒ Ignore
 ☐ Do not Use
 ☐ Use

Route Prioritization Type

☒ Sequence
 ☐ Proportion
 ☐ Round Robin
 ☐ All Proportion
 ☐ Least Cost Routing

Figure 78:

Route Prioritization Type For Equal Cost Routes: Sequence

☐ Use TAR Routes

TAR Route Prioritization Type

☒ Sequence
 ☐ Proportion
 ☐ Round Robin
 ☐ All Proportion
 ☐ Least Cost Routing

Route Prioritization Type For Equal Cost Routes: Sequence

Local Routes

☐ Pass Only Local Routes
 ☐ Prioritize Local Routes
 ☒ Do Nothing

Filter Criteria Routes

☐ Pass Only Filter Criteria Routes
 ☐ Prioritize Filter Criteria Routes
 ☒ Do Not Change Route Order

Flags

☐ Continue Number Translation
 ☐ Continue CNAM Translation
 ☐ No Connect Signal To Be Sent
 ☐ Use Configured NAPTR Order and Preference Value

Routes

Type	Endpoint 1	Endpoint 2	IP Peer	Sequence	Proport...	Status	TAR Ac...	TAR Lo...	DM/P...	Apply ...	Testing	Cost	Skip LR	STI T...	N...	N...
GSX Gateway	SIPREC_TG2	SBCSYAM1		0	0	In Service	Normal	0		Do Not...	Normal	1000000	Disab...	0	65...	65...

Figure 79:

Route

Type: GSX Gateway

Gateway: SBCSYAM1

Trunk Group: SIPREC\_TG2

IP Peer: <None>

Sequence: 0

Proportion: 0

Cost: 1000000

TAR Action: Normal

TAR Location: 0

NAPTR Order: 65536

NAPTR Preference: 65536

DMPM Rule: <None> ☐ Apply Later

Testing: ☒ Normal ☐ Test ☐ Non-Test

☒ In Service ☐ Skip Local Recursion

☐ Signing ☐ Local Tagging ☐ Verification

OK Cancel

## Routes

Routing allows you to send calls to the correct destination. You can use routing options based on your requirements. Configure the standard and specific routes (with usernames) to ensure that no matter how the called party is addressed (a number or username), the SBC routes the message to the Core. Create Route entries for standard Trunk Group routing with Matching Criteria and a Routing Label destination.

### Route 1

Figure 80:

Host: 172.16.100.216 @ 4330  
Master (SWe) - V14.01.00R000

View: Standard Route Close All Perspective: Full View

Entity Type: <None>

Not Applicable

Not Applicable

Not Applicable

☒ Call Parameter Filter Profile: <None>

☐ Call Parameter Filter Profile Group: <None>

Destination National: 5555511111

Destination Country: 1 - USA, Canada and Caribbean

Domain Name: <None>

☒ IP Address: 0 . 0 . 0 . 0

Partition: DEFAULT

Routing Label: SIPREC\_RL1

Cal Type

Transmission Medium

Speech

3.1 KHz Audio

7.0 KHz Audio

56 kbps

64 kbps

Packet

Multirate

384 kbps

1536 kbps

☒ All Call Type Bits

Time Range: ALL

## Route 2

Figure 81:

Entity Type: <None>

Not Applicable

Not Applicable

Not Applicable

☒ Call Parameter Filter Profile: <None>

☐ Call Parameter Filter Profile Group: <None>

Destination National: 2222222222

Destination Country: 1 - USA, Canada and Caribbean

Domain Name: <None>

☒ IP Address: 0 . 0 . 0 . 0

Partition: DEFAULT

Routing Label: SIPREC\_RL2

Cal Type

Transmission Medium

Speech

3.1 KHz Audio

7.0 KHz Audio

56 kbps

64 kbps

Packet

Multirate

384 kbps

1536 kbps

☒ All Call Type Bits

Time Range: ALL

## Configuring SIPRec

The PSX uses the following configurable objects when determining whether a call needs to be recorded or not:

- Recording Criteria contain the rules to match for invoking call recording (this is the same for SIPREC and MCT).
- SRS Groups contain multiple Recording profiles for SRS redundancy (up to 8).
  - Transport

- IP V4/V6 address port
- Encryption data (for SRTP)
- IP TG to be used by the SBC for RS session
- Contains data of multiple SRS servers
- Recording Cluster profile contains multiple SRS Groups for simultaneous recording (up to 4).

## NICE Trunk Group

Create Trunk Group in PSX for SIPRec with the same name created above using SBC CLI. Duplicate default IP Signaling Profile and Packet Service Profile and map it to NICE TG.

**Figure 82:**

Trunk Group:	SIPREC_TG4	<input type="checkbox"/> Unrestricted
Gateway:	SBCPOOJA	
Description:		
Auto Recall Profile:	<None>	
Call Processing Localization Variant:	North America	
Calling Area:	<None>	
Carrier:	0000	
Carrier Selection Priority:	<None>	
Country:	1 - USA, Canada and Caribbean	
DDI Range Profile:	<None>	
Destination Switch Type:	Access	
Direction:	Two Way	
Element Routing Priority Profile:	SIPREC	
Feature Control Profile:	DEFAULT_IP	
IP Signaling Profile:	SIPREC_IPSP	
LATA:	<None>	
Local Recursion Profile:	<None>	
Maximum Satellite Hops:	Three or More Satellite Hops	
Network Data Partition:	0	
Network Data Net:	0	
Next Hop Domain:	<None>	
Number Analysis Profile:	<None>	
Number Length Enforcement:	<None>	
Originating Carrier:	<None>	

**Figure 83:**

PPR Profile:	<None>
Pseudo Carrier:	<None>
Remote Sip Peer Type:	None
Region:	<None>
Routing Criteria Profile:	<None>
SCP Business Service Group:	0
Signaling Profile:	DEFAULT_IP_PROFILE
Signaling Flag:	GR394 ISUP
SIP Domain:	<None>
SIP Response Code Profile:	<None>
TDM Type:	Other
Tone And Announcement Profile:	<None>
Trunk Group COS:	
Trunk Group COS Profile:	<None>
Trunk Group Domain:	<None>
Trunk Number:	
Zone Index Profile:	<None>
ZZ Profile:	<None>
Charge Band Profile:	<None>
Enum Domain Profile:	<None>
Flexible Variable Rule:	<None>

**Figure 84:**

STI Profile:	<None>	
P-Origination-ID:	<input type="text"/>	<input type="checkbox"/> Autogenerate <input type="button" value="Clear"/>
RPH Signaling Profile:	<None>	
Beep Tone Profile:	<None>	
STI Generic Profile:	<None>	
IPSP Generic Profiles:	<None>	
Context Info	<input type="text"/>	
<b>Ingress</b>		
Charge Indicator:	None	
Default CPC:	<None>	
Default OLIP:	<None>	
Dial Plan Profile:	<None>	
Forced OLIP Value:	<None>	
In DM/PM Rule:	<None>	
Info Transfer Capability Profile:	<None>	
IP Version Preference:	IPv4 Only	
ONI:	<input type="text"/>	
JIP:	<input type="text"/>	
NPA:	<input type="text"/>	
Numbering Plan:	NANP_ACCESS	
In Policy Profile Group:	<None>	

**Figure 85:**

CVT Rule:	<None>										
Service Detect Policy Profile Group:	<None>										
<b>Flags</b> <table border="1"> <tr> <td><input type="checkbox"/> Allow Hex Digits In Cdpn</td> <td><input type="checkbox"/> Non-Zero Video Bandwidth Based Routing for H.323</td> </tr> <tr> <td><input type="checkbox"/> Discard NPDI</td> <td><input type="checkbox"/> Non-Zero Video Bandwidth Based Routing for SIP</td> </tr> <tr> <td><input type="checkbox"/> Discard RN</td> <td><input type="checkbox"/> Overlap Dialing</td> </tr> <tr> <td><input type="checkbox"/> HD Preferred Routing</td> <td><input type="checkbox"/> TNS Circuit Code Based Routing</td> </tr> <tr> <td><input type="checkbox"/> HD Supported Routing</td> <td><input type="checkbox"/> Use IPTG Routing (Hop By Hop Routing) For Ingress</td> </tr> </table>		<input type="checkbox"/> Allow Hex Digits In Cdpn	<input type="checkbox"/> Non-Zero Video Bandwidth Based Routing for H.323	<input type="checkbox"/> Discard NPDI	<input type="checkbox"/> Non-Zero Video Bandwidth Based Routing for SIP	<input type="checkbox"/> Discard RN	<input type="checkbox"/> Overlap Dialing	<input type="checkbox"/> HD Preferred Routing	<input type="checkbox"/> TNS Circuit Code Based Routing	<input type="checkbox"/> HD Supported Routing	<input type="checkbox"/> Use IPTG Routing (Hop By Hop Routing) For Ingress
<input type="checkbox"/> Allow Hex Digits In Cdpn	<input type="checkbox"/> Non-Zero Video Bandwidth Based Routing for H.323										
<input type="checkbox"/> Discard NPDI	<input type="checkbox"/> Non-Zero Video Bandwidth Based Routing for SIP										
<input type="checkbox"/> Discard RN	<input type="checkbox"/> Overlap Dialing										
<input type="checkbox"/> HD Preferred Routing	<input type="checkbox"/> TNS Circuit Code Based Routing										
<input type="checkbox"/> HD Supported Routing	<input type="checkbox"/> Use IPTG Routing (Hop By Hop Routing) For Ingress										
<b>Egress</b>											
Charge Indicator:	None										
Out DM/PM Rule:	<None>										
Out Policy Profile Group:	<None>										
CVT Rule:	<None>										
Trunk Context:	<input type="text"/>										
R-URI Host:	<input type="text"/> R-URI Host Port: <input type="text" value="0"/>										
<b>Flags</b> <table border="1"> <tr> <td><input type="checkbox"/> Disable Crankback</td> </tr> <tr> <td><input type="checkbox"/> Enable JIP Interwork</td> </tr> <tr> <td><input type="checkbox"/> Use Preferred Identity</td> </tr> <tr> <td><input type="checkbox"/> Send STI Verified Display Name</td> </tr> </table>		<input type="checkbox"/> Disable Crankback	<input type="checkbox"/> Enable JIP Interwork	<input type="checkbox"/> Use Preferred Identity	<input type="checkbox"/> Send STI Verified Display Name						
<input type="checkbox"/> Disable Crankback											
<input type="checkbox"/> Enable JIP Interwork											
<input type="checkbox"/> Use Preferred Identity											
<input type="checkbox"/> Send STI Verified Display Name											

**Figure 86:**



<b>Billing</b>	
Billing Plan:	<None>
Billing Information:	<None>
Default Billing Number:	
Nature Of Address:	<None>
Numbering Plan Indicator:	<None>
<b>Calling Party Number</b>	
Calling Party:	
Nature Of Address:	<None>
Numbering Plan Indicator:	<None>
Presentation:	<None>
Screening:	<None>
Default Presentation:	<None>
<b>Flags</b>	
<input checked="" type="checkbox"/> Do Not Use For Fallback Bearer Capability	<input type="checkbox"/> Out Of Service
<input type="checkbox"/> Escaped	<input type="checkbox"/> Satellite Trunk
	<input type="checkbox"/> Use Sac NonSac Call Types For ZZ Profile
<b>IPTG</b>	
IP Signaling Peer Group:	SIPREC_PEER2
	<input checked="" type="checkbox"/> IP Peer Supported
Packet Service Profile ID Group:	SIPREC_EGRESS1

**Figure 87:**

<input checked="" type="checkbox"/> Egress IP Signaling Profile:	SIPREC_IPSP
<b>Packet Service Profile</b>	
Preferred Packet Service Profile ID Group:	<None>
	<input type="checkbox"/> Destination Override
<b>Traffic Management Options</b>	
Trunk Group Reservation Level 1:	10
Trunk Group Reservation Level 2:	5
<b>VPN Information</b>	
Business Group:	<None>
Business Location:	<None>
	<input type="checkbox"/> Business Group From CLI
<b>Services</b>	
<input checked="" type="radio"/> Not Screened	<input type="radio"/> Screened - Normal
	<input type="radio"/> Screened - Fraud
Class Of Service:	<None>
Service Exception Profile:	<None>
<b>Use SIP in Core</b>	
Inter Gateway IP Signaling Profile:	<None>
Egress IP Signaling Profile:	<None>
<b>SIP Used in Core</b>	
Inter Gateway IP Signaling Profile:	<None>
Egress IP Signaling Profile:	<None>

## SRS Cluster

An SRS is the target to which the SBC sends session recordings. The SBC supports configuring multiple SRS' on the PSX using SRS Group Profiles. SRS Cluster contains multiple SRS Groups for simultaneous recording (up to 4).

**Figure 88:**

SRS Group Cluster Id: SRSCUSTER\_1

Description:

Sequence Number: 0

SRS Group Id: SRSPROFILE\_1

Add/Update

Sequence Number	SRS Group Id
0	SRSPROFILE_1

## SRS Group Profile

Provide NICE recorder, primary and secondary IPv4 or IPv6 address and port (5060). Also, mention the NICE TG name. The name of the NICE TG created in the SBC and the PSX should be the same, otherwise recording would not be initiated toward NICE. Transport type can be set to UDP/TCP/TLS. We have to configure appropriate transport at NICE for successful recordings. Please refer to [NICE transport configurations](#) for NICE specific configurations

To enable SRTP, we can choose CryptoSuite from the dropdown. For more details refer [Media Encryption](#)



1. NICE SRS IP and port details should be configured as per customer deployment.
2. Transport preference mentioned in SRS Group profile should match transport preferences in Trunk Group towards SIPRec zone.

Figure 89:

SRS Group Profile ID: SRSPROFILE\_1

Description:

SRS Group Properties

Number Of Simultaneous Stream: 1

Load Distribution: ☒ Sequence ☐ RoundRobin

SRS Server Properties

Sequence Number: 3

Trunkgroup ID: SIPREC\_TG4

Crypto Suite Profile: SIPREC\_CRYPT0

☒ IPv4 Address: 3 . 3 . 3 . 3 Port V4 Number: 5060

☐ IPv6 Address: 0 : 0 : 0 : 0 : 0 : 0 : 0 : 0 Port V6 Number: 0

☐ Server FQDN: Port Number: 0

SRS Server Transport: ☐ UDP ☐ TCP ☒ TLS

☒ Enable SRTP

Add/Update

Sequence Number	SRS IP/FQDN Address	Port	Transport	Trunkgroup ID	Crypto Suite ID
0	2.2.2.2	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0
1	3.3.3.3	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0
2	2.2.2.2	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0
3	3.3.3.3	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0

## Call Recording Criteria

Provide call criteria for recording which you wish to record, like calling number, called number, ingress and egress TG, SBC name, the leg you want to record, and either ingress or egress. Recorder type should be SIPRec. Enable the criteria. When a call is made, it shall be recorded if it falls under this criteria.

Figure 90:

Call Recording Criteria: CRC\_SIPREC1

SRS Group Cluster: SRSCUSTER\_1

Ingress Trunk Group Id: <None>

Egress Trunk Group Id: <None>

Calling Party Id:

Called Party Id:

☒ Next Hop IPv4 Signaling Address: 0 . 0 . 0 . 0

☐ Next Hop IPv6 Signaling Address: 0 : 0 : 0 : 0 : 0 : 0 : 0 : 0

☒ Previous Hop IPv4 Signaling Address: 0 . 0 . 0 . 0

☐ Previous Hop IPv6 Signaling Address: 0 : 0 : 0 : 0 : 0 : 0 : 0 : 0

GSX Name: SBCSYAM1

Recording Type: Ingress Leg

Recording Stop Criteria: 0 ☒ Manual ☐ Number Of Calls

Recording Duration: 0

Recorder Type: SIPRec

Beep Tone Profile: <None>

☒ Criteria Enabled

### Call Forking to two Active recorders

Configure Number of Simultaneous Stream to "2", for SBC to stream media simultaneously to two **Active** SRSs.


 Use this configuration only when you have two independent NICE recorder setups with both configured SRSs running in Active mode.

Figure 91:

SRS Group Profile ID: SRSPROFILE\_1

Description:

SRS Group Properties

Number Of Simultaneous Stream: 1

Load Distribution: ☒ Sequence ☐ RoundRobin

SRS Server Properties

Sequence Number: 3

Trunkgroup ID: SIPREC\_TG4

Crypto Suite Profile: SIPREC\_CRYPT0

☒ IPv4 Address: 3 . 3 . 3 . 3 Port V4 Number: 5060

☐ IPv6 Address: 0 : 0 : 0 : 0 : 0 : 0 : 0 : 0 Port V6 Number: 0

☐ Server FQDN: Port Number: 0

SRS Server Transport: ☐ UDP ☐ TCP ☒ TLS

☒ Enable SRTP

Add/Update

Sequence Number	SRS IP/FQDN Address	Port	Transport	Trunkgroup ID	Crypto Suite ID
0	2.2.2.2	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0
1	3.3.3.3	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0

### Redundancy withActive-Standby SRSs

With the SRS redundancy solution, the integration includes two SRS, where one is active (primary-SRS1)and the standby is inactive (secondary-SRS2). If the primary SRS fails, then the secondary SRS becomesactive.



Ribbon recommends NICE to be configured with Failback disabled. Refer to [NICE configuration for NoFailback mode](#) for additional NICE configuration changes.

With Failback disabled, If the primary SRS fails, the secondary SRS becomes active. When the primary SRS comes back up, the secondary SRS remains active and the primary server becomes inactive.

## Sequential Forking

When the number of simultaneous streams is set to 1, the SBC shall start streaming to active SRS with lowest sequence number[SRS1]. If the SRS1 goes down, the SBC blacklists the SRS1 and the SBC automatically uses the next active SRS - SRS2 in the SRS group. Refer to [NICE Configurations for Sequential Forking](#) for additional NICE configuration changes.



With below pathcheck profile configuration, the SBC blacklists unreachable SRS servers as well as Standby SRS servers[based on 5xx response for OPTIONS]. So, the SBC is responsible for detecting SRS failures and initiating a new session to SRS for both ongoing and new calls.

In the pathcheck profile associated to SRS IP peers, we configure **failureResponseCodes** parameter to define 5xx response codes from Standby SRS server to treat as failure response. So, the SBC blacklists Standby SRS to avoid creating new recording sessions to the inactive SRS.

```
set profiles services pathCheckProfile sip_recording1 protocol sipOptions
set profiles services pathCheckProfile sip_recording1 sendInterval 10
set profiles services pathCheckProfile sip_recording1 replyTimeoutCount 3
set profiles services pathCheckProfile sip_recording1 recoveryCount 1
set profiles services pathCheckProfile sip_recording1 failureResponseCodes [ all5xx ]
set profiles services pathCheckProfile sip_recording1 transportPreference preference1 tls-tcp
comm
```

Figure 92:

**SRS Group Profile ID:** SRSPROFILE\_1

**Description:**

**SRS Group Properties**

Number Of Simultaneous Stream: 1

Load Distribution: ☒ Sequence ☐ RoundRobin

**SRS Server Properties**

**Sequence Number:** 3

**Trunkgroup ID:** SIPREC\_TG4

**Crypto Suite Profile:** SIPREC\_CRYPT0

☒ IPv4 Address: 3 . 3 . 3 . 3 **Port V4 Number:** 5060

☐ IPv6 Address: 0 : 0 : 0 : 0 : 0 : 0 : 0 : 0 **Port V6 Number:** 0

☐ Server FQDN: **Port Number:** 0

SRS Server Transport: ☐ UDP ☐ TCP ☒ TLS

☒ Enable SRTP

**Add/Update**

Sequence Number	SRS IP/FQDN Address	Port	Transport	Trunkgroup ID	Crypto Suite ID
0	2.2.2.2	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0
1	3.3.3.3	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0
2	2.2.2.2	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0
3	3.3.3.3	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0

## Parallel Forking

When the number of simultaneous stream is set to "2", the SBC sends two streams to primary[SRS1] and secondary[SRS2] SRS.Redundancy is handled by the SRS internally to detect SRS failure and handle the existing sessions. The SBC connects with SRS1 with active SDP with Active recording and SRS2 with inactive SDP.If SRS1 goes down, SRS2 sends a re-INVITE with active SDP (AIR IP details)to SBC to continue recording via SRS2. Refer [NICE Configurations for Parallel Forking](#) for additional NICE configuration changes.

In the pathcheck profile associated to SRS IP peers, the SBC blacklists only if there is no response from SRS. Any response from SRS is considered as an active response.

```

set profiles services pathCheckProfile sip_recording1 protocol sipOptions
set profiles services pathCheckProfile sip_recording1 sendInterval 10
set profiles services pathCheckProfile sip_recording1 replyTimeoutCount 3
set profiles services pathCheckProfile sip_recording1 recoveryCount 1
set profiles services pathCheckProfile sip_recording1 transportPreference preference1 tls-tcp
comm

```

**Figure 93:**

**SRS Group Profile ID:** SRSPROFILE\_1

**Description:**

**SRS Group Properties**

**Number Of Simultaneous Stream:** 1

**Load Distribution:** ☒ Sequence ☐ RoundRobin

**SRS Server Properties**

**Sequence Number:** 3

**Trunkgroup ID:** SIPREC\_TG4

**Crypto Suite Profile:** SIPREC\_CRYPT0

☒ IPv4 Address: 3 . 3 . 3 . 3 **Port V4 Number:** 5060

☐ IPv6 Address: 0 : 0 : 0 : 0 : 0 : 0 : 0 : 0 **Port V6 Number:** 0

☐ Server FQDN: **Port Number:** 0

**SRS Server Transport:** ☐ UDP ☐ TCP ☒ TLS

☒ Enable SRTP

**Add/Update**

Sequence Number	SRS IP/FQDN Address	Port	Transport	Trunkgroup ID	Crypto Suite ID
0	2.2.2.2	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0
1	3.3.3.3	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0

## Quad Recording

TheSBC is enhanced to support simultaneously recording SIP egress and ingress legs during a session, for a total of four recordings (four simultaneous streams: two in the ingress leg, and two in the egress leg).

TheSBC provisions the SIP recordings towards all four recorders, two from Ingress tap point and another two from egress tap point. (Due to NP limitations, four simultaneous recordings cannot be triggered on the same call leg.)



- The SBC supports sending the recording streams to up to four SRS servers simultaneously.
- Each recording criteria can be configured with a Recording Cluster. A Recording Cluster can have up to four SRS Groups.
- For Quad SIPREC, there are four recordings triggered. Two recordings are triggered on the Ingress leg and two on the Egress leg.
- If there is more than one SRS Group configured, it is recommended to set `recordingType` to "both legs" or "all legs".
- When SIPREC is selected as the Recorder Type, and Recording Type is selected as both legs and all legs, the SBC by default records the ingress leg.

Create four SRS profiles with one SRS entry in each profile.



Please note we need four NICE recorder setups with all four configured SRSs running in Active mode.

**Figure 94:**

SRS Group Cluster Id: SRSCLUSTER\_1

Description:

Sequence Number: 3

SRS Group Id: SRSPROFILE\_4

Add/Update

Sequence Number	SRS Group Id
0	SRSPROFILE_1
1	SRSPROFILE_2
2	SRSPROFILE_3
3	SRSPROFILE_4

**Figure 95:**

SRS Group Profile ID: SRSPROFILE\_1

Description:

SRS Group Properties

Number Of Simultaneous Stream: 1

Load Distribution: ☒ Sequence ☐ RoundRobin

SRS Server Properties

Sequence Number: 1

Trunkgroup ID: SIPREC\_TG4

Crypto Suite Profile: SIPREC\_CRYPT0

☒ IPv4 Address: 3 . 3 . 3 . 3 Port V4 Number: 5060

☐ IPv6 Address: 0 : 0 : 0 : 0 : 0 : 0 : 0 : 0 Port V6 Number: 0

☐ Server FQDN: Port Number: 0

SRS Server Transport: ☐ UDP ☐ TCP ☒ TLS

☒ Enable SRTP

Add/Update

Sequence Number	SRS IP/FQDN Address	Port	Transport	Trunkgroup ID	Crypto Suite ID
0	2.2.2.2	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0

## Media Encryption

The Secure Real-time Transport Protocol (Secure RTP or SRTP) is an IETF cryptographic protocol used to provide secure communications over untrusted networks as described in RFC 3711. SRTP provides confidentiality, message authentication, and replay protection to Internet media traffic such as audio and video. TheSBC SWe Core supports Secure RTP and its associated secure real-time transport control protocol (Secure RTCP) for IPv4/IPv6 addressing for both audio and video streams.

## Towards Endpoint

To enable sRTP towards endpoints, Crypto suite profiles must be configured in Packet service profiles mapped towards Ingress and Egress Trunks.

**Figure 96:**

Secure RTP/RTCP

Crypto Suite Profile: SIPREC\_CRYPT0

Flags

☒ Allow Fallback ☒ Enable SRTP

☐ Reset ROC On Session Key Change ☐ Reset Enc/Dec/ROC on Decryption Key Change

☐ Update Crypto On Modify ☐ Allow Pass Through

Add crypto suites to the crypto profile and save it.

**Figure 97:**

**Crypto Suite Profile:** SIPREC\_CRYPT0

Description:

**Crypto Suites**

**Sequence:** 0

Crypto Suite: AES CM 128 HMAC SHA1 32

Session Parameter Flags

☐ Unauthenticated SRTP ☐ Unencrypted SRTP

☐ Unencrypted SRTCP

Add/Update

Sequence	Crypto Suite
0	AES CM 128 HMAC SHA1 32

## Towards NICE SIP Recorder

To enable encryption towards SIPRec, Crypto suite profiles are attached to SRS Group Profiles.

Check Enable sRTP check box in SRS Profile and select Crypto Suite Profile from the drop down list.

**Figure 98:**

**SRS Group Profile ID:** SRSPROFILE\_1

Description:

**SRS Group Properties**

Number Of Simultaneous Stream: 1

Load Distribution: ☒ Sequence ☐ RoundRobin

**SRS Server Properties**

**Sequence Number:** 3

**Trunkgroup ID:** SIPREC\_TG4

**Crypto Suite Profile:** SIPREC\_CRYPT0

☒ IPv4 Address: 3 . 3 . 3 . 3 Port V4 Number: 5060

☐ IPv6 Address: 0 : 0 : 0 : 0 : 0 : 0 : 0 : 0 Port V6 Number: 0

☐ Server FQDN: Port Number: 0

SRS Server Transport: ☐ UDP ☐ TCP ☒ TLS

☒ Enable SRTP

Add/Update

Sequence Number	SRS IP/FQDN Address	Port	Transport	Trunkgroup ID	Crypto Suite ID
0	2.2.2.2	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0
1	3.3.3.3	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0
2	2.2.2.2	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0
3	3.3.3.3	5060	TLS	SIPREC_TG4	SIPREC_CRYPT0

Add crypto suites to the crypto profile and attach it to the SRS group profile.

NICE supports two sRTP crypto suites AES\_CM\_128\_HMAC\_SHA1\_80 and AES\_CM\_128\_HMAC\_SHA1\_32.

**Figure 99:**

**Crypto Suite Profile:** SIPREC\_CRYPT0

Description:

**Crypto Suites**

**Sequence:** 0

Crypto Suite: AES CM 128 HMAC SHA1 32

Session Parameter Flags

☐ Unauthenticated SRTP ☐ Unencrypted SRTP

☐ Unencrypted SRTCP

Add/Update

Sequence	Crypto Suite
0	AES CM 128 HMAC SHA1 32

# Ribbon SBC SWe Core High Availability

## Info

During this interop, SBC SWe was configured in HA mode with the below configuration for High Availability.

In an HA configuration, the two SBC VMs are connected to each other using the HA ports on the respective VMs. The HA logical ports must be in the same network and routable using the switch and they must be connected to a switch. Failure of the connection is via link detection and also TIPC keep-alives.

## HA Configuration Link Detection Group

The Link Detection Group allows you to group interfaces and associated Link Monitors together and track link verification failures within the group. A Link Detection Group (LDG) is configured with a unique name and a failover threshold. The LDG tracks the number of link verification failures that have occurred among the Link Monitors configured.

Create Link Detection Groups for both pkt0 and pkt1 interfaces.

```
set addressContext default linkDetectionGroup pkt0_act_ldg ceName SBP00JA1
set addressContext default linkDetectionGroup pkt0_act_ldg type ip
set addressContext default linkDetectionGroup pkt0_act_ldg threshold 1
set addressContext default linkDetectionGroup pkt0_act_ldg state enabled
set addressContext default linkDetectionGroup pkt0_act_ldg linkMonitor pkt0_act_lm interfaceGroup IG1
set addressContext default linkDetectionGroup pkt0_act_ldg linkMonitor pkt0_act_lm interface IF1
set addressContext default linkDetectionGroup pkt0_act_ldg linkMonitor pkt0_act_lm destination
<pkt0_default_gateway>
set addressContext default linkDetectionGroup pkt0_act_ldg linkMonitor pkt0_act_lm state enabled
set addressContext default linkDetectionGroup pkt0_stb_ldg ceName SBP00JA2
set addressContext default linkDetectionGroup pkt0_stb_ldg type ip
set addressContext default linkDetectionGroup pkt0_stb_ldg threshold 1
set addressContext default linkDetectionGroup pkt0_stb_ldg state enabled
set addressContext default linkDetectionGroup pkt0_stb_ldg linkMonitor pkt0_stb_lm interfaceGroup IG1
set addressContext default linkDetectionGroup pkt0_stb_ldg linkMonitor pkt0_stb_lm interface IF1
set addressContext default linkDetectionGroup pkt0_stb_ldg linkMonitor pkt0_stb_lm destination
<pkt0_default_gateway>
set addressContext default linkDetectionGroup pkt0_stb_ldg linkMonitor pkt0_stb_lm state enabled
set addressContext default linkDetectionGroup pkt1_act_ldg ceName SBP00JA1
set addressContext default linkDetectionGroup pkt1_act_ldg type ip
set addressContext default linkDetectionGroup pkt1_act_ldg threshold 1
set addressContext default linkDetectionGroup pkt1_act_ldg state enabled
set addressContext default linkDetectionGroup pkt1_act_ldg linkMonitor pkt1_act_lm interfaceGroup IG2
set addressContext default linkDetectionGroup pkt1_act_ldg linkMonitor pkt1_act_lm interface IF2
set addressContext default linkDetectionGroup pkt1_act_ldg linkMonitor pkt1_act_lm destination
<pkt1_default_gateway>
set addressContext default linkDetectionGroup pkt1_act_ldg linkMonitor pkt1_act_lm state enabled
set addressContext default linkDetectionGroup pkt1_stb_ldg ceName SBP00JA2
set addressContext default linkDetectionGroup pkt1_stb_ldg type ip
set addressContext default linkDetectionGroup pkt1_stb_ldg threshold 1
set addressContext default linkDetectionGroup pkt1_stb_ldg state enabled
set addressContext default linkDetectionGroup pkt1_stb_ldg linkMonitor pkt1_stb_lm interfaceGroup IG2
set addressContext default linkDetectionGroup pkt1_stb_ldg linkMonitor pkt1_stb_lm interface IF2
set addressContext default linkDetectionGroup pkt1_stb_ldg linkMonitor pkt1_stb_lm destination
<pkt1_default_gateway>
set addressContext default linkDetectionGroup pkt1_stb_ldg linkMonitor pkt1_stb_lm state enabled
comm
```

## NICE Configuration

For detailed NICE configurations, please visit official NICE support page <http://www.extranice.com/>.

As a part of this document, we have highlighted specific NICE configuration changes that were used in our testing.



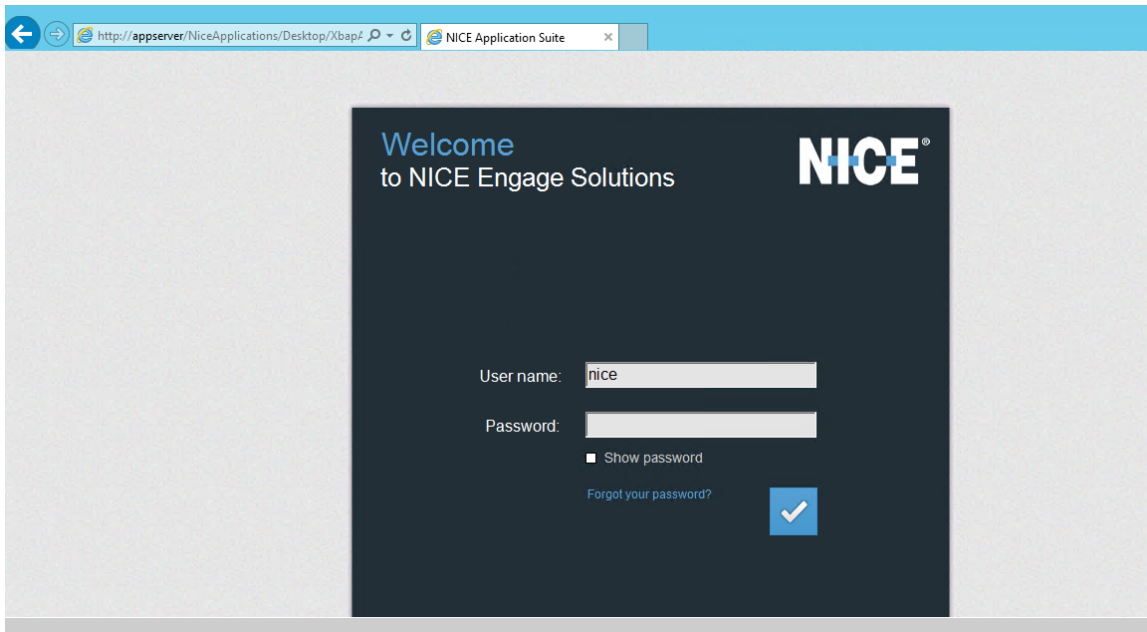
Please note that the configurations mentioned below were used in our lab environment for testing purposes. Each customer may have unique needs and configurations. Ribbon recommends that customers work with NICE engineers for NICE configurations to best meet their requirements.

## Application Server



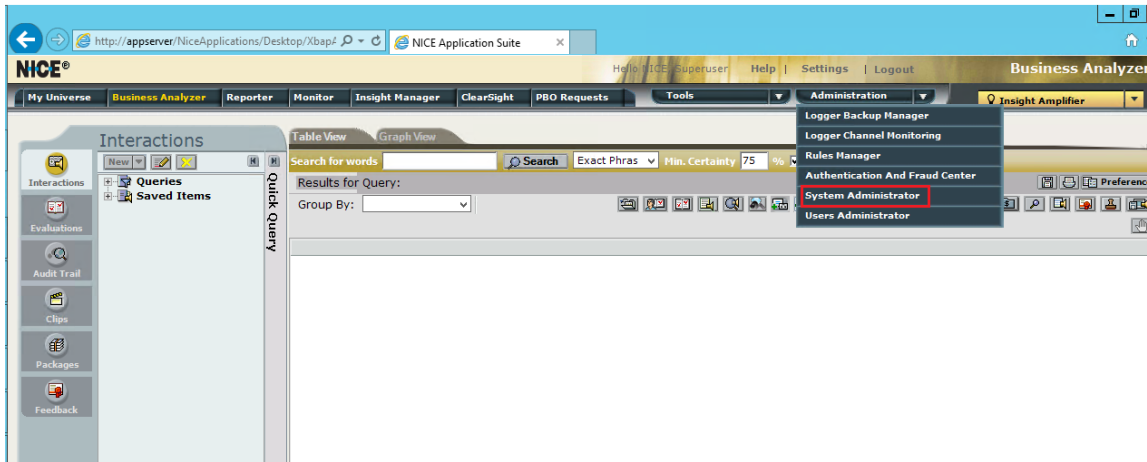
Use the below login page to access NICE application server for all the NICE configurations.

**Figure 100:**



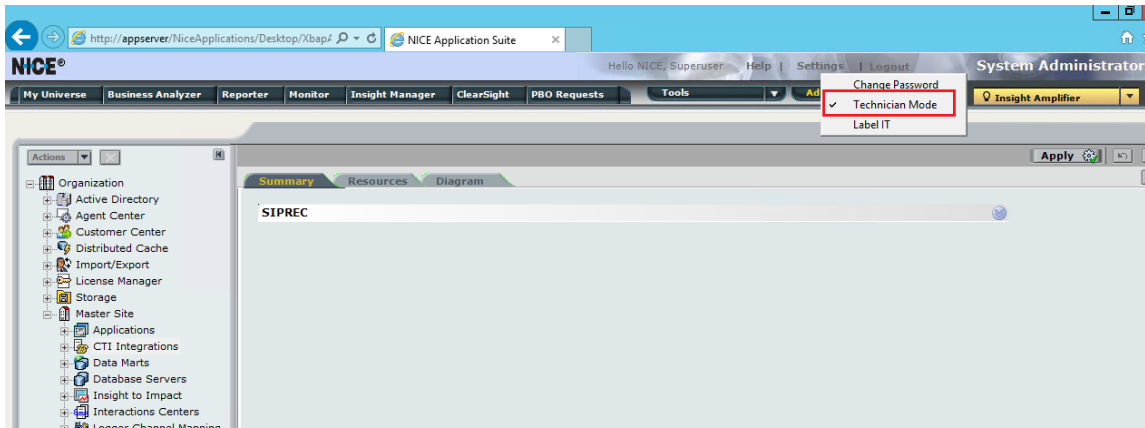
After login, Select System Administrator from the dropdown as mentioned below to check NICE configurations.

**Figure 101:**



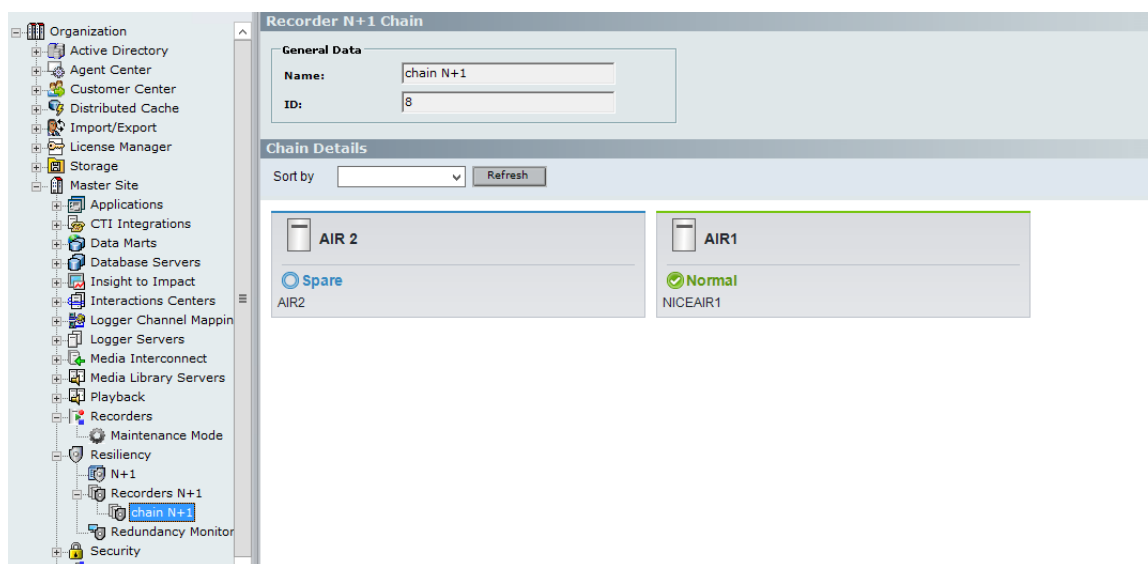
Once logged in as System Administrator, Check Enable Technician Mode from the drop down as mentioned below to edit any configurations.

**Figure 102:**



To check Active and Standby AIR servers, go to Master Site > Resiliency > Recorders N+1 > chain N+1.

**Figure 103:**



## Metadata Support

### Support for Metadata type 'sonus'

When siprecmetadata profile is not configured, by default theSBC supports backward compatibility and pre-defined metadata for passing proprietary call specific information from the SRC to the SRS.

In order to configure NICE server to support default Ribbon SBC configurations, Go to:

- Master Site > CTI Integrations > Media Provider Controllers >Additional Media Provider Controller Parameters >MetadataType > sonus
- Click Save
- In the CTI Integrations branch, click Apply



- Please repeat the above steps for both the VRSP servers.
- Restart NICE Integration Dispatch Service on both the VRSP servers.

### Support for Metadata type 'RFC7865'

When siprecmetadata profile version is set to 1, Ribbon SBC supports RFC 7865.

In order to configure NICE server to support RFC7865, Go to:

- Master Site > CTI Integrations > Media Provider Controllers >Additional Media Provider Controller Parameters >MetadataType > RFC7865
- Click Save
- In the CTI Integrations branch, click Apply



- Please repeat the above steps for both the VRSP servers.
- Restart NICE Integration Dispatch Service on both the VRSP servers.

Figure 104:

**Media Provider Controller General Information**

**Media Provider Controller Type**

**General Details**

**Attach Connection Manager**

**Additional Media Provider Controller Parameters**

☐ Display Read Only Information Mandatory fields are marked with an asterisk (\*)

Parameter Name	Parameter Value
VRSP Version	Ver_2
Unit Assembly	Integrations.N
Metadata Type	sonus
SipRefreshMethod	Update
AdApiPort	41042
FailoverReinviteDelay	0

Description: Metadata type to set the data translator type.

**Media Provider Controller Reporting Level**

**Set Parameter Value**

**MPC Additional Parameter**

**Set Parameter Value**

Name: Metadata Type

Value: **sonus**

acme  
audiocodes  
base7  
Draft1  
Draft15  
Draft17  
sonus  
RFC7865

## VRSP NoFailback mode

In order to change configuration at NICE server, Go to:

- Master Site > CTI integration > Media Provider Controller tab > VRSP[A/S]CTI integration > Media Provider Controller tab > VRSP[A/S] > RunningMode > NOFAILBACK
- Click Save
- In the CTI Integrations branch, click Apply



- Please repeat the above steps for both the VRSP servers.
- Restart NICE Integration Dispatch Service on both the VRSP servers.

Figure 105:

RedundancyIsEnabled	Yes
SrvPosition	Primary
RunningMode	NOFAILBACK
RedundancyRemoteIpAddress	172.16.106.221
RedundancyRemotePort	50501

Figure 106:

RedundancyIsEnabled	Yes
SrvPosition	Secondary
RunningMode	NOFAILBACK
RedundancyRemoteIpAddress	172.16.106.223
RedundancyRemotePort	50501

## Transport Configurations

### VRSP configurations

For UDP/TCP, Go to:

- Master Site > CTI Integrations > Media Provider Controllers > Additional Media Provider Controller Parameters > SipStackTlsEnabled > NO
- Click Save
- In the CTI Integrations branch, click Apply

For TLS, Go to:

- Master Site > CTI Integrations > Media Provider Controllers > Additional Media Provider Controller Parameters > SipStackTlsEnabled > YES
- Master Site > CTI Integrations > Media Provider Controllers > Additional Media Provider Controller Parameters > SipStackTlsCertificateSerialNumber > serial number of the NICE VRSP certificate
- Click Save
- In the CTI Integrations branch, click Apply



- Please repeat the above steps for both the VRSP servers.

Figure 107:

Parameter Name	Parameter Value
SessionTimerSessionExpires	1800
SipStackSubscribeExpires	1800
MemoryNumberOfPages	3000
MemoryPageSize	1024
DataCenterLocation	Default
<b>SipStackTlsEnabled</b>	<b>No</b>
SipStackTlsPort	5061
SipStackTlsIpAddress	

Description: Enable TLS connection.

Figure 108:

Parameter Name	Parameter Value
DataCenterLocation	Default
<b>SipStackTlsEnabled</b>	<b>Yes</b>
SipStackTlsPort	5061
SipStackTlsIpAddress	
SipStackTlsCertificateSerialNumber	00 f7 c2 b7 be 42 94 56 0d
SipStackTlsCertificateStoreLocation	LocalMachine
SipStackTlsCertificateExpirationDate	Offline

Description:

## AIR configurations for UDP with RTP

In order to change transport settings at NICE server, Go to:

- Master Site > Recorders -> AIR[A/S] > Advanced tab > IP Capture > SIP transport mode > UDP
- Master Site > Recorders -> AIR[A/S] > Advanced tab > IP Capture > SRTP enabled > False
- Click Save

- In the CTI Integrations branch, click Apply



- Please repeat the above steps for both the AIR servers.

Figure 109:

Parameter Name	Value
SIP support for re-invite messages	True
SIP timer mode	1
SIP transport mode	UDP
SRTP enabled	False
Summation wait time (milliseconds)	1000
Support Late Packet Arrival	False

## AIR configurations for TLS with sRTP

In order to change configuration at NICE server, Go to:

- Master Site > Recorders > AIR[A/S] > Advanced tab > IP Capture > SIP transport mode > TLS
- Master Site > Recorders > AIR[A/S] > Advanced tab > IP Capture > SRTP enabled > True
- Master Site > Recorders > AIR[A/S] > Advanced tab > IP Capture > Certificate serial > serial number of the NICE AIR certificate
- Click Save
- In the CTI Integrations branch, click Apply



Please repeat the above steps for both AIR servers.

Figure 110:

Parameter Name	Value
SIP stack port	5064
SIP support for re-invite messages	True
SIP timer mode	1
SIP transport mode	TLS
SRTP enabled	True
Summation wait time (milliseconds)	1000

Figure 111:

Parameter Name	Value
AAC LATM dynamic payload types	
AAC-LD dynamic payload types	
Audio file cache size	4096
Certificate serial	b6 03 06 3b c7 71 55 87 40 a1 12 3c 49 f8 ...
Default Target Compression	G729
Dialer Session Duration - Total Recording (...)	300



For Transport changes to be effective:

- Restart NICE Integration Dispatch Service on both the VRSP servers.
- Restart NICE Interactions Center RCM service on Interactions Center server.
- Restart NICE IP Capture and NICE Recorder Administrator services on both AIR servers.

## Sequential Forking

In order to change configuration at NICE server, Go to:

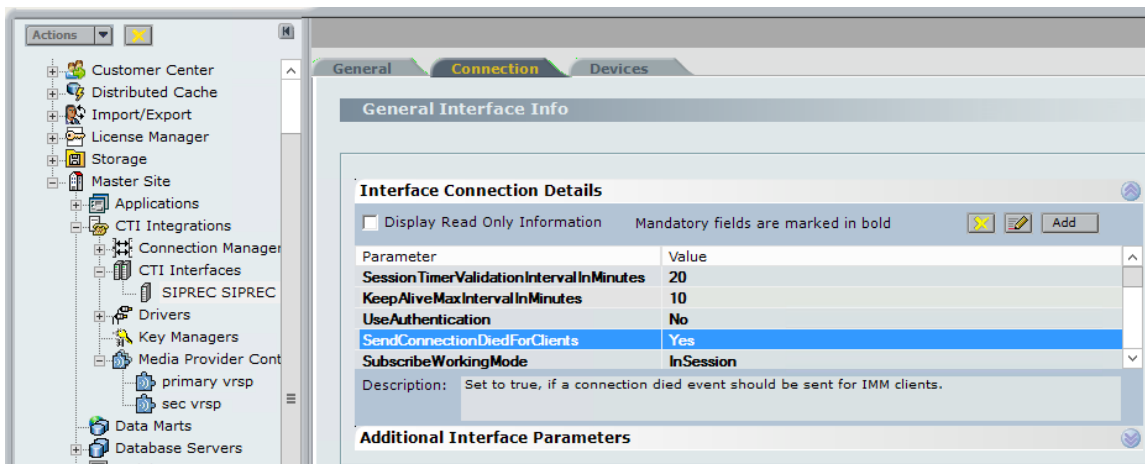
- Master Site > CTI Integrations > Media Provider Controller tab > VRSP[A/S] > Additional Media Provider Controller Parameters > RunningMode > NoFailback
- Master Site > CTI Integrations > CTI Interfaces > Connection > Interface Connection Details > SendConnectionDiedForClients > Yes
- Click Save
- In the CTI Integrations branch, click Apply



For Transport changes to be effective:

- Restart NICE Integration Dispatch Service on both the VRSP servers.
- Restart NICE IP Capture and NICE Recorder Administrator services on both AIR servers.

Figure 112:



## Parallel Forking

In order to change configurations at NICE server, Go to:

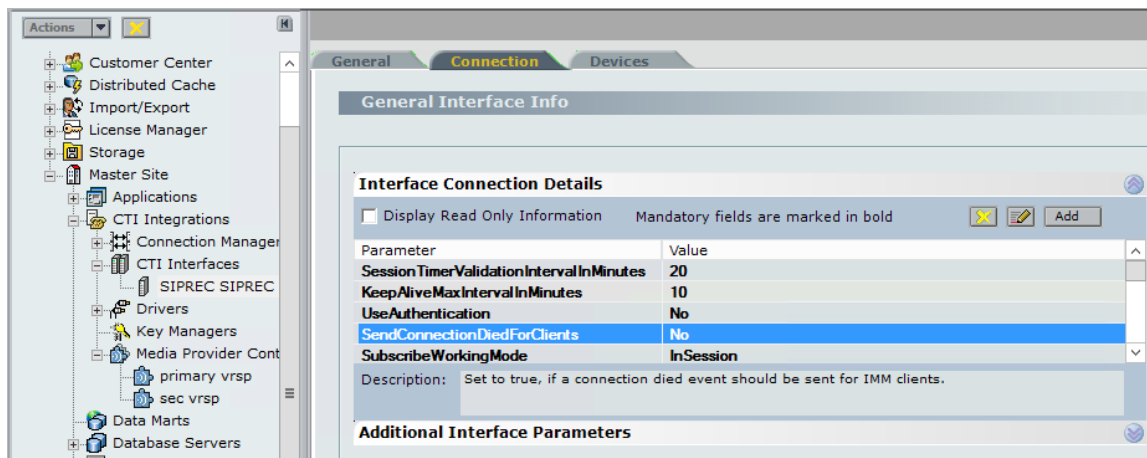
- Master Site > CTI Integrations > Media Provider Controller tab > VRSP[A/S] > Additional Media Provider Controller Parameters > RunningMode > NoFailback
- Master Site > CTI Integrations > CTI Interfaces > Connection > Interface Connection Details > SendConnectionDiedForClients > No
- Master Site > Integration Centers > IC\_Server > Configuration > Call Server > op\_MaxOpenCallDuration / op\_MaxOpenCompoundCallDuration > Desired timeout for long call duration.
- Click Save
- In the CTI Integrations branch, click Apply



For Transport changes to be effective:

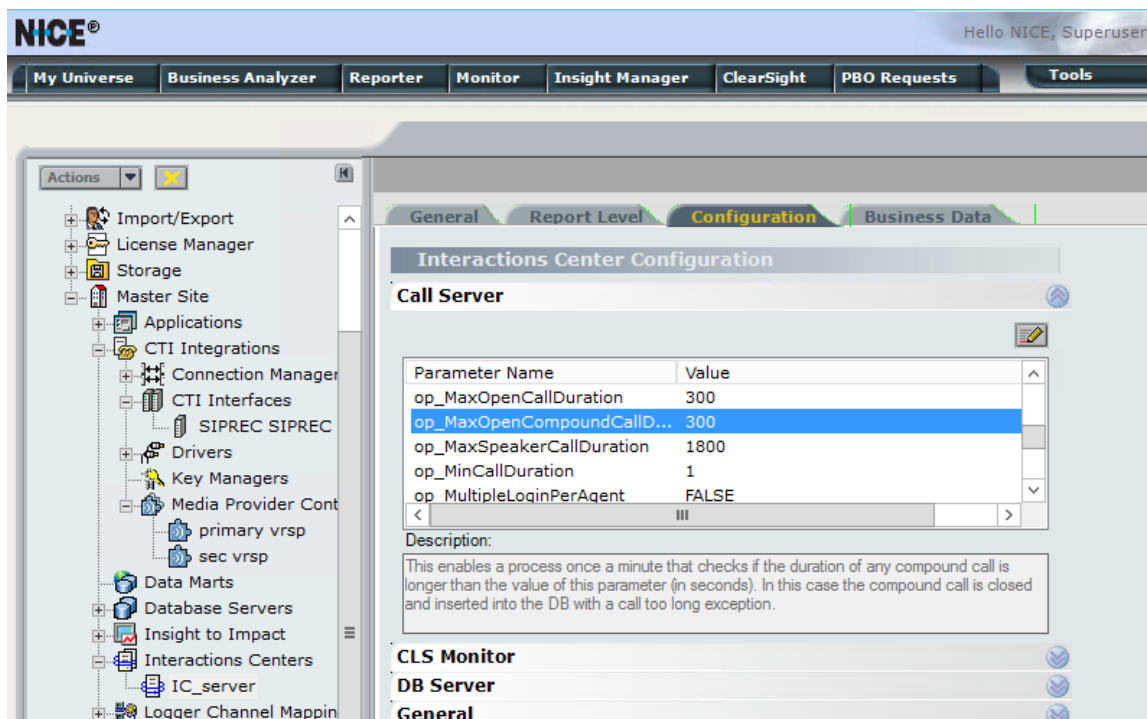
- Restart NICE Integration Dispatch Service on both the VRSP servers.
- Restart NICE Interactions Center Core and NICE Interactions Center RCM services on Interactions Center server.
- Restart NICE IP Capture and NICE Recorder Administrator services on both AIR servers.

Figure 113:



**i** Please note the timeouts captured in the snapshots were configured solely for the purpose of testing. Please tune this timeout as per specific business needs.

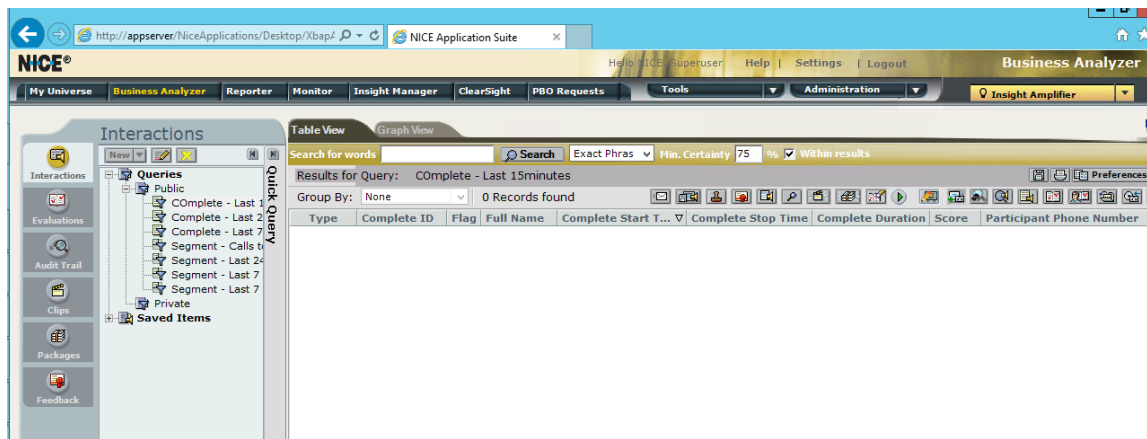
Figure 114:



## NICE Business Analyzer

Use NICE Business Analyzer to view/query/listen to recordings created.

Figure 115:



## Supplementary Services & Features Coverage

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

Sr. No	Supplementary Services/ Features	Coverage
1	Basic Call Setup & Termination	✓
2	Call Recording via CLI	✓
3	DTMF - RFC2833/ Inband	✓
4	DTMF - SIP INFO	✗
5	Call Hold/ Resume	✓
6	Call Transfer (Blind/ Unattended)	✓
7	Call Transfer (Attended)	✓
8	Session Refresh	✓
9	Call Forward No Answer	✓
10	Conference	✓
11	Transcoding	✓
12	Music On Hold	✓
13	TLS with SRTP	✓
14	SIPRec Call Forking	✓
15	Quad Recording	✓
16	HA SBC switchover	✓
17	SRS Redundancy - Sequential Forking	✓
18	SRS Redundancy - Parallel Forking	✓

### Legend

Supported	✓
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## Caveats

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### Ribbon:

- SIPRec leg goes to Inactive state after call transfer with REFER processed on SBC while recording type is set to either "Egress" or "Ingress."
- SBC sends two different session\_id's for single call towards Active and Standby NICE SRS servers. During NICE VRSP failover scenarios, NICE recorder is unable to map the two sessions to a single interaction. As a workaround to avoid any recording loss, at NICE, we configure op\_MaxOpenCompoundCallDuration/"op\_MaxOpenCallDuration". NICE will push open interactions handled by failed SRS server to NBA as a new file after this configured timeout [default 5 hours].

### Nice:

- Upon NICE VRSP failover, it may take up to three minutes (default) for AIR to refresh the session and retrieve keys from secondary VRSP. This may result in failure to decrypt and open any new calls for up to three minutes ("white noise" + exception on the interaction).

## Support

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For any support related queries about this guide, please contact your local Ribbon representative, or use the details below:

- Sales and Support: 1-833-742-2661
- Other Queries: 1-877-412-8867
- Website: <https://ribboncommunications.com/services/ribbon-support-portal>

## References

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For detailed information about Ribbon products & solutions, please visit: <https://ribboncommunications.com/products>

## Conclusion

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This Interoperability Guide describes successful configuration of Ribbon SBC SWe Core& PSX with NICE SIP Recorder.

All features and capabilities tested are detailed within this document - any limitations, notes or observations are also recorded in order to provide the reader with an accurate understanding of what has been covered, and what has not.

Configuration guidance is provided to enable the reader to replicate the same base setup - there may be additional configuration changes required to suit the exact deployment environment.