

Ribbon SBC Edge R9.0 SIP Interface Interop with Identity Hub : Interoperability Guide



Table of Contents

- Interoperable Vendors
- Copyright
- Document Overview
 - About Ribbon SBC Edge
 - About SIP Interface
 - About Identity Hub Reputation Scoring Process
- Scope
- Non-Goals
- Audience
- Prerequisites
- Product and Device Details
- Network Topology
 - Deployment Topology
 - IOT Lab Topology
 - Signaling and Media Flow
- Document Workflow
- Section A : Ribbon SBC Edge Configuration
 - Installing Ribbon SBC Edge on VMware
 - Accessing Ribbon SBC Edge
 - Networking Interfaces Information
 - Configure Static Routes
 - License and TLS Certificates
 - View License
 - Import Trusted Root CA Certificates
 - Ribbon SBC Edge Configurations for PSTN, IDH & Enterprise side
 - Modify Media List
 - SIP Profile
 - TLS Profile
 - SIP Server Table
 - Signaling Group
 - Outbound Signaling Group towards IDH
 - Cause Code Reroutes
 - Transformation Table
 - Create Transformation Table
 - Create Transformation Table for PSTN to Enterprise
 - Create Transformation Table for Enterprise to PSTN
 - Create Transformation Table for Emergency Call
 - Call Routing Table
 - Create Call Routing Table for PSTN to Enterprise
 - Create Call Routing Table for Enterprise to PSTN
 - Emergency
- Section B : Ribbon Identity Hub Configuration
 - Identity Hub Reputation Scoring Process
 - Possible SIP Responses
- Supplementary Services and Features Coverage
- Caveats
- Support
- References
- Conclusion

Interoperable Vendors



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Document Overview

This document depicts the configuration details for Ribbon's SBC Edge interworking & compliance with IDH SIP Interface.

About Ribbon SBC Edge

The Ribbon Session Border Controller Edge provides best-in-class communications security. The SBC Edge dramatically simplifies the deployment of robust communications security services for SIP Trunking, Direct Routing, and Cloud UC services.

About SIP Interface

The Client SIP Node (SBC Edge) needs specific configuration to be able to access SIP interface towards the Identity Hub for Reputation scoring service.

About Identity Hub Reputation Scoring Process

The SIP client initiates a request by routing the SIP INVITE request for a call to the Identity Hub service FQDN. The INVITE request must include a FROM header to identify the calling party and a To header to identify the called party. Based on workflow selector and workflow configuration, Identity Hub selects the service graph and begins reputation scoring.

Scope

This document provides configuration best practices for deploying Ribbon SBC Edge when connecting with Identity Hub SIP Interface. 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.

Non-Goals

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

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

Prerequisites

The following aspects are required before proceeding with the interop:

- Ribbon SBC Edge on VMware
- Ribbon SBC Edge license
 - This interop requires the acquisition and application of SIP sessions, as documented in [Working with Licenses](#).
- Public IP Addresses and Port to communicate with Identity Hub Solution
- TLS certificates for SBC Edge
 - For more details, please visit: [Working with Certificates](#).

Product and Device Details

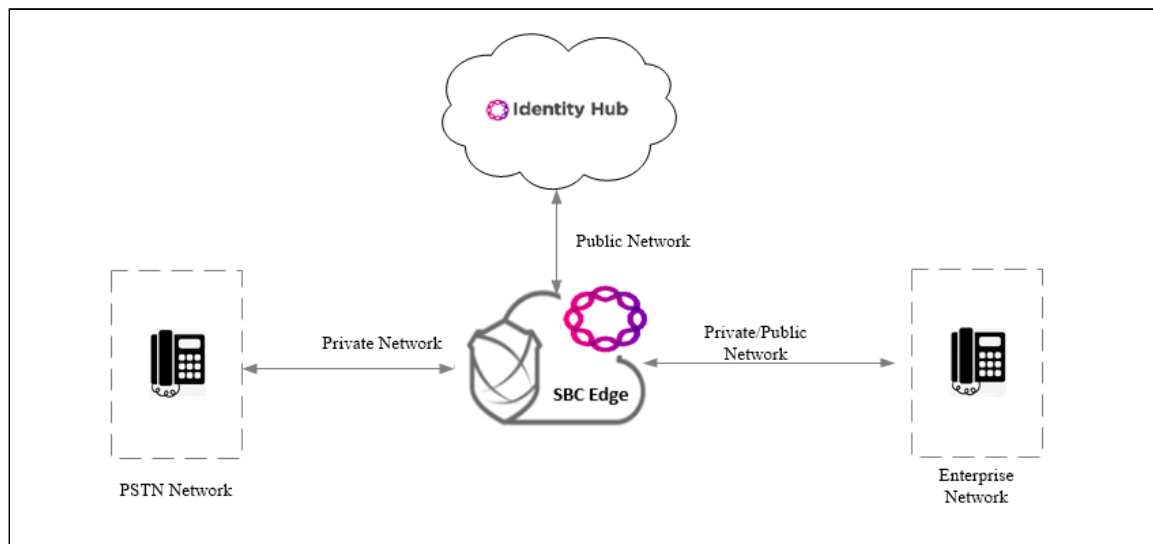
The configuration uses the following equipment and software:

Table 1: Requirements

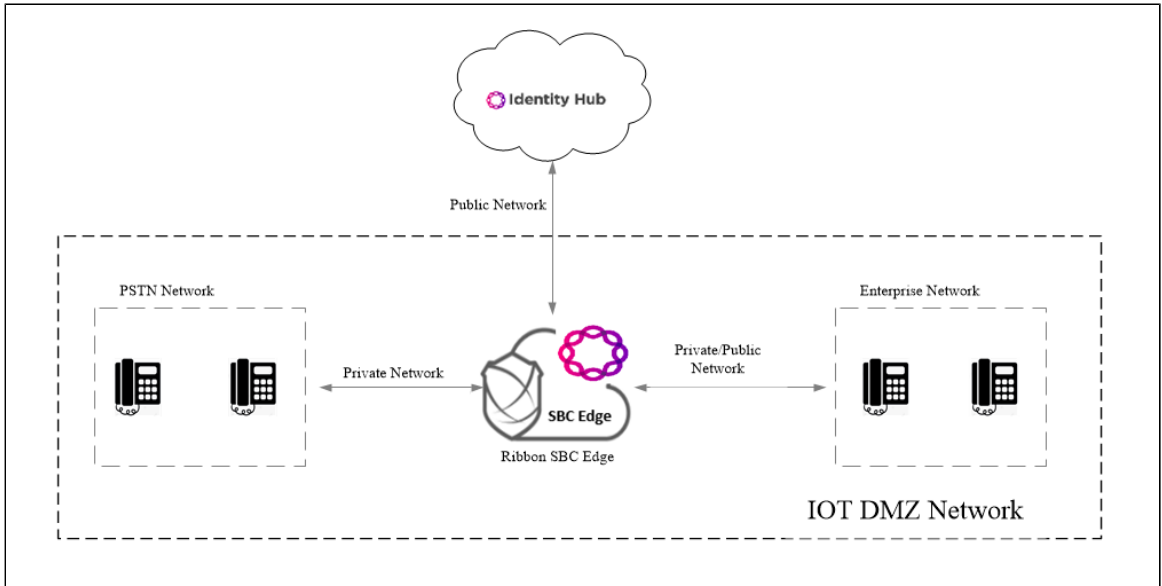
Product	Equipment/ Devices	Software/ Firmware Version
Ribbon Communications	Ribbon SBC Edge	9.0.9
	Identity Hub	23.01.00
Third Party Phones	Zoiper Softphone	5.5.8
	Polycom	5.9.1.0615
	Phoner Lite	2.96
Administration and Debugging Tools	Wireshark	3.2.7
	LX Tool	2.1.0.6

Network Topology

Deployment Topology

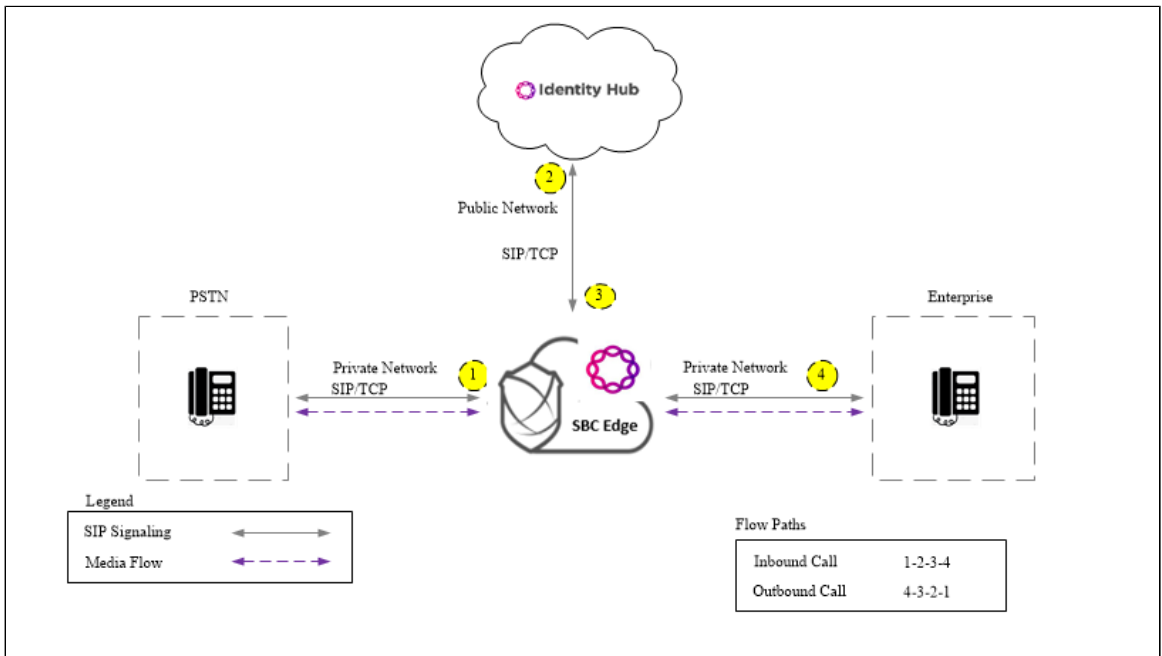


IOT Lab Topology

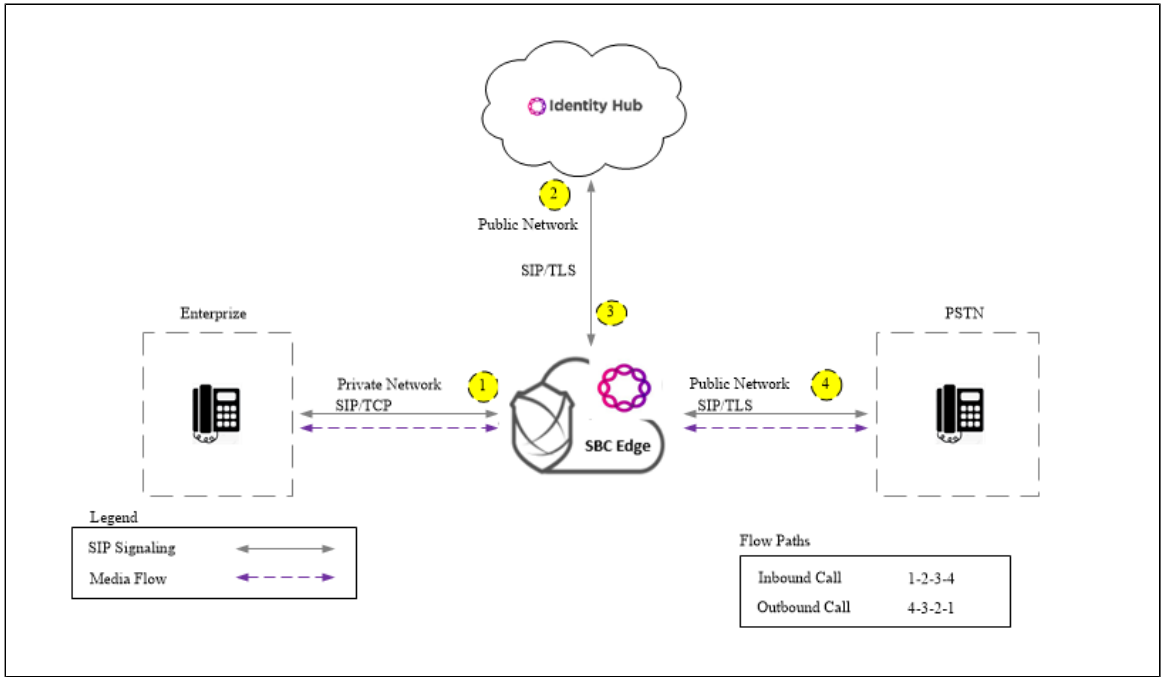


Signaling and Media Flow

TCP Call Flow

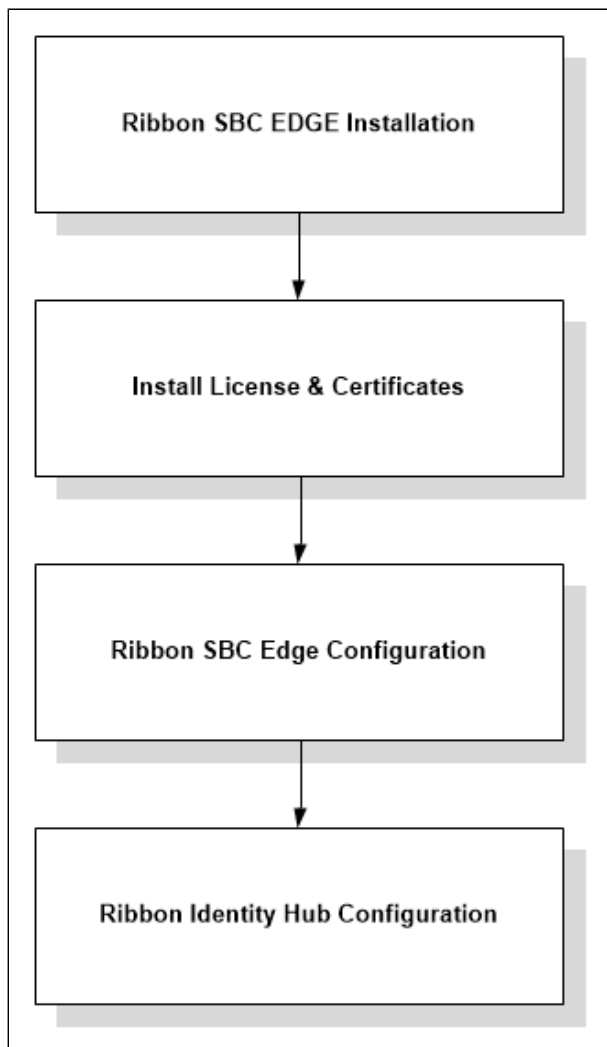


TCP-TLS Call Flow



Document Workflow

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



Section A: Ribbon SBC Edge Configuration

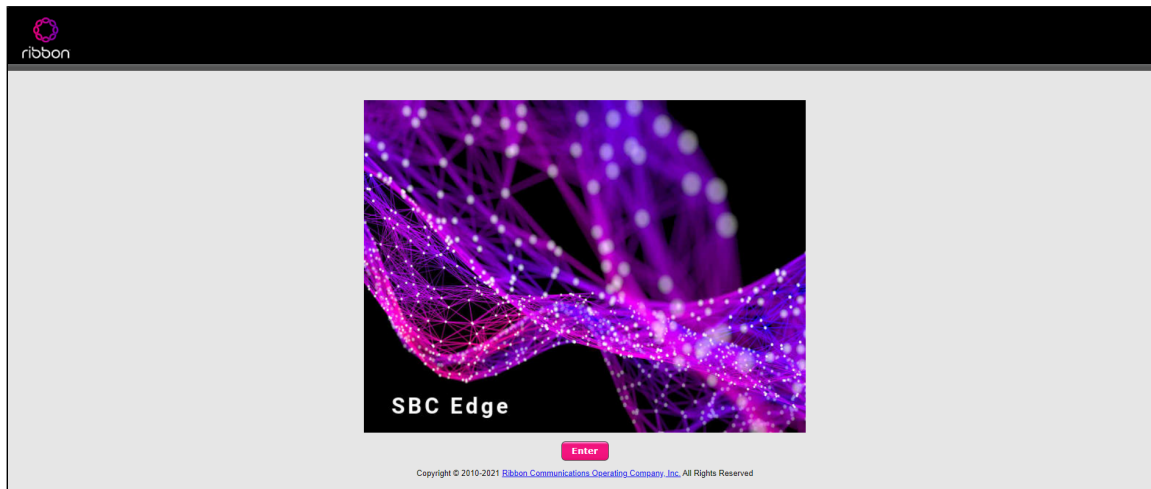
Installing Ribbon SBC Edge on VMware

The SBC Edge is available for deployment on VMware. To deploy an SBC Edge instance, refer to: <https://doc.rbn.com/display/UXDOC90/Installing+SBC+SWe+Lite+on+VMware+ESXi>.

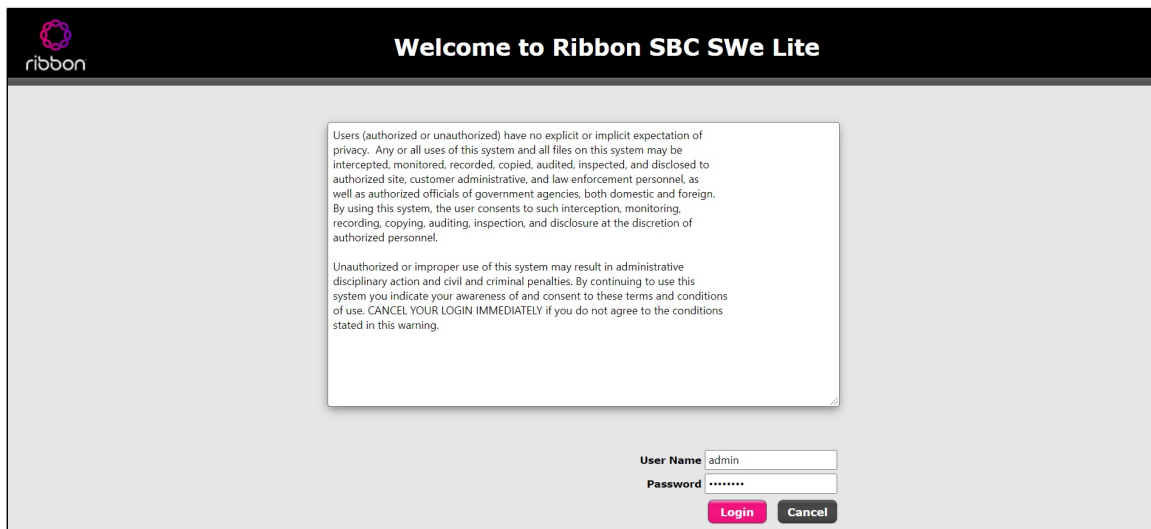
Once the Edge instance is successfully created on VMware, retrieve the allocated NAT Public IPs, Ethernet IPs, and the Management IP. For more details, visit the link given in the References section.

Accessing Ribbon SBC Edge

1. Open any browser and enter the SBC IP address.



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



Networking Interfaces Information

The Ribbon SBC Edge supports five system-created logical interfaces (known as **Administrative IP**, **Ethernet 1 IP**, **Ethernet 2 IP**, **Ethernet 3 IP**, and **Ethernet 4 IP**). In addition to the system-created logical interfaces, the Ribbon SBC Edge supports user-created VLAN logical sub-interfaces.

Ethernet 1 IP and Ethernet 2 IP are used for this interop.

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

Interface Name	IPv4 Address	IPv6 Address	Description	Admin State
Admin IP				Enabled
Ethernet 1 IP	115.100.100.100			Enabled
Ethernet 2 IP	172.30.100.100			Enabled

Ethernet 1 IP

Configure this Ethernet 2 interface as follows as per the requirement. This interface will face the IDH SIP Interface Solution and the Public Network.

The screenshot displays a network configuration page for 'Ethernet 1 IP'. On the left is a navigation tree with categories like Call Routing, Signaling Groups, Networking Interfaces, System, and Security. The main area is titled 'Interface Name' and 'IPv4 Address'. Below this, there are two main sections: 'Identification/Status' and 'Networking'. The 'Identification/Status' section contains fields for 'Interface Name' (Ethernet 1 IP), 'I/F Index' (8), 'Alias' (empty), 'Description' (empty), and 'Admin State' (Enabled). The 'Networking' section contains 'MAC Address' (redacted), 'IP Addressing Mode' (IPv4), and an 'IPv4 Information' section. The 'IPv4 Information' section includes 'IP Assign Method' (Static), 'Primary Address' (115.), 'Primary Netmask' (255.255.255.192), and 'Media Next Hop IP' (115.). An 'Apply' button is at the bottom right.

Ethernet 2 IP

Ethernet 1 IP is assigned an IP address used for transporting all the VOIP media packets (for example, RTP, SRTP) and all the protocol packets (for example, SIP, RTCP, TLS). In the default software, **Ethernet 1 IP** is enabled, and an IPv4 address is acquired through a connected DHCP server, or you can assign a static IP as well.

Configure Static Routes

Static routes are used to create communication to remote networks. In a production environment, static routes are mainly configured for routing from a specific network to another network that you can only access through one point or one interface (single path access or default route).

Destination IP	Specifies the destination IP address.
Mask	Specifies the network mask of the destination host or subnet. If the 'Destination IP Address' field and 'Mask' field are both 0.0.0.0, the static route is called the 'default static route'.
Gateway	Specifies the IP address of the next-hop router to use for this static route.
Metric	Specifies the cost of this route, and therefore, indirectly specifies the preference of the route. Lower values indicate more preferred routes. The typical value is 1 for most static routes, indicating that static routes are preferred to dynamic routes.

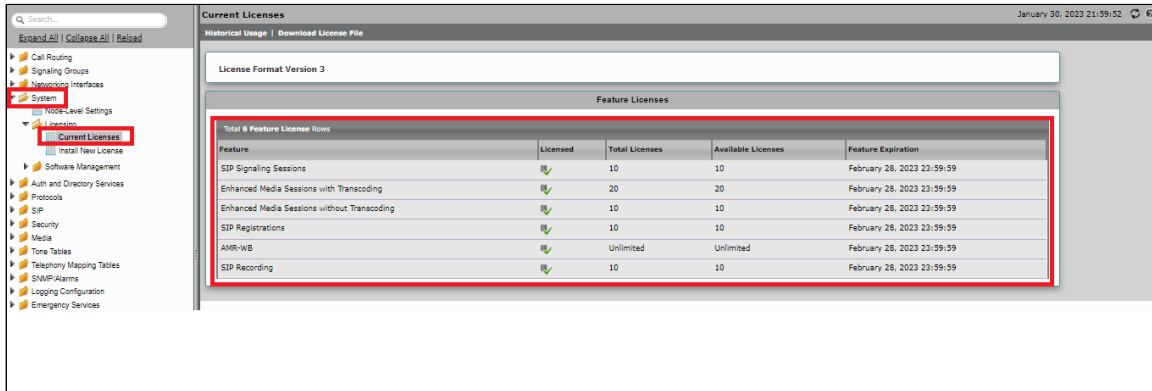
Row ID	Destination IP	Mask	Gateway	Administrative Distance	Primary Key
1	0.0.0.0	0.0.0.0	11	1	1
2	172.x.x.x	255.x.x.x	172.x.x.x	1	2
3	10.x.x.x	255.x.x.x	172.x.x.x	1	3

License and TLS Certificates

View License

This section describes how to view the status of each license along with a copy of the license keys installed on your SBC. The **Feature Licenses** panel enables you to verify whether a feature is licensed, along with the number of remaining licenses available for a given feature at run-time.

From the **Settingstab**, navigate to **System > Licensing > Current Licenses**.



Feature	Licensed	Total Licenses	Available Licenses	Feature Expiration
SIP Signaling Sessions	✔	10	10	February 28, 2023 23:59:59
Enhanced Media Sessions with Transcoding	✔	20	20	February 28, 2023 23:59:59
Enhanced Media Sessions without Transcoding	✔	10	10	February 28, 2023 23:59:59
SIP Registrations	✔	10	10	February 28, 2023 23:59:59
AMR-WB	✔	Unlimited	Unlimited	February 28, 2023 23:59:59
SIP Recording	✔	10	10	February 28, 2023 23:59:59

Import Trusted Root CA Certificates


A Trusted CA Certificate is a certificate issued by a trusted certificate authority. Trusted CA Certificates are imported to the SBC Edgeto establish its authenticity on the network.

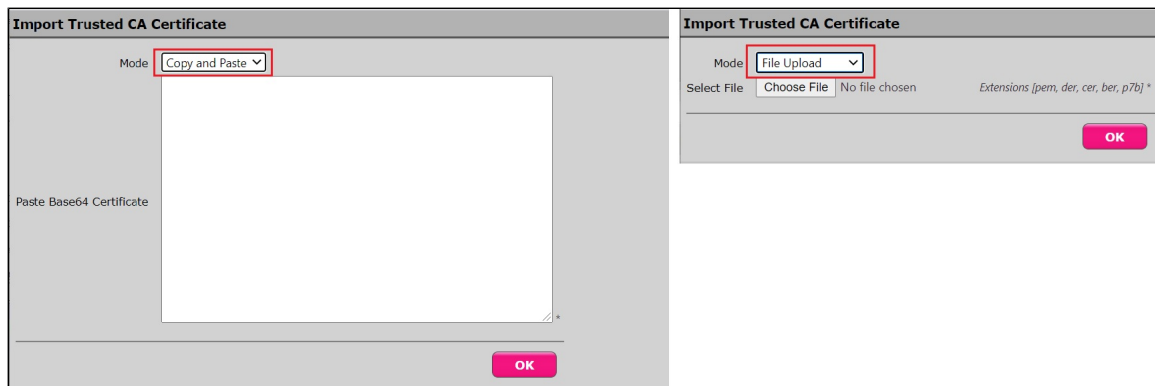
From the **Settingstab**, navigate to **Security > SBC Certificates > Trusted CA Certificates**.



- Generate SBC Edge CSR
- SBC Primary Certificate
- SBC Supplementary Certificates
- Trusted CA Certificates

This section describes the process of importing Trusted Root CA Certificates, using either the **File Uploader** or **Copy and Paste** methods.

1. To import a Trusted CA Certificate, click the Import Trusted CA Certificate () icon.
2. Select either **Copy and Paste** or **File Upload** from the **Mode** menu.
3. If you choose **File Upload**, use the **Select File** button to find the file.
4. Click **OK**.



Follow the steps above to import certificates for this interop.

Note When the **Verify Status** field in the Certificate panel indicates Expired or Expiring Soon, replace the Trusted CA Certificate. You must delete the old certificate before importing a new certificate successfully.

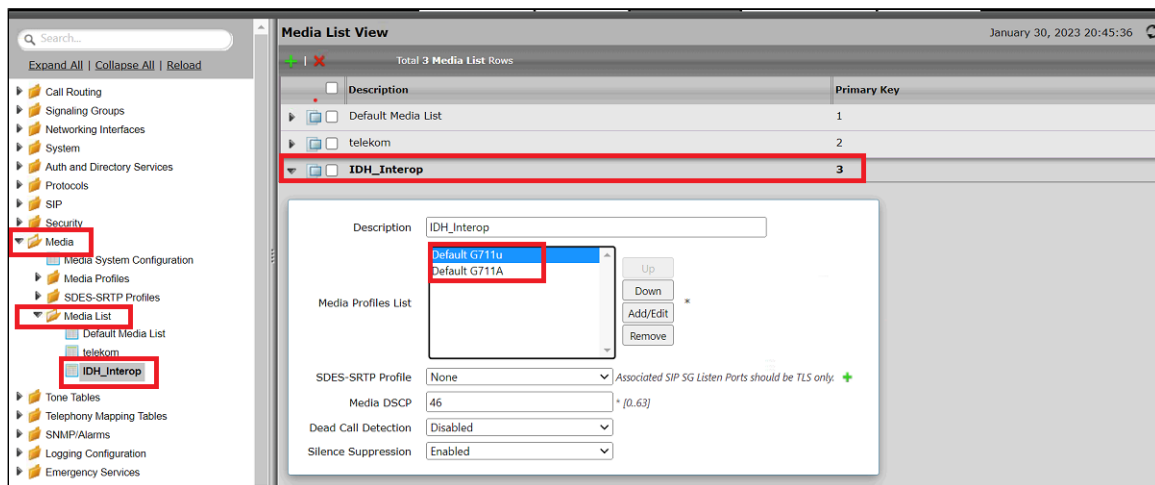
Ribbon SBC Edge Configurations for PSTN, IDH & Enterprise side

This configuration uses web-based Easy Configuration Wizard for quick start provisioning towards the PSTN/carrier network as well as towards the Microsoft Teams cloud.

Modify Media List

Modify the media list that was created using the Easy Configuration Wizard.

Select **Settings > Media > Media List > IDH Interop**



SIP Profile

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

1. Select **Settings > SIP > SIP Profiles**.
2. Create a new SIP profile with desired name "**IDH: SIP Interface**" and with the session timer enabled. The Minimum Acceptable Timer is 600, and the Offered Session Timer is 3600.
3. Configure the required fields as shown in the screenshot below.

TLS Profile

The TLS profile defines the crypto parameters for the SIP protocol.

1. Select **Settings > Security > TLS Profile**.
2. Click the **+** icon to create a new TLS profile.
3. Create a profile with desired name "**IDH_TLS_Certificate**" and Set TLS Protocol to **TLS 1.2 Only**.
4. Select the correct valid public certificate from the drop down menu and finally, ensure **Validate Client FQDN** is Enabled/Disabled.

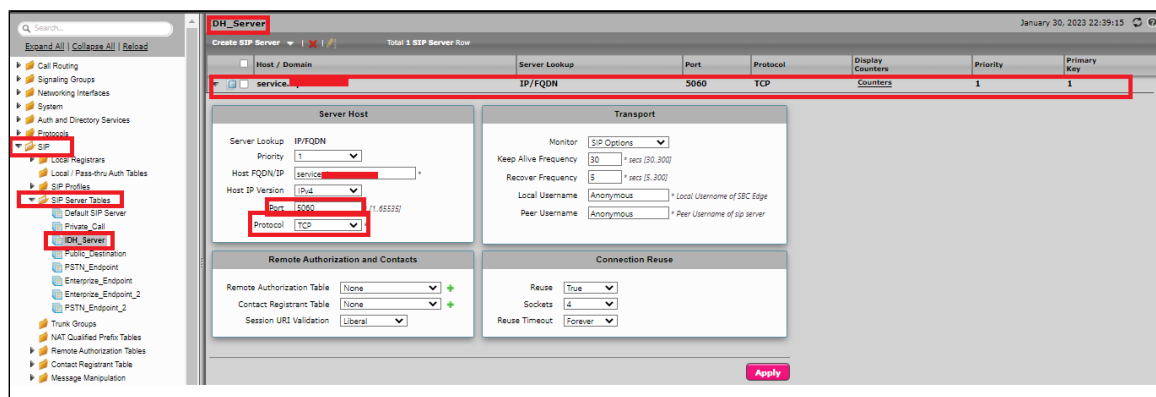
SIP Server Table

Select **Settings > SIP > SIP Server Tables**

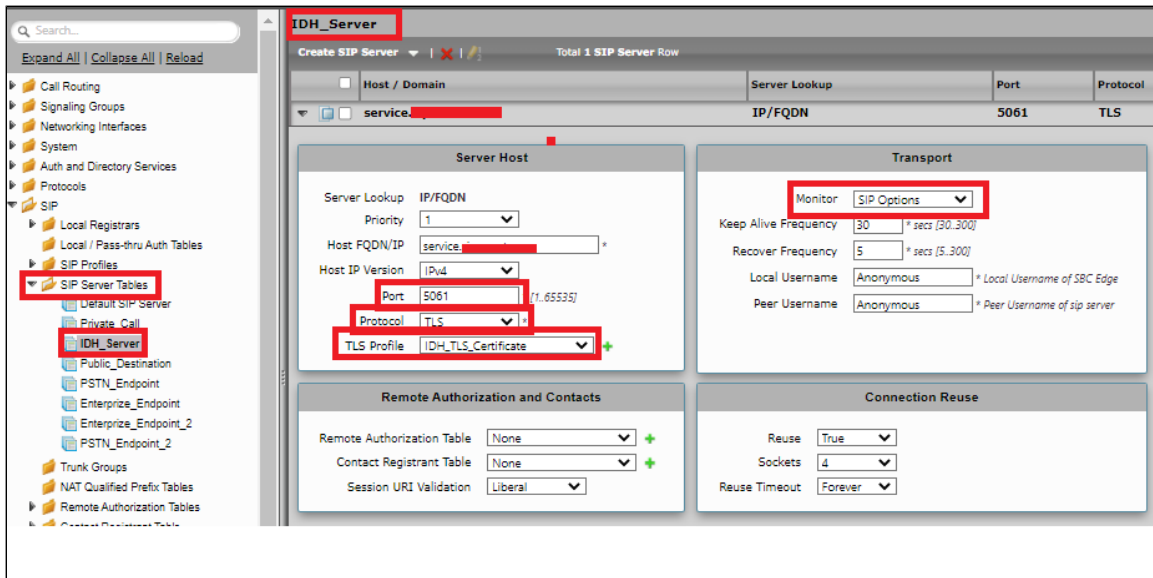
SIP Server Tables contain information about the SIP devices connected to the SBC Edge . The entries in the tables provide information about the IP Addresses, ports, and protocols used to communicate with each server. The table entries also contain links to counters that are useful for troubleshooting.

SIP Server Table for IDH Server

1. Create a SIP Server Table with desired name: "IDH_Server".
2. Add domain name provided by the IDH Solution.
3. Enter the correct signaling port number this will be 5061 for TLS or 5060 for TCP/UDP.
4. Select the signaling protocol you require. The TLS is shown.
5. For TLS, you will need to select the TLS Profile you created earlier.

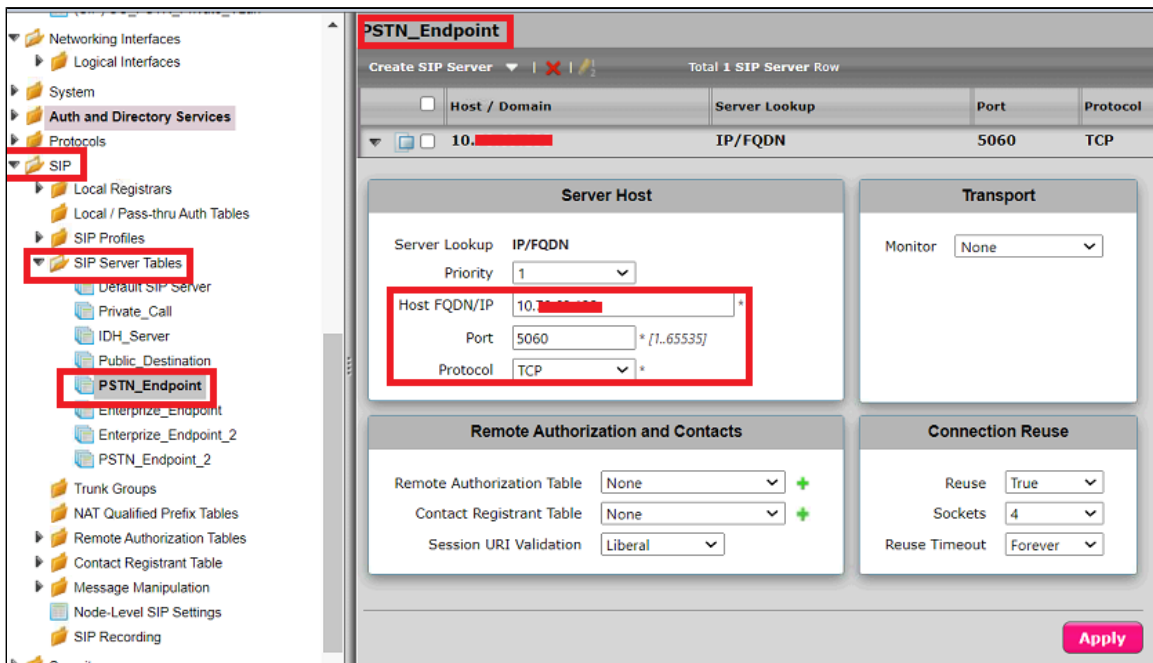


Enable OPTIONS	An OPTIONS message is sent to the server. When this option is selected, additional configuration items are displayed
Keep Alive Frequency	Specifies how often, in seconds, theSBC Edge queries the server with an OPTIONS message to determine the server's availability. Visible only whenSIP Optionsis selected from theMonitorfield. If the server does not respond, theSBC Edge marks the Signaling Group as down. When the server begins to respond to the OPTIONS messages again, it is marked as up.In this case, Keep Alive Frequency is set to 30 seconds.
Recover Frequency	Specifies frequency in seconds to check server to determine whether it has become available. Recovery Frequency is set to 5 seconds for this interop.
Local Username	Local username of the SBC Edge system. Default entry: Anonymous . Visible only when SIP Options is selected from the Mo onitorfield.
Peer Username	Username of the SIP Server. Visible only when SIP Options is selected from the Monitor field. The user can changeLocal and Peer Usernames according to their wishes.



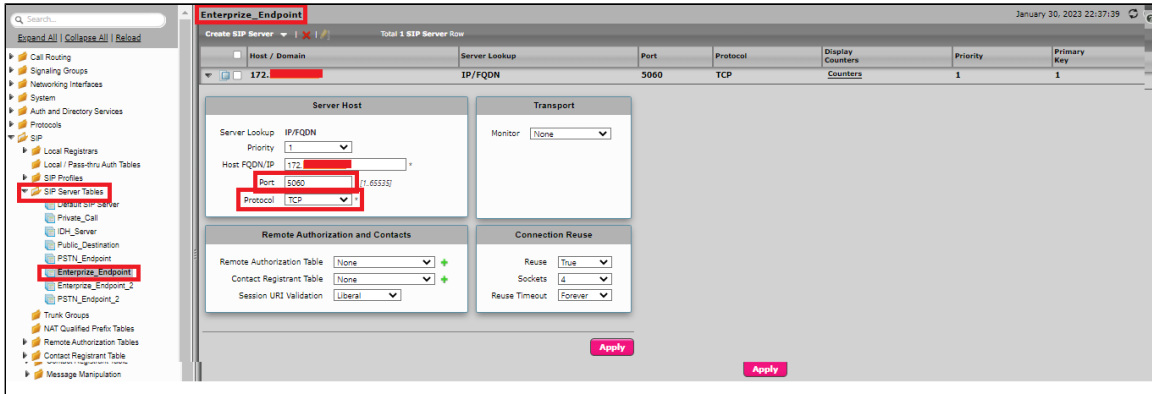
SIP Server Table for PSTN

1. Create a SIP Server Table with desired name **"PSTN_Endpoint"**
2. Add domain name provided by the IDH Solution.
3. Enter the correct signaling port number this will be 5061 for TLS or 5060 for TCP/UDP.
4. Select the signaling protocol you require. The TLS is shown.
5. For TLS, you will need to select the TLS Profile you created earlier.



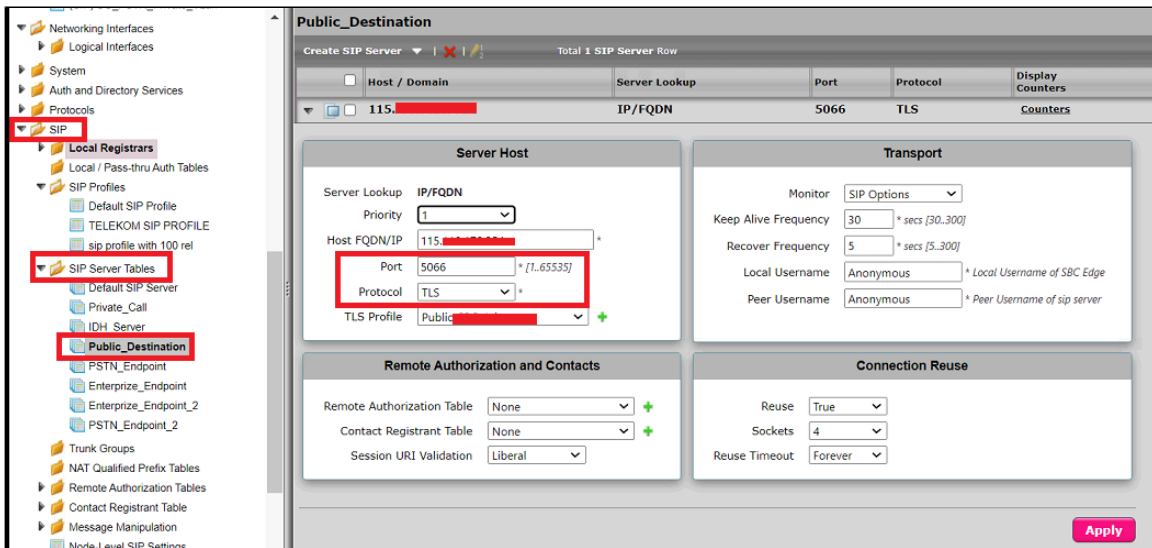
SIP Server Table for Enterprise

1. Create a SIP Server Table with desired name **"Enterprise_Endpoint"**.
2. Add domain name provided by the IDH Solution.
3. Enter the correct signaling port number this will be 5061 for TLS or 5060 for TCP/UDP.
4. Select the signaling protocol you require. The TLS is shown.
5. For TLS, you will need to select the TLS Profile you created earlier.



SIP Server Table for Public PSTN

1. Create a SIP Server Table with desired name "Public Destination"
2. Add domain name provided by the IDH Solution.
3. Enter the correct signaling port number this will be 5061 for TLS or 5060 for TCP/UDP.
4. Select the signaling protocol you require. The TLS is shown.
5. For TLS, you will need to select the TLS Profile you created earlier.



Note

Enable OPTIONS if required for the other SIP Server Tables.

Signaling Group

Inbound Signaling Group

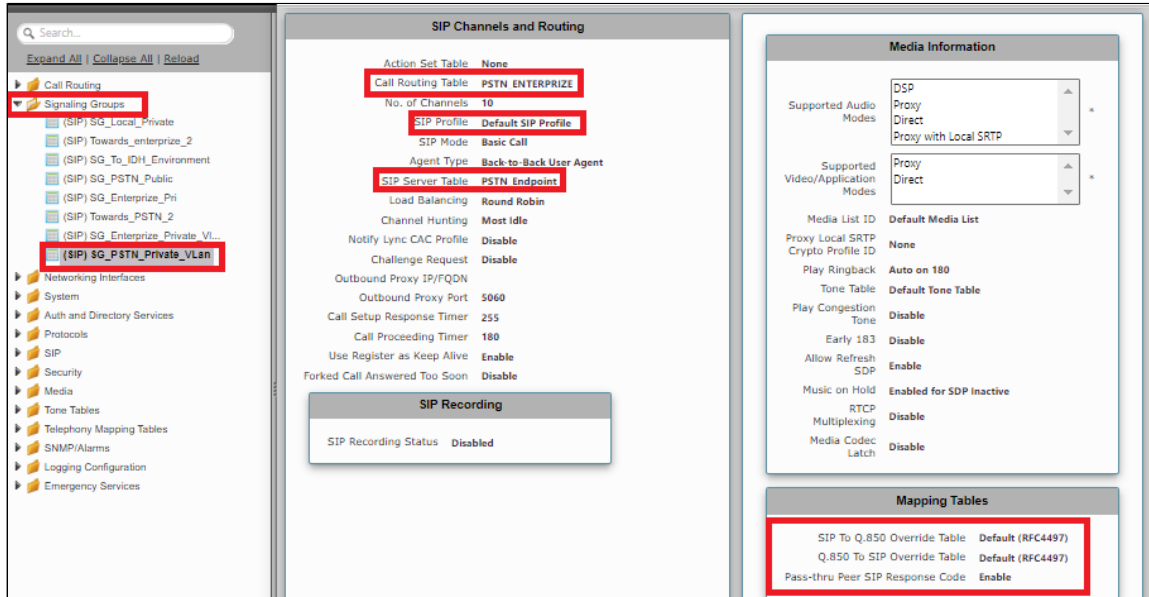
Signaling Groups allow grouping telephony channels together for the purposes of routing and shared configuration. They are the entity where calls are routed, as well as the location that the Call Routes are selected.

1. Select **Settings > Signaling Groups**.
2. Create an entry in the signaling group named: "**SG_PSTN_Private_Vlan**".
3. Choose "**Default SIP Profile**" under Sip Profile.
4. Choose Call Routing as "**PSTN ENTERPRIZE**".

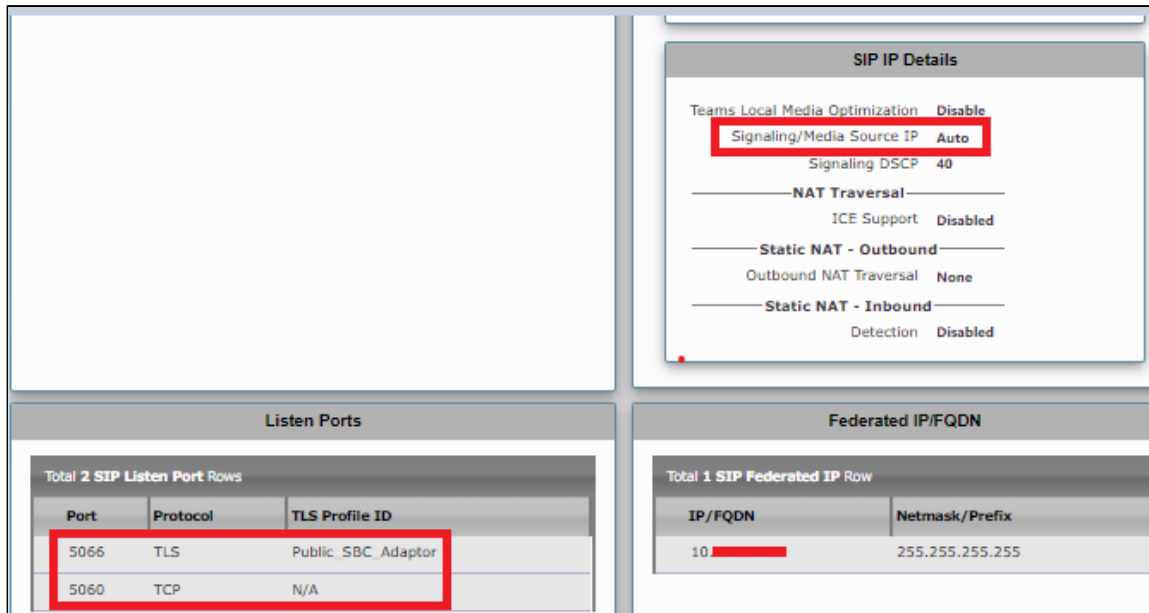


Initially, choose "Default" for the call Route. Create the Route, as shown in the call Routing section, and then update the call Route to "**PSTN_ENTERPRIZE**".

- Choose Agent type as "Back-to-Back user agent" and Media List as "IDH Interop".
- Choose SIP Server Table as "PSTN Endpoint".



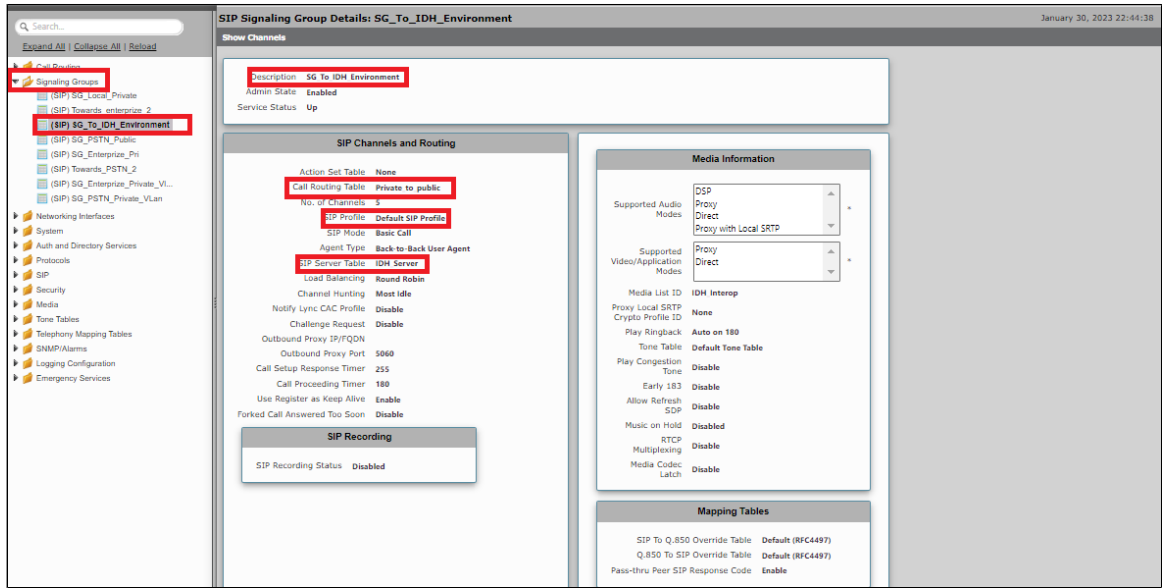
- Update the Federated IP/FQDN with endpoint IPv4.
- Add a listening port for TLS.



Outbound Signaling Group towards IDH

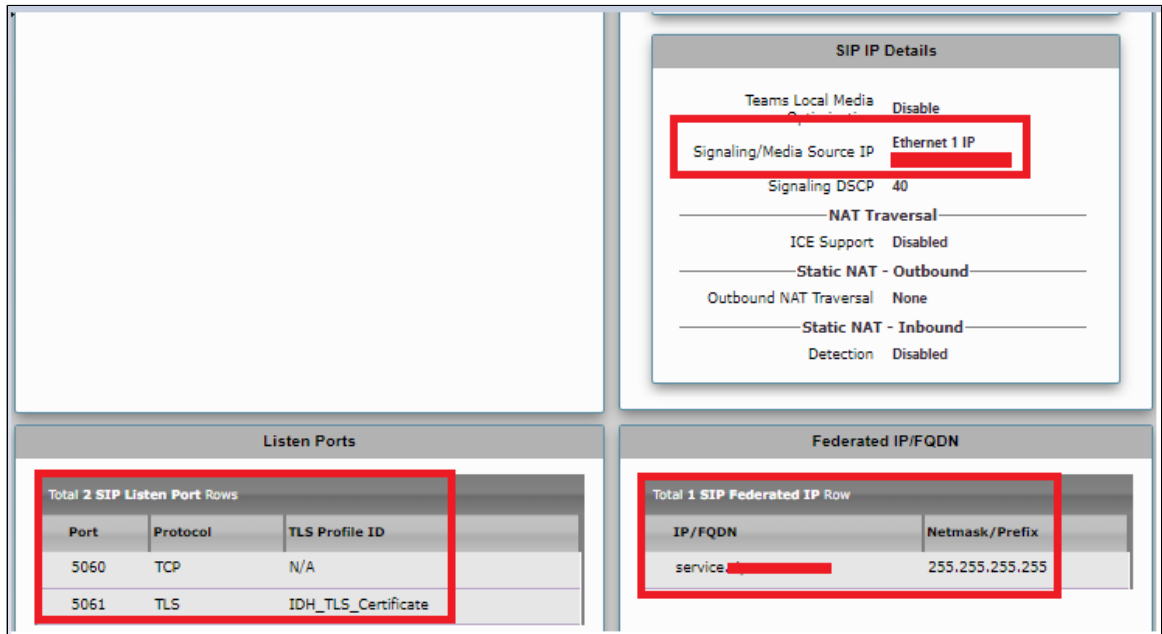
- Select **Settings > Signaling Groups**.
- Create an entry in the signaling group named: "**SG_To_IDH_Environment**".
- Choose "**Default SIP Profile**" under SIP Profile.
- Choose Call Routing as "**Default Route table**".
- Choose Agent type as "Back-to-Back user agent" and Media List as "**IDH Interop**".

6. Choose SIP Server Table as "IDH Server".



7. Update the Federated IP/FQDN with endpoint IPv4.

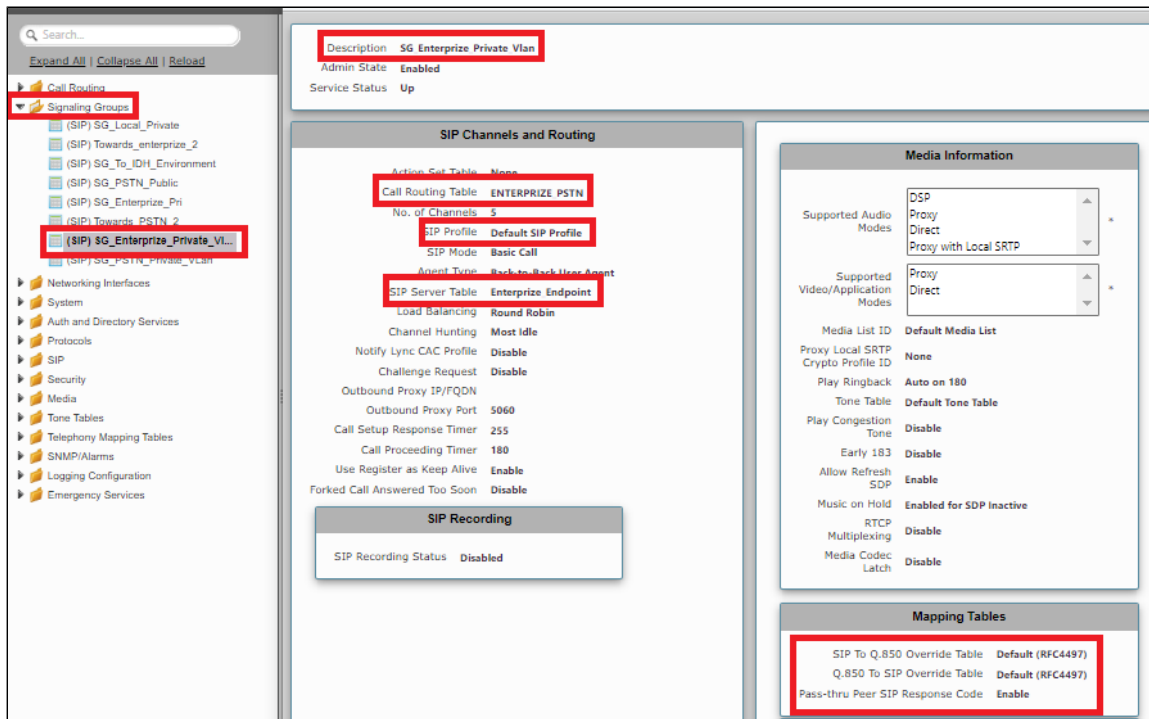
8. Add a listening port for TLS.



Outbound Signaling Group towards Enterprise Private

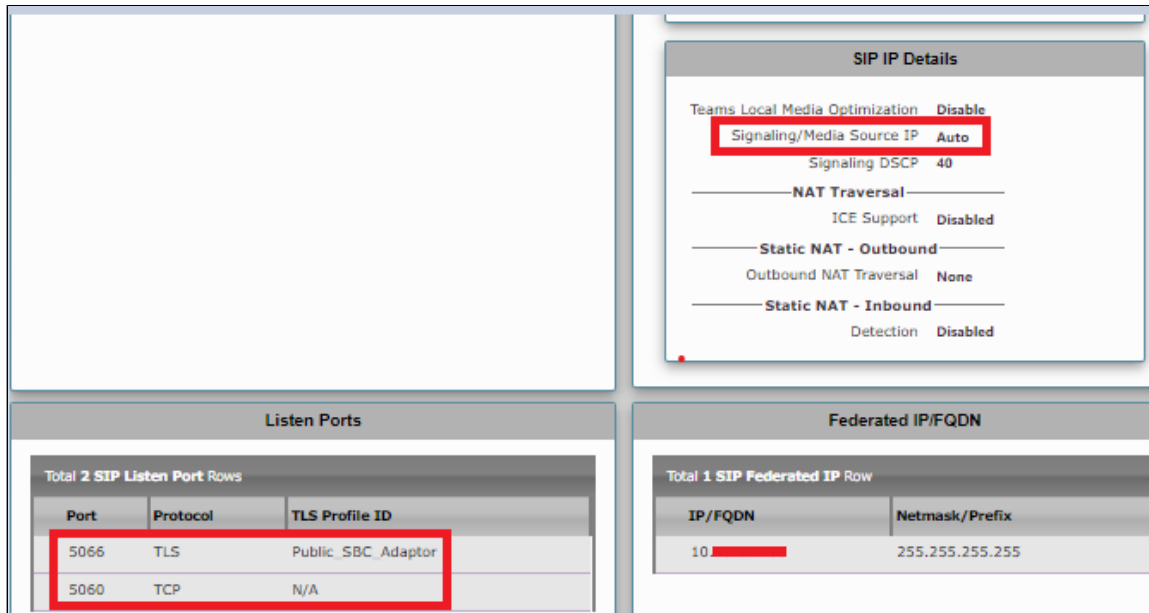
1. Select **Settings > Signaling Groups**.
2. Create an entry in the signaling group named: **"SG_Enterprise_Private_Vlan"**.
3. Choose **"Default SIP Profile"** under SIP Profile.
4. Choose Call Routing as **"ENTERPRIZE_PSTN"**.
5. Choose Agent type as **"Back-to-Back user agent"** and Media List as **"IDH Interop"**.

6. Choose SIP Server Table as "Enterprise Endpoint".



7. Update the Federated IP/FQDN with endpoint IPv4.

8. Add a listening port for TLS.



Inbound Signaling Group Enterprise Private

1. Select **Settings > Signaling Groups**.
2. Create an entry in the signaling group named: "**SG_Enterprise_Private**".
3. Choose "**Default SIP Profile**" under SIP Profile.
4. Choose Call Routing as "**Private_to_Public**".
5. Choose Agent type as "Back-to-Back user agent" and Media List as "**IDH Interop**".

6. Choose SIP Server Table as "Public Destination".

The screenshot shows a configuration interface with four main sections:

- SIP IP Details:** Contains settings for Teams Local Media Optimization (Disable), Signaling/Media Source IP (Auto), Signaling DSCP (40), NAT Traversal, ICE Support (Disabled), Static NAT - Outbound, Outbound NAT Traversal (None), Static NAT - Inbound, and Detection (Disabled).
- Listen Ports:** A table with 2 rows. The first row (5066, TLS, Public_SBC_Adaptor) is highlighted with a red box.
- Federated IP/FQDN:** A table with 1 row (10, 255.255.255.255).

7. Update the Federated IP/FQD with endpoint IPv4.

This screenshot is identical to the one above, showing the same configuration panels for SIP IP Details, Listen Ports, and Federated IP/FQDN.

Outbound Signaling Group - PSTN Public

1. Select **Settings > Signaling Groups**.
2. Create an entry in the signaling group named: "**SG_PSTN_Public**".
3. Choose "**Default SIP Profile**" under SIP Profile.
4. Choose Call Routing as "**Public_to_Private**".
5. Choose Agent type as "Back-to-Back user agent" and media list as "**IDH Interop**".

6. Choose SIP Server Table as "Public Destination".

Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

- (SIP) SG_Local_Private
- (SIP) Towards_enterprise_2
- (SIP) SG_To_IDH_Environment
- (SIP) SG_PSTN_Public**
- (SIP) SG_Enterprise_Private
- (SIP) Towards_PSTN_2
- (SIP) SG_Enterprise_Private_VL...
- (SIP) SG_PSTN_Private_VLan

Networking Interfaces

- System
- Auth and Directory Services
- Protocols
- SIP
- Security
- Media
- Tone Tables
- Telephony Mapping Tables
- SNMP/Alarms
- Logging Configuration
- Emergency Services

Description: SG_PSTN_Public
Admin State: Disabled
Service Status: Down

SIP Channels and Routing

Action Set Table: None
Call Routing Table: Public_to_Private
No. of Channels: 3
SIP Profile: **Default SIP Profile**
SIP Mode: Basic Call
Agent Type: Back-to-Back User Agent
SIP Server Table: Public Destination
Load Balancing: Round Robin
Channel Hunting: Most Idle
Notify Lync CAC Profile: Disable
Challenge Request: Disable
Outbound Proxy IP/FQDN:
Outbound Proxy Port: 5060
Call Setup Response Timer: 255
Call Proceeding Timer: 180
Use Register as Keep Alive: Enable
Forked Call Answered Too Soon: Disable

SIP Recording

SIP Recording Status: Disabled

Media Information

Supported Audio Modes: DSP, Proxy, Direct, Proxy with Local SRTP
Supported Video/Application Modes: Proxy, Direct
Media List ID: IDH_Interop
Proxy Local SRTP: None
Crypto Profile ID:
Play Ringback: Auto on 180
Tone Table: Default Tone Table
Play Congestion: Disable
Early 183: Disable
Allow Refresh: Disable
Music on Hold: Disabled
RTCP Multiplexing: Disable
Media Codec Latch: Disable

Mapping Tables

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

7. Update the Federated IP/FQD with endpoint IPv4.

8. Add a listening port for TLS.

SIP IP Details

Teams Local Media Optimization: Disable
Signaling/Media Source IP: **Ethernet 1 IP (115.255.255.255)**
Signaling DSCP: 40
NAT Traversal:
ICE Support: Disabled
Static NAT - Outbound:
Outbound NAT Traversal: None
Static NAT - Inbound:
Detection: Disabled

Listen Ports

Total 2 SIP Listen Port Rows

Port	Protocol	TLS Profile ID
5060	TCP	N/A
5066	TLS	Public: [redacted]

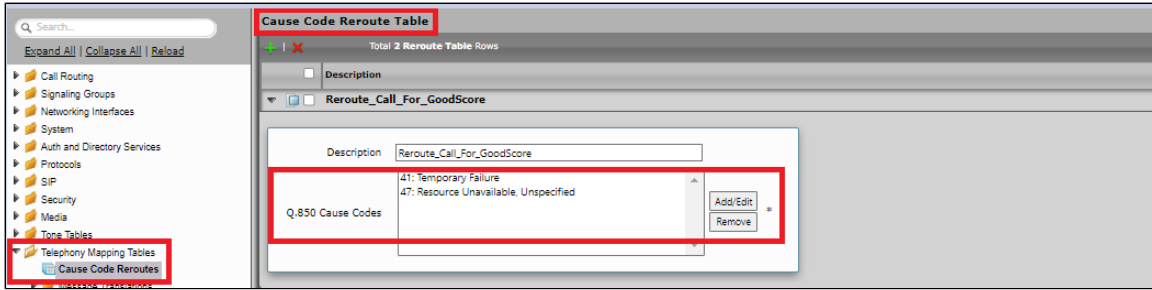
Federated IP/FQDN


Total 1 SIP Federated IP Row

IP/FQDN	Netmask/Prefix
115.255.255.255	255.255.255.255

Cause Code Reroutes

1. Select **Telephony Mapping Tables > Cause Code Reroutes**.
2. Click the Cause Code Reroute table (+) icon to create a new entry.
3. Provide the required description.
4. Add the **Q.850 Cause Codes** as '47:Resource Unavailable Unspecified' & '41:Temporary Failure'.
5. Click **Apply**.



 Cause Code Reroutes must be used in the call routing table "**PSTN_ENTERPRIZE**" and "**ENTERPRIZE_PSTN**" to reroute a call when receiving "**503 Service Unavailable**" and "**480 Temporarily not available**" responses from the **IDH SIP Interface**.

Transformation Table

Transformation Tables facilitate the conversion of names, numbers and other fields when routing a call. They can, for example, convert a public PSTN number into a private extension number, or into a SIP address (URI). Every entry in a Call Routing Table requires a Transformation Table, and they are selected from there. In addition, Transformation tables are configurable as a reusable pool that Action sets can reference.

From the **Settings > Call Routing > Transformation**.

Create Transformation Table

Each Transformation Table contains a list of entries considered as routing rules to execute on. Each rule is executed in order until the end of the table is reached or when a mandatory entry fails to execute.

- Create transformation table as "**Towards_IDH_Without_Country_Code**".

Follow the procedure described below to configure Transformation Tables and the Entries.

1. Click the **Create (+)** icon.
 2. Enter a descriptive name in the **Description** text field.
 3. Click **OK**.
- Creating an Entry to a Message Transformation Table.

For this interop, the entries are created based on the numbers associated with each endpoint. Users are free to select their own variables or Regular expressions. The UCC prime number can be configured.

1. Click the **Create (+)** icon next to the table created in the previous step.
2. Provide the details below:

Admin State	Enabled-The default state is Enabled.
Match Type	Optional: Optional entries must match at least one of that Input Field type. When a call arrives at a Transformation Table, the incoming message contains a number of Informational Elements (IEs). These IEs include important call information such as: Called Address/Number, Called Extension, Calling Name, Redirecting Number and others. Each Informational Element is processed row by row in the Transformation Table.
Value (Input /Output)	Specifies the value to match against for the selected type. Depending on the type selected, values are free-form or selected from a menu.

3. Click **Apply**.

Call Routing Entry: Towards IDH

Route Details

Description	Towards IDH
Admin State	Enabled
Route Priority	1
Call Priority	Normal
Number/Name Transformation Table	Towards_IDH_Without_Country_Code
Time of Day Restriction	None

Destination Information

Destination Type	Normal
Message Translation Table	None
Cause Code Reroutes	Reroute_Call_For_GoodScore
Cancel Others upon Forwarding	Disabled
Fork Call	No
Destination Signaling Groups	(SIP) SG_To_IDH_Environment *
Enable Maximum Call Duration	Disabled

Media	
Audio Stream Mode	DSP
Video/Application Stream Mode	Disabled
Media Transcoding	Disabled
Media List	IDH_Interop

Quality of Service	
Quality Metrics Number of Calls	10
Quality Metrics Time Before Retry	10
Min. ASR Threshold	0
Enable Min MOS Threshold	Disabled
Enable Max. R/T Delay	Enabled
Max. R/T Delay	65535
Enable Max. Jitter	Enabled
Max. Jitter	3000

From the **Settings > Call Routing > Transformation**.

Create Transformation Table for PSTN to Enterprise

Each Transformation Table contains a list of entries considered as routing rules to execute on. Each rule is executed in order until the end of the table is reached or when a mandatory entry fails to execute.

- Create transformation table as "**PSTN_Enterprise**".

Follow the procedure described below to configure Transformation Tables and the Entries.

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

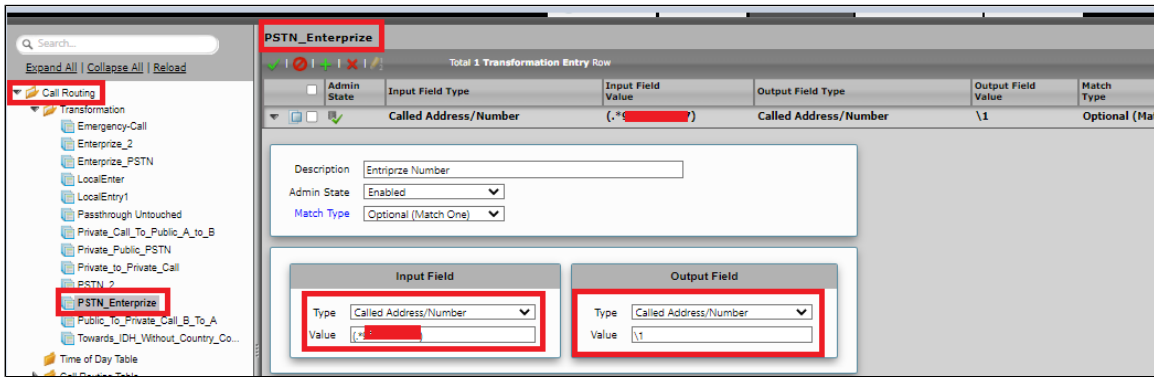
- Creating an Entry to a Message Transformation Table.

For this interop, the entries are created based on the numbers associated with each endpoint. Users are free to select their own variables or Regular expressions. The UCC prime number can be configured.

1. Click the **Create (+)** icon next to the table created in the previous step.
2. Provide the details below:

Admin State	Enabled-The default state is Enabled.
Match Type	Optional: Optional entries must match at least one of that Input Field type. When a call arrives at a Transformation Table, the incoming message contains a number of Informational Elements (IEs). These IEs include important call information such as: Called Address/Number, Called Extension, Calling Name, Redirecting Number and others. Each Informational Element is processed row by row in the Transformation Table.
Value (Input/Output)	Specifies the value to match against for the selected type. Depending on the type selected, values are free-form or selected from a menu.

3. Click **Apply**.



Create Transformation Table for Enterprise to PSTN

Each Transformation Table contains a list of entries considered as routing rules to execute on. Each rule is executed in order until the end of the table is reached or when a Mandatory entry fails to execute.

- Create transformation table as "Enterprise_PSTN".

Follow the procedure described below to configure Transformation Tables and the Entries.

1. Click the **Create (+)** icon.
 2. Enter a descriptive name in the **Description** text field.
 3. Click **OK**.
- Creating an Entry to a Message Transformation Table.

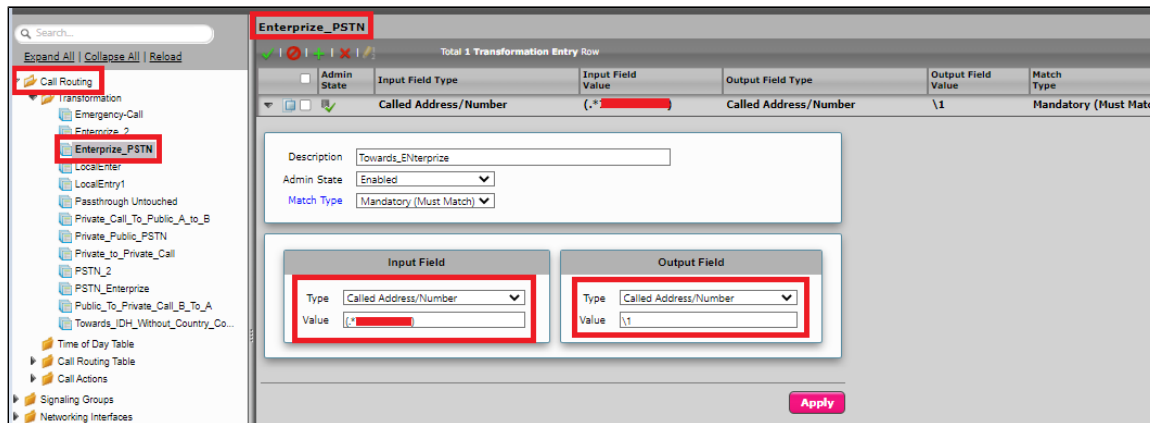
For this interop, the entries are created based on the numbers associated with each endpoint. Users are free to select their own variables or Regular expressions. The UCC prime number can be configured.

1. Click the **Create (+)** icon next to the table created in the previous step.
2. Provide the details below:

Admin State	Enabled-The default state is Enabled.
Match Type	Optional: Optional entries must match at least one of that Input Field type. When a call arrives at a Transformation Table, the incoming message contains a number of Informational Elements (IEs). These IEs include important call information such as: Called Address/Number, Called Extension,

	Calling Name, Redirecting Number and others. Each Informational Element is processed row by row in the Transformation Table.
Value (Input /Output)	Specifies the value to match against for the selected type. Depending on the type selected, values are free-form or selected from a menu.

3. Click **Apply**.



Create Transformation Table for Emergency Call

Each Transformation Table contains a list of entries considered as routing rules to execute on. Each rule is executed in order until the end of the table is reached or when a Mandatory entry fails to execute.

- Create transformation table as "**Emergency Call**".

Follow the procedure described below to configure Transformation Tables and the Entries.

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

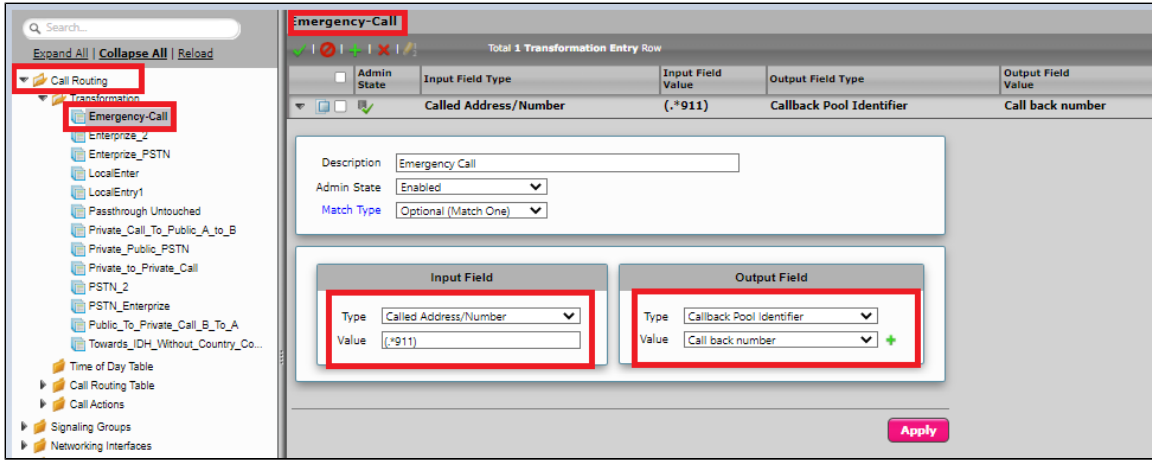
- Creating an Entry to a Message Transformation Table.

For this interop, the entries are created based on the numbers associated with each endpoint. Users are free to select their own variables or Regular expressions. The UCC prime number can be configured.

1. Click the **Create (+)** icon next to the table created in the previous step.
2. Provide the details below :

Admin State	Enabled-The default state is Enabled.
Match Type	Optional: Optional entries must match at least one of that Input Field type. When a call arrives at a Transformation Table, the incoming message contains a number of Informational Elements (IEs). These IEs include important call information such as: Called Address/Number, Called Extension, Calling Name, Redirecting Number and others. Each Informational Element is processed row by row in the Transformation Table.
Value (Input /Output)	Specifies the value to match against for the selected type. Depending on the type selected, values are free-form or selected from a menu.

3. Click **Apply**.



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

Call Routing Table

Call Routing allows carrying of calls between Signaling Groups. Routes are defined by Call Routing Tables, which allow for a flexible configuration that calls carry, and how to translate them.

Select **Settings > Call Routing > Call Routing Table**.

Create Call Routing Table for PSTN to Enterprise

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

- Create Call Routing table as "**PSTN_Enterprise**"

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

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

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

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

To create an entry:

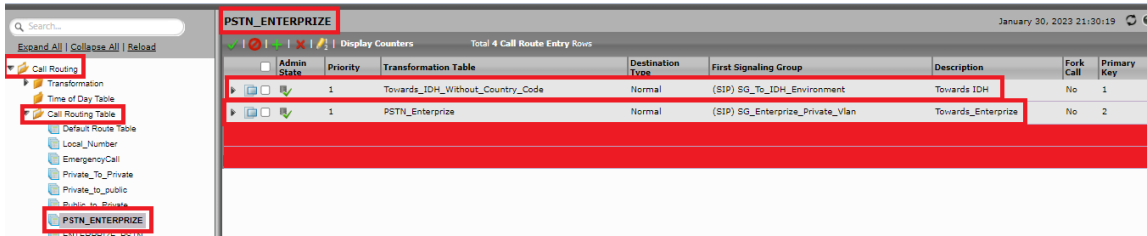
1. Click the **Create Routing Entry (+)** icon.
2. Set the following fields:

Admin State	Enabled - Enables the call route entry for routing the call, displays in configuration header as .
Route Priority	Priority of the route from 1 (highest) to 10 (lowest). Higher priority routes are matched against before lower priority routes, regardless of the order of the routes in the table.
Number/Name Transformation Table	Specifies the Transformation Table to use for this routing entry. This drop-down list is populated from the entries in the Transformation Table.

**Destination
Signaling
Groups**

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

3. Click **Apply**.



Route1 details: Number/Name Transformation Table with "Towards_IDH_without_Counry_Code" and Destination Signaling Groups with "SG_To_IDH_Environment".

Route2 details: Number/Name Transformation Table with "PSTN Enterprize" and Destination Signaling Groups with "SG_Enterprize_Private_Vlan".

Call Routing Entry: Towards_Enterprize

Route Details

Description Towards_Enterprize
 Admin State Enabled
 Route Priority 1
 Call Priority Normal
 Number/Name Transformation Table PSTN_Enterprize
 Time of Day Restriction None

Destination Information

Destination Type Normal
 Message Translation Table None
 Cause Code Reroutes None
 Cancel Others upon Forwarding Disabled
 Fork Call No
 Destination Signaling Groups (SIP) SG_Enterprize_Private_Vlan *
 Enable Maximum Call Duration Disabled

Media

Audio Stream Mode DSP
 Video/Application Stream Mode Disabled
 Media Transcoding Disabled
 Media List IDH_Interop

Quality of Service

Quality Metrics Number of Calls 10
 Quality Metrics Time Before Retry 10
 Min. ASR Threshold 0
 Enable Min MOS Threshold Disabled
 Enable Max. R/T Delay Enabled
 Max. R/T Delay 65535
 Enable Max. Jitter Enabled
 Max. Jitter 3000

Create Call Routing Table for Enterprise to PSTN

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

- Create Call Routing table as "PSTN_Enterprize"

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


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

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

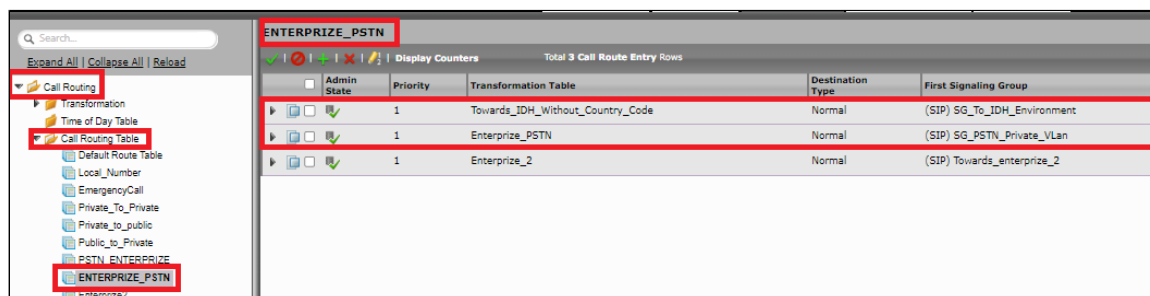
Each call routing entry describes how to route the call and also points to a Transformation Table that defines the conversion of names, numbers and other fields when routing a call.

To create an entry:

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

Admin State	Enabled - Enables the call route entry for routing the call, displays in configuration header as  .
Route Priority	Priority of the route from 1 (highest) to 10 (lowest). Higher priority routes are matched against before lower priority routes, regardless of the order of the routes in the table.
Number /Name Transformation Table	Specifies the Transformation Table to use for this routing entry. This drop-down list is populated from the entries in the Transformation Table.
Destination Signaling Groups	Specifies the Signaling Groups used as the destination of calls. The first operational Signaling Group from the list is chosen to place the call. Click the Add/Edit button to select the destination signaling group.
Audio Stream Mode	DSP(default entry): The SBC uses DSP resources for media handling (transcoding), but does not facilitate the capabilities/features between endpoints that are not supported within the SBC (codec/capability mismatch). When the DSP is configured, the Signaling Groups enabled to support DSP are attempted in order.

3. Click **Apply**.



Route1 details: Number/Name Transformation Table with "Towards_IDH_without_Counry_Code" and Destination Signaling Groups with "SG_To_IDH_Environment".

Call Routing Entry: Towards IDH

Route Details

Description	Towards IDH
Admin State	Enabled
Route Priority	1
Call Priority	Normal
Number/Name Transformation Table	Towards_IDH_Without_Country_Code
Time of Day Restriction	None

Destination Information

Destination Type	Normal
Message Translation Table	None
Cause Code Reroutes	Reroute_Call_For_GoodScore
Cancel Others upon Forwarding	Disabled
Fork Call	No
Destination Signaling Groups	(SIP) SG_To_IDH_Environment *
Enable Maximum Call Duration	Disabled

Media	
Audio Stream Mode	DSP
Video/Application Stream Mode	Disabled
Media Transcoding	Disabled
Media List	IDH_Interop

Quality of Service	
Quality Metrics Number of Calls	10
Quality Metrics Time Before Retry	10
Min. ASR Threshold	0
Enable Min MOS Threshold	Disabled
Enable Max. R/T Delay	Enabled
Max. R/T Delay	65535
Enable Max. Jitter	Enabled
Max. Jitter	3000

Route2 details:Number/Name Transformation Table with "Enterprize_PSTN" and Destination Signaling Groups with "SG_PSTN_Private_Vlan"

Call Routing Entry: ToPrivate

Route Details

Description	ToPrivate
Admin State	Enabled
Route Priority	1
Call Priority	Normal
Number/Name Transformation Table	Enterprize_PSTN
Time of Day Restriction	None

Destination Information

Destination Type	Normal
Message Translation Table	None
Cause Code Reroutes	None
Cancel Others upon Forwarding	Disabled
Fork Call	No
Destination Signaling Groups	(SIP) SG_PSTN_Private_VLan *
Enable Maximum Call Duration	Disabled

Media

Audio Stream Mode	DSP
Video/Application Stream Mode	Disabled
Media Transcoding	Enabled
Media List	None

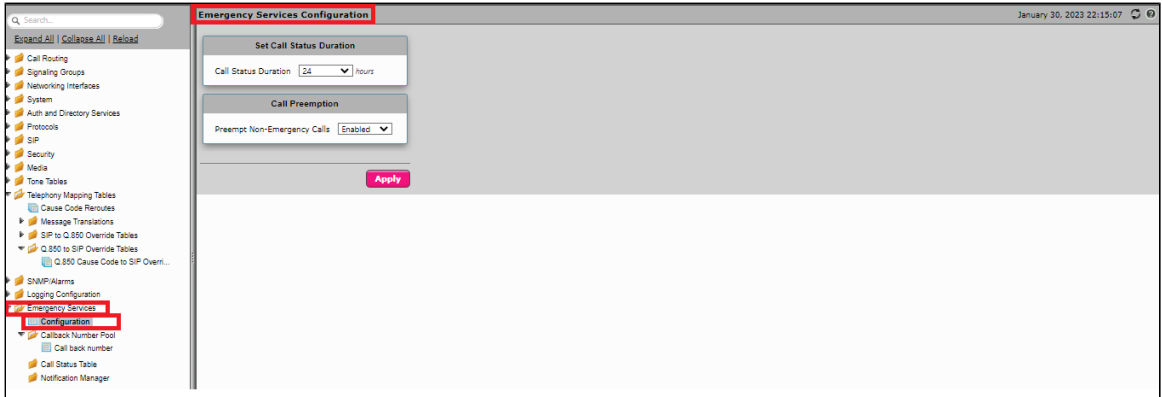
Quality of Service

Quality Metrics Number of Calls	10
Quality Metrics Time Before Retry	10
Min. ASR Threshold	0
Enable Min MOS Threshold	Disabled
Enable Max. R/T Delay	Enabled
Max. R/T Delay	65535
Enable Max. Jitter	Enabled
Max. Jitter	3000

Emergency

The Emergency Services functionality is an essential and important feature in support of the E911 and 911 feature in the SBC.

Emergency Calls are logged in the emergency call status table whenever the SBC identifies the call as an emergency and raises an alarm to alert the operator. Call Forking or Call routing on failure are provided as part of existing call routing functionality.

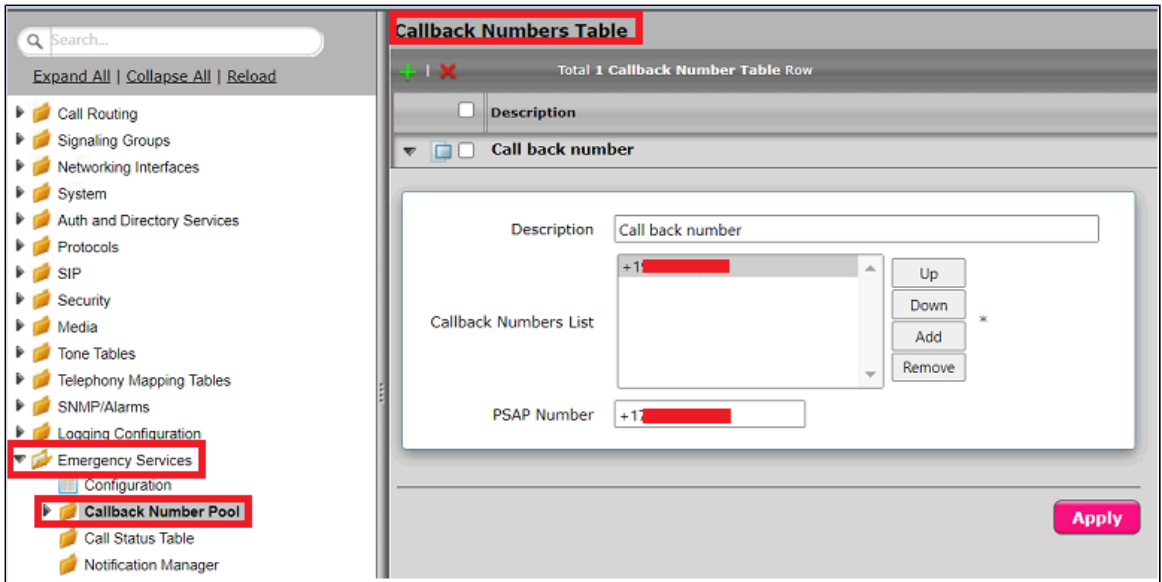


Callback Number

Callback number lists contain up to 16 numbers. These numbers replace the actual number of the calling party when the call is sent to the Emergency Services Operator (PSAP). If the PSAP operator calls back (to the callback number) the call is routed using a Transformation Table entry back to the original calling extension.

Creating a Callback Number List

1. Click the Create Callback Numbers List (+) icon at the top of the Callback Numbers Table page.



Callback Numbers List	A list of numbers which are substituted in a Transformation for the actual number from which the call was initiated. These numbers are used in a round-robin scheme.
PSAP Number	Use this field to specify the Public Safety Answering Point (PSAP) number. This is the number to which the call is connected when an extension calls 911. If no value is specified in this field, the call is connected to the local 911 facility.

Section B : Ribbon Identity Hub Configuration

Identity Hub Reputation Scoring Process

The SIP client initiates a request by routing the SIP INVITE request for a call to the Identity Hub service FQDN. The INVITE request must include a From header to identify the calling party and a To header to identify the called party. Based on the workflow selector and the workflow configuration, the Identity Hub selects the service graph and begins reputation scoring.

Following the scoring process, the Identity Hub returns different response codes to the SIP client based on the fraud score. If the fraud score is below the suspect-call threshold value of 80, then the service returns a 503 SIP response code. If the fraud score is 80 or above, then the service returns a 608 SIP response code. Note that the 503 and 608 values are the default responses. When setting up your service with Ribbon, you have the option to use 480 as an alternative response to non-fraud calls, and either 603 or 607 as an alternative response to fraud calls.

The SIP client then either resumes routing a good call, or terminates a fraud call and relays the 6xx response back to the originator.

Possible SIP Responses

The SIP service API can return the following responses:

- 503 (default), 480 call is unlikely to be fraudulent (fraud score less than 80)
- 608 (default), 603, 607 call is likely fraudulent (fraud score is 80 or above)
- 405 Method Not Allowed non-INVITE request is not allowed
- Other 4xx, 5xx service failure or error

For Reputation scoring using SIP interface related configurations, refer to: <https://doc.rbn.com/display/IDHDOC20/Reputation+Scoring+using+the+SIP+Interface>.

Supplementary Services and 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	Call Setup and Termination over TCP	✓
2	Call Setup and Termination over TLS	✓
3	Call Transfer (Blind)	✓
4	Call Transfer (Attended)	✓
5	Call hold and Resume (with MOH enable)	✓
6	Call hold and Resume (without MOH enable)	✓
7	AnonymousCall	✓
8	Call Forward	✓
9	Emergency Call	✓

Legend

Supported	✓
Not Supported	✗

Caveats

- The Call forward issue has been observed during testing with SIP interface towards the Identity Hub. As per the current design of Ribbon SBC Edge, the call routing table can perform cause code reroute only once. This is a known constraint for Ribbon SBC Edge and will be addressed in the upcoming release.
- The SBC Edge initiates a 2nd INVITE towards the IDH solution after getting a 5xx response for the 1st INVITE. This issue will be addressed in the upcoming SBC release.

Support

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/about-us>

References

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

Conclusion

This Interoperability Guide describes the configuration steps required for **Ribbon SBC Edge SIP Interface** to successfully interoperate with **IDH**. All features and serviceability test cases were completed and passed with the exceptions/observations noted in test results. 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 is/is not covered.

Configuration guidance is provided to enable the reader to replicate the same base setup. Any additional configuration changes are possibly required to suit the exact customer deployment environment.

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