
Ribbon SBC Edge 1K_2K_SWe Edge R9.0 Interop with Google Voice SIP Link : Interoperability Guide



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Interoperable Vendors



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Document Overview

This document outlines the configuration best practices for Ribbon SBC SWe Edge interworking with Google Voice SIP Link.

About Ribbon SBC SWe Edge

The Ribbon Session Border Controller Software Edition Lite (SBC SWe Edge) provides best-in-class communications security. The SBC SWe Edge dramatically simplifies the deployment of robust communications security services for SIP Trunking, Direct Routing, and Cloud UC services. SBC SWe Edge operates natively in the Azure and AWS Cloud as well as on virtual machine platforms including Microsoft Hyper-V, VMware and Linux KVM.

About Google Voice

Google Voice is a telephone service that provides a U.S. phone number to Google Account customers in the U.S. and Google Works customers in Canada, Denmark, France, the Netherlands, Portugal, Spain, Sweden, Switzerland and the United Kingdom. Calls are forwarded to the phone number that each user must configure in the account web portal. Users can answer and receive calls on any of the phones configured to ring in the web portal. While answering a call, the user can switch between the configured phones. Subscribers in the United States can make outgoing calls to domestic and international destinations. The service is configured and maintained by users in a web-based application, similar in style to Google's email service Gmail, or Android and iOS applications on smartphones or tablets.

Scope/Non-Goals

This document provides configuration best practices for deploying Ribbon's SBC SWe Edge for Google Voice SIP Link 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

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

To perform this interop, you need to

- use graphical user interface (GUI) or command line interface (CLI) of the Ribbon product.
- understand the basic concepts of TCP/UDP/TLS and IP/Routing.
- have SIP/RTP/SRTP 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.

Pre-Requisites

The following aspects are required before proceeding with the interop:

- Ribbon SBC SWe Edge
- Ribbon SBC SWe Edge license
 - This interop requires the acquisition and application of SIP sessions, as documented at [Working with Licenses](#)
- Public IP addresses
- TLS certificates for SBC SWe Edge
 - For more details, please visit [Working with Certificates](#)
- Google Workspace and Domain
 - Google Voice Premier license for the users
 - For more details, contact [Google support](#)

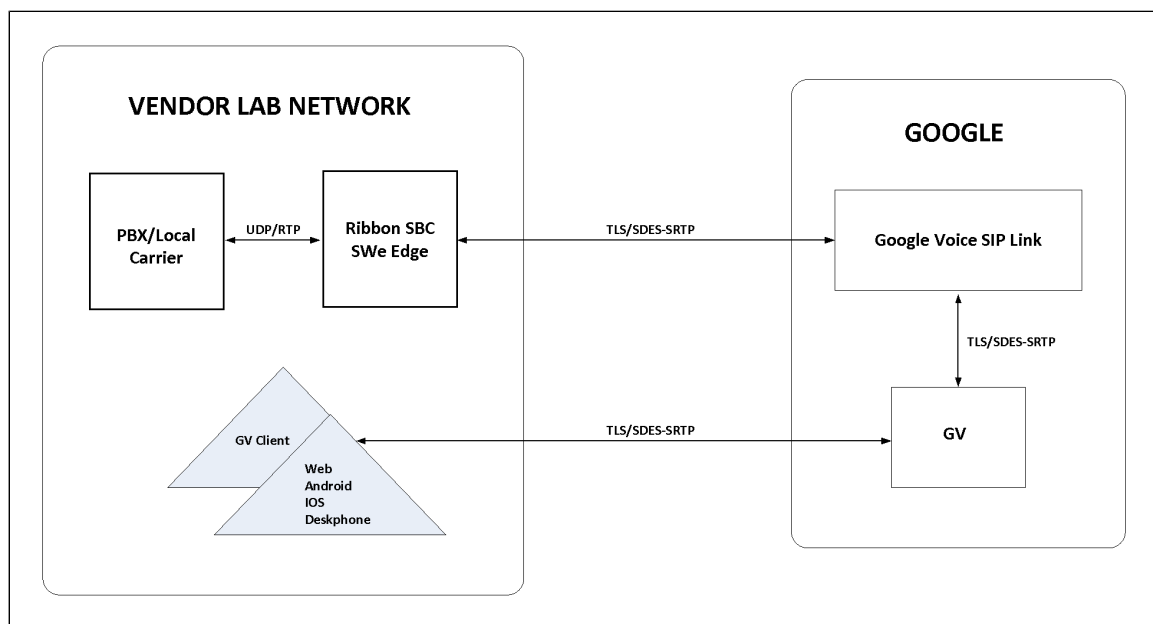
Product and Device Details

The configuration uses the following equipment and software:

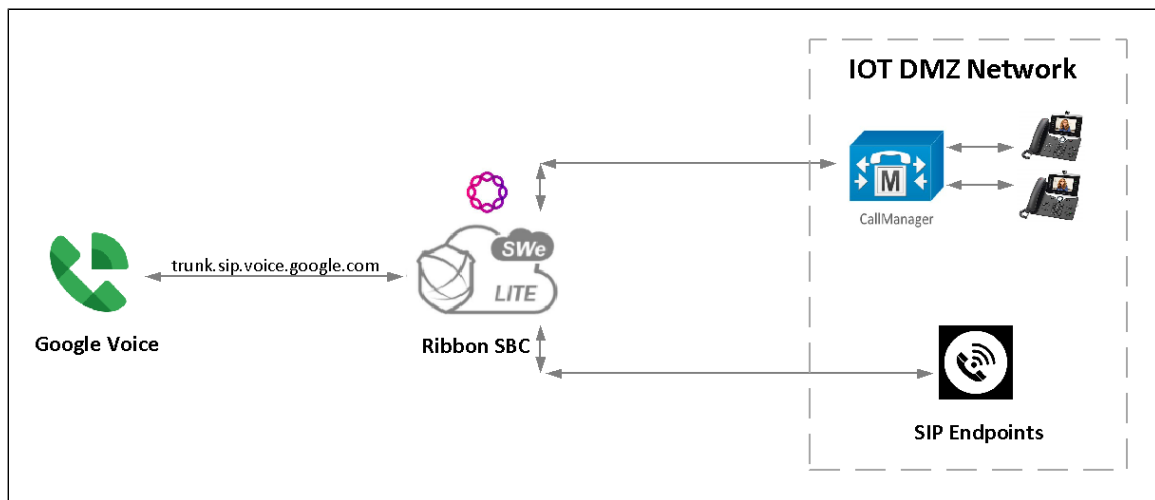
Product	Equipment/Service	Software Version
Ribbon SBC	Ribbon SBC SWe Edge	9.0.7
Google Voice SIP Link	Telephone Service	NA
Third-party Equipment	Cisco Unified Communications Manager	12.5.1.11900-146
Administration and Debugging Tools	Wireshark	3.4.9
	LX Tool	2.1.0.6

Network Topology and E2E Flow Diagrams

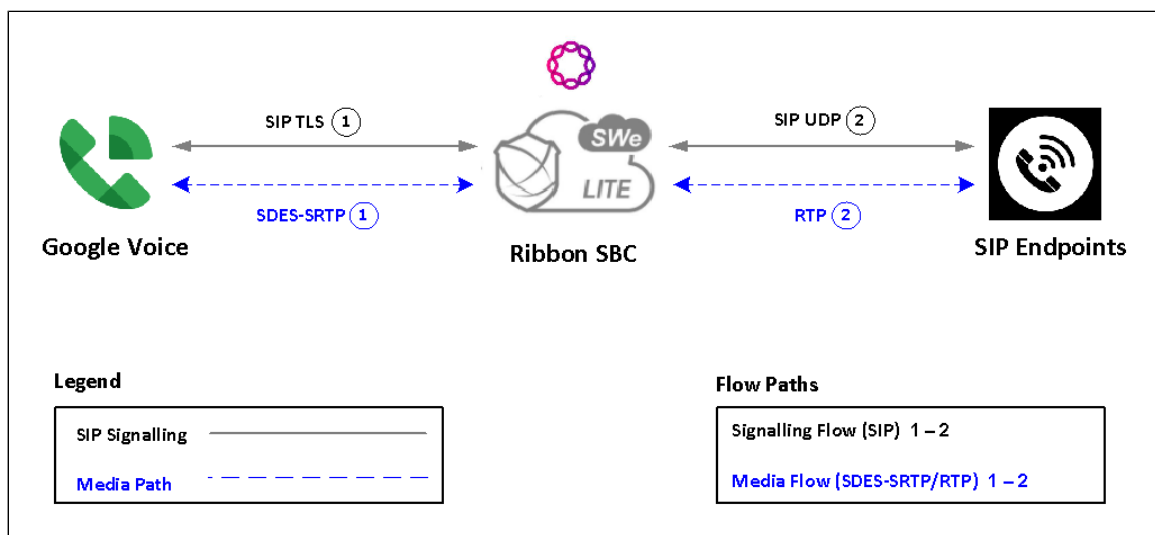
Deployment Topology



Interoperability Test Lab Topology

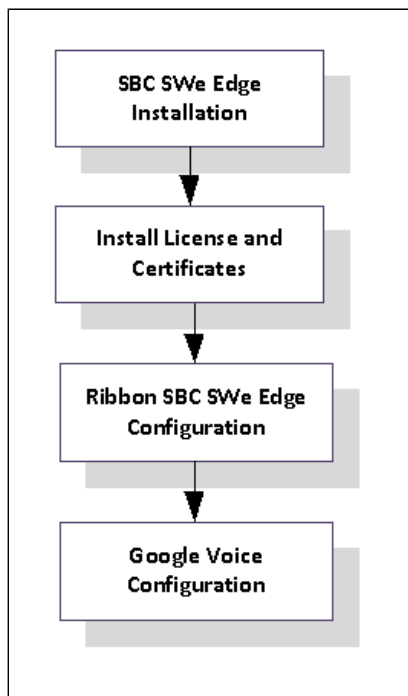


Call Flow Diagram



Document Workflow

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



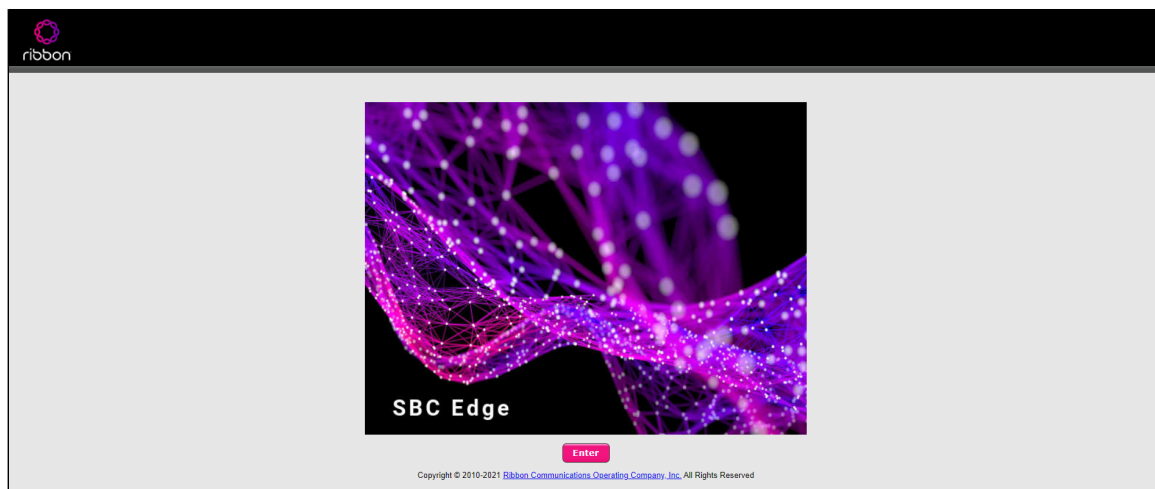
Installing Ribbon SBC SWe Edge

To deploy Ribbon SBC SWe Edge instance, refer to [Installing SBC SWe Edge](#).

Ribbon SBC SWe Edge Configuration

Accessing SBC SWe Edge

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



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

Welcome to Ribbon SBC SWe Lite

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

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

User Name

Password

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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 **Settings** tab, navigate to **System > Licensing > Current Licenses**.

Current Licenses

License Format Version 3

Feature	Licensed	Total Licenses	Available Licenses	Feature Expiration
SIP Signaling Sessions		100	100	April 03, 2022 23:59:59
Enhanced Media Sessions with Transcoding		100	100	April 03, 2022 23:59:59
Enhanced Media Sessions without Transcoding		100	100	April 03, 2022 23:59:59
SIP Registrations		100	100	April 03, 2022 23:59:59
AHR-WB		Unlimited	Unlimited	April 03, 2022 23:59:59
SIP Recording		100	100	April 03, 2022 23:59:59

For more details on Licenses, refer to [SWe Edge License](#)

SBC Certificate

From the **Settings** tab, navigate to **Security > SBC Certificates > Generate SBC Edge Certificates**.

1. Provide the Common Name of the SBC that includes Host and Domain.
2. Set the Key Length to 2048 bits.
3. Provide the location information.
4. Click OK.
5. The CSR will be generated and displayed in the result text box.

Generate Certificate Signing Request

Subject Distinguished Name

Common Name * Hostname or FQDN

Subject Alternative Name DNS comma-separated FQDN list

Email Address

ISO Country Code

State/Province

Locality e.g.: City

Organization e.g.: Company

Organizational Unit e.g.: Department

Key Length

OK

After generating the CSR on Ribbon SBC, provide it to the Certificate Authority. CA would generally provide the following certificates:

- SBC Certificate
- CA's Root Certificate
- Intermediate Certificate

SBC Certificates Index

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

There are two ways to import SBC Primary Certificate as described below:

To import an X.509 signed certificate:

1. Select X.509 Signed Certificate from the Import menu at the top of the page.
2. Chose the import mode (Copy and Paste or File Upload) from the Mode pull-down menu.
3. If you chose File Upload, use the Browse button to find the file and click OK.
4. If you choose Copy and Paste, open the file in a text editor, paste the contents into the Paste Base64 Certificate text field and click OK.

To import a PKCS12 Certificate and Key:

1. Select PKCS12 Certificate and Key from the Import menu at the top of the page.
2. Enter the password used to export the certificate in the Password field.
3. Browse for the PKCS certificate and key file and click OK.

Import X.509 Server Certificate

Mode: Copy and Paste

Paste Base64 Certificate

OK

Import X.509 Server Certificate

Mode: File Upload

Select File: Choose File No file chosen Extensions [pem, der, cer, ber, p7b] *

OK

Import PKCS12 Server Certificate

Password: *

Select File: Choose File No file chosen Extensions [.pfx or .p12] *

OK

Trusted CA Certificates

A Trusted CA Certificate is a certificate issued by a Trusted Certificate Authority. Trusted CA Certificates are imported to the SBC SWe Edge to establish its authenticity on the network.

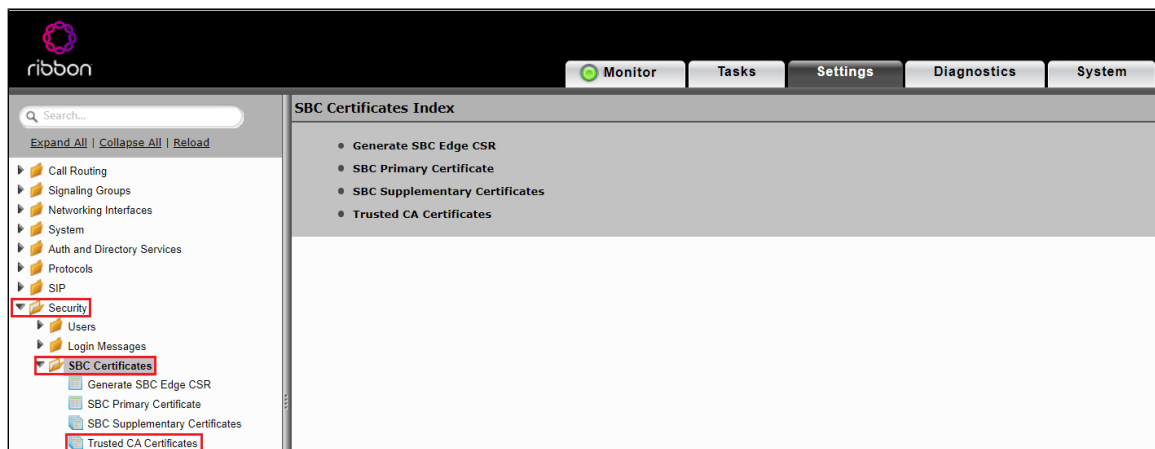
- For TLS to work, a Trusted CA (Certificate Authority) is required. For this interop, GoDaddy is used as Trusted CA.
- Add an entry in the Public DNS to resolve Ribbon SBC SWe Edge FQDN to Public IP Address.
- Ensure to have the following certificates as part of the root certificate trust:
 - GTS Root R1
 - GlobalSign Root CA (if required)




Note

Refer to Google Voice SIP Link documentation for other compatible CAs.

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



This section describes the process of importing Trusted Root CA Certificates using either the File Upload or Copy and Paste method.

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

Import Trusted CA Certificate

Mode

Copy and Paste

Paste Base64 Certificate

OK

Import Trusted CA Certificate

Mode

File Upload

Select File

Choose File

 No file chosen

Extensions [pem, der, cer, ber, p7b] *

OK

Follow the steps above to import GTS Root R1 and GlobalSign Root CA certificates from Google Voice.

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.

 Warning

Most Certificate Vendors sign the SBC Edge certificate with an intermediate certificate authority. There is at least one, but there could be several intermediate CAs in the certificate chain. When importing the Trusted Root CA Certificates, import the root CA certificate and all Intermediate CA certificates. Failure to import all certificates in the chain causes the import of the SBC Edge certificate to fail. Please refer to [Unable To Get Local Issuer Certificate](#) for more information.

Networking Interfaces


The SBC SWe 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 SWe Edge supports user created VLAN logical sub-interfaces.

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

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

Administrative IP

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



Monitor

Tasks

Settings

Diagnostics

System

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

- Call Routing
- Signaling Groups
- Networking Interfaces**
 - Logical Interfaces**
 - Admin IP
 - Ethernet 1 IP
 - Ethernet 2 IP
 - Ethernet 3 IP
 - Ethernet 4 IP

Logical Interfaces

Create VLAN T/F

Total 5 LogicalInterface Rows

	Interface Name	IPv4 Address	IPv6 Address	Description	Admin State	Display	Primary Key
<input type="checkbox"/>	Admin IP	10.54. <input type="text" value=""/>			Enabled	Counters	35
<input type="checkbox"/>	Ethernet 1 IP	10.54. <input type="text" value=""/>			Enabled	Counters	36
<input type="checkbox"/>	Ethernet 2 IP	115.110. <input type="text" value=""/>			Enabled	Counters	37
<input type="checkbox"/>	Ethernet 3 IP	10.10.10.10			Enabled	Counters	38
<input type="checkbox"/>	Ethernet 4 IP	20.20.20.20			Enabled	Counters	39

Ethernet 1 IP

Ethernet 1 IP is assigned an IP address used for transporting all the VOIP media packets (for example, RTP, SRTP) and all protocol packets (for example, SIP, RTCP, TLS). DNS servers of the customer's network should map the SBC SWe Edge system hostname to this IP address. In the default software, **Ethernet 1 IP** is enabled and an IPv4 address is acquired via a connected DHCP server. This IP address is used for performing Initial Setup on the SBC SWe Edge.

Logical Interfaces

Create VLAN I/F | Total 5 LogicalInterface Rows

Interface Name	IPv4 Address	IPv6 Address	Description	Admin State	Display	Primary Key
Admin IP	10.54.x.x			Enabled	Counters	35
Ethernet 1 IP	10.54.x.x			Enabled	Counters	36

Identification/Status

Interface Name: Ethernet 1 IP
 I/F Index: 8
 Alias:
 Description:
 Admin State: Enabled

Networking

MAC Address:
 IP Addressing Mode: IPv4

IPv4 Information

IP Assign Method: Static
 Primary Address: 10.54.x.x
 Primary Netmask: 255.255.255.0
 Media Next Hop IP: 10.54.x.x

Ethernet 2 IP

After initial configuration, you may configure this logical interface using the Settings or Tasks tabs in the WebUI or you can use the IP address configured during Initial Setup.

Logical Interfaces

Create VLAN I/F | Total 5 LogicalInterface Rows

Interface Name	IPv4 Address	IPv6 Address	Description	Admin State	Display	Primary Key
Admin IP	10.54.x.x			Enabled	Counters	35
Ethernet 1 IP	10.54.x.x			Enabled	Counters	36
Ethernet 2 IP	115.110.x.x			Enabled	Counters	37

Identification/Status

Interface Name: Ethernet 2 IP
 I/F Index: 9
 Alias:
 Description:
 Admin State: Enabled

Networking

MAC Address:
 IP Addressing Mode: IPv4

IPv4 Information

IP Assign Method: Static
 Primary Address: 115.110.x.x
 Primary Netmask: 255.255.255.192
 Media Next Hop IP: 115.110.x.x

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).

Derive the Private IP address and Gateway for each interface on AWS.

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'.

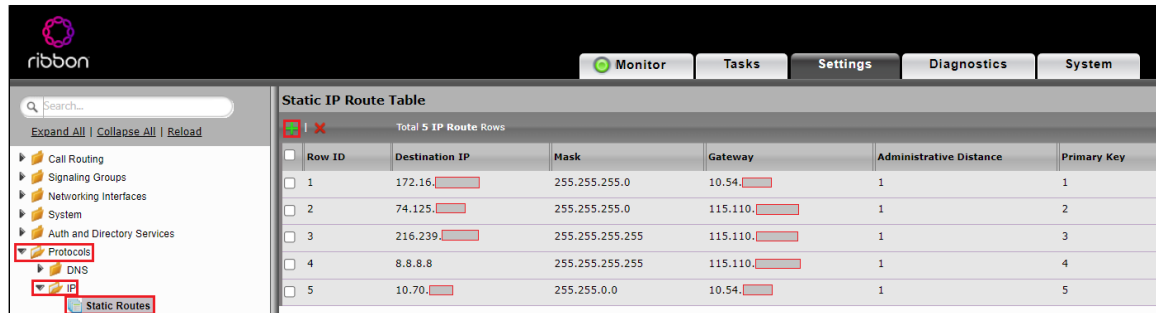
Gateway

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.

From the **Settings** tab, navigate to **Protocols > IP > Static Routes**. Click the **+** icon to add the entries.



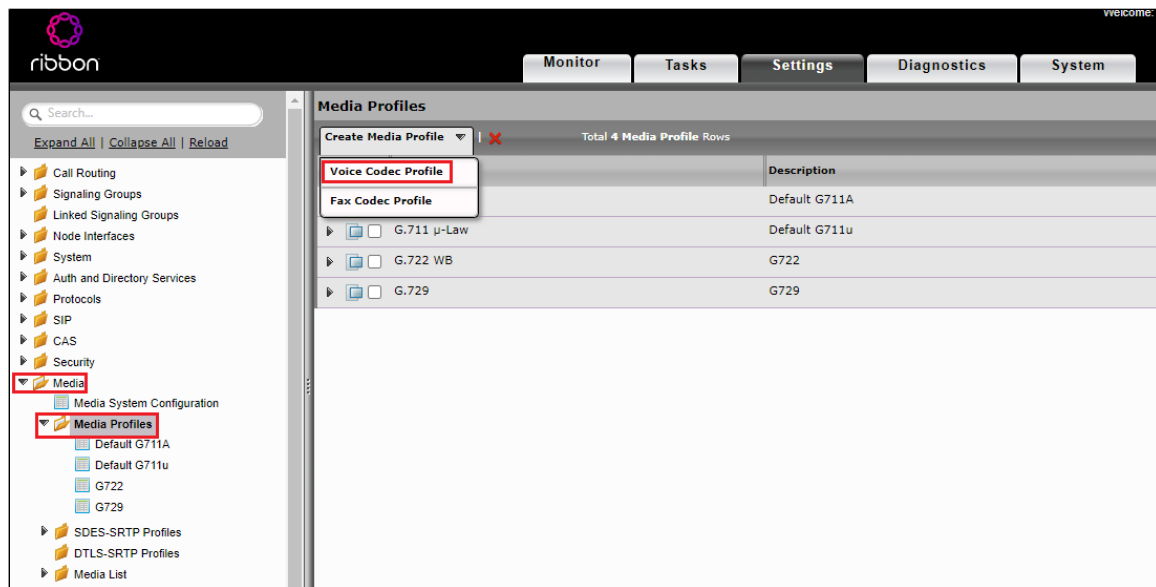
Row ID	Destination IP	Mask	Gateway	Administrative Distance	Primary Key
1	172.16.1.0	255.255.255.0	10.54.1.1	1	1
2	74.125.1.0	255.255.255.0	115.110.1.1	1	2
3	216.239.1.0	255.255.255.255	115.110.1.1	1	3
4	8.8.8.8	255.255.255.255	115.110.1.1	1	4
5	10.70.1.0	255.255.0.0	10.54.1.1	1	5

Global Configuration

Media Profiles

Media Profiles allow you to specify the individual voice and fax compression codecs and their associated settings, for inclusion in a Media List. Different codecs provide varying levels of compression, allowing one to reduce bandwidth requirements at the expense of voice quality.

From the **Settings** tab, navigate to **Media > Media Profiles**. From the **Create Media Profile** drop-down, select **Voice Codec Profile**.



Create Media Profile	Description
Voice Codec Profile	Default G711A
Fax Codec Profile	Default G711u
G.711 µ-Law	Default G711u
G.722 WB	G722
G.729	G729

The codecs G711A and G711U are configured on the SBC SWe Edge by default. Configure OPUS and G722 by following the steps provided below:



Note

OPUS is supported on the Ribbon SBC SWe Edge but not on the SBC 1K. During the 1K configuration, ignore the step below that describes the procedure to configure OPUS codec.

For OPUS:

1. Provide the profile's description.
2. Select OPUS from the Codec drop-down menu.

3. Configure 111 as the Payload Type.
4. Click OK.

For G722:

1. Provide the profile's description.
2. Select G.722 from the Codec drop-down menu.
3. Click OK.

Create Voice Codec Profile

Voice Codec Configuration

Description

Codec OPUS ▼

Rate b/s [6000..51000]

Payload Size ms

Payload Type 111 [96..127]

Voice Bit Rate VBR ▼

Use FEC False ▼

Use DTX False ▼

OK

Create Voice Codec Profile

Voice Codec Configuration

Description

Codec G.722 ▼

Rate 64000 b/s

Payload Size 20 ms

OK

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. In addition, Transformation tables are configurable as a reusable pool that Action Sets can reference.

From the Settings tab, navigate to **Call Routing > Transformation**. Click the + icon to create a Transformation Table.

1. Provide a name for the Transformation Table in the Description field.
2. Click OK.

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

- Call Routing
- Transformation
- Time of Day Table
- Call Routing Table
- Call Actions
- Signaling Groups
- Networking Interfaces
- System
- Auth and Directory Services
- Protocols
- SIP
- Security
- Media
- Tone Tables
- Telephony Mapping Tables
- SNMP/Alarms
- Logging Configuration
- Emergency Services

Monitor
Tasks
Settings
Diagnostics
System

Transformation

Total 1 Transformation Table Row

Description	Primary Key
Passthrough Untouched	1

Create Transformation Table


Row ID

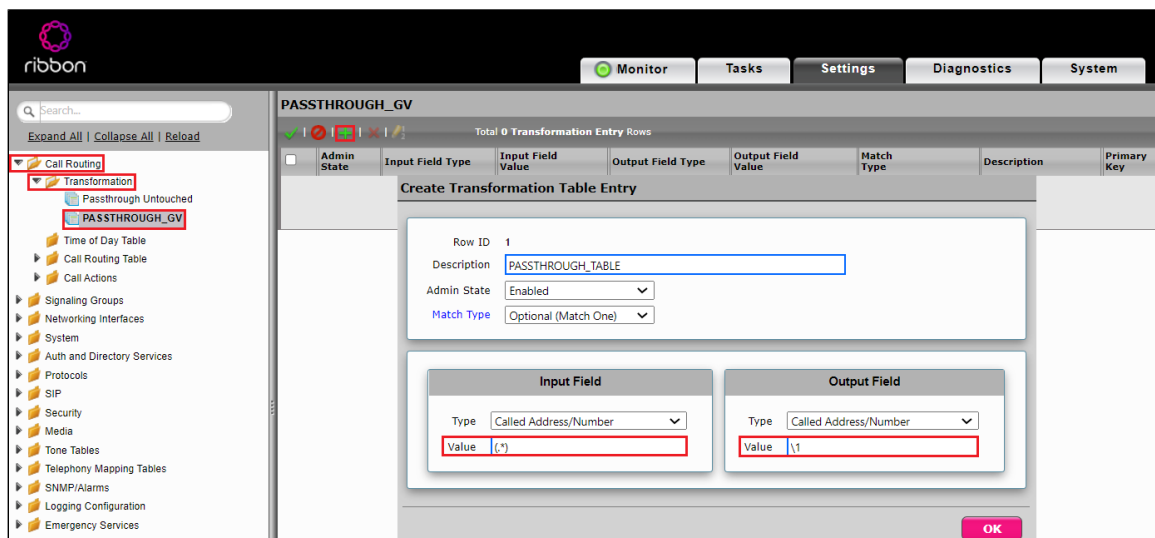
Description

OK

Transformation Table Entry

1. Click on the Transformation Table created in the previous step.


2. Click the  icon to create an entry.
3. Provide the values in Input and Output fields.
4. Click OK.



The screenshot shows the 'PASSTHROUGH_GV' configuration page in the Ribbon Communications interface. The left sidebar shows a tree view with 'Call Routing' expanded and 'PASSTHROUGH_GV' selected. The main area displays a 'Create Transformation Table Entry' form. The form includes fields for Row ID (1), Description (PASSTHROUGH_TABLE), Admin State (Enabled), and Match Type (Optional (Match One)). Below these are two sections: 'Input Field' and 'Output Field'. The 'Input Field' section has a Type dropdown set to 'Called Address/Number' and a Value field containing '(*)'. The 'Output Field' section has a Type dropdown set to 'Called Address/Number' and a Value field containing '\1'. An 'OK' button is at the bottom right.

SBC SWe Edge Configuration for PSTN side

Media List - PSTN

From the Settings tab, navigate to **Media > Media List**. Click the  icon at the top of the Media List View page.

1. Provide a name for the profile.
2. Attach the Media Profiles by clicking Add/Edit.
3. Enable Dead Call Detection.
4. From the DTMF drop-down menu, select RFC2833.
5. Click OK.

The screenshot displays the 'Media List View' in the Ribbon Communications settings. The left sidebar shows a tree view with 'Media' expanded, and 'Media List' selected. The main panel shows a table with 3 rows: 'Default Media List' (Primary Key 1), 'GOOGLE' (Primary Key 2), and 'PSTN' (Primary Key 3). The 'PSTN' row is expanded, showing configuration details:

- Description:** PSTN
- Media Profiles List:** A list containing 'Default G711u', 'Default G711A', 'OPUS', and 'G722'. The 'Add/Edit' button is highlighted.
- SDES-SRTP Profile:** None
- Media DSCP:** 46
- Dead Call Detection:** Enabled
- Silence Suppression:** Enabled

Below the main configuration, there are sections for 'Digit Relay' and 'Passthrough/Tone Detection'.

Digit Relay:

- Digit (DTMF) Relay Type:** RFC 2833
- Digit Relay Payload Type:** 101

Passthrough/Tone Detection:

- Modem Passthrough:** Enabled
- Fax Passthrough:** Enabled
- Fax Tone Detection:** Disabled

Message Manipulation - PSTN

The Message Manipulation feature comprises two primary components that work in concert to modify SIP messages. Those components are Condition Rules and Rule Tables. SIP Message rules per table include all SIP rule types: Header, Request, Status and Raw.

The Message Manipulation PSTN_RULE is used for the following purposes:

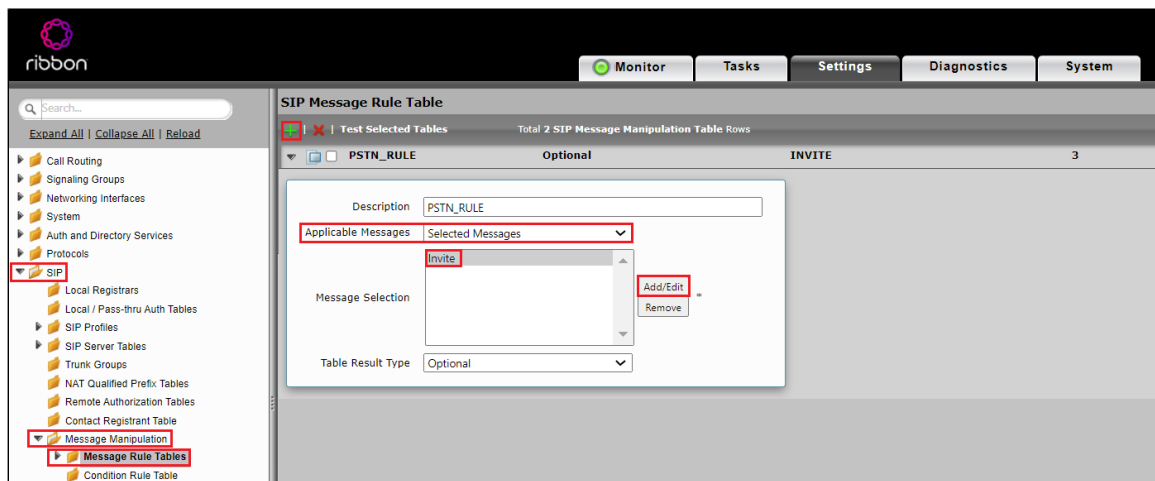
- To replace a=inactive with a=sendonly, as inactive is not supported on Google Voice
- To remove P-Preferred-Identity header to facilitate SWe Edge to send P-Asserted-Identity header to Google Voice instead of relaying P-Preferred-Identity received from PSTN

Message Rule Table

Message Rule can be added to: all messages, all requests, all responses or selected messages.

From the **Settings** tab, navigate to **SIP > Message Manipulation > Message Rule Table**. Click the **+** icon to create a Message Rule Table.

- Provide a description for the Rule Table.
- Apply Message Rule to the Selected messages and choose Invite from the Message Selection list.
- Click OK.

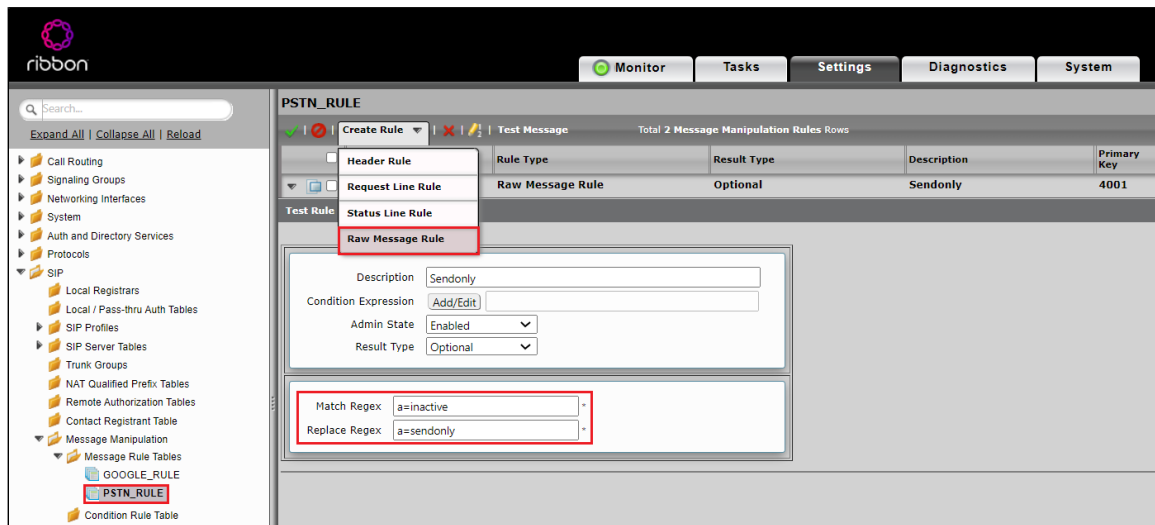


Message Rule Table Entry

Raw Message Rule:

Raw rules allow you to manipulate any string in the entire message: request, headers and payload. If the condition rule evaluates true, the MME will search the message for content matching the "Match Regex" and replace it with the content specified in the "Replace Regex".

1. Click on the Message Rule Table PSTN_RULE.
2. From the Create Rule drop-down menu, select Raw Message Rule.
3. Provide a name for the entry.
4. Replace a=inactive with a=sendonly using regex.



Header Rule:

1. Click on the Message Rule Table PSTN_RULE.
2. From the Create Rule drop-down menu, select Header Rule.
3. Provide a name for the entry.
4. Remove P-Preferred-Identity header using the Action "Remove".

PSTN_RULE

Test Message Total 2 Message Manipulation Rules Rows

	Rule Type	Result Type	Description	Primary Key
Request Line Rule	Raw Message Rule	Optional	Sendonly	4001
Status Line Rule	Header Rule	Optional	REMOVE PPI	1

Test Rule: Raw Message Rule

Modal Window Fields:

- Description: REMOVE PPI
- Condition Expression: Add/Edit
- Admin State: Enabled
- Result Type: Optional
- Header Action: Remove
- Header Name: P-Preferred-Identity
- Header Ordinal Number: 1st

SIP Profile - PSTN

SIP Profiles control how SBC Edge communicates with SIP devices. They control important characteristics such as Session Timers, SIP Header Customization, SIP Timers, MIME Payloads and Option Tags.

From the **Settings** tab, navigate to **SIP > SIP Profiles**. Click the **+** icon to create a new SIP Profile.

1. Provide a name for the profile in the Description field.
2. Enable Session Timer. This field specifies whether or not to use Session Timer to verify the SIP session. The remainder of the options in this panel are visible only after enabling Session Timer.
3. Set Minimum Acceptable Timer to 90 and Offered Session Timer to 1800.
4. In the Options Tags panel, set the Timer field to Required and the Update field to Supported.
5. Click OK.

SIP Profile Table

Total 3 SIP Profile Rows

Description	Primary Key
Default SIP Profile	1
GOOGLE_SIP_PROFILE	2
PSTN_SIP_PROFILE	3

Description: PSTN_SIP_PROFILE

Session Timer

Session Timer: Enable

Minimum Acceptable Timer: 90 * secs [90..7200]

Offered Session Timer: 1800 * secs [90..7200]

Terminate On Refresh Failure: False

MIME Payloads

ELIN Identifier: LOC

PIDF-LO Passthrough: Enable

Unknown Subtype Passthrough: Disable

Header Customization

FQDN in From Header: Disable

FQDN in Contact Header: Disable

Send Assert Header: Trusted Only

SBC Edge Diagnostics Header: Enable

Trusted Interface: Enable

UA Header: Ribbon SBC Edge

Calling Info Source: RFC Standard

Diversion Header Selection: Last

Record Route Header: RFC 3261 Standard

Options Tags

100rel: Supported

Path: Not Present

Timer: Required

Update: Supported

Timers

Transport Timeout Timer: 5000 ms [5000..32000]

Maximum Retransmissions: RFC Standard

Redundancy Retry Timer: 180000 ms [5000..180000]

RFC Timers

Timer T1: 500 ms [100..10000]

Timer T2: 4000 ms [1000..80000](>= T1)

Timer T4: 5000 ms [1000..100000]

Timer D: 32000 ms [5000..640000]

Timer B: 32000 ms

Timer F: 32000 ms

Timer H: 32000 ms (64*TimerT1)

Timer J: 4000 ms [4000..640000]

SDP Customization

Send Number of Audio Channels: False

Connection Info in Media Section: True

Origin Field Username: SBC default: SBC

Session Name: VoipCall default: VoipCall

Digit Transmission Preference: RFC 2833/Voice

SDP Handling Preference: Legacy Audio/Fax

SIP Server Table - PSTN

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. The SIP Server supports either an FQDN or IP Address (V4 or V6).

From the **Settings** tab, navigate to **SIP > SIP Server Tables**. Click the **+** icon to create a new SIP Server Table.

1. Provide a name for the SIP Server.
2. From the Type drop-down menu, choose SIP Server.
3. Click OK.

SIP Server Tables

Total 3 SIP Server Table Rows

Description	Primary Key
Default SIP Server	1
PSTN	2

Description: PSTN

Type: SIP Server

SIP Server Table Entry

1. Click on the SIP Server Table created in the previous step.
2. From the Create SIP Server drop-down menu, select IP/FQDN.
3. Provide IP Address and Port Number of the PSTN endpoint.
4. Enable OPTION pings by selecting SIP Options from the Monitor field.
5. Click OK.

ribbon

Monitor Tasks Settings Diagnostics System

Search...

Expand All Collapse All Reload

Call Routing
Signaling Groups
Networking Interfaces
System
Auth and Directory Services
Protocols
SIP
Local Registrars
Local / Pass-thru Auth Tables
SIP Profiles
SIP Server Tables
Default SIP Server
PSTN
GOOGLE
Trunk Groups
NAT Qualified Prefix Tables
Remote Authorization Table
Contact Registrant Table
Message Manipulation
Node-Level SIP Settings
SIP Recording
Security
Media
Tone Tables
Telephony Mapping Tables
SNMP/Alarms
Logging Configuration
Emergency Services

PSTN

Create SIP Server Total 0 SIP Server Rows

IP/FQDN	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
DNS-SRV						

-- Table is empty --

Create SIP Server Entry

Server Host

Row ID 1
Server Lookup IP/FQDN
Priority 1
Host FQDN/IP 10.54
Port 5060
Protocol UDP

Transport

Monitor SIP Options
Keep Alive Frequency 30
Recover Frequency 5
Local Username PSTN
Peer Username PSTN

Remote Authorization and Contacts

Remote Authorization Table None
Contact Registrant Table None
Session URI Validation Liberal

OK

Call Routing Table - PSTN

Call Routing allows calls to be carried between Signaling Groups, thus allowing calls to be carried between ports and between protocols (like ISDN to SIP). Routes are defined by Call Routing Tables, which allow flexible configuration of how calls are to be carried and how they are translated. These tables are the central connection points of the system, linking [Transformation Tables](#), [Message Translations](#), [Cause Code Reroute Tables](#), [Media Lists](#) and the Signaling Groups.

From the **Settings** tab, navigate to **Call Routing > Call Routing Table**. Click the **+** icon to create a Call Routing Table.

1. Provide a name for the Routing Table.
2. Click OK.

ribbon

Monitor Tasks Settings Diagnostics System

Search...

Expand All Collapse All Reload

Call Routing
Transformation
Time of Day Table
Call Routing Table
Call Actions
Signaling Groups
Networking Interfaces
System

Call Routing Tables

Total 3 Call Routing Table Rows

Description	Primary Key
Default Route Table	1
PSTN_TO_GV	2

Description PSTN_TO_GV

SIP Signaling Group - PSTN

Signaling groups allow telephony channels to be grouped together for the purposes of routing and shared configuration. They are the entity to which calls are routed, as well as the location from which [Call Routes](#) are selected. They are also the location from which [Tone Tables](#) and [Action Sets](#) are selected.

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

1. Attach the Call Routing Table ([PSTN_TO_GV](#)).

2. Attach the SIP Profile ([PSTN_SIP_PROFILE](#)).
3. Attach the SIP Server Table ([PSTN](#)).
4. Attach the Media List ID ([PSTN](#)).
5. Associate the appropriate IP address in the "Signaling/Media Source IP" field.
 - a. This specifies the Logical IP address at which SIP messages are received.
 - b. This address is used as the source IP for all SIP messages leaving the SBC SWe Edge or SBC 1000/2000 through this Signaling Group
6. Configure Protocol and Listen Ports in the "Listen Ports" panel.
7. Create an entry in the Federated IP/FQDN panel.
 - a. Federated IP addresses and FQDNs specified in a SIP Signaling Group are whitelisted.
 - b. The Federated IP/FQDN feature acts as an access control by defining from which server a SIP Signaling Group will accept messages.
8. Enable Message Manipulation and attach the profile [PSTN_RULE](#) to Inbound Message Manipulation Table List.
 - a. This option allows the SBC to manipulate SIP messages using previously configured Message Tables.
9. Click OK.

The screenshot displays the Ribbon SBC configuration interface. The top navigation bar includes tabs for Monitor, Tasks, Settings, Diagnostics, and System. The left sidebar shows a tree view of configuration categories, with 'Signaling Groups' highlighted. The main content area is titled 'Signaling Group Table' and shows a table with one row for 'PSTN_SG'. Below the table, there are several configuration sections:

- SIP Channels and Routing:** Includes settings for Action Set Table (None), Call Routing Table (PSTN_TO_GV), No. of Channels (60), SIP Profile (PSTN_SIP_PROFILE), SIP Mode (Basic Call), Agent Type (Back-to-Back User Agent), SIP Server Table (PSTN), Load Balancing (Round Robin), Channel Hunting (Most Idle), Notify Lync CAC Profile (Disable), Challenge Request (Disable), Outbound Proxy IP/FQDN (5060), Call Setup Response Timer (255), Call Proceeding Timer (180), Use Register as Keep Alive (Enable), and Forked Call Answered Too Soon (Disable).
- SIP Recording:** Includes SIP Recording Status (Disabled).
- Media Information:** Includes Supported Audio Modes (Proxy, Direct, Proxy with Local SRTP), Supported Video/Application Modes (Proxy, Direct), Media List ID (PSTN), Proxy Local SRTP (None), Crypto Profile ID (None), Play Ringback (Auto on 180), Tone Table (Default Tone Table), Play Congestion Tone (Disable), Early 183 (Disable), Allow Refresh SDP (Enable), Music on Hold (Disabled), and RTPC Multiplexing (Disable).
- Mapping Tables:** Includes SIP To Q.850 Override Table (Default (RFC4497)), Q.850 To SIP Override Table (Default (RFC4497)), and Pass-thru Peer SIP Response Code (Enable).
- SIP IP Details:** Includes Teams Local Media Optimization (Disable), Signaling/Media Source IP (Ethernet 1 IP (10.54.10.63)), Signaling DSCP (40), NAT Traversal (Disabled), ICE Support (Disabled), Static NAT - Outbound (None), Outbound NAT Traversal (None), Static NAT - Inbound (None), and Detection (Disabled).
- Listen Ports:** Includes a table with one row for UDP on port 5060.
- Federated IP/FQDN:** Includes a table with one row for IP/FQDN 10.54.10.63 with Netmask/Prefix 255.255.255.255.
- Message Manipulation:** Includes Message Manipulation (Enabled), Inbound Message Manipulation (PSTN_RULE), and Outbound Message Manipulation (empty).

SBC SWe Edge Configuration for Google Voice SIP Link side DNS

From the **Settings** tab, navigate to **System > Node-Level Settings**.

1. From the Use Primary DNS drop-down menu, select Yes.
2. Provide the Primary DNS IP address.
3. Select the Ethernet facing Google Voice SIP Link from the Primary Source drop-down menu.
4. Click Apply.

Node-Level Settings

Set Date/Time | Backup Config | Restore Config | Clear DNS Cache

Host Information

Host Name: SWeLite-Google

Domain Name: rbbnriot.com

System Information

System Description:

System Location:

System Contact:

Time Management

Time Zone: (GMT+5:30) India, Sri Lanka

Network Time Protocol

Use NTP: Yes

NTP Server: 172.16.1.1 * IPv4/6 Address or FQDN

NTP Server Authentication: Disabled

NTP Server 2: No

Country Level Information

Country Code: None

Domain Name Service

Use Primary DNS: Yes

Primary Server IP: 8.8.8.8 * xxx.xxx.x.xxx

Primary Source: Ethernet 2 IP (115.110.1.1)

Use Secondary DNS: No

EdgeView

EdgeView: No

TLS Profile

TLS Profiles are used by SIP Signaling Groups when the TLS transport type is selected for incoming and outgoing SIP trunks (Listen Ports), and in SIP Server Tables when TLS is selected as the Server Host protocol.

From the **Settings** tab, navigate to **Security > TLS Profiles**. Click the **+** icon to create a new TLS profile.

1. From TLS Protocol drop-down menu, select TLS 1.0-1.2.
2. Add the cipher suites that are supported on Google Voice SIP Link (TLS_ECDHE_RSA_WITH_AES_256_GCM_SHA384 and TLS_ECDHE_RSA_WITH_AES_128_GCM_SHA256).
3. Disable the Validate Server and Client FQDN fields.
4. Click OK.

TLS Profile

Total 2 TLS Profile Rows

Description	Primary Key
Default TLS Profile	1
GOOGLE_TLS	2

Description: GOOGLE_TLS

TLS Parameters

Common Attributes

TLS Protocol: TLS 1.0-1.2

Mutual Authentication: Enabled

Handshake Inactivity Timeout: 30 secs [1..30]

Certificate: SBC Edge Certificate

Client Attributes

Client Cipher List:

- TLS_ECDHE_RSA_WITH_AES_256_GCM_SHA384
- TLS_ECDHE_RSA_WITH_AES_128_GCM_SHA256
- TLS_ECDHE_RSA_WITH_AES_256_CBC_SHA384
- TLS_ECDHE_RSA_WITH_AES_128_CBC_SHA256
- TLS_ECDHE_RSA_WITH_3DES_EDE_CBC_SHA
- TLS_RSA_WITH_AES_256_CBC_SHA256
- TLS_RSA_WITH_AES_128_CBC_SHA256
- TLS_RSA_WITH_AES256_CBC_SHA
- TLS_RSA_WITH_3DES_EDE_CBC_SHA

Validate Server FQDN: Disabled

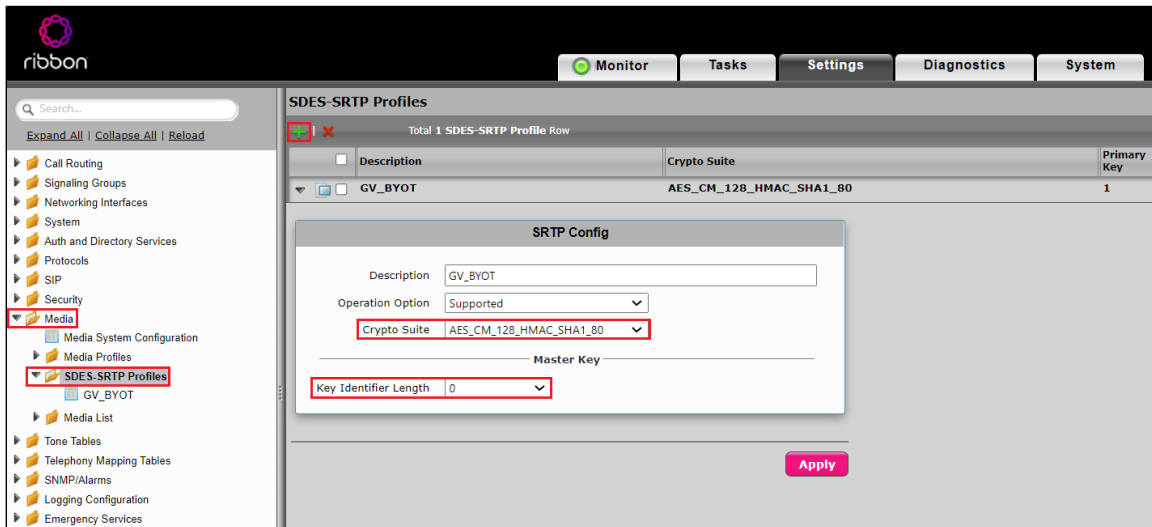
Validate Client FQDN: Disabled

SDES-SRTP Profile

SDES-SRTP Profiles define a cryptographic context which is used in SRTP negotiation. SDES-SRTP Profiles are required for enabling media encryption and are applied to Media Lists.

From the **Settings** tab, navigate to **Media > SDES-SRTP Profiles**. Click the **+** icon to create a new SDES-SRTP profile.

1. Provide a name for the profile in the Description field.
2. Attach the Crypto suite "AES_CM_128_HMAC_SHA1_80", a crypto suite algorithm which uses the 128 bit AES-CM encryption key and a 80 bit HMAC_SHA1 message authentication tag length.
3. Set the Key Identifier Length to 0 to disable the MKI in SDP.
4. Click OK.



Note

Google Voice does not support MKI, hence the Key Identifier Length must be set to 0 on the Ribbon SBC SWe Edge.

Media List - GV

From the Settings tab, navigate to **Media > Media List**. Click the **+** icon at the top of the Media List View page

1. Provide a name for the profile.
2. Attach the Media Profiles by clicking Add/Edit.
3. Attach the SDES-SRTP profile ([GV_BYOT](#)).
4. Enable Dead Call Detection.
5. From the DTMF drop-down menu, select RFC2833.
6. Click OK.

Media List View

Total 3 Media List Rows

Description	Primary Key
Default Media List	1
GOOGLE	2

Description: GOOGLE

Media Profiles List: Default G711u, Default G711A, OPUS, G722

SDP-SRTP Profile: GV_BYOT

Media DSCP: 46

Dead Call Detection: Enabled

Silence Suppression: Enabled

Digit Relay

Digit (DTMF) Relay Type: RFC 2833

Digit Relay Payload Type: 101

Passthrough/Tone Detection

Modem Passthrough: Enabled

Fax Passthrough: Enabled

Fax Tone Detection: Disabled

Message Manipulation - GV

The Message Manipulation GOOGLE_RULE is used for the following purposes:

- To add the header "X-Google-Pbx-Trunk-Secret-Key" for Google Voice. The value of this header is generated when the SIP Trunk is created.
- To change the request URI of specific request messages to Google specified FQDN, trunk.sip.voice.google.com.
- To modify the FQDN in the To header to trunk.sip.voice.google.com.

Message Rule Table

From the **Settings** tab, navigate to **SIP > Message Manipulation > Message Rule Table**. Click the **+** icon to create a Message Rule Table.

- Provide a description for the Rule Table.
- Apply Message Rule to the selected messages and choose Invite, Cancel, Options and ACK from the Message Selection list.
- Click OK.

SIP Message Rule Table

Total 2 SIP Message Manipulation Table Rows

Description	Result Type	Message Type	Primary Key
GOOGLE_RULE	Optional	INVITE CANCEL	1

Description: GOOGLE_RULE

Applicable Messages: Selected Messages

Message Selection: Invite, Cancel, Options, ACK

Table Result Type: Optional

Message Rule Table Entry

Header Rule:

1. Click on the Message Rule Table GOOGLE_RULE.
2. From the Create Rule drop-down menu, select **Header Rule**.
3. Provide a name for the entry.
4. Add the header "X-Google-Pbx-Trunk-Secret-Key".
5. To add the value, select **Add** from the Header Value drop-down menu and provide the literal value of the header.
6. Click **OK**.

The screenshot shows the Ribbon Communications web interface. On the left is a sidebar with a tree view of system components. The top navigation bar includes tabs for Monitor, Tasks, Settings, Diagnostics, and System. The main content area displays the configuration for a Message Rule Table named 'GOOGLE_RULE'. A table lists the rules, and a 'Create Rule' dropdown menu is open, showing options like Header Rule, Request Line Rule, Status Line Rule, and Raw Message Rule. The 'Header Rule' option is selected. Below this, a form for the 'Header Rule' is displayed with fields for Description, Condition Expression, Admin State, Result Type, Header Action, and Header Name. The 'Header Action' is set to 'Add' and the 'Header Name' is 'X-Google-Pbx-Trunk-Secret-Key'. Below the form, there is a section for 'Header Value' with a dropdown set to 'Add' and a text field containing the value '43871d45-d275-4e90-b800-cd'. At the bottom, an 'Edit Message Field' dialog is open, showing 'Type of Value' set to 'Literal' and 'Value' set to '43871d45-d275-4e90-b800-cd'. The 'OK' button is highlighted.

Request Line Rule:

1. Click on the Message Rule Table GOOGLE_RULE.
2. From the Create Rule drop-down menu, select Request Line Rule.
3. Provide a name for the entry.
4. Replace the FQDN "siplink.telephony.goog" with "trunk.sip.voice.google.com" using regex.
5. Click OK.

The screenshot shows the Ribbon Communications interface. On the left is a navigation tree with categories like Call Routing, Signaling Groups, System, Auth and Directory Services, Protocols, SIP, and Security. The 'SIP' category is expanded, and 'Message Manipulation' is selected. Under 'Message Manipulation', 'Message Rule Tables' is expanded, and 'GOOGLE_RULE' is highlighted. The main panel displays the 'GOOGLE_RULE' configuration. At the top, there's a 'Create Rule' dropdown menu with options: Header Rule, Request Line Rule (selected), Status Line Rule, and Raw Message Rule. Below this is a table with 5 columns: Rule Type, Result Type, Description, and Primary Key. The table contains 2 rows: 'Request Line Rule' (Header Rule, Optional, Google_header, 1) and 'Status Line Rule' (Request Line Rule, Optional, Request_uri_FQDN, 2001). Below the table is a 'Test Rule' section. The 'Request Line Rule' is selected, and the 'Edit Message Field' dialog is open. The dialog has fields for 'Type of Value' (Regex), 'Match Regex' (siplink.telephony.goog), and 'Replace Regex' (trunk.sip.voice.google.com). There are 'OK' and 'Cancel' buttons at the bottom.

Header Rule:

1. Click on the Message Rule Table GOOGLE_RULE.
2. From the Create Rule drop-down menu, select Header Rule.
3. Provide a name for the entry.
4. Select Header Action as Modify and choose To from the Header Name list.
5. Replace the FQDN "siplink.telephony.goog" with "trunk.sip.voice.google.com" using regex.
6. Click OK.

The screenshot shows the Ribbon Communications interface. On the left is a navigation tree with categories like Call Routing, Signaling Groups, System, Auth and Directory Services, Protocols, SIP, and Security. The 'SIP' category is expanded, and 'Message Manipulation' is selected. Under 'Message Manipulation', 'Message Rule Tables' is expanded, and 'GOOGLE_RULE' is highlighted. The main panel displays the 'GOOGLE_RULE' configuration. At the top, there's a 'Create Rule' dropdown menu with options: Header Rule (selected), Request Line Rule, Status Line Rule, and Raw Message Rule. Below this is a table with 5 columns: Rule Type, Result Type, Description, and Primary Key. The table contains 3 rows: 'Request Line Rule' (Header Rule, Optional, Google_header, 1), 'Status Line Rule' (Request Line Rule, Optional, Request_uri_FQDN, 2001), and 'Raw Message Rule' (Header Rule, Optional, FQDN for To, 2). Below the table is a 'Test Rule' section. The 'Raw Message Rule' is selected, and the 'Edit Message Field' dialog is open. The dialog has fields for 'Type of Value' (Regex), 'Match Regex' (siplink.telephony.goog), and 'Replace Regex' (trunk.sip.voice.google.com). There are 'OK' and 'Cancel' buttons at the bottom.

SIP Profile - GV

From the **Settings** tab, navigate to **SIP > SIP Profiles**. Click the **+** icon to create a new SIP Profile.

1. Provide a name for the profile in the Description field.
2. Enable Session Timer.
3. Set the Minimum Acceptable Timer to 90 and the Offered Session Timer to 1800.
4. In the Options Tags panel, set the Timer field to Required and the Update field to Supported.
5. Click OK.

The screenshot displays the 'SIP Profile Table' configuration page in the Ribbon interface. The left sidebar contains a navigation tree with 'SIP Profiles' selected. The main configuration area is titled 'SIP Profile Table' and shows a list of profiles. The 'GOOGLE_SIP_PROFILE' is selected, and its configuration is displayed in a form. The 'Session Timer' section is highlighted with a red box, showing 'Session Timer' set to 'Enable', 'Minimum Acceptable Timer' set to '90', and 'Offered Session Timer' set to '1800'. The 'Options Tags' section is also highlighted with a red box, showing 'Timer' set to 'Required' and 'Update' set to 'Supported'.



Note

The session will always be refreshed by Ribbon SBC SWe Edge as per the Google Voice requirement.

SIP Server Table - GV

From the **Settings** tab, navigate to **SIP > SIP Server Tables**. Click the **+** icon to create a new SIP Server Table.

1. Provide a name for the SIP Server.
2. From the Type drop-down menu, choose SIP Server.
3. Click OK.

SIP Server Tables

Total 3 SIP Server Table Rows

Description	Primary Key
Default SIP Server	1
PSTN	2
GOOGLE	3

Description:

Type:

SIP Server Table Entry

1. Click on the SIP Server Table created in the previous step.
2. From the Create SIP Server drop-down menu, select IP/FQDN.
3. Provide the IP Address and the Port Number of the PSTN endpoint.
4. Enable OPTION pings by selecting SIP Options from the Monitor field.
5. Click OK.

GOOGLE

Create SIP Server: Total 1 SIP Server Row

Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
siplink.telephony.goog	IP/FQDN	5672	TLS	Counters	1	1

Server Host

Server Lookup:

Priority:

Host FQDN/IP:

Host IP Version:

Port: * [1..65535]

Protocol:

TLS Profile:

Transport

Monitor:

Keep Alive Frequency: * secs [30..300]

Recover Frequency: * secs [5..300]

Local Username: * Local Username of SBC Edge

Peer Username: * Peer Username of sip server

Remote Authorization and Contacts

Remote Authorization Table:

Contact Registrant Table:

Session URI Validation:

Connection Reuse

Reuse:

Sockets:

Reuse Timeout:



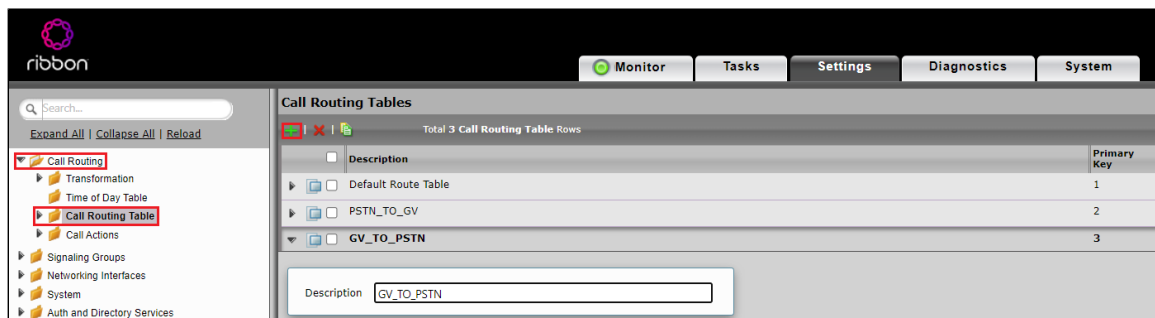
Note

For production, the Google Voice (GV) hostname is siplink.telephony.goog.

Call Routing Table - GV

From the **Settings** tab, navigate to **Call Routing** > **Call Routing Table**. Click the **+** icon to create a Call Routing Table.

1. Provide a name for the Routing Table.
2. Click OK.



SIP Signaling Group - GV

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

1. Attach the Call Routing Table ([GV_TO_PSTN](#)).
2. Attach the SIP Profile ([GOOGLE_SIP_PROFILE](#)).
3. Attach the SIP Server Table ([GOOGLE](#)).
4. Attach the Media List ID ([GOOGLE](#)).
5. Select the SDES-SRTP Profile [GV_BYOT](#) in the Proxy Local SRTP Crypto Profile ID field.
6. Associate the appropriate IP address in the "Signaling/Media Source IP" field.
7. Configure the Protocol, TLS Listen Ports and TLS Profile ([GOOGLE_TLS](#)) in the "Listen Ports" panel.
8. Provide the Google Voice SIP Link's FQDN or IP address in the Federated IP/FQDN panel.
9. Enable Message Manipulation and attach the profile [GOOGLE_RULE](#) to the Outbound Message Manipulation Table List.
10. Click OK.



Note

Ignore step 5 if you are configuring SBC 1K.



Warning

Ensure the "TLS Listen Port" towards the Google Voice Server is always set to 5061 in order to accept the incoming messages from Google Voice Server using an already established TLS connection with the Ribbon SBC.

The screenshot displays the Ribbon Communications configuration interface. The left sidebar shows a navigation tree with 'Signaling Groups' highlighted. The main area is titled 'Signaling Group Table' and shows a table with two rows: PSTN_SG and GOOGLE_SG. The GOOGLE_SG row is selected, and its configuration is shown in the right-hand panels.

Signaling Group Table

Type	Description	Admin State	Service Status	Display	Primary Key
SIP	PSTN_SG	Up	Up	Counters Channels Sessions	1
SIP	GOOGLE_SG	Up	Up	Counters Channels Sessions	2

GOOGLE_SG Configuration

Description: GOOGLE_SG
Admin State: Enabled
Service Status: Up

SIP Channels and Routing

- Action Set Table: None
- Call Routing Table: GV_TO_PSTN
- No. of Channels: 60
- SIP Profile: GOOGLE_SIP_PROFILE
- SIP Mode: Basic Call
- Agent Type: Back-to-Back User Agent
- SIP Server Table: GOOGLE
- 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: GOOGLE
- Proxy Local SRTP Crypto Profile ID: GV_BYOT
- Play Ringback: Auto on 180
- Tone Table: Default Tone Table
- Play Congestion Tone: Disable
- Early 183: Disable
- Allow Refresh SDP: Enable
- Music on Hold: Disabled
- RTCP Multiplexing: Enable

Mapping Tables

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

SIP IP Details

- Teams Local Media Optimization: Disable
- Signaling/Media Source IP: Ethernet 2 IP (115.110.)
- Signaling DSCP: 40
- NAT Traversal:
 - ICE Support: Disabled
 - Static NAT - Outbound: Outbound NAT Traversal: None
 - Static NAT - Inbound: Detection: Disabled

Listen Ports

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

Federated IP/FQDN

IP/FQDN	Netmask/Prefix
siplink.telephony.goog	255.255.255.255

Message Manipulation

Message Manipulation: Enabled

Inbound Message Manipulation

Message Table List:

Outbound Message Manipulation

Message Table List: GOOGLE_RULE

Call Routing Table Entry

Call Routing entries must to be created after creating SIP Signaling Groups as Destination SGs need to be attached to these entries.

PSTN to GV:

1. Click the **Create Routing Entry** () icon.
2. Attach the Transformation Table ([PASSTHROUGH_GV](#)).
3. Add the Destination Signaling Group which in this case is [GOOGLE_SG](#).

- In the Media panel, select DSP from the Audio Stream Mode and enable Media Transcoding.
- Click OK.

The screenshot displays the Ribbon Communications configuration interface. On the left is a navigation tree with categories like Call Routing, Signaling Groups, and Media. The main area is titled 'PSTN_TO_GV' and shows a table with one entry: 'PASSTHROUGH_GV' with priority 1. Below the table are three panels: 'Route Details', 'Destination Information', and 'Media'. The 'Route Details' panel shows fields for Description, Admin State, Route Priority, Call Priority, Number/Name Transformation Table (set to PASSTHROUGH_GV), and Time of Day Restriction. The 'Destination Information' panel shows Destination Type (Normal), Message Translation Table, Cause Code Reroutes, Cancel Others upon Forwarding, Fork Call, Destination Signaling Groups (set to (SIP) GOOGLE_SG), and Enable Maximum Call Duration. The 'Media' panel shows Audio Stream Mode (set to DSP), Video/Application Stream Mode (Disabled), Media Transcoding (Enabled), and Media List (Default Media List). A 'Quality of Service' panel on the right contains various metrics and thresholds.

Admin State	Priority	Transformation Table	Destination Type	First Signaling Group	Description	Fork Call	Primary Key
<input checked="" type="checkbox"/>	1	PASSTHROUGH_GV	Normal	(SIP) GOOGLE_SG	PSTN_TO_GV	No	1

Route Details

Description: PSTN_TO_GV
Admin State: Enabled
Route Priority: 1
Call Priority: Normal
Number/Name Transformation Table: PASSTHROUGH_GV
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) GOOGLE_SG
Enable Maximum Call Duration: Disabled

Media

Audio Stream Mode: DSP
Video/Application Stream Mode: Disabled
Media Transcoding: Enabled
Media List: Default Media List

Quality of Service

Quality Metrics Number of Calls: 10 [1..100]
Quality Metrics Time Before Retry: 10 [1..60] min.
Min. ASR Threshold: 0 % [0..100]
Enable Min MOS Threshold: Disabled
Enable Max. R/T Delay: Enabled
Max. R/T Delay: 65535 ms [1..65535]
Enable Max. Jitter: Enabled
Max. Jitter: 3000 ms [1..3000]

GV to PSTN :

- Click the **Create Routing Entry** () icon.
- Attach the Transformation Table ([PASSTHROUGH_GV](#)).
- Add the Destination Signaling Group [PSTN_SG](#).
- In the Media panel, select DSP from the Audio Stream Mode and enable Media Transcoding.
- Click OK.

Google Voice Configuration

For configuration on Google Voice, visit support.google.com/a?p=siplink.

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	Auto Attendant	✓
2	DTMF - RFC2833	✓
3	Basic Call Setup & Termination	✓
4	Calls to/from GV Android Client, Web Client and Desk-phone (OBi based)	✓
5	Long Duration Calls	✓
6	Session Timers	✓
7	Voice Mail Deposit and Retrieval	✓
8	4xx/5xx Response Handling	✓

9	Ring Group	✓
10	Call Hold/Resume	✓
11	Call Transfer (Attended)	✓
12	Call Transfer (Blind/ Unattended)	✓
13	Call Forwarding Unconditional	✓
14	Call Forward No Answer	✓
15	Call Cancel/Reject	✓
16	Short Code Dialing	✗

Legend

Supported	✓
Not Supported	✗

Caveats

The following items should be noted in relation to this Interop - these are either limitations, untested elements, or useful information pertaining to the Interoperability.

- Short Code calls are not supported on Google Voice clients.
- When GV rejects or does not answer the call from PSTN, the call is expected to connect to GV Voice Mail after 30 seconds. However, the SWe Edge sends a CANCEL to GV to terminate the call before it connects.

These issues will be addressed by GV/Ribbon in their upcoming releases.

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/services/ribbon-support-portal>

References

For detailed information about Ribbon products & solutions, please visit:

<https://ribboncommunications.com/products>

Conclusion

This Interoperability Guide describes successful configuration for Google Voice SIP Link interop involving the Ribbon SBC SWe Edge.

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.

