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

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Interoperable Vendors

Copyright

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

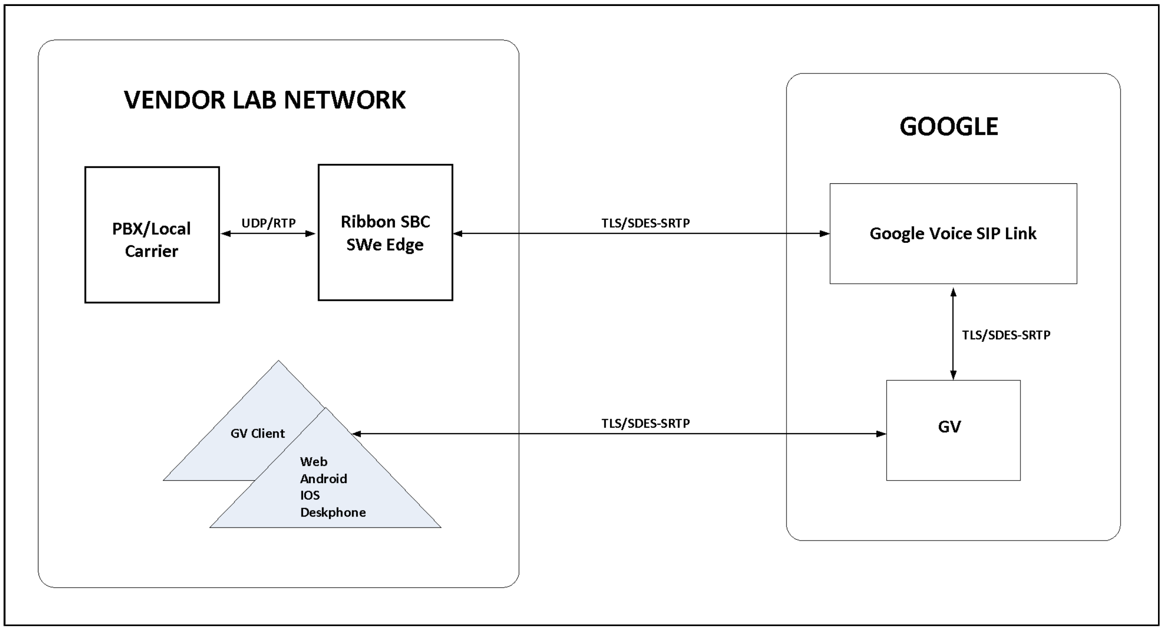
Product and Device Details

The configuration uses the following equipment and software:

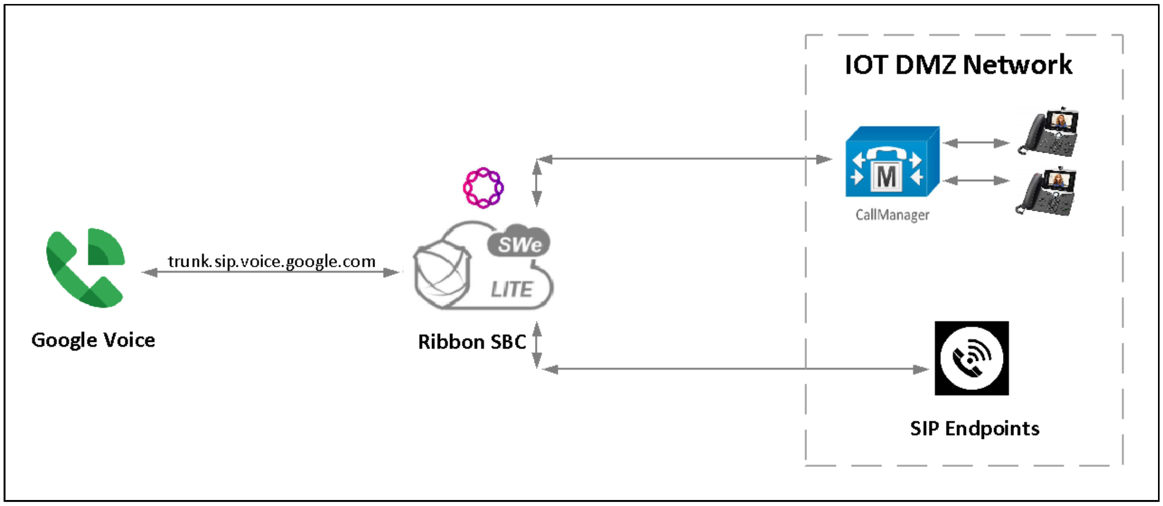
Product	Equipment/Service	Software Version
Ribbon SBC	Ribbon SBC SWe Edge	9.0.7
Google Voice SIP Link	Telephone Service	NA
Third-party Equipment	Cisco Unified Communications Manager	12.5.1.11900-146
Administration and Debugging Tools	Wireshark LX Tool	3.4.9 2.1.0.6

Network Topology and E2E Flow Diagrams

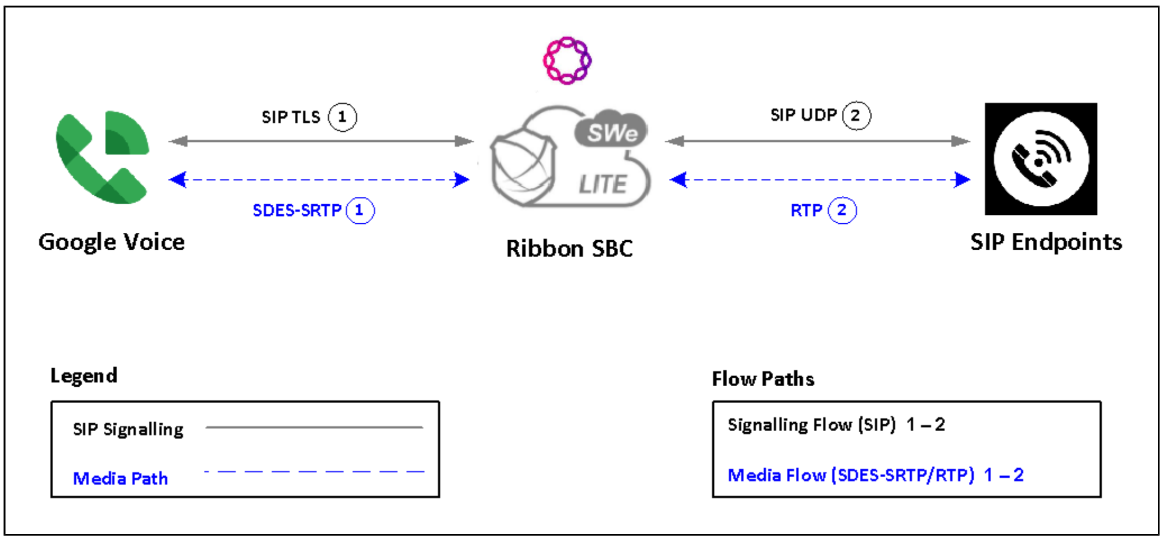
Deployment Topology



Interoperability Test Lab Topology

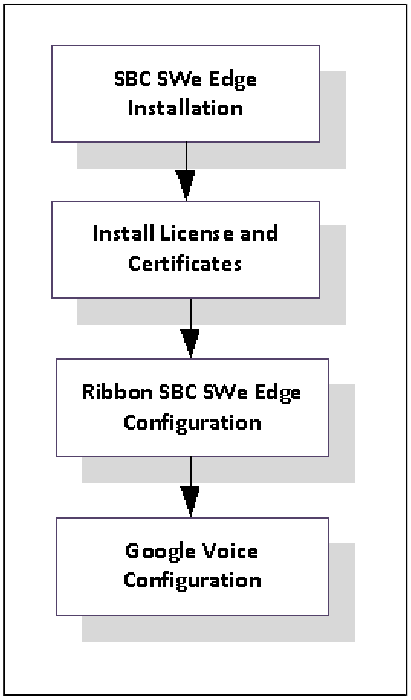


Call Flow Diagram



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

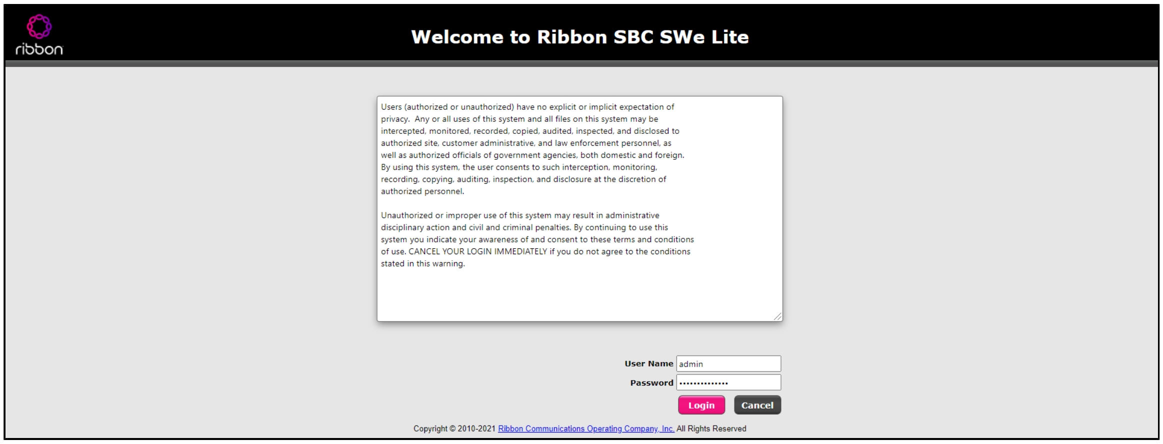
To deploy Ribbon SBC SWe Edge instance, refer to Installing SBC SWe Edge.

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.

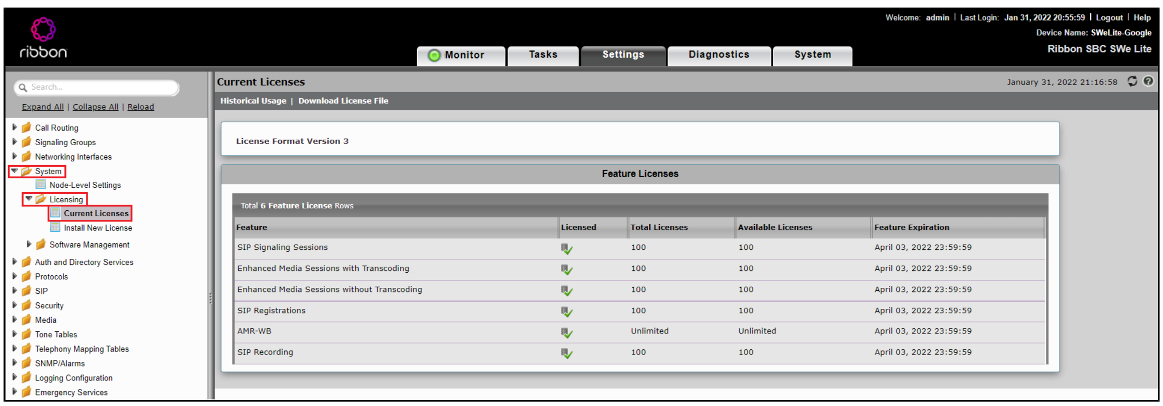


License and TLS Certificates

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** pan el 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.**

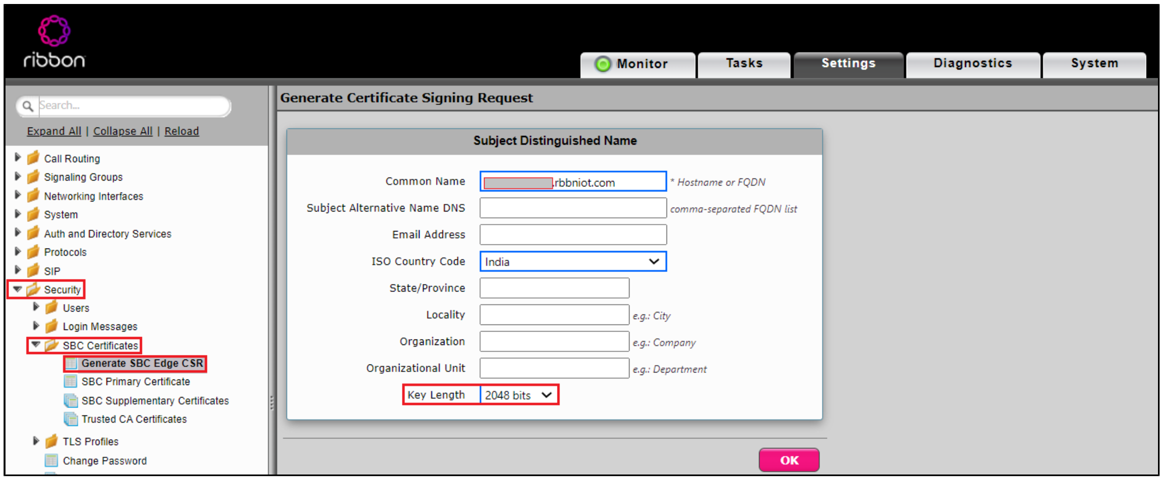


For more details on Licenses, refer to SWe Edge License

SBC Certificate

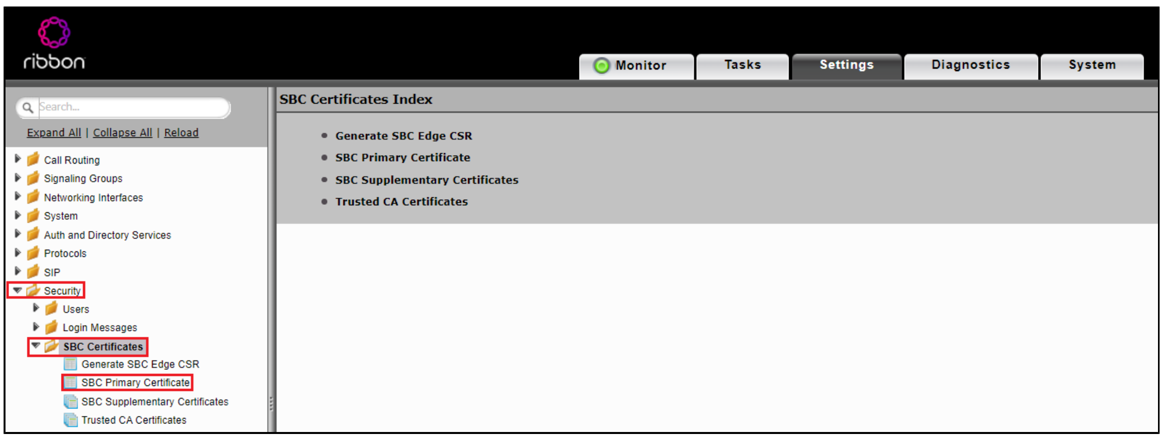
From the **Settings** tab, navigate to **Security > SBC Certificates > Generate SBC Edge Certificates.**

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.



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



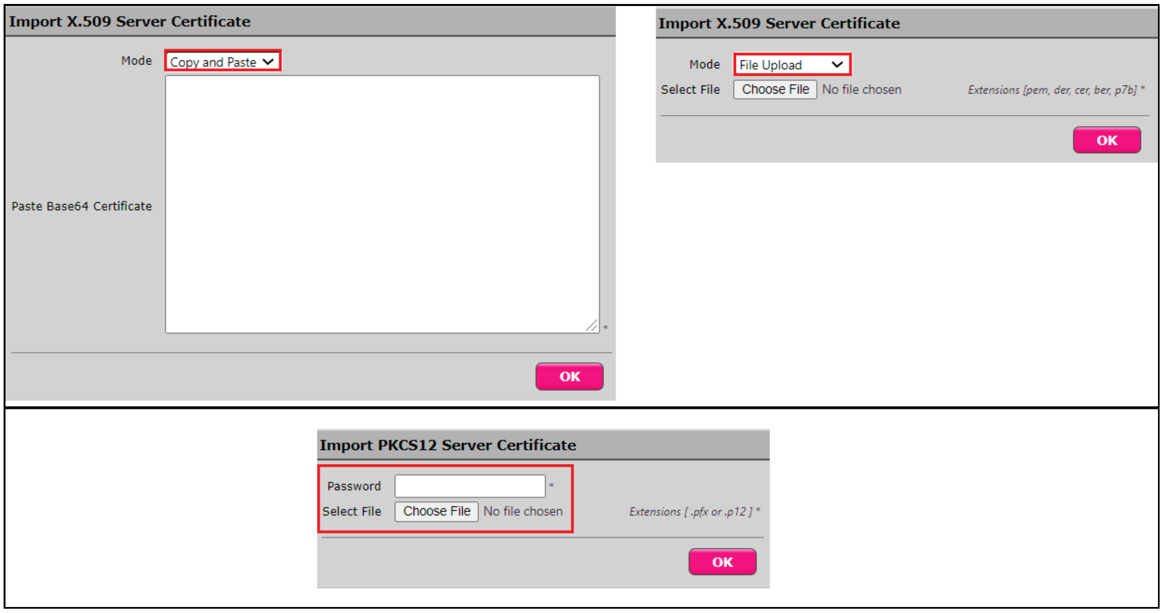
There are two ways to import SBC Primary Certificate as described below:

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.



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.

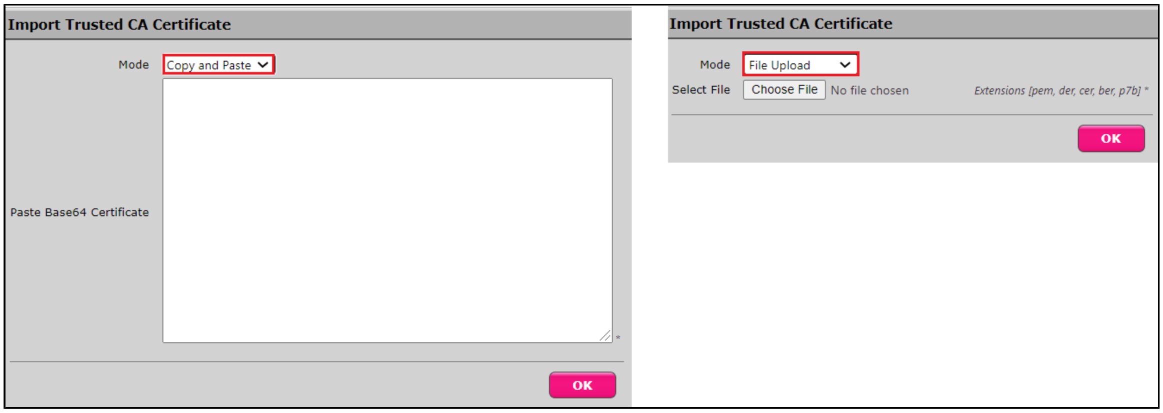
- For TLS to work, a Trusted CA (Certificate Authority) is required. For this interop, GoDaddy is used as Trusted CA.
- Add an entry in the Public DNS to resolve Ribbon SBC SWe Edge FQDN to Public IP Address.
- Ensure to have the following certificates as part of the root certificate trust:
  - GTS Root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

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 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

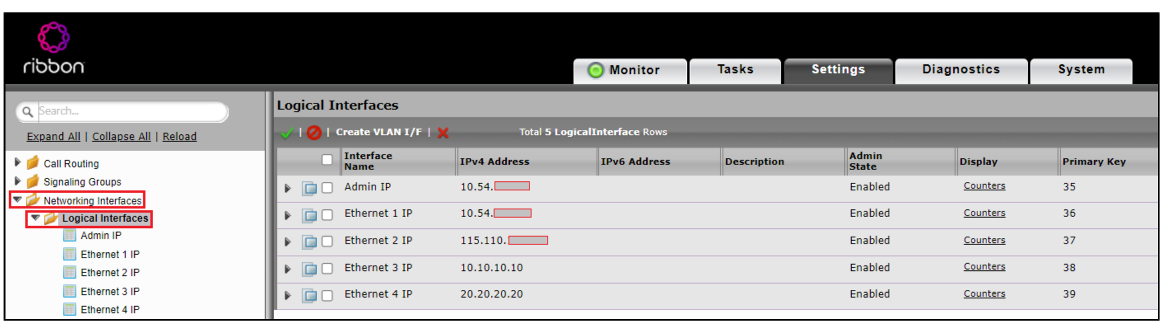
The SBC SWe Edge supports five system created logical interfaces known as Administrative IP, Ethernet 1 IP, Ethernet 2 IP, Ethernet 3 IP, and Ethernet 4 IP. In addition to the system created logical interfaces, the Ribbon SBC SWe Edge supports user created VLAN logical sub-interfaces.

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

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

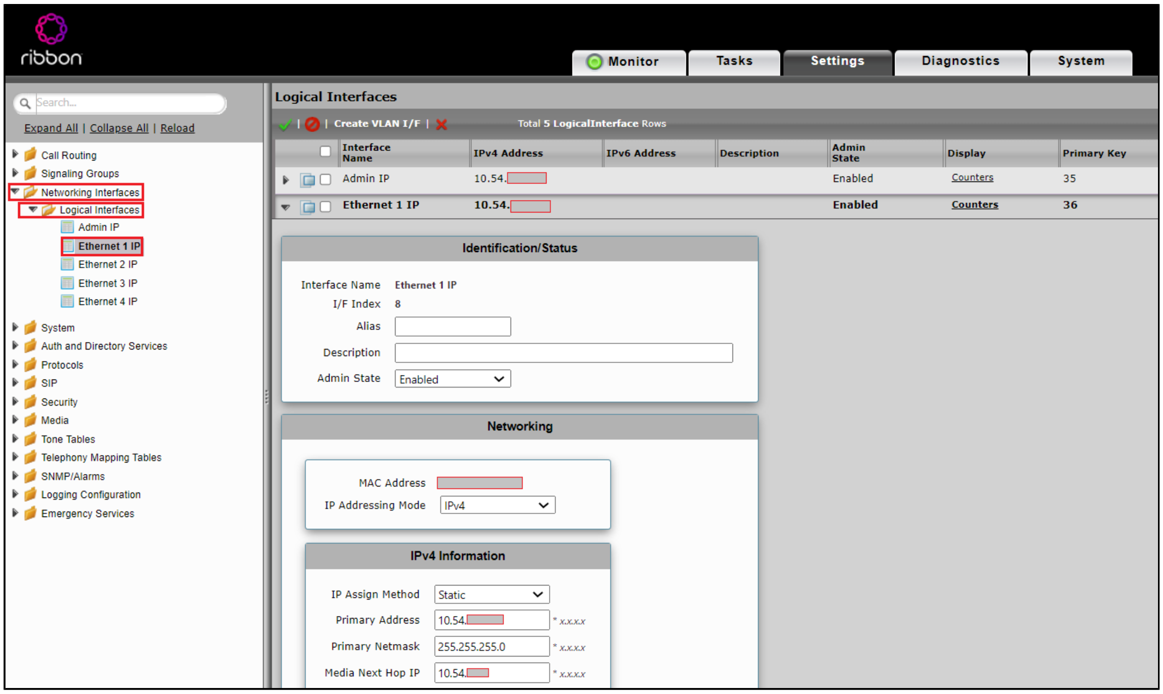
Administrative IP

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.



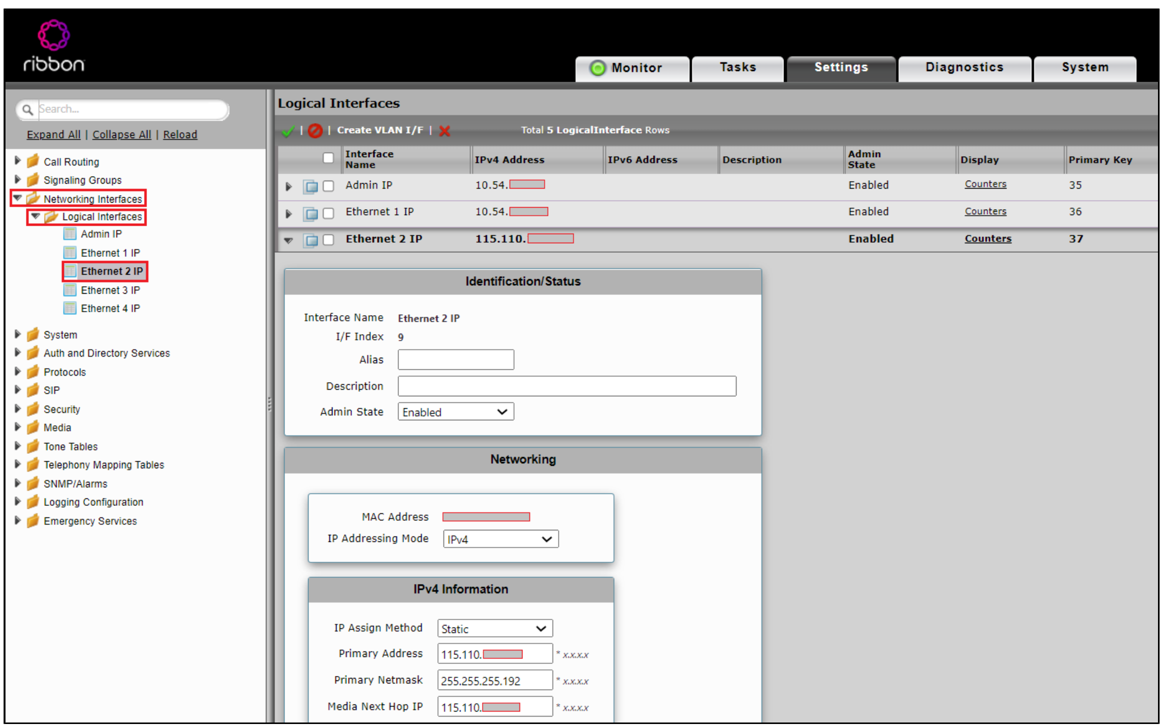
Ethernet 1 IP

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.



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.



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.

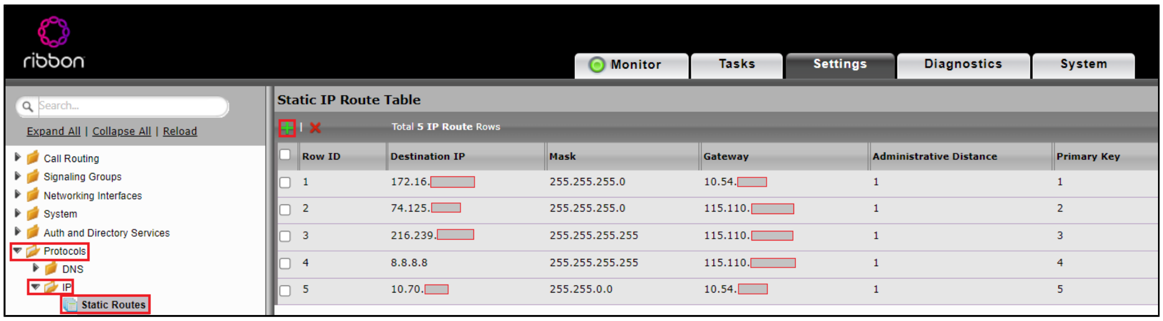
Destination IP
Specifies the destination IP address.

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.

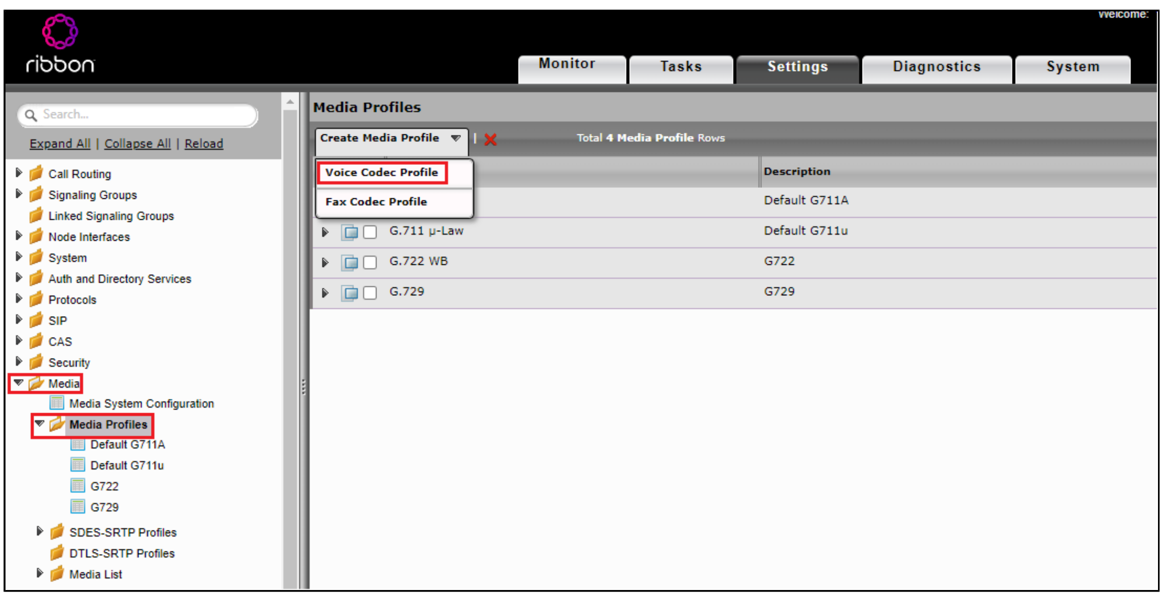


Global Configuration

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.**



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.

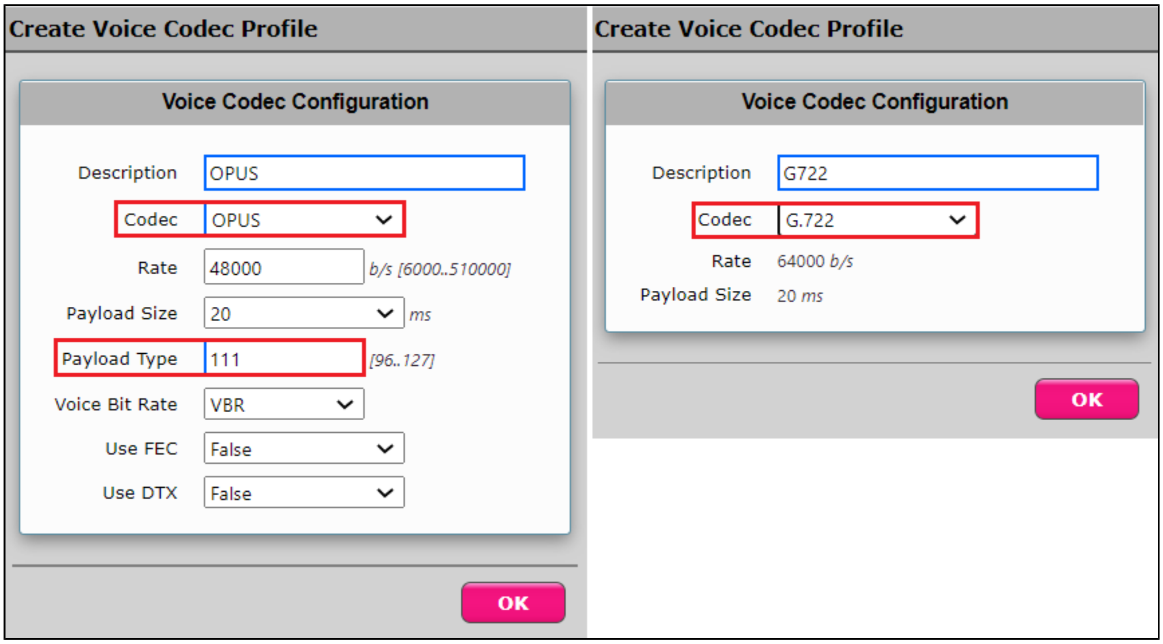
For OPUS:

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.

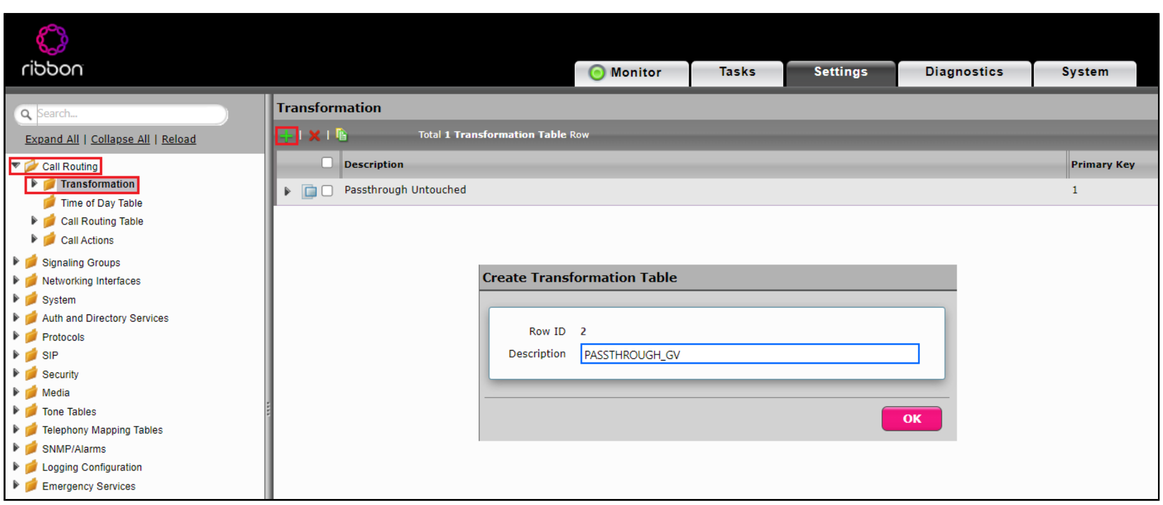


Transformation Table

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.

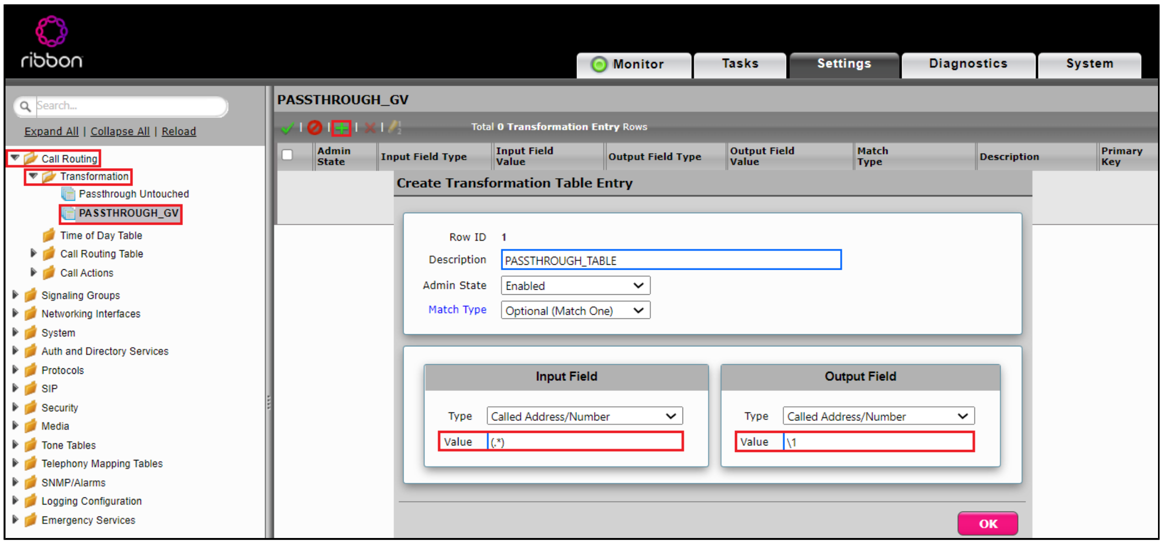
1. Provide a name for the Transformation Table in the Description field.
2. Click OK.



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.

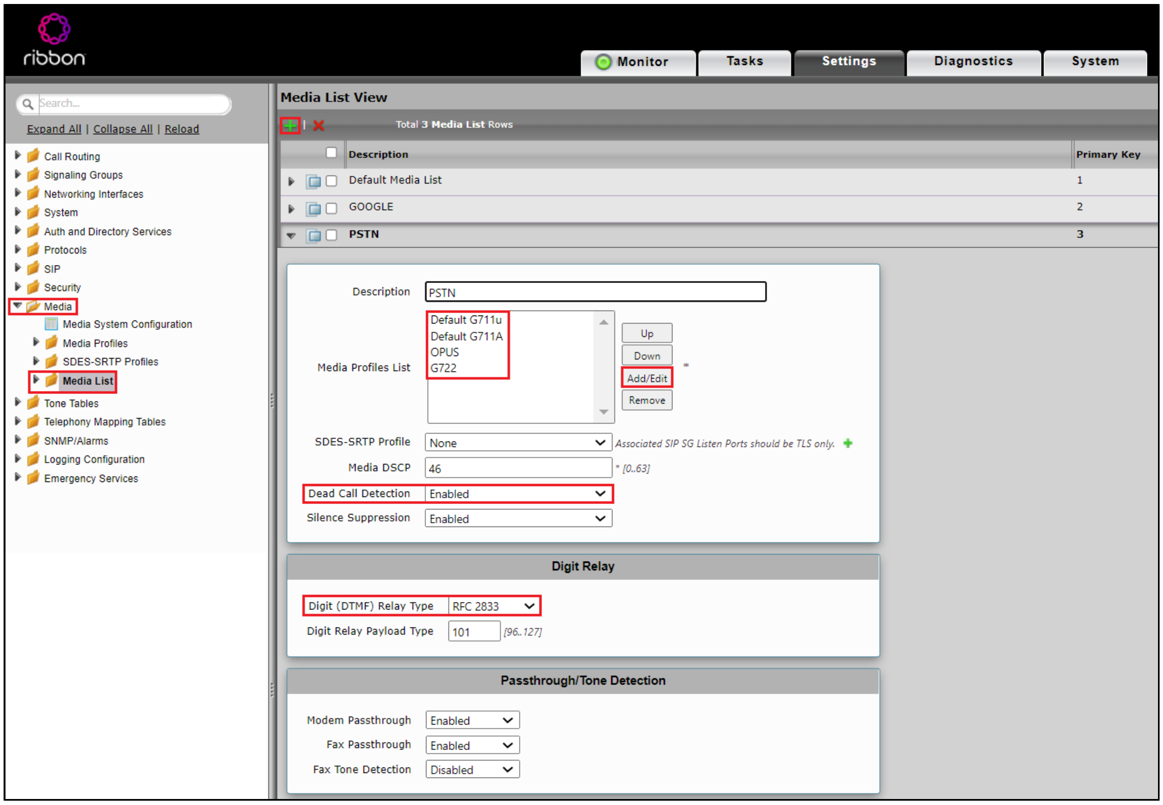


SBC SWe Edge Configuration for PSTN side

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.



Message Manipulation - PSTN

The Message Manipulation feature comprises two primary components that work in concert to modify SIP messages. Those component 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:

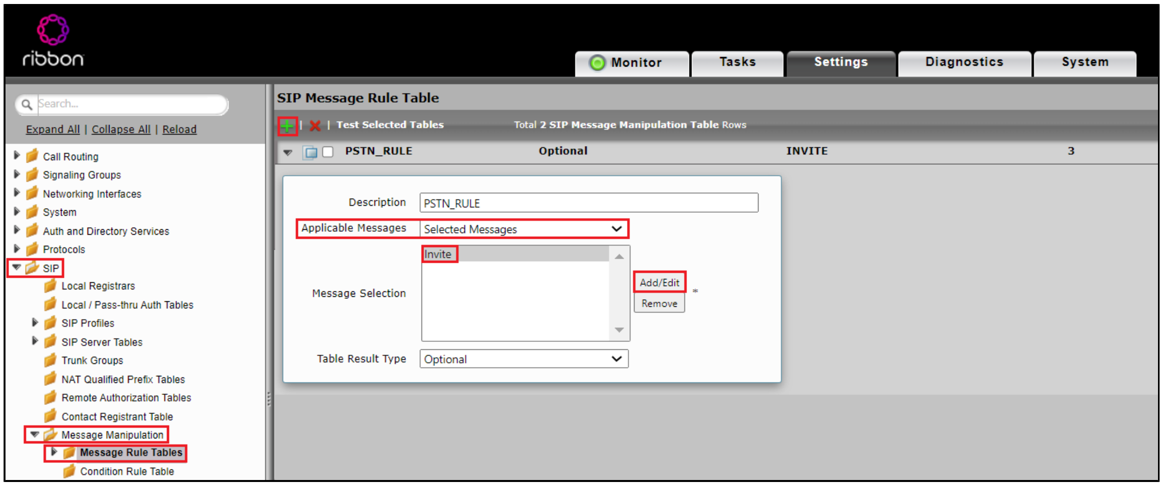
- To replace a=inactive with a=sendonly, as inactive is not supported on Google Voice
- To remove P-Preferred-Identity header to facilitate SWe Edge to send P-Asserted-Identity header to Google Voice instead of relaying P-Preferred-Identity received from PSTN

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.

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

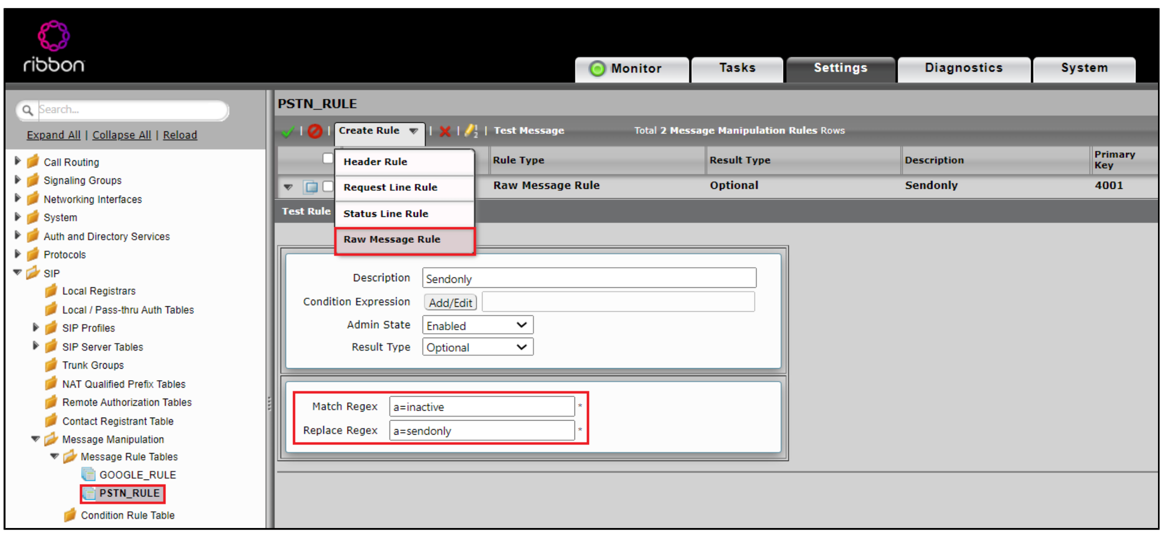


Message Rule Table Entry

Raw Message Rule:

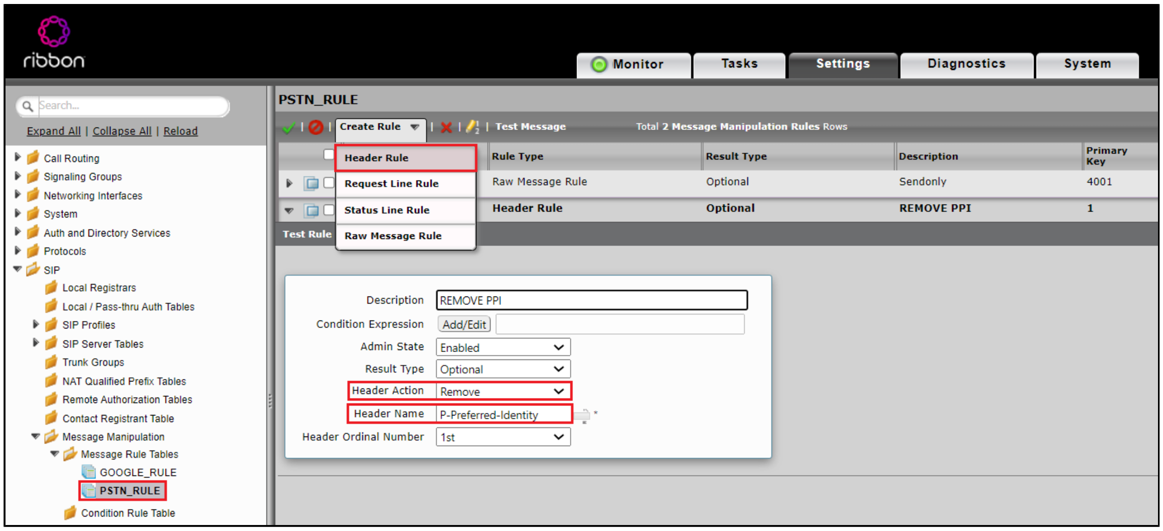
Raw rules allow you to manipulate any string in the entire message: request, headers and payload. If the condition rule evaluates true, the MME will search the message for content matching the "Match Regex" and replace it with the content specified in the "Replace Regex".

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".

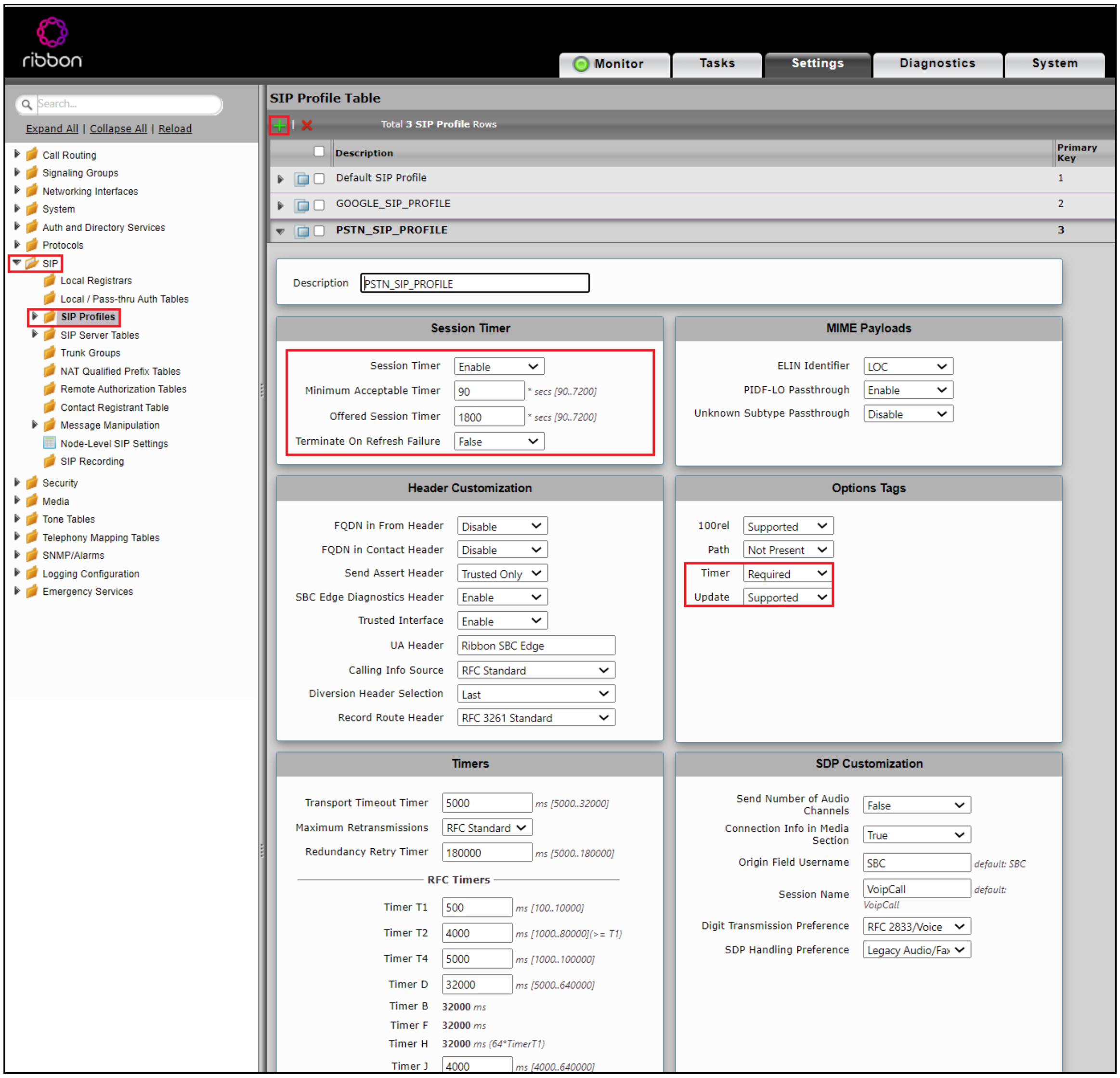


SIP Profile - PSTN

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.

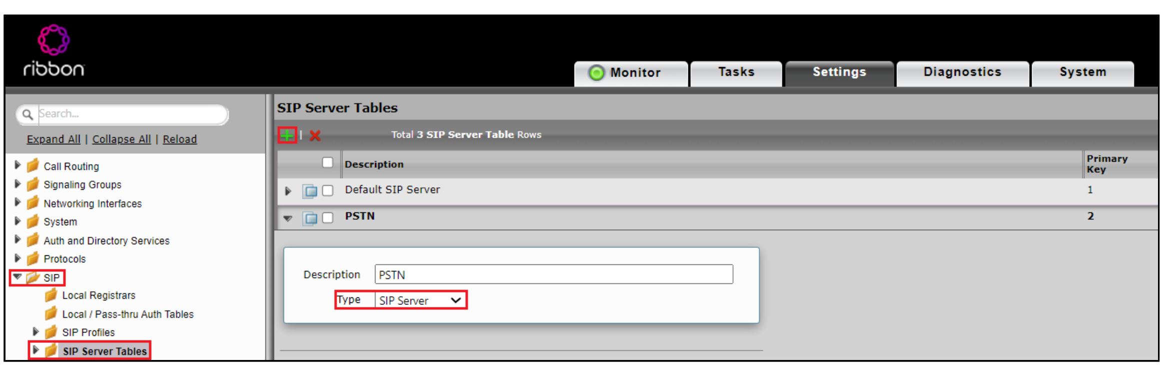


SIP Server Table - PSTN

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).

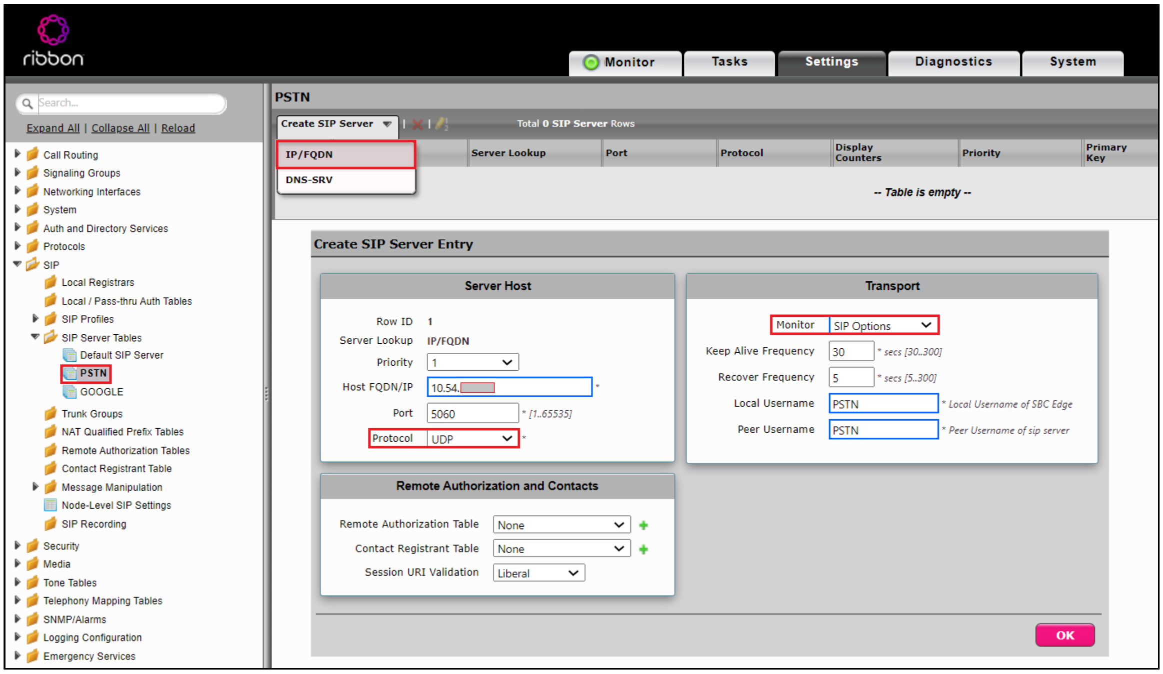
From the **Settings** tab, navigate to **SIP > SIP Server Tables.** Click the + icon to create a new SIP Server Table.

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



SIP Server 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.

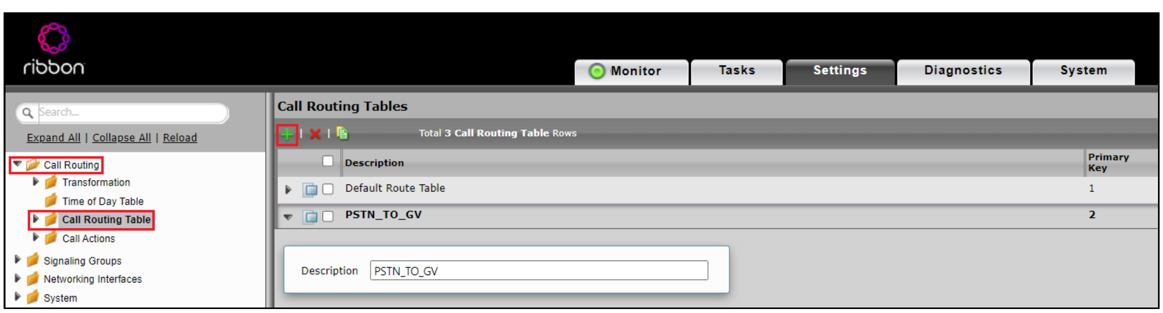


Call Routing Table - PSTN

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.



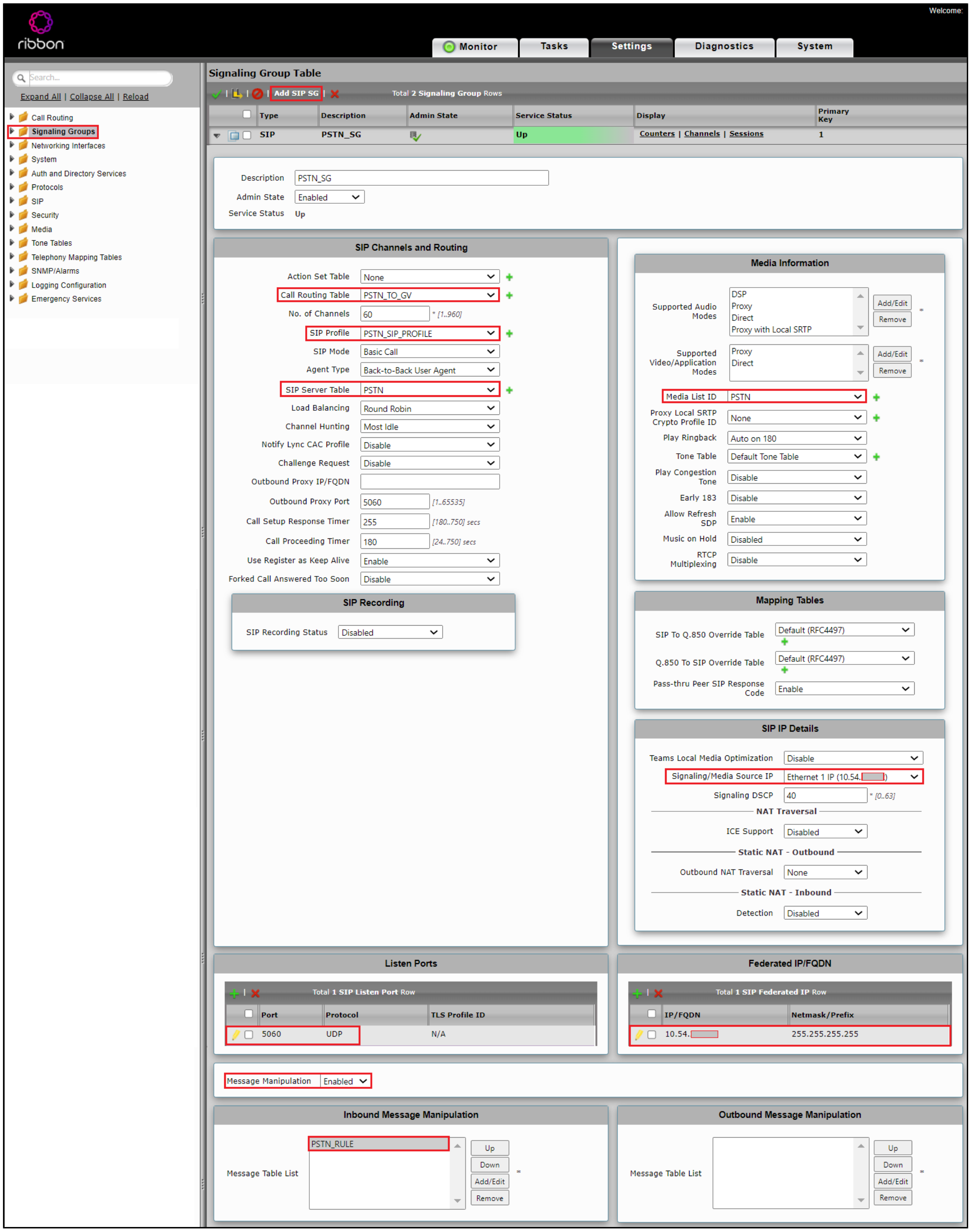
SIP Signaling Group - PSTN

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.

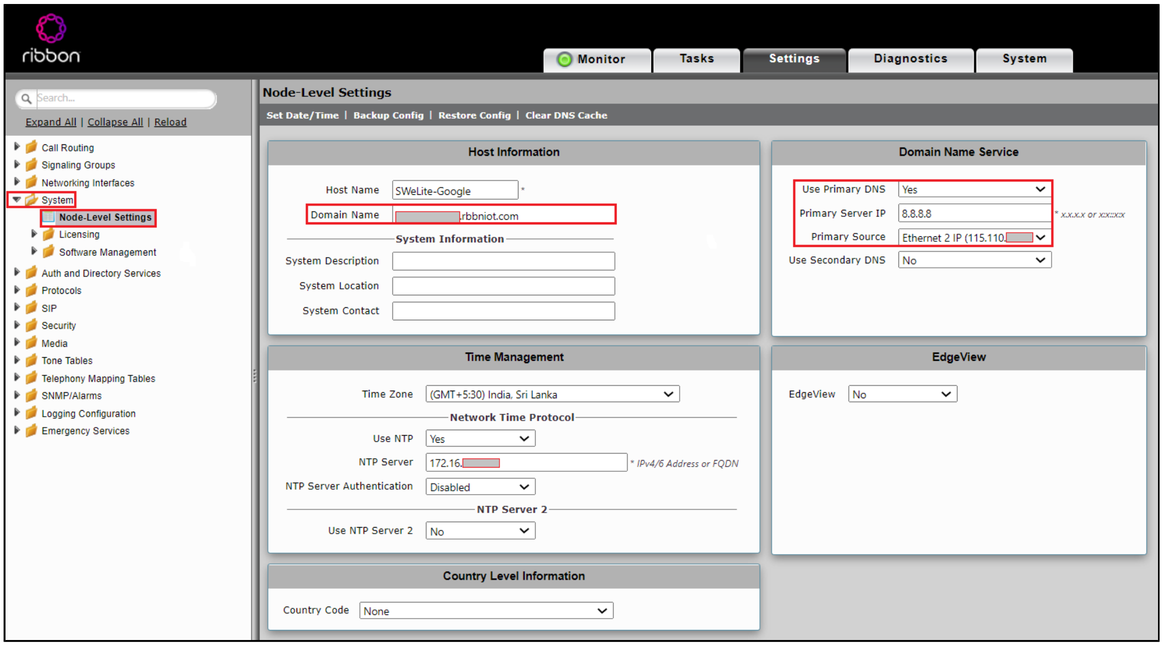


SBC SWe Edge Configuration for Google Voice SIP Link side

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.

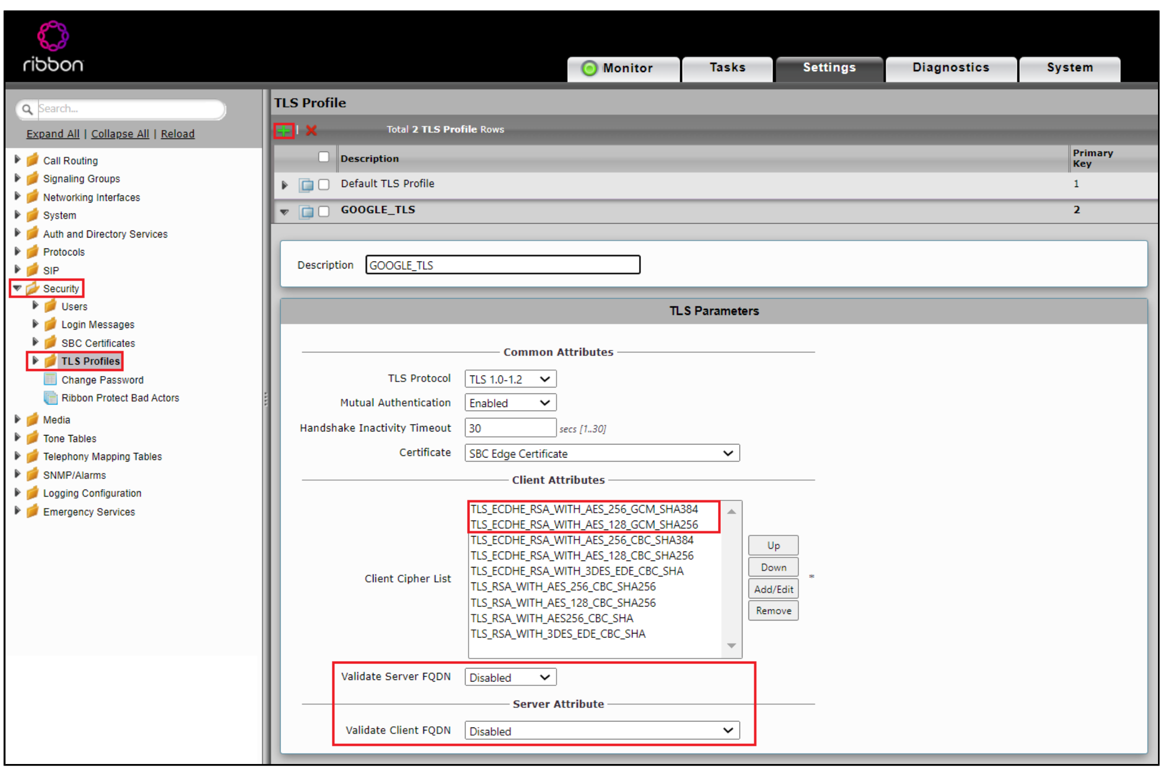


TLS Profile

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.

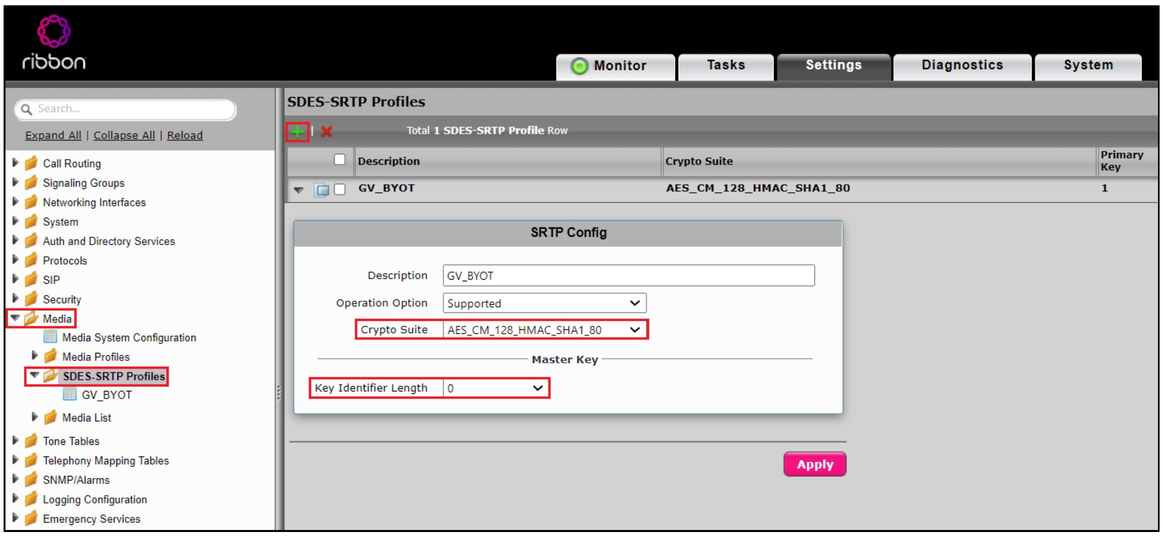


SDES-SRTP Profile

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_SHAI_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.

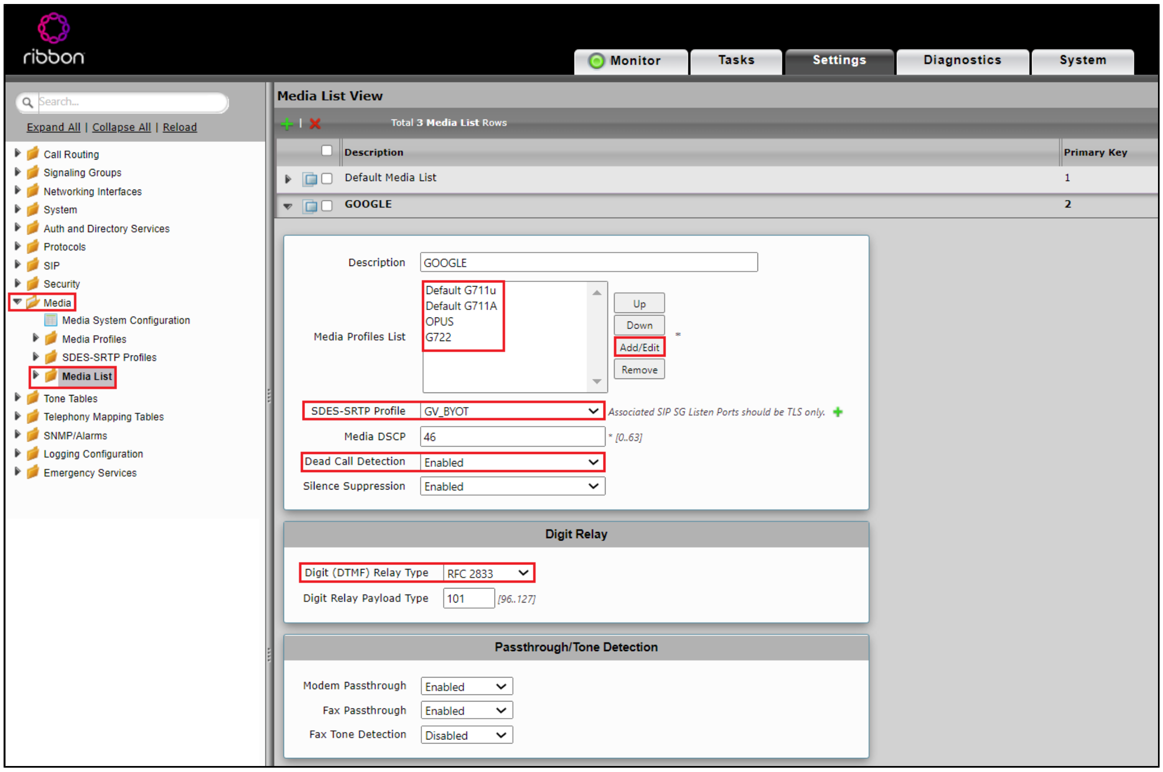


Note
Google Voice does not support MKI, hence the Key Identifier Length must be set to 0 on the Ribbon SBC SWe Edge.

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.



Message Manipulation - GV

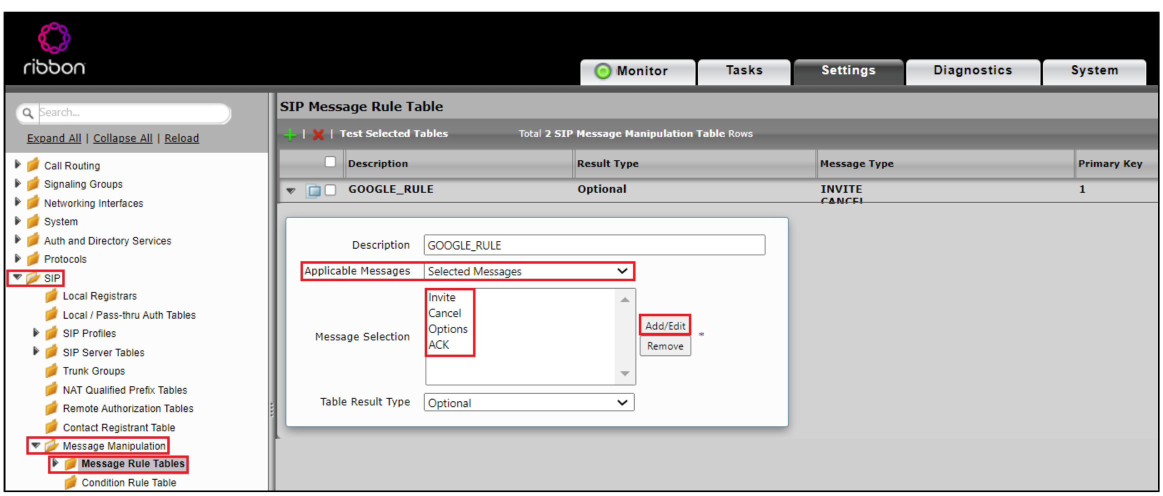
The Message Manipulation GOOGLE_RULE is used for the following purposes:

- 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.

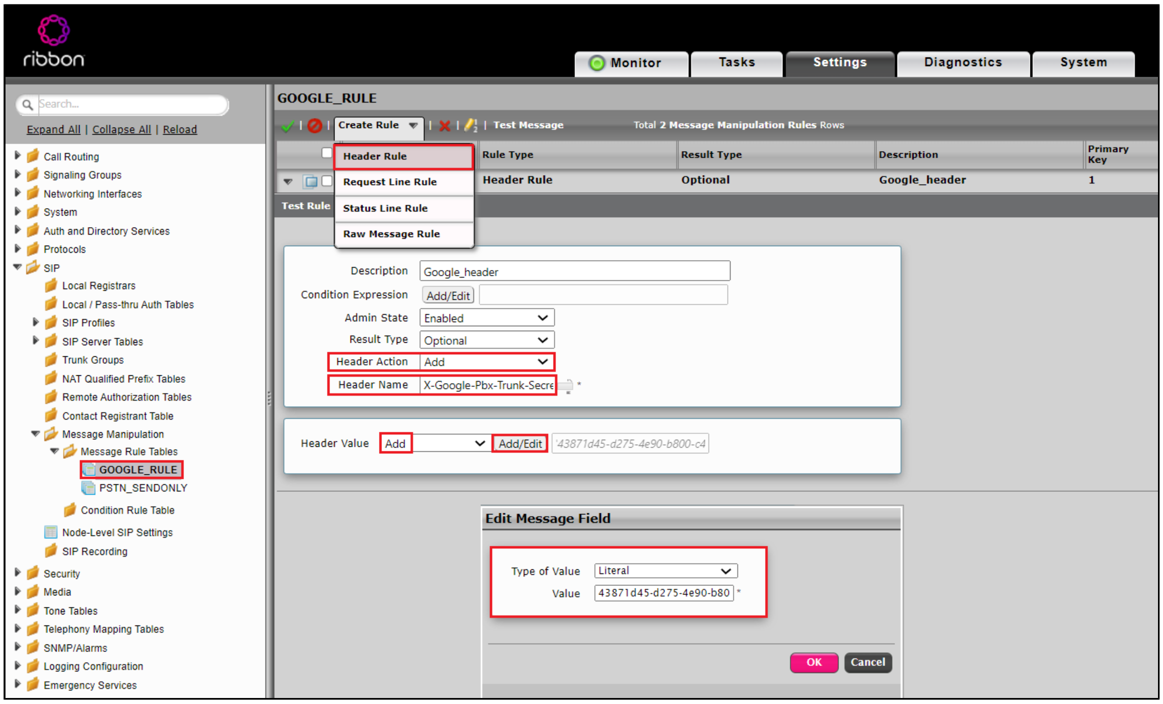
1. Provide a description for the Rule Table.
2. Apply Message Rule to the selected messages and choose Invite, Cancel, Options and ACK from the Message Selection list.
3. Click OK.



Message Rule Table Entry

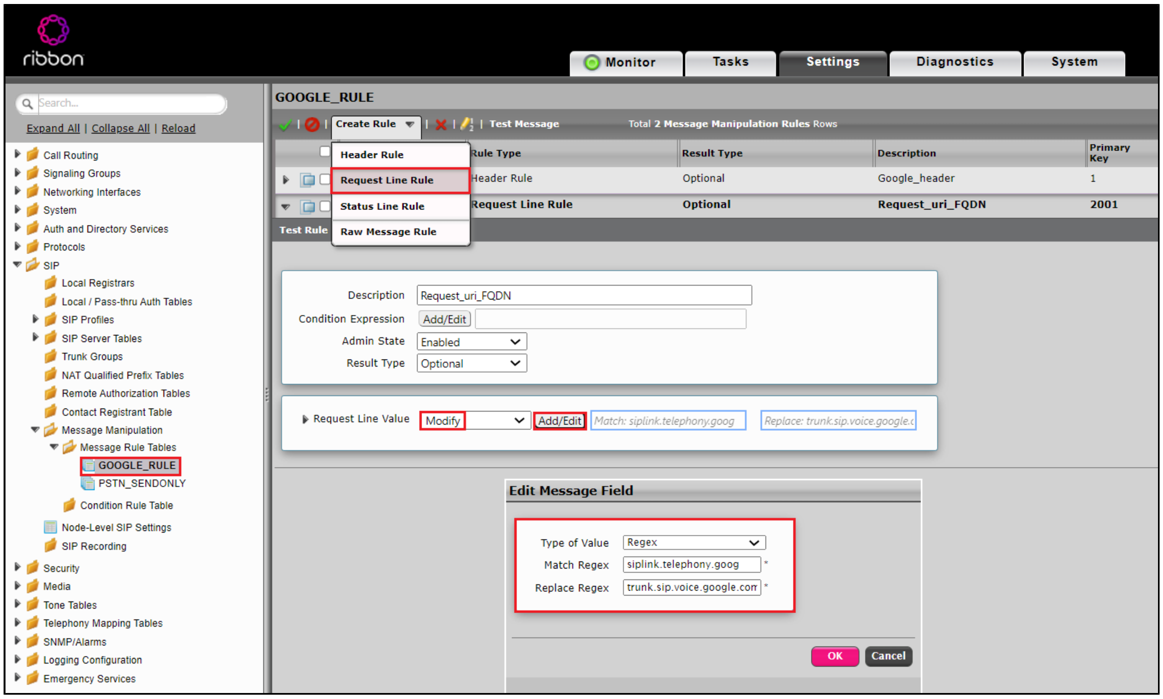
Header Rule:

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**.



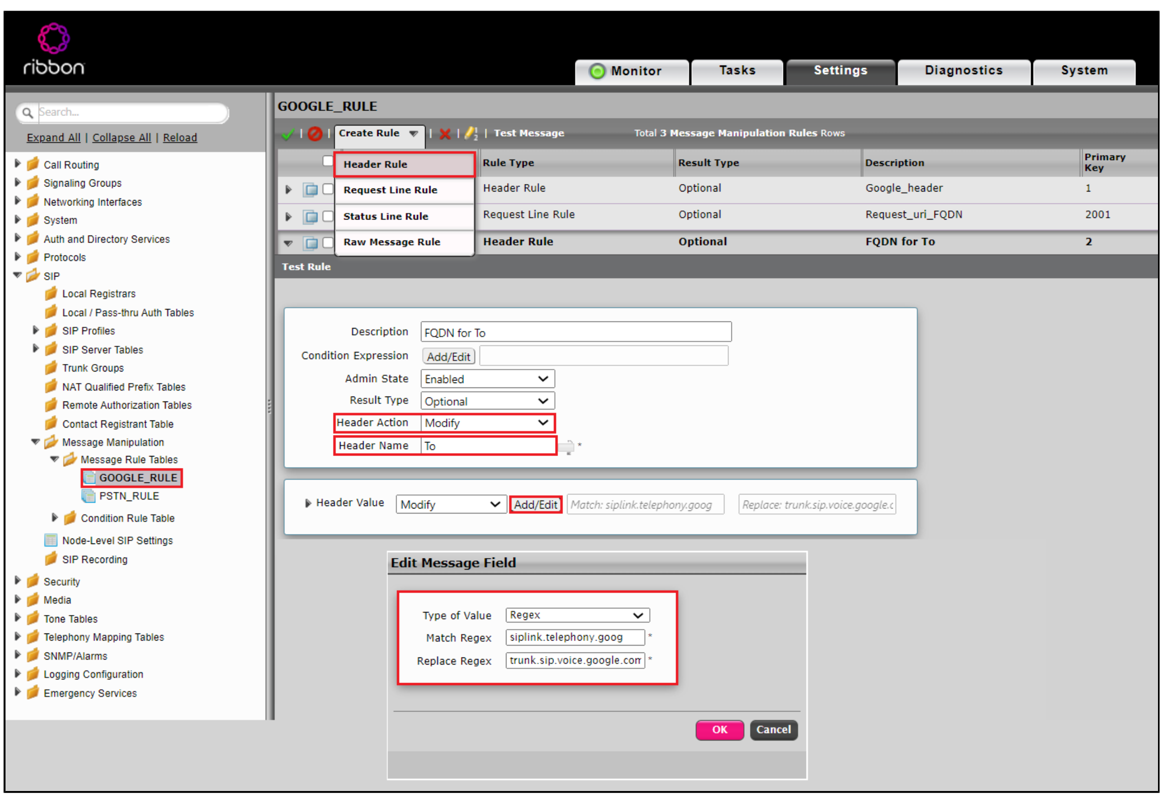
Request Line Rule:

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.



Header Rule:

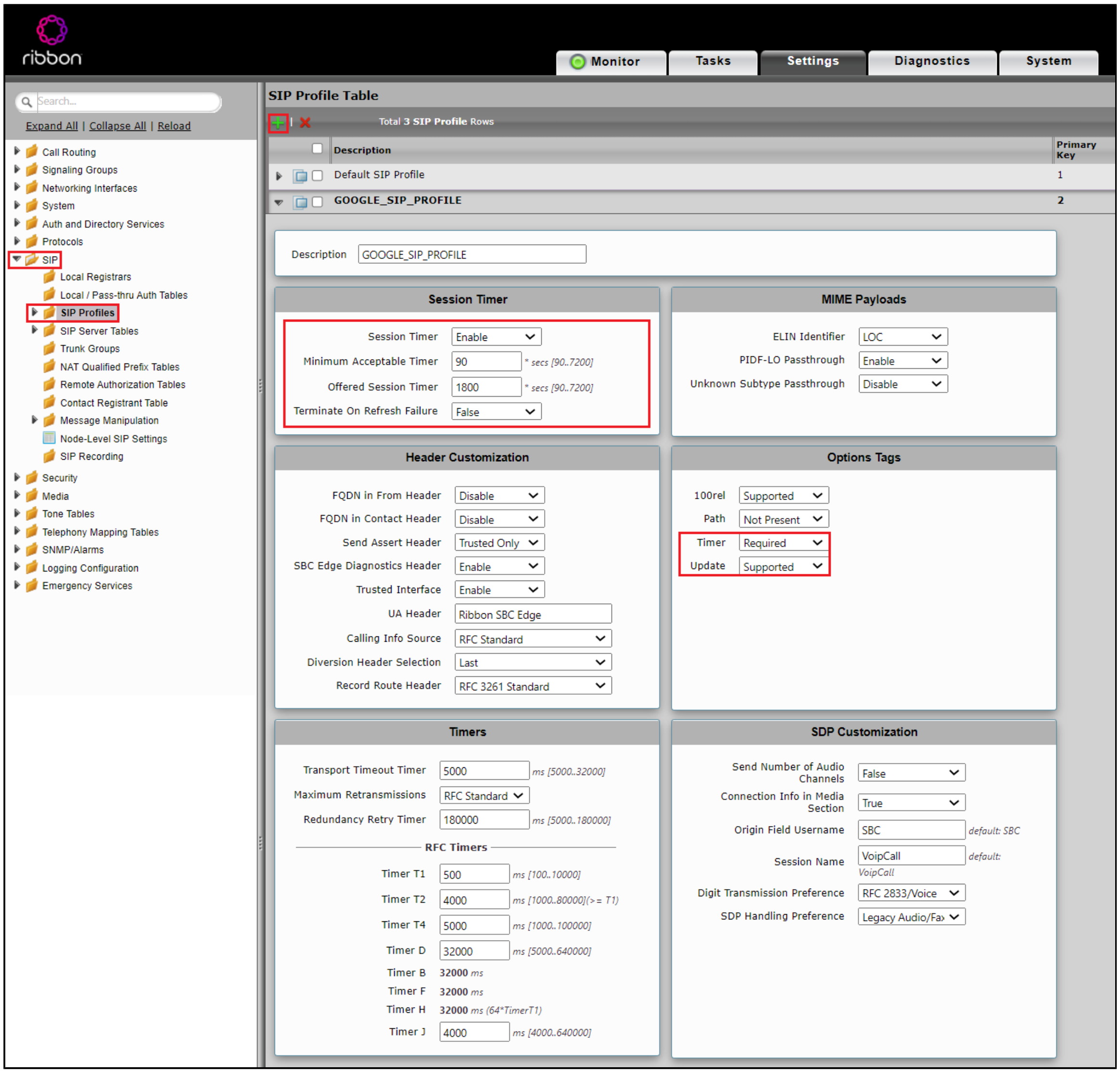
1. Click on the Message Rule Table GOOGLE_RULE.
2. From the Create Rule drop-down menu, select Header Rule.
3. Provide a name for the entry.
4. Select Header Action as Modify and choose To from the Header Name list.
5. Replace the FQDN "siplink.telephony.goog" with "trunk.sip.voice.google.com" using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



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.

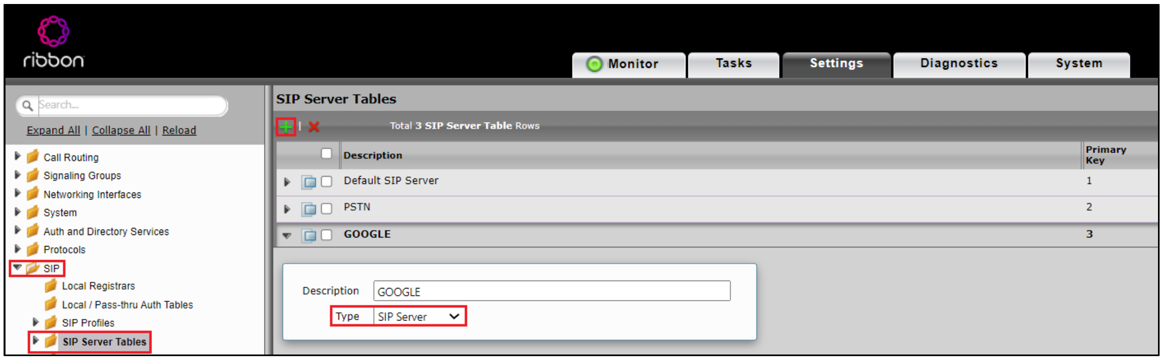


> **Note**
> The session will always be refreshed by Ribbon SBC SWe Edge as per the Google Voice requirement.

SIP Server Table - GV

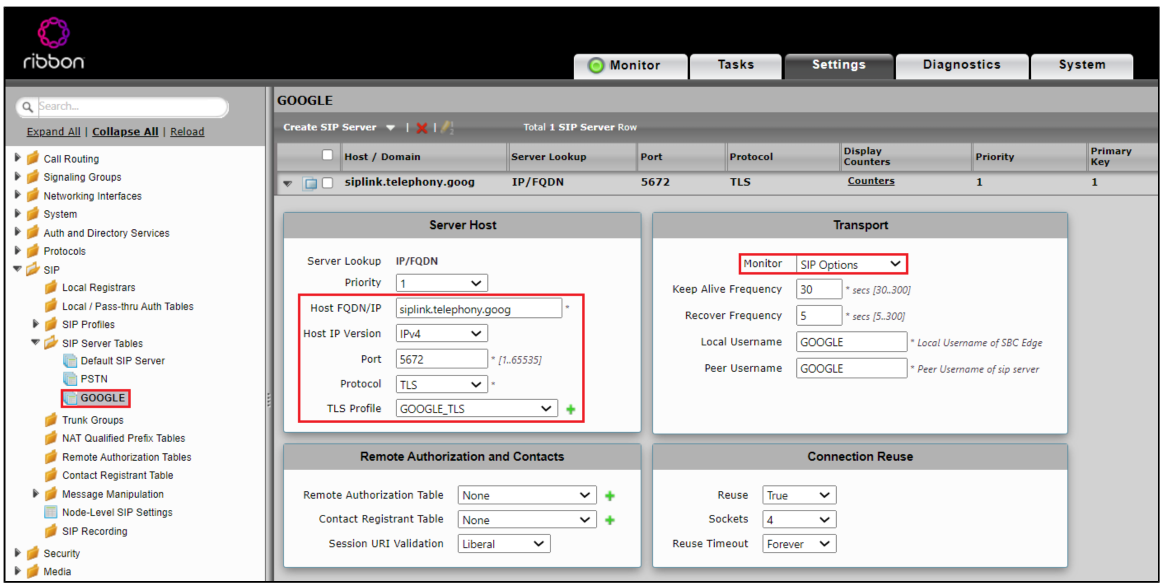
From the **Settings** tab, navigate to **SIP > SIP Server Tables.** Click the icon to create a new SIP Server Table.

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



SIP Server 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.

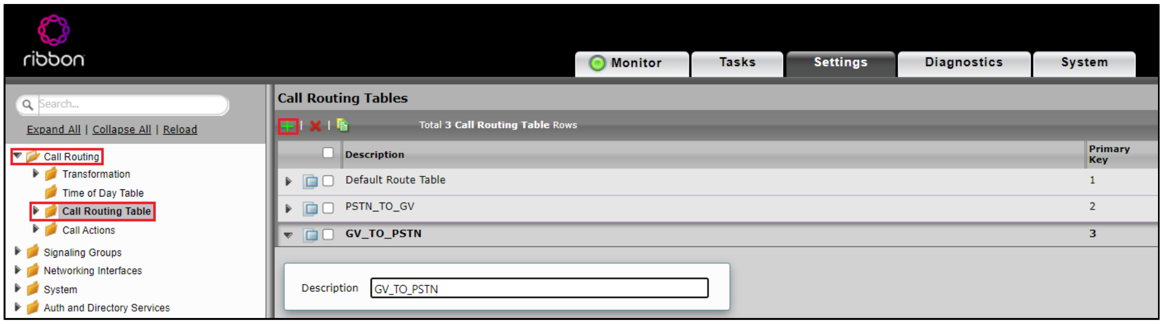


Note
For production, the Google Voice (GV) hostname is siplink.telephony.goog.

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.



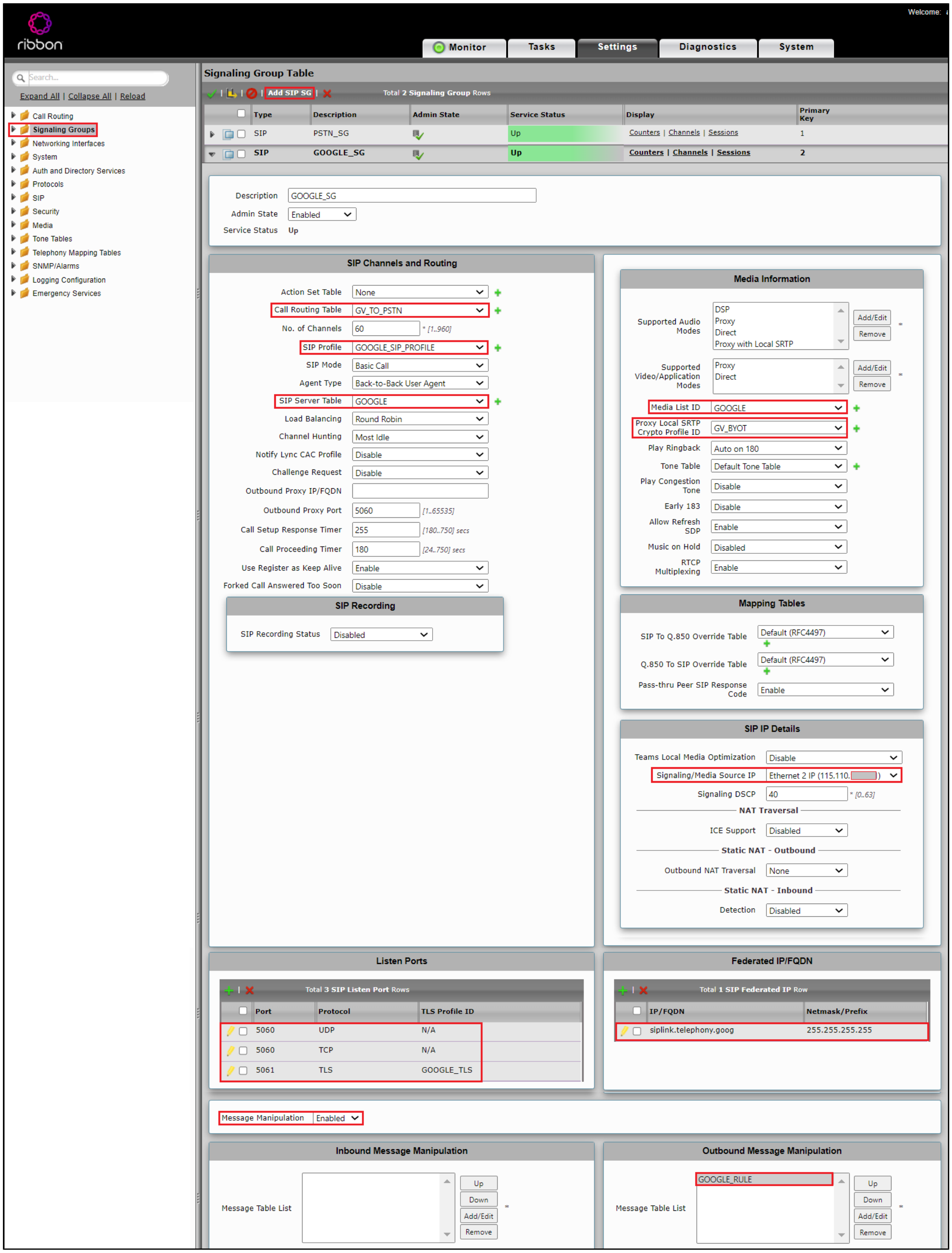
SIP Signaling Group - GV

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.



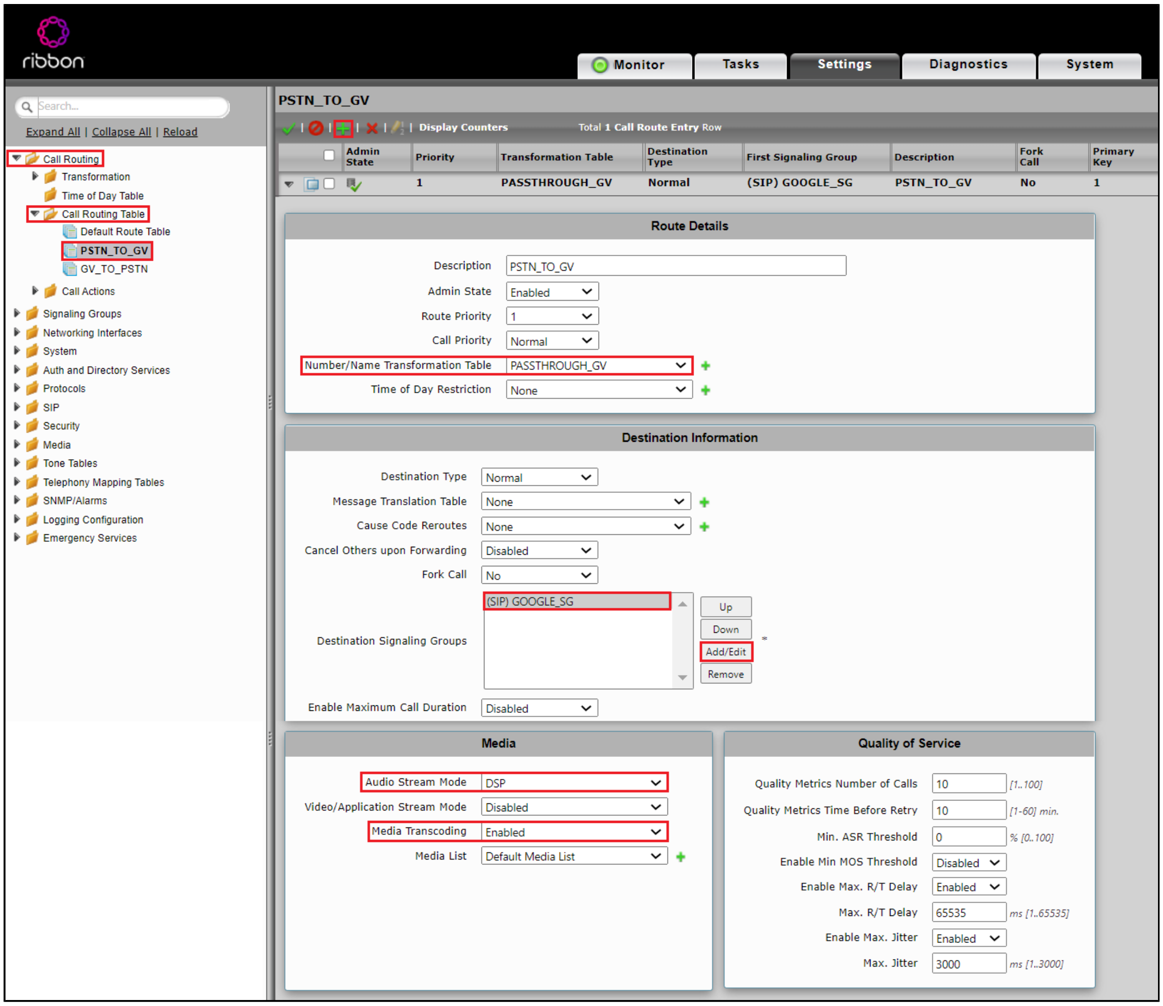
Call Routing Table Entry

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:

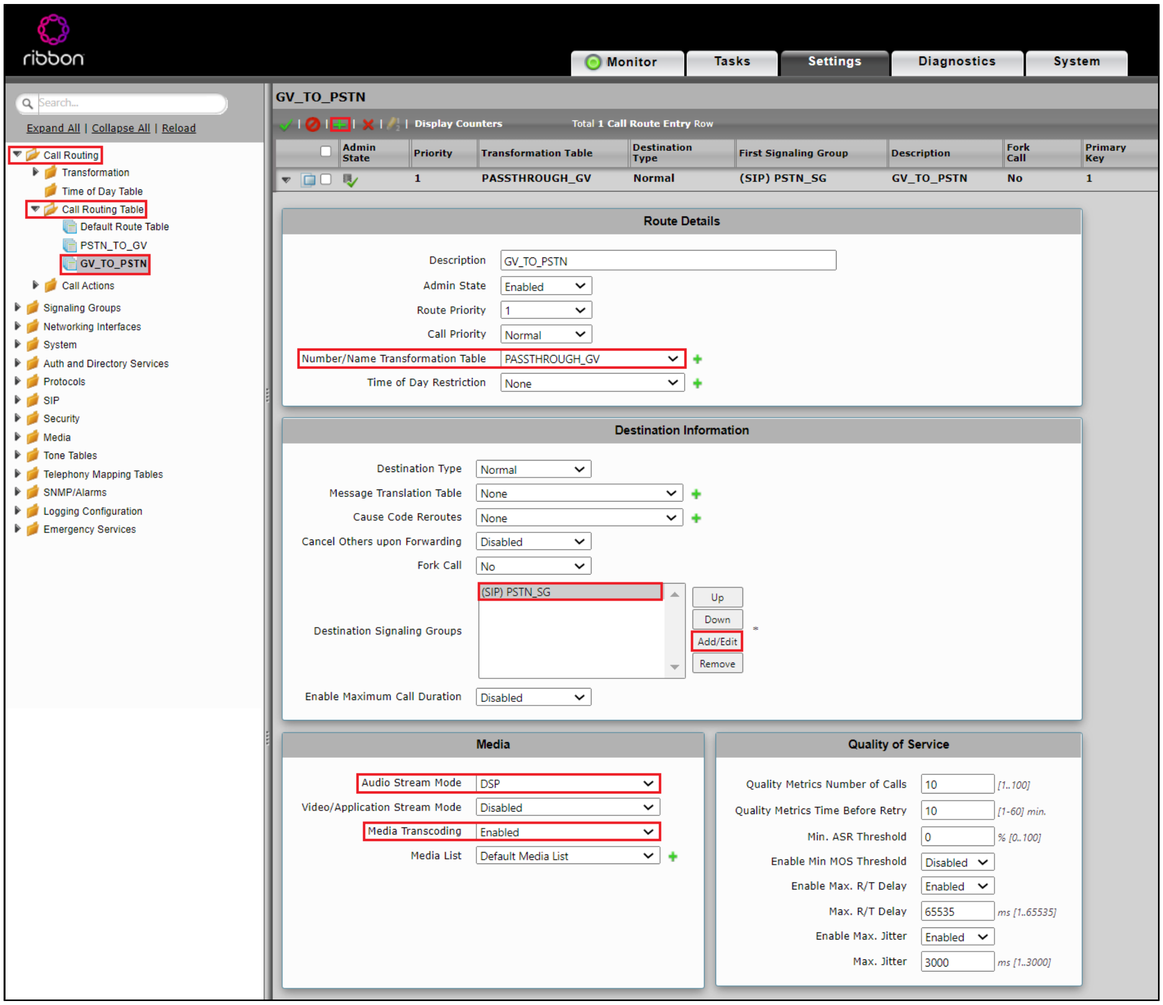
1. Click the **Create Routing Entry** ( ) icon.
2. Attach the Transformation Table (PASSTHROUGH_GV).
3. Add the Destination Signaling Group which in this case is GOOGLE_SG.

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



GV to PSTN :

1. Click the **Create Routing Entry** (+) icon.
2. Attach the Transformation Table (PASSTHROUGH_GV).
3. Add the Destination Signaling Group PSTN_SG.
4. In the Media panel, select DSP from the Audio Stream Mode and enable Media Transcoding.
5. 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.