
Ribbon SBC Edge 1K_2K R11.0 PRI 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 Edge 1000 PRI interworking with Google Voice SIP Link.

About Ribbon SBC Edge 1000

The Ribbon SBC Edge 1000 provides best-in-class communications security. The SBC Edge 1000 dramatically simplifies the deployment of robust communications security services for SIP Trunking, Direct Routing, and Cloud UC services. The SBC 1000 are hardware appliance-based platforms that are part of the Ribbon SBC Edge Portfolio, which addresses the security and interoperability challenges associated with SIP-based communications. The SBC 1000 includes options for Foreign Exchange Office (FXO)/Foreign Exchange Subscriber (FXS) ports and T1/E1 Channel-associated Signaling (CAS)/Primary Rate Interface (PRI) ports. The SBC 1000 is ideally suited for small to medium size organizations and branch offices.

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 to 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 Edge 1000 PRI interop for Google Voice SIP Link. 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.
- understand the basic concepts of T1/E1/ISDN.
- have basic knowledge on 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.

Prerequisites

The following aspects are required before proceeding with the interop:

- Ribbon SBC Edge 1000
- Ribbon SBC Edge 1000 license
 - This interop requires the acquisition and application of SIP sessions, as documented at [Working with Licenses](#)
 - Requires the license for DSI and FXI ports (ISDN ports).
- Public IP addresses
- TLS certificates for SBC Edge 1000
 - 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](#)

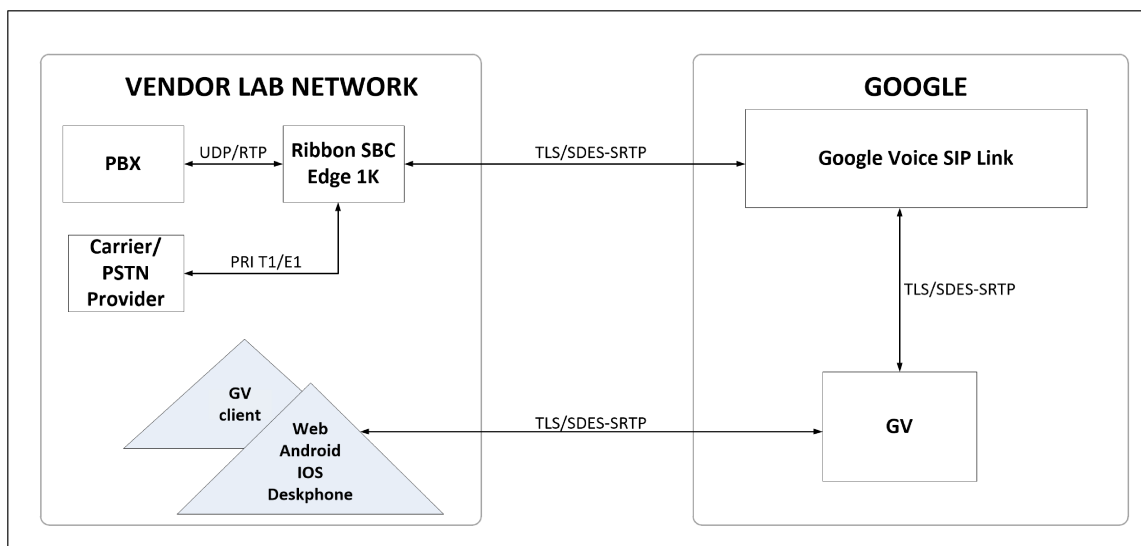
Product and Device Details

The configuration uses the following equipment and software:

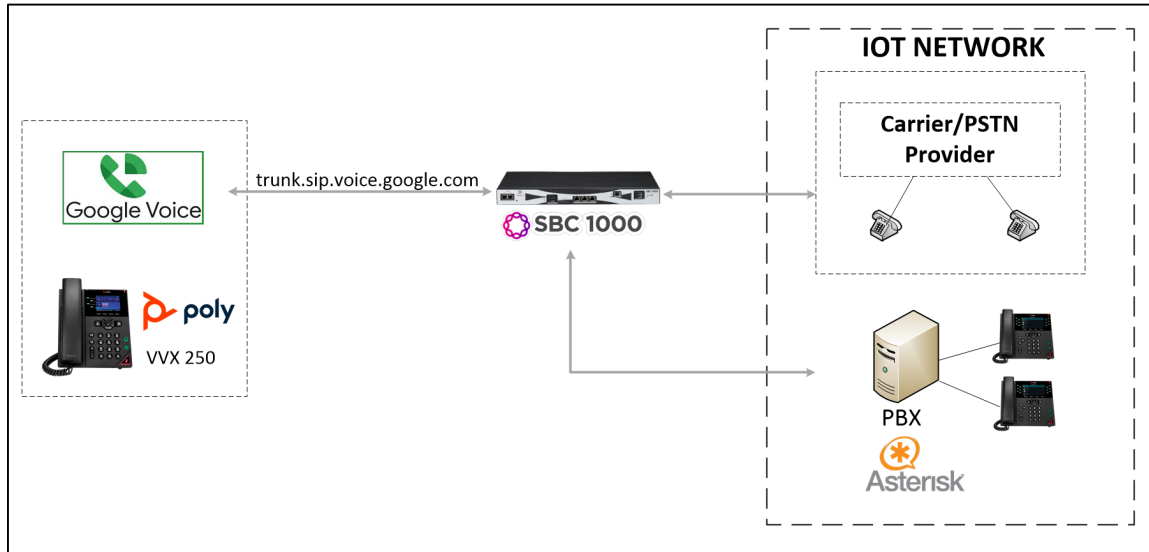
Product	Equipment/Service	Software Version
Ribbon SBC	Ribbon SBC Edge 1000	11.0.1 Build 634
Google Voice SIP Link	Telephone Service	NA
Third-party PBX	Asterisk	16.0.26
Third-party Phone	Poly VVX 250 Edition	6.4.3.10318
Administration and Debugging Tools	Wireshark	3.4.9
	LX Tool	2.1.0.6

Network Topology and E2E Flow Diagrams

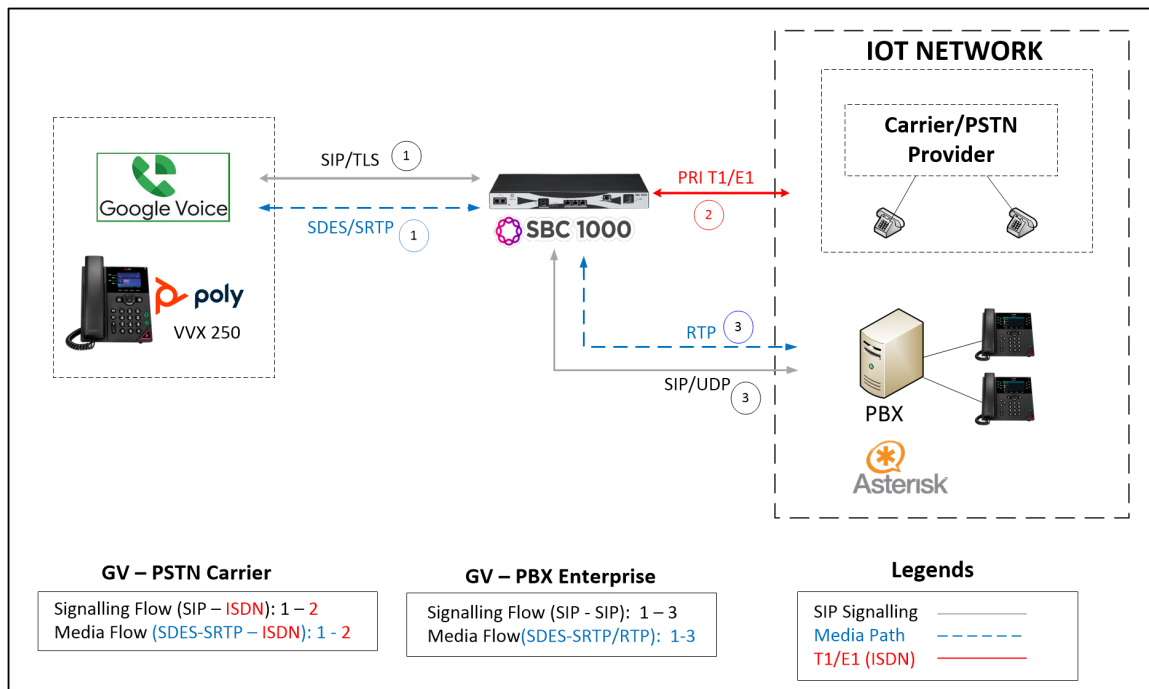
Deployment Topology



Interoperability Test Lab Topology

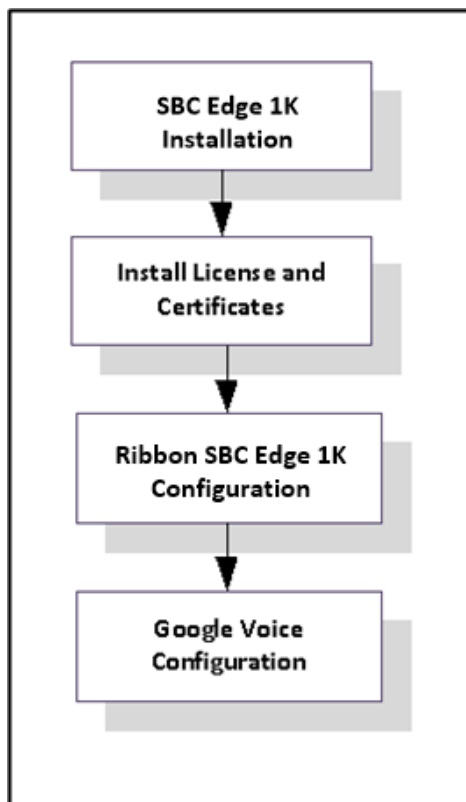


Call Flow Diagram



Document Workflow

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



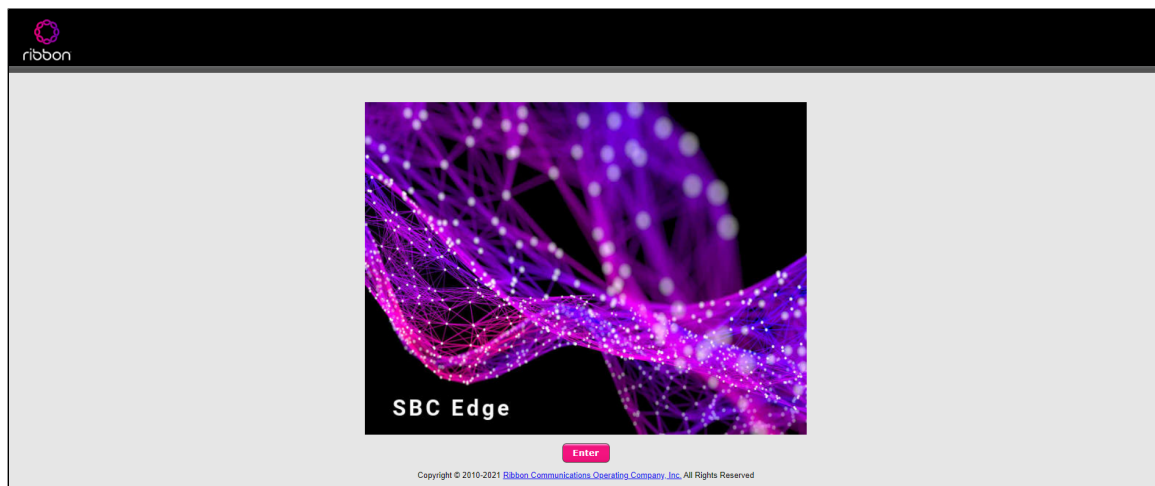
Installing Ribbon SBC Edge 1000

To deploy the Ribbon SBC Edge 1000 instance, refer to [Installing SBC 1000/2000](#)

Ribbon SBC Edge 1000 Configuration

Accessing SBC Edge 1000

Open any browser and enter the SBC Edge 1000 IP address.



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

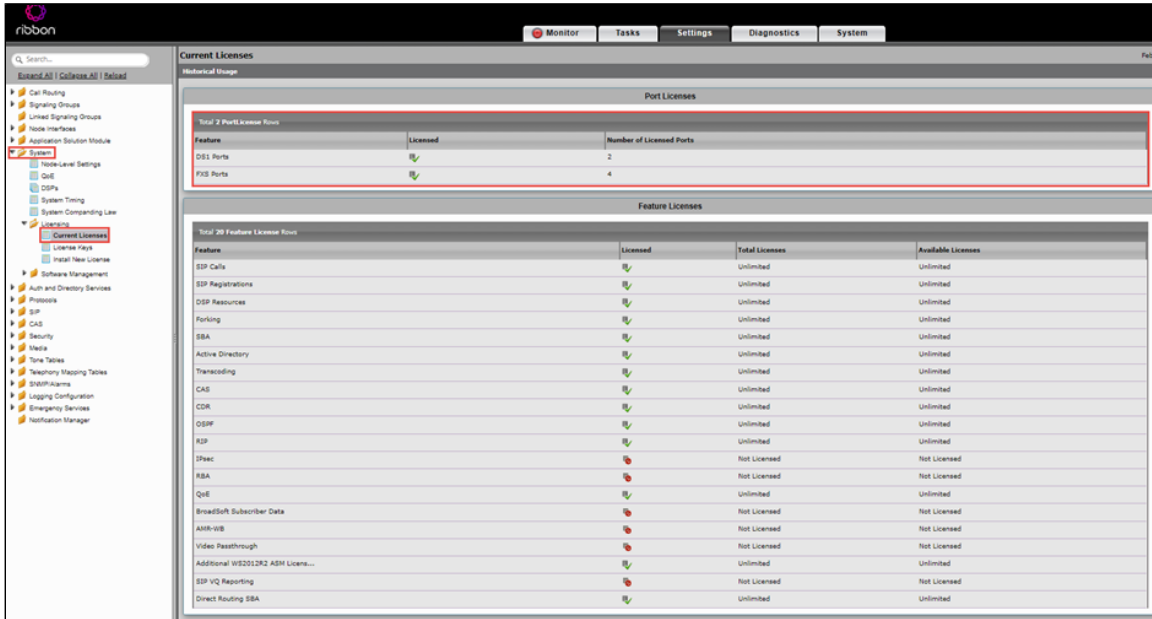


License and TLS Certificates

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** 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**.



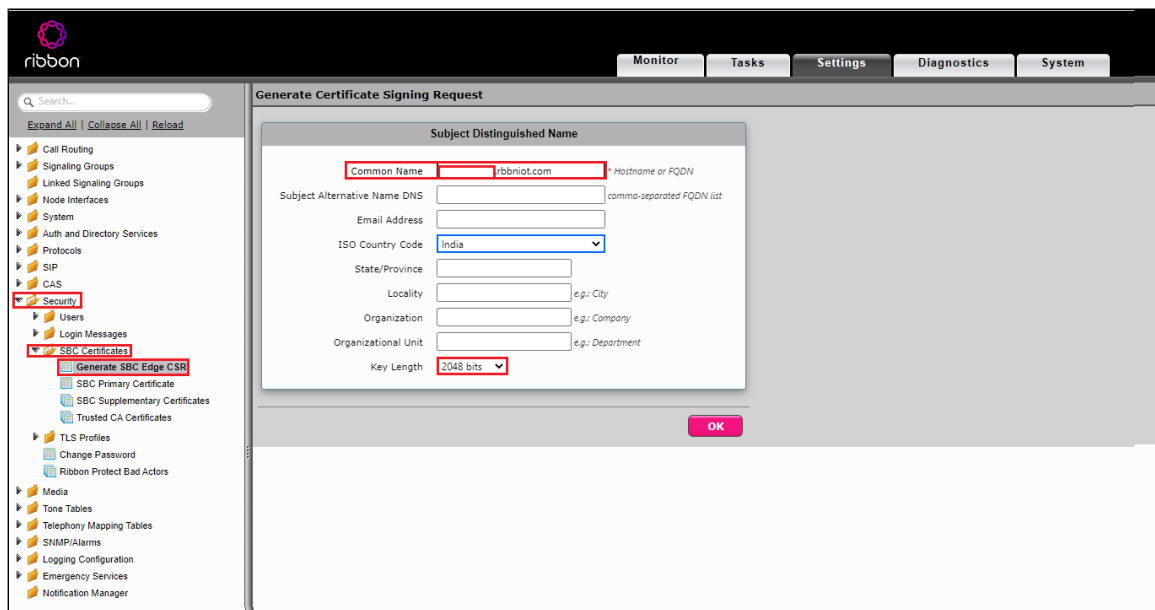
This interop requires license for ISDN ports (DSI/FSX ports).

For more details on Licenses, refer to [Working with Licenses](#).

SBC Certificate

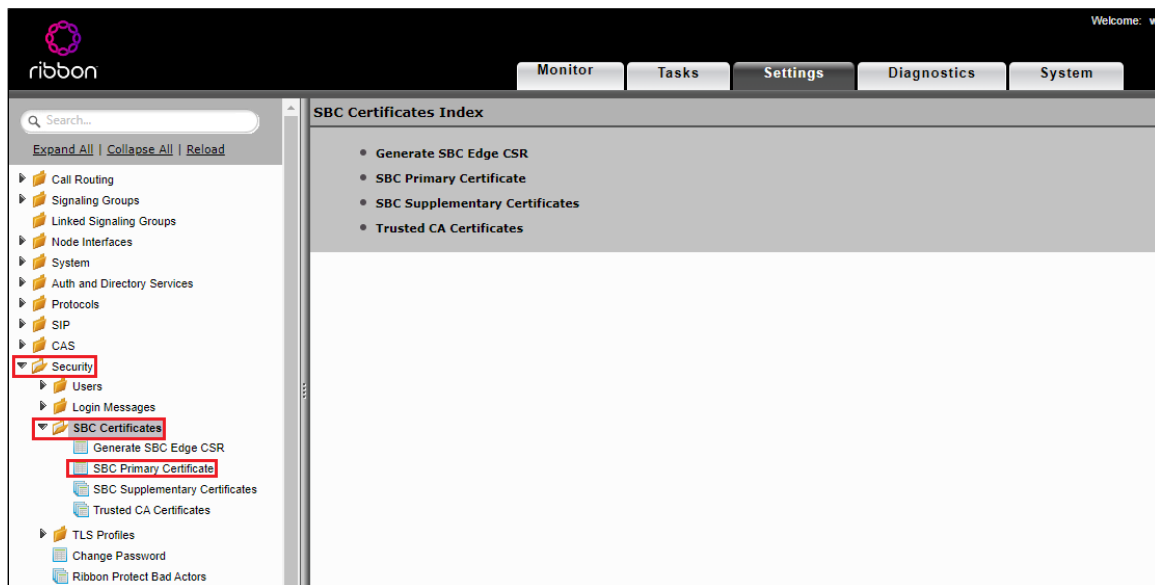
From the **Settingstab**, navigate to **Security > SBC Certificates > Generate SBC Edge Certificates**.

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



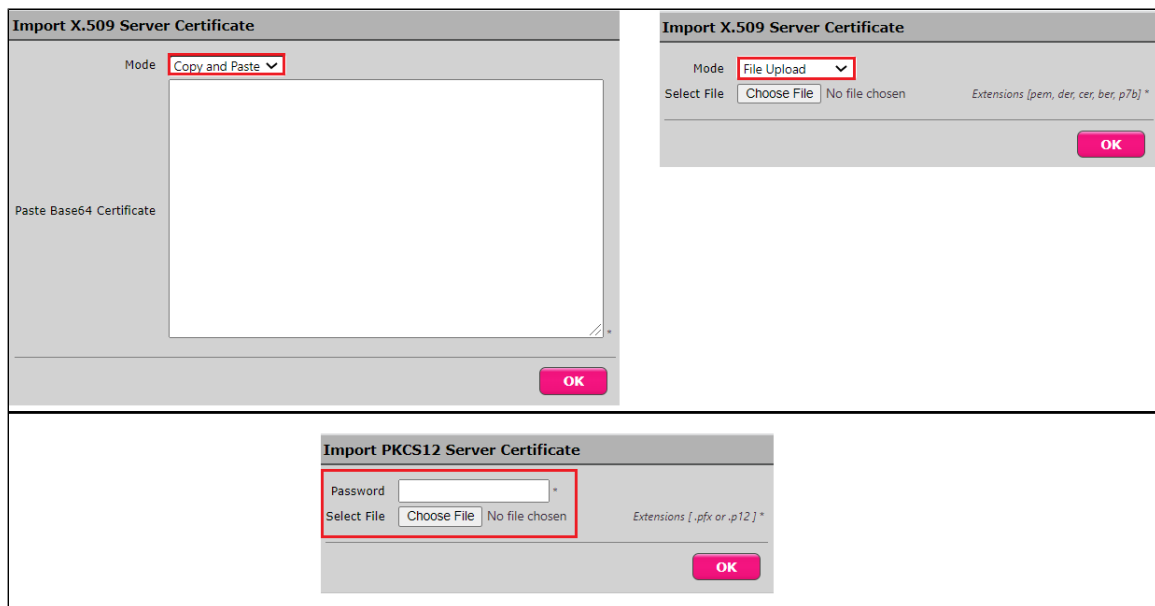
There are two ways to import SBC Primary Certificate as described below:

To import an X.509 signed certificate:

1. Select X.509 Signed Certificate from the Import menu at the top of the page.
2. Choose 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 Edge 1000 to establish its authenticity on the network.

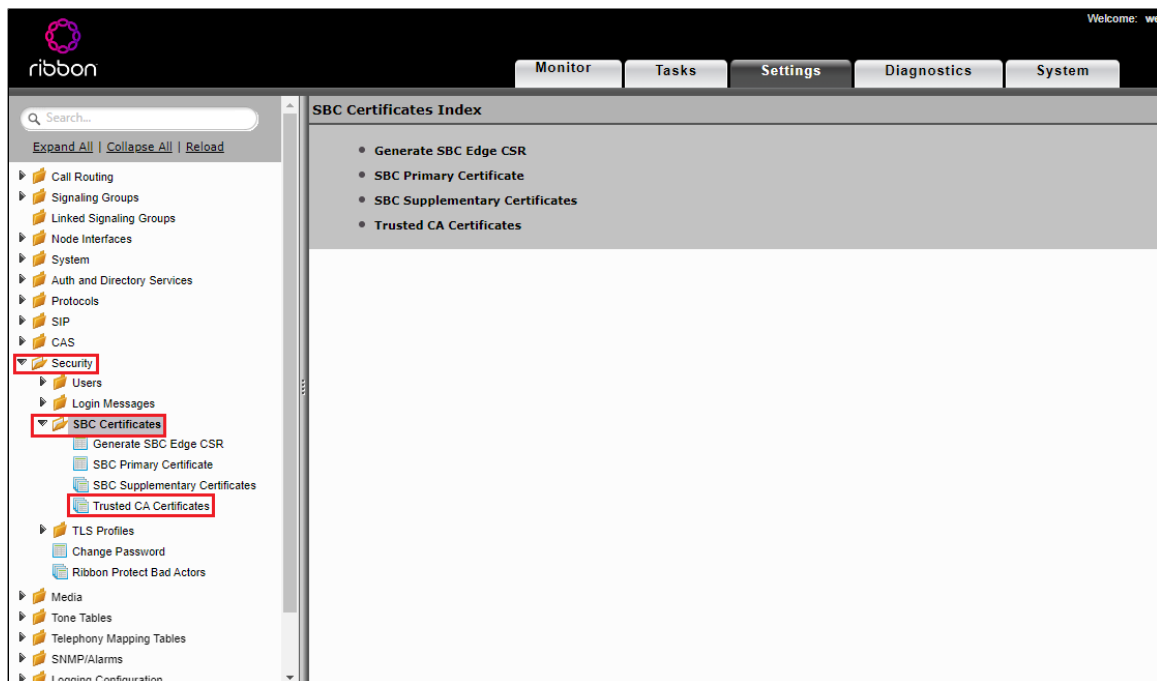
- For TLS to work, a Trusted CA (Certificate Authority) is required. For this interop, GoDaddy is used as the Trusted CA.
- Add an entry in the Public DNS to resolve Ribbon SBC Edge 1000 FQDN to Public IP Address.
- Ensure to have the following certificates as part of the root certificate trust:
 - GTS Root R1
 - GlobalSign Root CA (if required)




Note

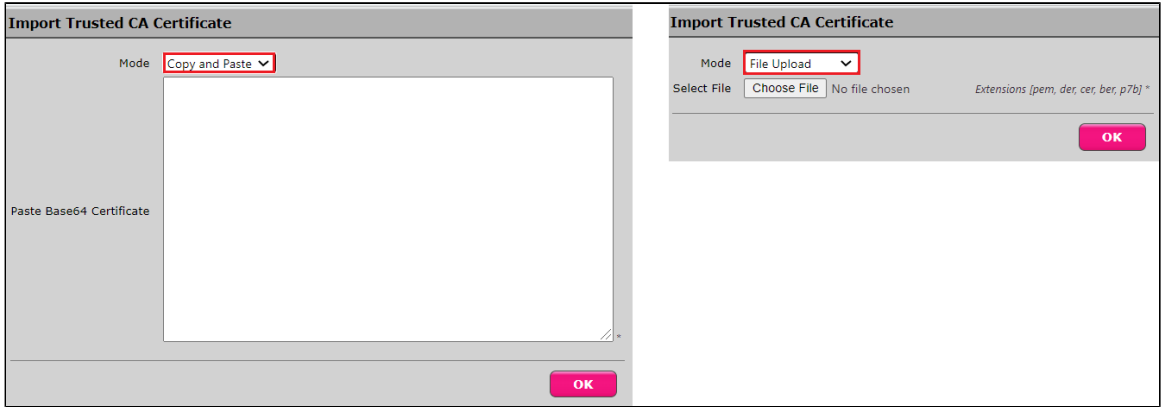
Refer to Google Voice SIP Link documentation for other compatible CAs.

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 Uploader 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.



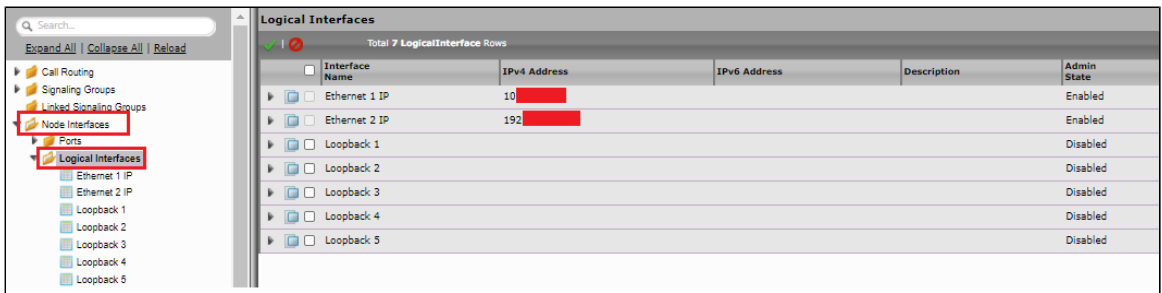
Warning

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

Configure Ethernet 1 and Ethernet 2 of the SBC 1000/2000 with the IP as follows:

Navigate to **Node Interfaces > Logical Interfaces**.



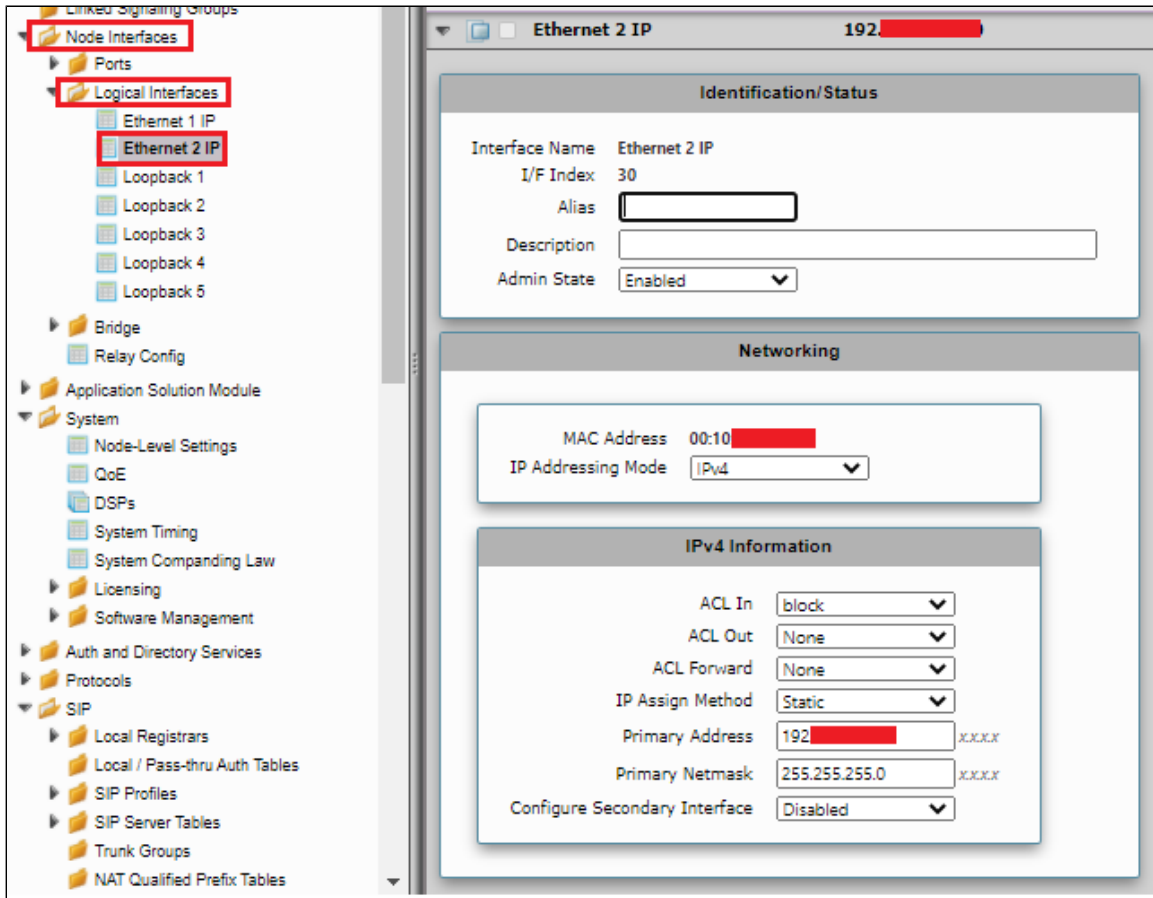
Ethernet 1 IP

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 Edge 1000 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 Edge 1000.

The screenshot displays a network configuration web interface. On the left is a navigation tree with categories like Call Routing, Signaling Groups, Node Interfaces, Ports, Logical Interfaces, Bridge, Application Solution Module, System, and SIP. The 'Logical Interfaces' section is expanded, showing 'Ethernet 1 IP' selected. The main panel is titled 'Logical Interfaces' and shows a table with one row: 'Ethernet 1 IP' with an IPv4 address of '10.0.0.1'. Below the table are three configuration sections: 'Identification/Status', 'Networking', and 'IPv4 Information'. The 'Identification/Status' section includes fields for Interface Name (Ethernet 1 IP), I/F Index (29), Alias, Description, and Admin State (Enabled). The 'Networking' section includes MAC Address (00:10:00:00:00:00) and IP Addressing Mode (IPv4). The 'IPv4 Information' section includes ACL In, ACL Out, and ACL Forward (all set to None), IP Assign Method (Static), Primary Address (10.0.0.1), Primary Netmask (255.255.255.0), and Configure Secondary Interface (Disabled).

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).

Destination IP

Destination IP specifies the destination IP address.

Mask

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

Gateway specifies the IP address of the next-hop router to use for this static route.

Metric

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.

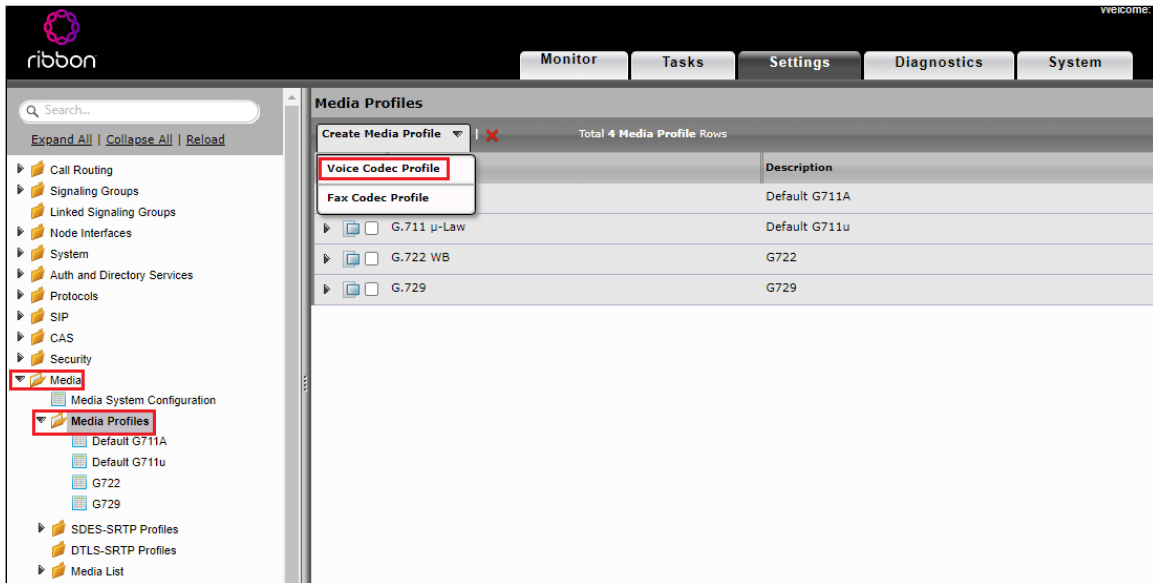
Row ID	Destination IP	Mask	Gateway	Administrative Distance	Primary Key
1	172.16.0.0	255.255.255.0	10.54.0.0	1	1
2	74.125.0.0	255.255.255.0	115.110.0.0	1	2
3	216.239.0.0	255.255.255.255	115.110.0.0	1	3
4	8.8.8.8	255.255.255.255	115.110.0.0	1	4
5	10.70.0.0	255.255.0.0	10.54.0.0	1	5

Global Configuration

Media Profiles

Media Profiles allow you to specify the individual voice and fax compression codecs and their associated settings, for inclusion in aMedia 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 screenshot shows the Ribbon Communications web interface. The 'Settings' tab is selected. The left navigation pane shows the 'Media' folder expanded, with 'Media Profiles' selected. The main content area displays the 'Media Profiles' configuration page. At the top, there is a 'Create Media Profile' dropdown menu with 'Voice Codec Profile' selected. Below this, there is a table with the following data:

	Description
<input type="checkbox"/> Voice Codec Profile	Default G711A
<input type="checkbox"/> Fax Codec Profile	Default G711u
<input type="checkbox"/> G.711 u-Law	Default G711u
<input type="checkbox"/> G.722 WB	G722
<input type="checkbox"/> G.729	G729

The codecs G711A and G711U are configured on the SBC Edge 1000 by default. Configure G722 by following the steps provided below:

For G722:

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

For G729:

1. Provide the profile's description.
2. Select G.729 from the Codec drop-down menu.
3. Click **OK**.

Create Voice Codec Profile

Voice Codec Configuration

Description

Codec

Payload Size ms

OK

Create Voice Codec Profile

Voice Codec Configuration

Description

Codec

Rate 64000 b/s

Payload Size 20 ms

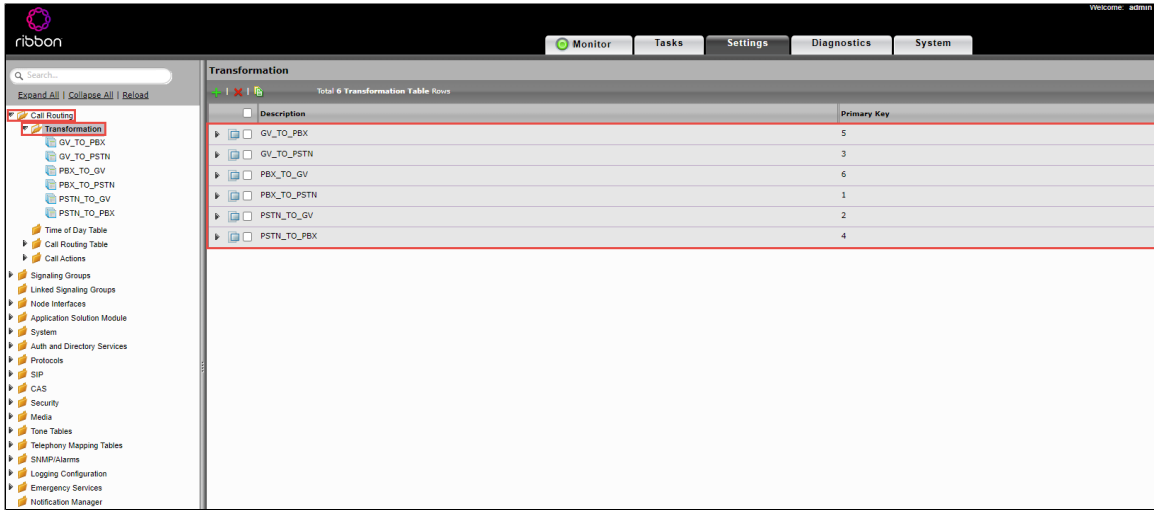
OK

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**.

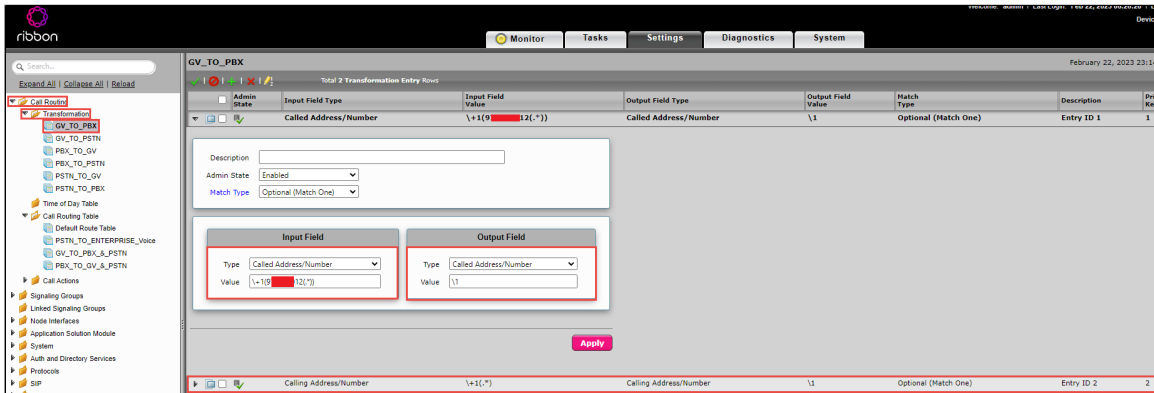


Transformation Table Entry

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**.

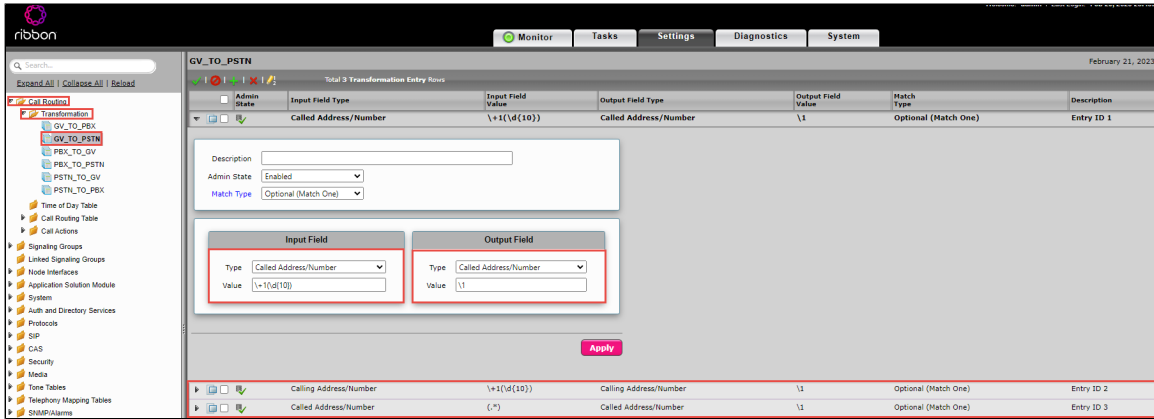
Transformation Table Entry for GV_TO_PBX

1. Provide the DID range for PBX as value in the Input Field.
2. Click **OK**.



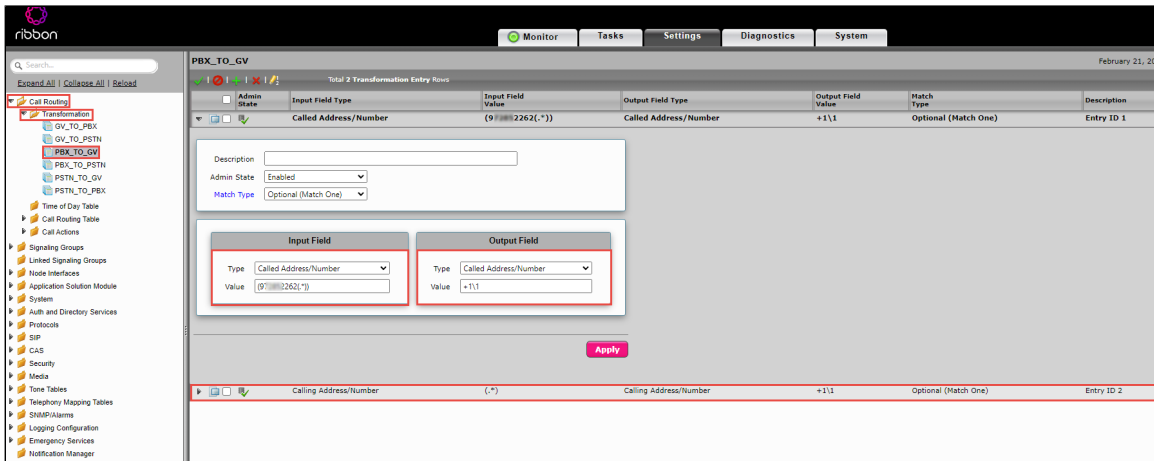
Transformation Table Entry for GV_TO_PSTN

1. Provide the DID number range of PSTN as value in the Input Field. Here all 10 digit numbers are allowed.
2. Click **OK**.



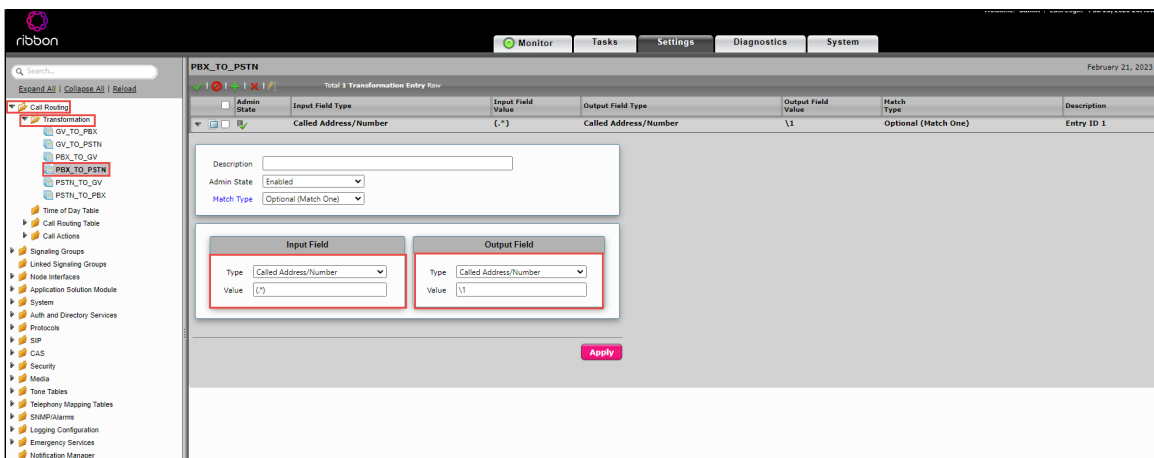
Transformation Table Entry for PBX_TO_GV

1. Provide the DID number range of GV as value in the Input Field.
2. Click OK.



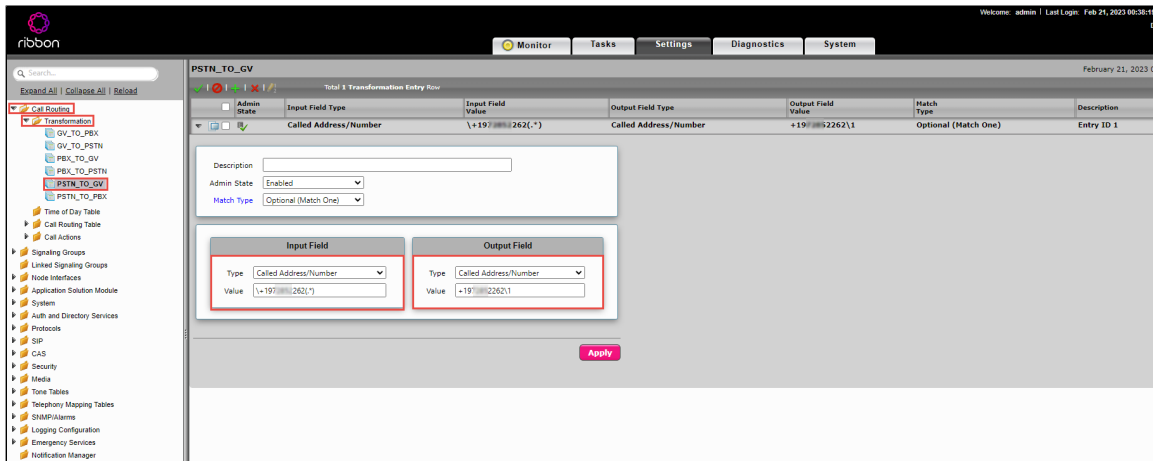
Transformation Table Entry for PBX_To_PSTN

1. Provide the DID number range of PSTN as value in the Input Field. Here all the Numbers/Address are allowed.
2. Click OK.



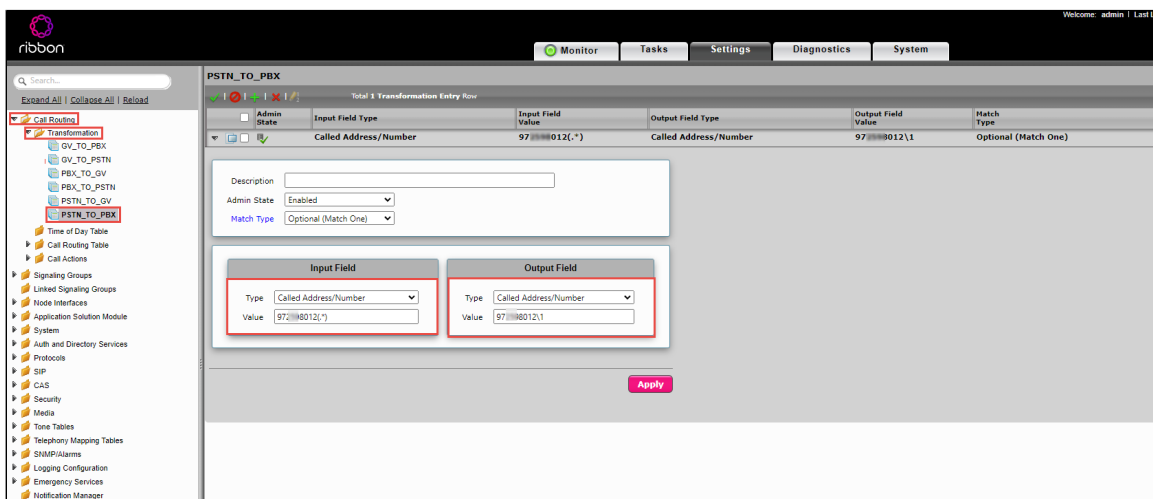
Transformation Table Entry for PSTN_TO_GV

1. Provide the DID number range of GV as value in the Input Field.
2. Click OK.



Transformation Table Entry for PSTN_TO_PBX

1. Provide the DID number range of PBX as value in the Input Field.
2. Click **OK**.

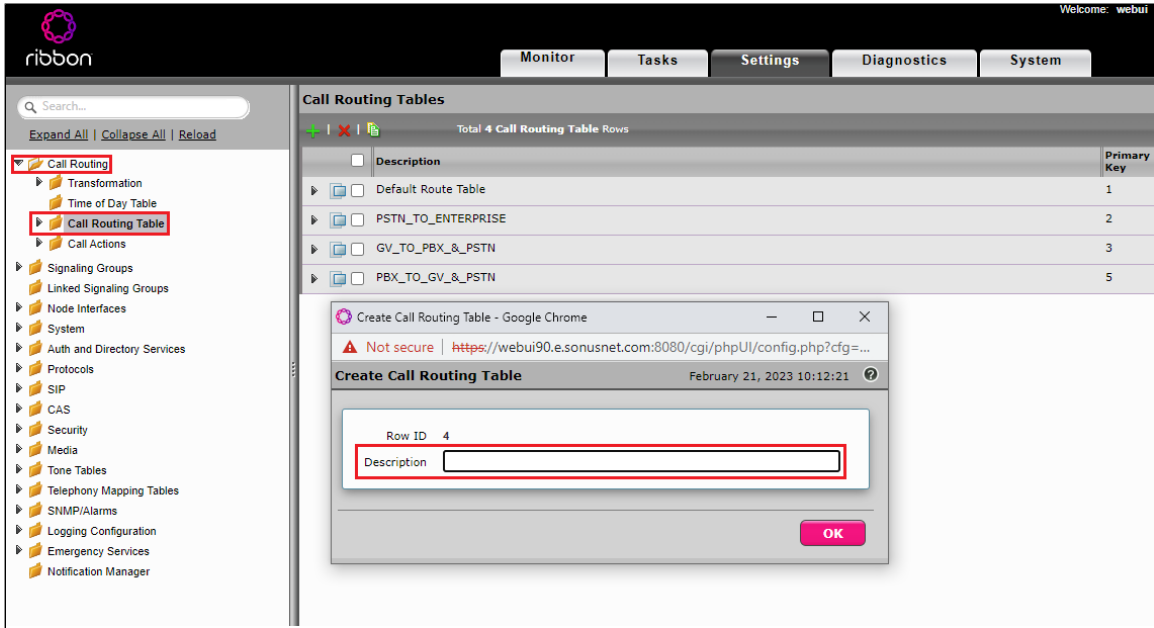


Call Routing Table

Call Routing allows calls to be carried between Signaling Groups, thus allowing calls to be carried between ports and between protocols (such as 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.

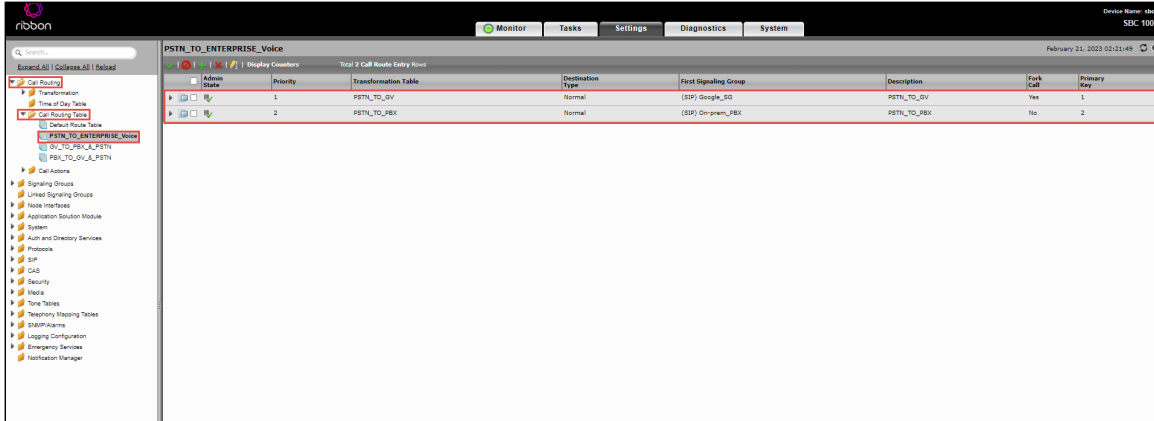
From the **Settings** tab, navigate to **Call Routing > Call Routing Table**. Click the **+** icon to create a Call Routing Table.

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



Call Routing Table Entry

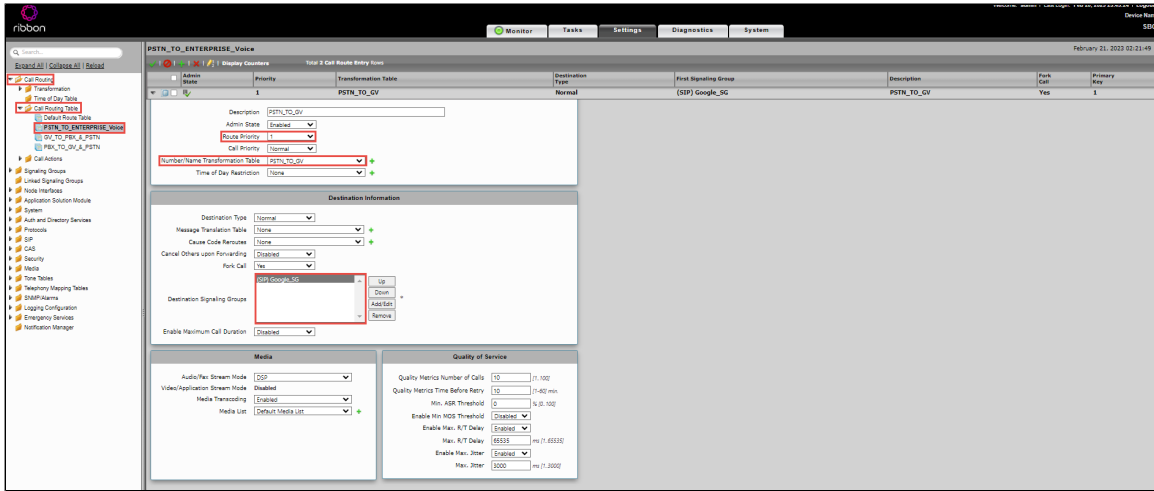
PSTN to ENTERPRISE_Voice:



Entry 1 (PSTN_TO_GV)

1. Click the **Create Routing Entry (+)** icon.
2. Attach the Transformation Table (**PSTN_TO_GV**) with priority 1.
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**.

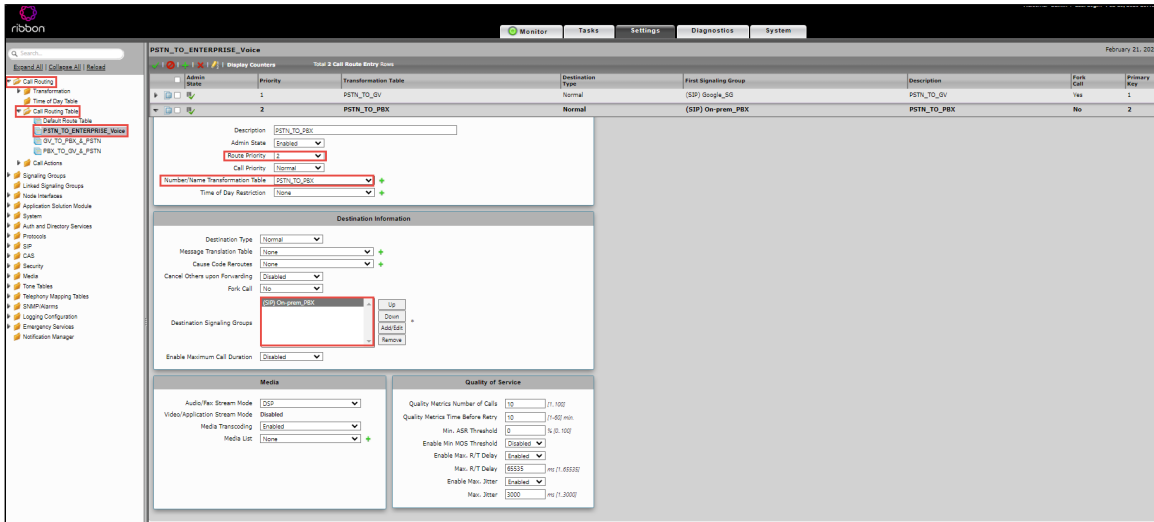
Call Matching PSTN_TO_GV transformation table will be routed to the Google_SG.



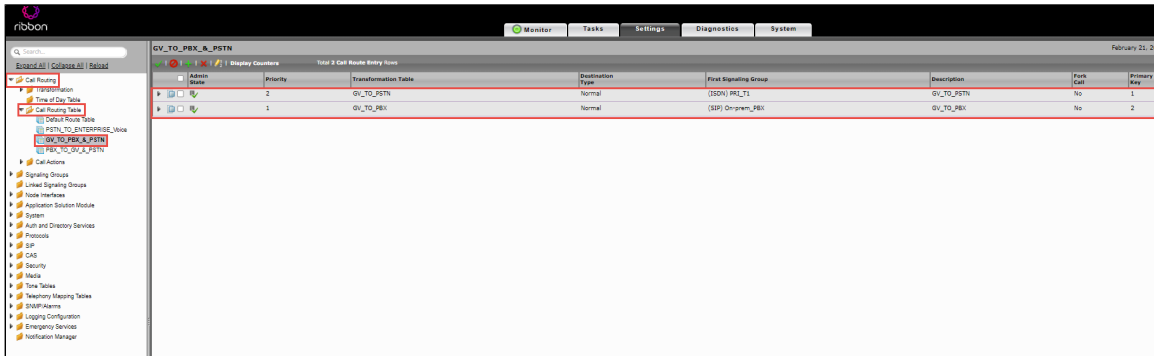
Entry 2 (PSNT_TO_PBX)

1. Click the **Create Routing Entry (+)** icon.
2. Attach the Transformation Table (**PSTN_TO_PBX**) with priority 2.
3. Add the Destination Signaling Group which in this case is **On-prem_PBX**.
4. In the Media panel, select **DSP** from the Audio Stream Mode and enable Media Transcoding.
5. Click **OK**.

Call matching PSTN_TO_PBX transformation table will be routed to the On-prem_PBX SG.



GV to PBX_ & PSTN :



Entry 1 (GV_TO_PBX)

1. Click the **Create Routing Entry (+)** icon.
2. Attach the Transformation Table (**GV_TO_PSTN**) with priority 2.
3. Add the Destination Signaling Group (**PRI_T1**).
4. In the Media panel, select **DSP** from the Audio Stream Mode and enable Media Transcoding.
5. Click **OK**.

Call Matching **GV_TO_PSTN** transformation table will be routed to the **PRI_T1** SG.

Admin State	Priority	Transformation Table	Destination Type	First Signaling Group	Description	Fork Call	Primary Key
Enabled	2	GV_TO_PSTN	Normal	(ISDN) PRI_T1	GV_TO_PSTN	No	1

Destination Information

Destination Type: Normal
Message Translation Table: None
Cause Code Reroutes: None
Cancel Others upon Forwarding: Disabled
Fork Call: No
Destination Signaling Groups: (ISDN) PRI_T1

Media

Audio/Fax 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. RTT Delay: Enabled [0.033] ms [1..65536]
Enable Max. Jitter: Enabled [0.000] ms [1..3000]

Entry 2 (GV_TO_PBX)

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

Call Matching **GV_TO_PBX** transformation table will be routed to the **On-prem_PBX** SG.

Admin State	Priority	Transformation Table	Destination Type	First Signaling Group	Description	Fork Call	Primary Key
Enabled	2	GV_TO_PSTN	Normal	(ISDN) PRI_T1	GV_TO_PSTN	No	1
Enabled	1	GV_TO_PBX	Normal	(SIP) On-prem_PBX	GV_TO_PBX	No	2

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) On-prem_PBX

Media

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

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. RTT Delay: Enabled [0.033] ms [1..65536]
Enable Max. Jitter: Enabled [0.000] ms [1..3000]

PBX_TO_GV_&_PSTN:

Admin State	Priority	Transformation Table	Destination Type	First Signaling Group	Description	Fork Call	Primary Key
<input type="checkbox"/>	2	PBX_TO_PSTN	Normal	(ISDN) PRI_T1	PBX_PSTN	No	1
<input type="checkbox"/>	1	PBX_TO_GV	Normal	(SIP) Google_SG	PBX_TO_GV	No	2

Entry 1 (PBX_TO_PSTN)

1. Click the **Create Routing Entry** (+) icon.
2. Attach the Transformation Table (**PBX_TO_PSTN**) with priority 2.
3. Add the Destination Signaling Group (**PRI_T1**).
4. In the Media panel, select DSP from the Audio Stream Mode and enable Media Transcoding.
5. Click **OK**.

Call Matching PBX_TO_PSTN transformation table will be routed to the PRI_T1 SG.

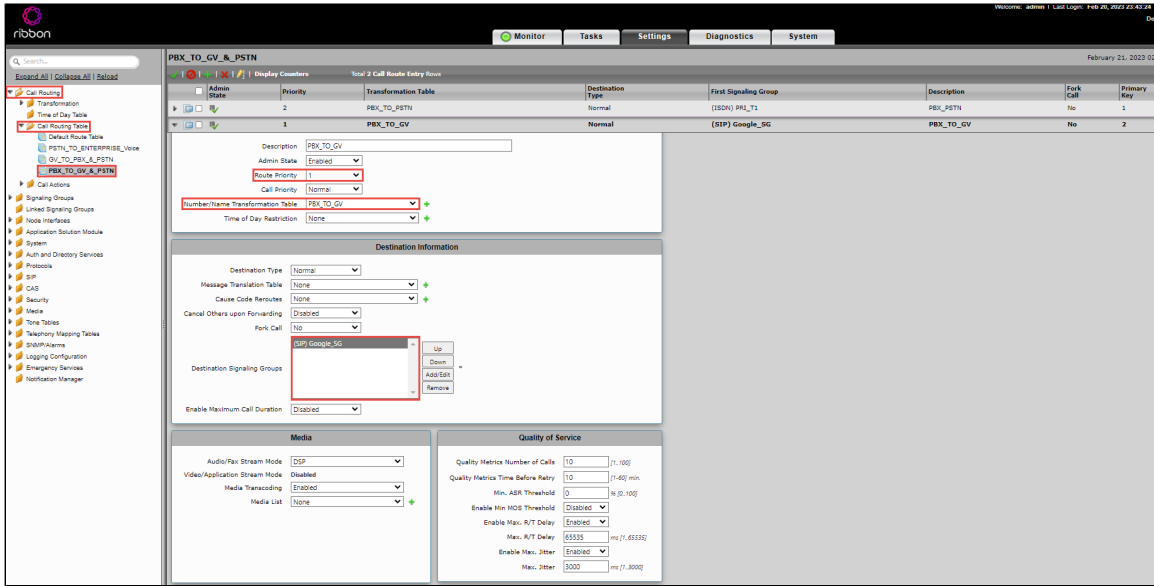
Entry 1 (PBX_TO_PSTN) Configuration Details:

- Description:** PBX_PSTN
- Admin State:** Enabled
- Route Priority:** 2
- Call Priority:** Normal
- Number/Name Transformation Table:** PBX_TO_PSTN
- Time of Day Restriction:** None
- Destination Information:**
 - Destination Type:** Normal
 - Message Translation Table:** None
 - Cause Code Reroutes:** None
 - Cancel Others upon Forwarding:** Disabled
 - Force Call:** No
 - Destination Signaling Groups:** (ISDN) PRI_T1
 - Enable Maximum Call Duration:** Disabled
- Media:**
 - Audio/Fax Stream Mode:** DSP
 - Video/Application Stream Mode:** Disabled
 - Media Transcoding:** Enabled
 - Media List:** None
- Quality of Service:**
 - Quality Metrics Number of Calls:** 10 (0-100)
 - Quality Metrics Time Before Retry:** 10 (1-60) min
 - Min. ASR Threshold:** 0 (0-100)
 - Enable Min. HOS 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)

Entry 2 (PBX_TO_GV)

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

Call Matching PBX_TO_GV transformation table will be routed to the Google_SG.

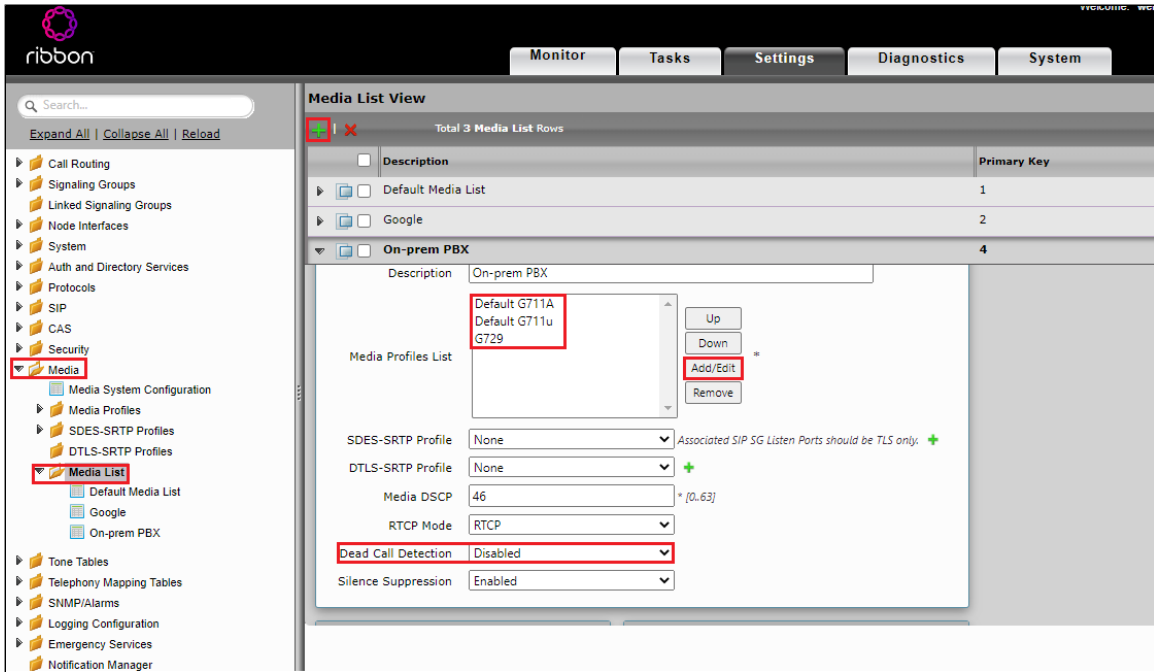


SBC Edge 1000 Configuration for PBX side

Media List - PBX

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 in the Description field.
2. Attach the Media Profiles by clicking Add/Edit.
3. Enable Dead Call Detection.
4. Click **OK**.

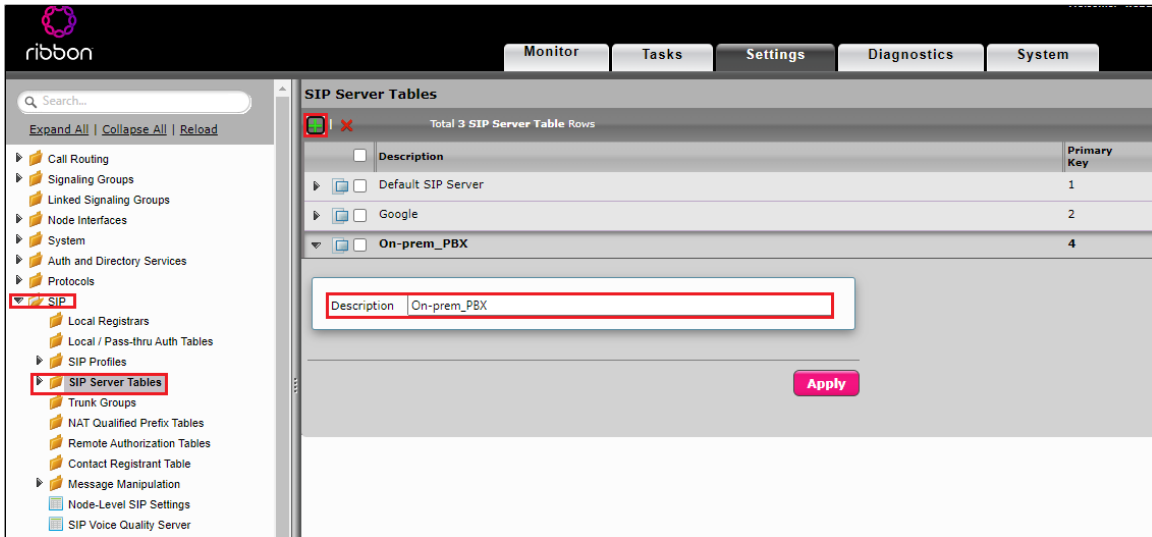


SIP Server Table - PBX

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).

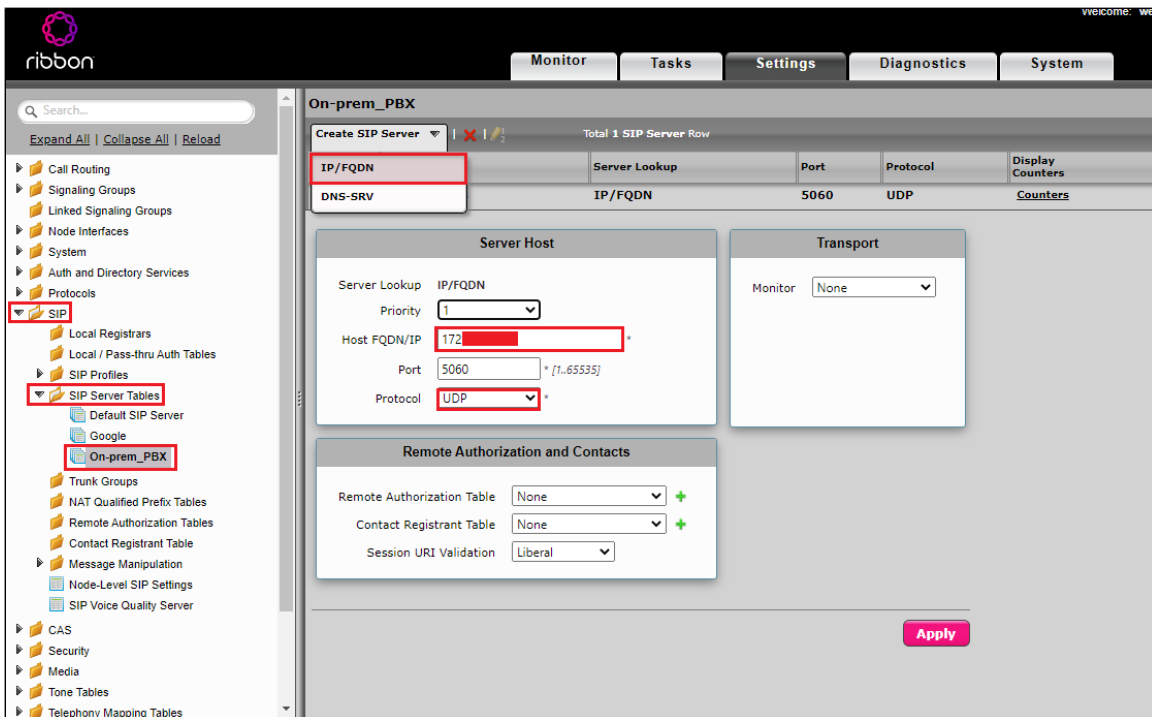
From the **Settings** tab, navigate to **SIP > SIP Server Tables**. Click the **+** icon to create a new SIP Server Table.

1. Provide a name for the SIP Server in the Description field.
2. 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 PBX.
4. Click **OK**.



SIP Signaling Group - PBX

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 ([PBX_TO_GV_&_PSTN](#)).
2. Attach the SIP Profile (Default SIP Profile).
3. Attach the SIP Server Table ([on-prem_PBX](#)).
4. Attach the Media List ID ([on-prem_PBX](#)).
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 1000 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. Click **OK**.

Description: On-prem_PBX
Admin State: Enabled
Service Status: Up

SIP Channels and Routing

Action Set Table: None

Call Routing Table: PBX_TO_GV_&_PSTN

No. of Channels: 30

SIP Profile: Default SIP Profile

SIP Mode: Basic Call

Agent Type: Back-to-Back User Agent

Interop Mode: Standard

SIP Server Table: On-prem_PBX

Load Balancing: Round Robin

Notify Lync CAC Profile: Disable

Challenge Request: Disable

Outbound Proxy IP/FQDN:

Outbound Proxy Port: 5060

No Channel Available Override: 34: No Circuit/Channel Available

Call Setup Response Timer: 255

Call Proceeding Timer: 180

QoE Reporting: Disabled

Use Register as Keep Alive: Enable

Forked Call Answered Too Soon: Disable

Media Information

Supported Audio/Fax Modes: DSP, Proxy, Direct

Supported Video/Application Modes: Disabled

Media List ID: On-prem_PBX

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: Disabled

Media Codec Latch: Enable

Listen Ports

Listen Port: UDP-5060, TCP-5060

Mapping Tables

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

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

Pass-thru Peer SIP Response Code: Enable

SIP IP Details

Teams Local Media Optimization: Disable

Signaling/Media Source IP: Ethernet 1 IP (10.1.1.144)

Signaling DSCP: 40

NAT Traversal: Disabled

Static NAT - Outbound: None

Static NAT - Inbound: Disabled

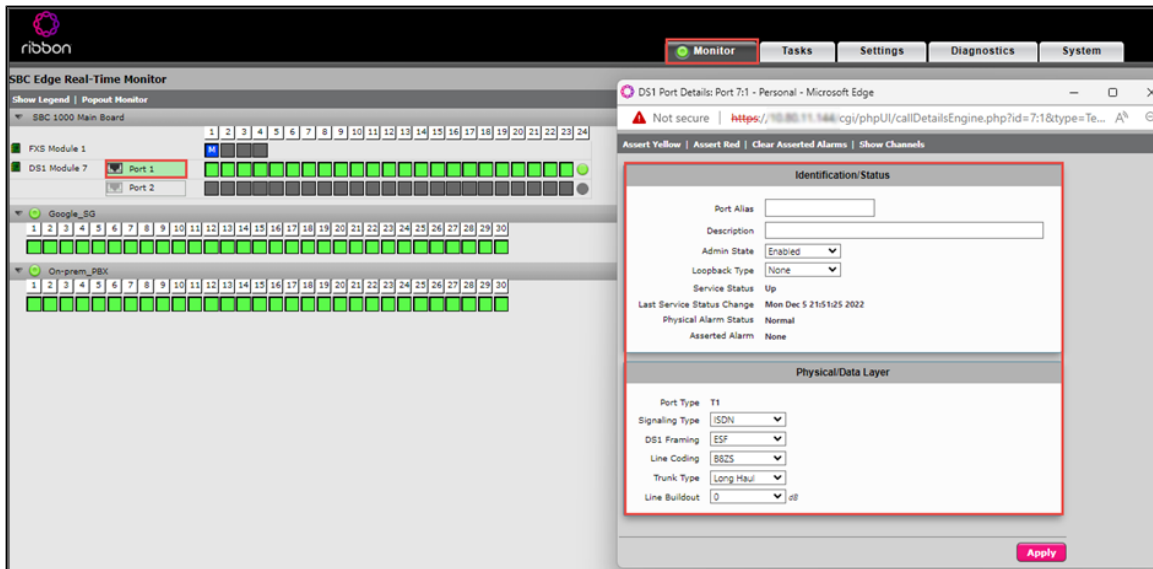
Federated IP/FQDN

IP/FQDN	Netmask/Prefix
172.31.1.53	255.255.255.255

Message Manipulation: Disabled

SBC Edge 1000 Configuration for T1/PRI side DSI Port Configuration

From the **Monitoring** tab, Select the **DS1** port and make a configuration according to the service provider Trunk type, Framing, and Line coding.



SIP Signaling Group - PRI/PSTN

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

1. Attach the Call Routing Table ([PSTN_TO_ENTERPRISE_Voice](#)).
2. Attach the Port Name (T1 Port 7:1).

Description: PRI_T1
Admin State: Enabled
Service Status: Up

Channels and Routing

Channel Hunting: Most Idle
Direction: Bidirectional
Tone Table: Default Tone Table Ringback * +
Action Set Table: None +
Call Routing Table: PSTN_TO_ENTERPRISE_Voice * +
No Channel Available Override: 34: No Circuit/Channel Available
Play Inband Message Post-Disconnect: No
Call Setup Response Timer: 255 [180..750] secs

Port and Protocol

Port Name: (T1) Port 7:1 *
Fractional: No
Switch Variant: NI2
ISDN Side: User
Play Ringback: Auto on Alert
Service Msg Capability: Enabled
Stop Far-End T310: Disabled
Indicated Channel: Exclusive

Switch Specific Parameters

Send Calling Name: Enabled
Add PI To Setup: None
Early Media for PI: 2(Dest not ISDN): Enabled
Include Channel Interface Identifier: Disabled
Channel Number Bit: Set

Timeout/Timer Settings

T301	180	[1..600] secs
T302	15	[1..255] secs
T303	4	[1..255] secs
T305	30	[1..255] secs
T308	4	[1..255] secs
T309	6	[1..255] secs
T310	30	[1..255] secs
T313	4	[1..255] secs
T314	4	[1..255] secs
T316	120	[1..255] secs
T322	4	[1..255] secs
T3M1/T323	120	[1..255] secs

SBC Edge 1000 Configuration for Google Voice SIP Link side

DNS

From the **Settingstab**, 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**.

The screenshot shows the 'Node-Level Settings' page in the Ribbon Communications interface. The left sidebar contains a navigation tree with 'System' expanded and 'Node-Level Settings' selected. The main content area is divided into several sections:

- Host Information:** Host Name: sbc7, Domain Name: sbc7lab.com.
- System Information:** System Description, System Location, System Contact (all empty).
- Domain Name Service:** Use Primary DNS: Yes, Primary Server IP: 8.8.8.8, Primary Source: Ethernet 2 IP (192.168.1.158), Use Secondary DNS: No, Enable DNS Service: Yes.
- Time Management:** Time Zone: (GMT-6:00) Central (US/Canada), Use NTP: Yes, NTP Server: 10.10.10.5, NTP Server Authentication: Disabled, NTP Server 2: (empty), Use NTP Server 2: No.
- Ribbon Application Management Platform (RAMP):** Connect to RAMP: No.
- System LEDs:** Power LED: Green, Alarm LED: Amber, Ready LED: Green, Locator LED: On Green.
- Country Level Information:** Country Code: None.

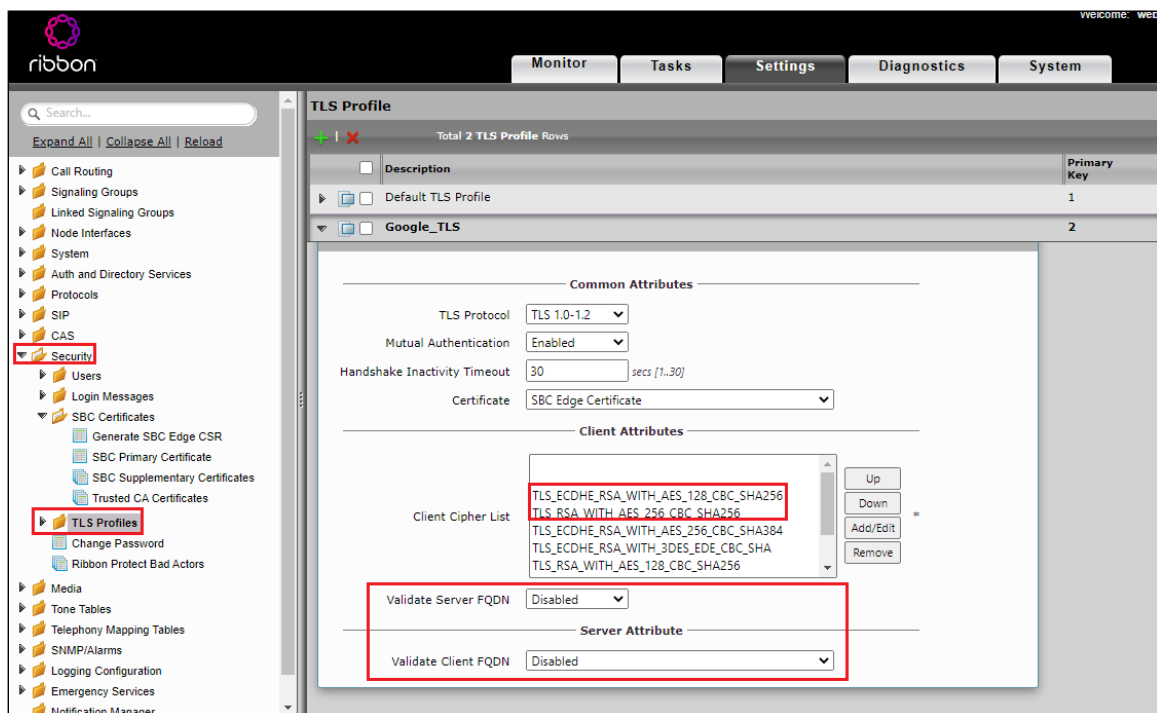
An 'Apply' button is located at the bottom right of the page.

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**.

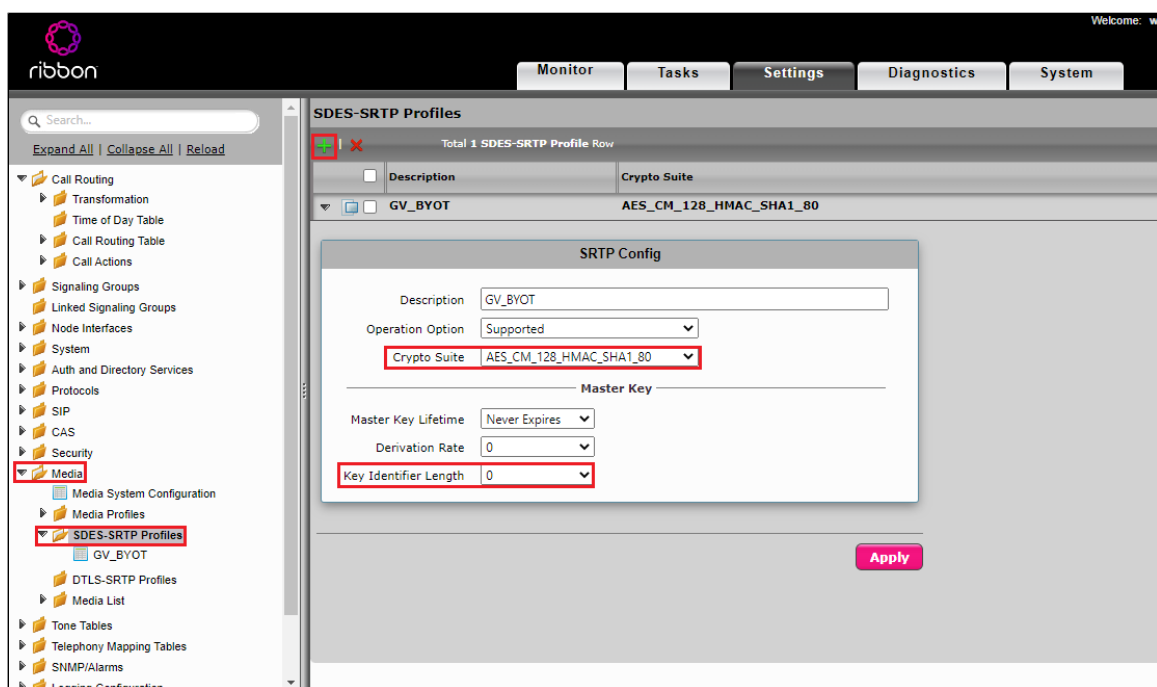


SDES-SRTP Profile

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_SHA1_80", 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 Edge 1000.

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 in the Description field.
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**.

The screenshot shows the 'Media List View' page in the Ribbon SBC Edge 1000 settings. The left sidebar shows the navigation tree with 'Media' and 'Media List' highlighted. The main panel shows a table with 3 rows: 'Default Media List' (Primary Key 1) and 'Google' (Primary Key 2). The 'Google' profile is expanded, showing the following configuration:

- Description: Google
- Media Profiles List: Default G711u, Default G711A, G22 (Add/Edit button highlighted)
- SDES-SRTP Profile: GV_BYOT (Associated SIP SG Listen Ports should be TLS only)
- DTLS-SRTP Profile: None
- Media DSCP: 46
- RTCP Mode: RTCP
- Dead Call Detection: Enabled
- Silence Suppression: Enabled

The screenshot shows the 'Media List View' page in the Ribbon SBC Edge 1000 settings, showing the configuration for the 'Google' profile. The left sidebar shows the navigation tree with 'Media' and 'Media List' highlighted. The main panel shows the configuration for the 'Google' profile, including the following settings:

- Dead Call Detection: Enabled
- Silence Suppression: Enabled
- Gain Control:
 - Receive Gain: 0 [-14, +6] dB
 - Transmit Gain: 0 [-14, +6] dB
- Digit Relay:
 - Digit (DTMF) Relay Type: RFC 2833
 - Digit Relay Payload Type: 101 [96, 127]
- Passthrough/Tone Detection:
 - Modem Passthrough: Enabled
 - Fax Passthrough: Enabled
 - CNG Tone Detection: Disabled
 - Fax Tone Detection: Enabled

Message Manipulation - GV

The Message Manipulation feature comprises two primary components that work in concert to modify SIP messages. These 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 GOOGLE_RULE is used for the following purposes:

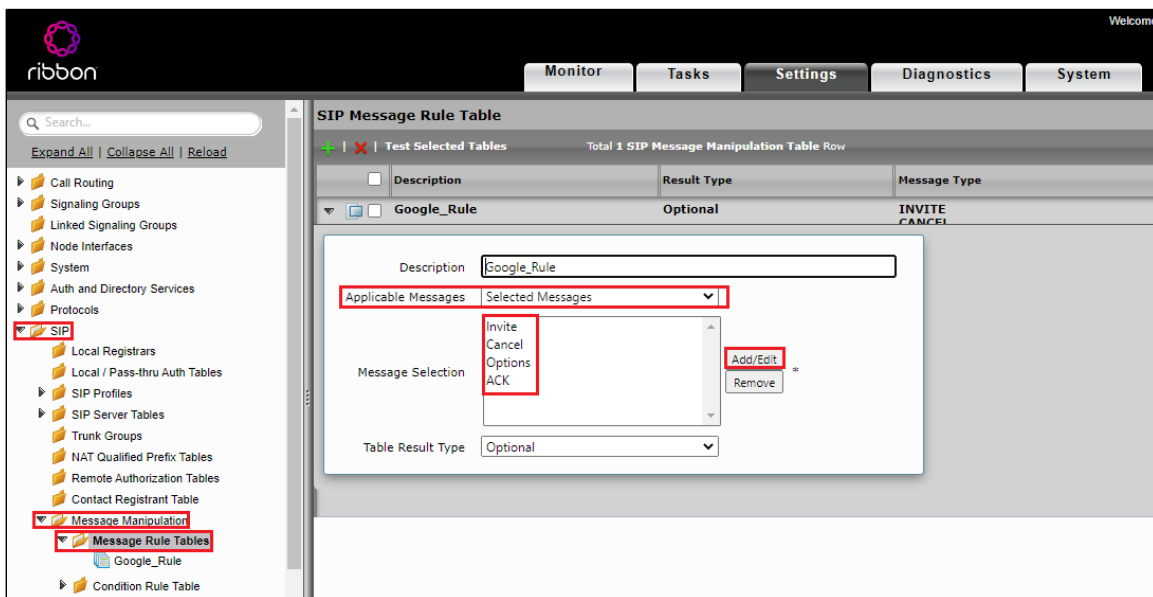
- To add the header X-Google-Pbx-Trunk-Secret-Key for Google Voice. The value of this header is generated when the SIP Trunk is created.
- To change the request URI of specific request messages to Google specified FQDN, trunk.sip.voice.google.com.

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.

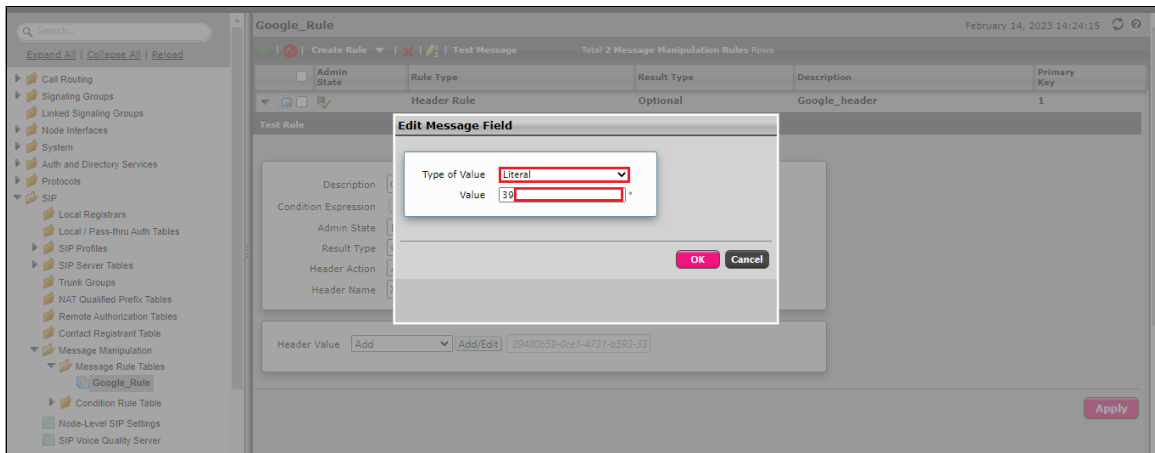
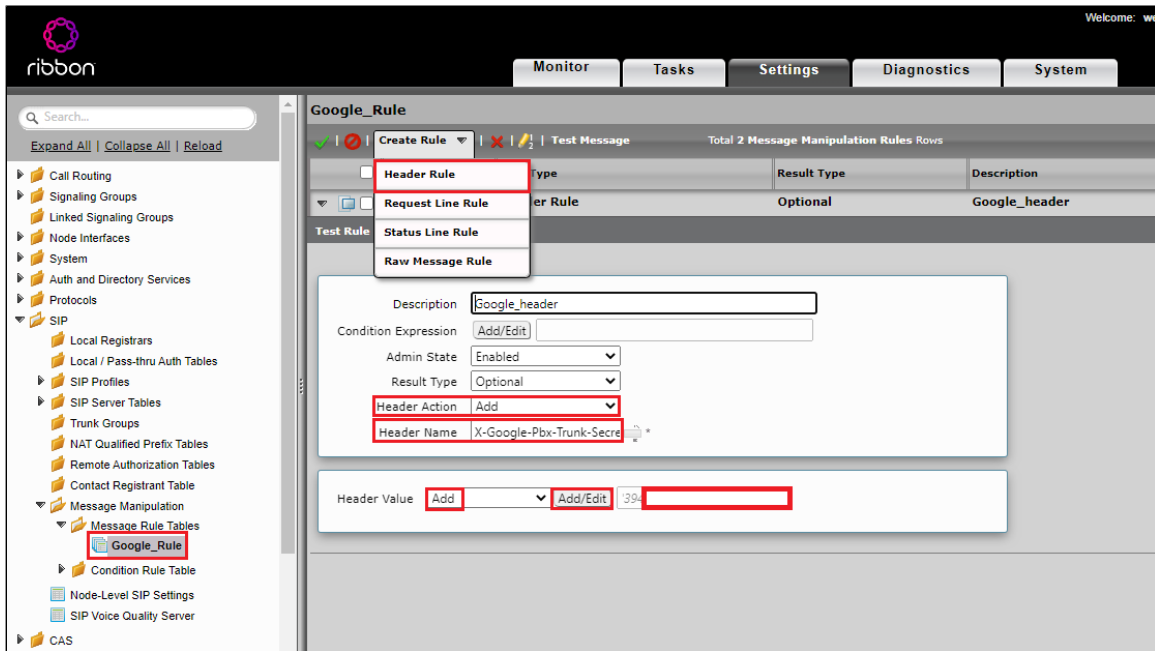
1. Provide a description for the Rule Table.
2. Apply Message Rule to the selected messages and choose Invite, Cancel, Options and ACK from the Message Selection list.
3. Click **OK**.



Message Rule Table Entry

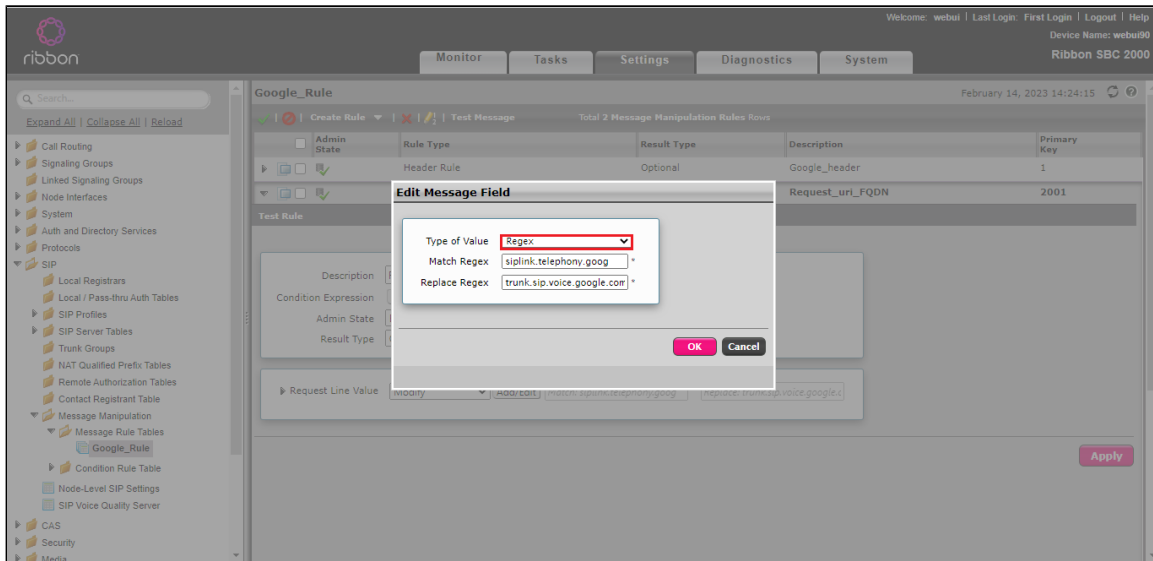
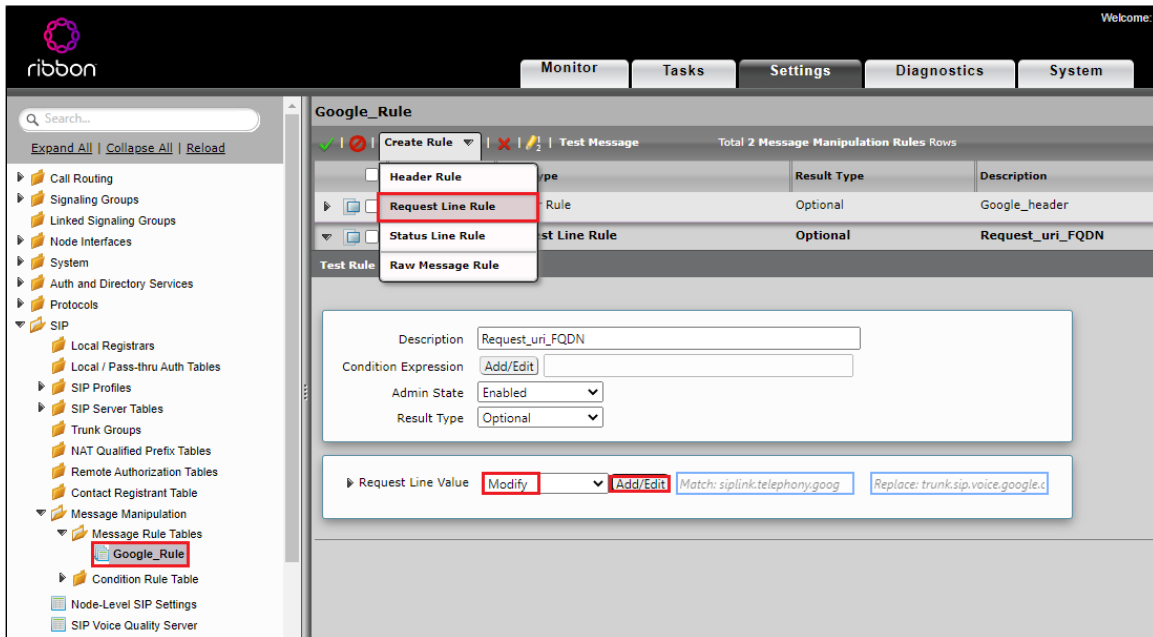
Header Rule:

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 RuleTable **GOOGLE_RULE**.
2. From the Create Rule drop-down menu, select Request Line Rule.
3. Provide a name for the entry in the Description field.
4. Replace the FQDN "siplink.telephony.goog" with "trunk.sip.voice.google.com" using regex.
5. 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 600 and the Offered Session Timer to 3600.
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 2 SIP Profile Rows

Description	Primary Key
Default SIP Profile	1
Google	2

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: SBC Edge FQDN
- FQDN in Contact Header: SBC FQDN
- 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: 33000 ms (5000, 64000)
- Timer B: 32000 ms
- Timer F: 32000 ms
- Timer H: 32000 ms (5*Timer T1)
- Timer J: 4000 ms (4000, 64000)

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



Note

The session will always be refreshed by Ribbon SBC Edge 1000 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 in the Description field.
2. Click **OK**.

SIP Server Tables

Total 3 SIP Server Table Rows

Description	Primary Key
Default SIP Server	1
Google	2
On-prem_PBX	4

Description: Google

Apply

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.

SIP Signaling Group - GV

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

1. Attach the Call Routing Table ([GV_TO_PBX_&_PSTN](#)).
2. Attach the SIP Profile ([Google](#)).
3. Attach the SIP Server Table ([Google](#)).
4. Attach the Media List ID ([Google](#)).
5. Associate the appropriate IP address in the "Signaling/Media Source IP" field.
6. Configure the Protocol, TLS Listen Ports and TLS Profile ([Google_TLS](#)) in the "Listen Ports" panel.
7. Provide the Google Voice SIP Link's FQDN or IP address in the Federated IP/FQDN panel.
8. Enable Message Manipulation and attach the profile [Google_Rule](#) to the Outbound Message Manipulation Table List.
9. Click **OK**.

Description

Admin State

Service Status

SIP Channels and Routing

Action Set Table

Call Routing Table

No. of Channels * [1..960]

SIP Profile

SIP Mode

Agent Type

Interop Mode

SIP Server Table

Load Balancing

Notify Lync CAC Profile

Challenge Request

Outbound Proxy IP/FQDN

Outbound Proxy Port [1..65535]

No Channel Available Override

Call Setup Response Timer [180..750] secs

Call Proceeding Timer [24..750] secs

QoE Reporting

Use Register as Keep Alive

Forked Call Answered Too Soon

Media Information

Supported Audio/Fax Modes

Supported Video/Application Modes

Media List ID

Play Ringback

Tone Table

Play Congestion Tone

Early 183

Allow Refresh SDP

Music on Hold

RTCP Multiplexing

Media Codec Latch

Mapping Tables

SIP To Q.850 Override Table

Q.850 To SIP Override Table

Pass-thru Peer SIP Response Code

SIP IP Details

Teams Local Media Optimization

Signaling/Media Source IP

Signaling DSCP * [0..63]

NAT Traversal

ICE Support

Static NAT - Outbound

Outbound NAT Traversal

Static NAT - Inbound

Detection

Listen Ports

Listen Port

Federated IP/FQDN

Total 1 SIP Federated IP Row

IP/FQDN	Netmask/Prefix
<input type="checkbox"/> siplink.telephony.goog	255.255.255.255

Inbound Message Manipulation

Message Table List

Outbound Message Manipulation

Message Table List

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	Basic calls	✓
2	Call Hold and Resume	✓
3	Call Transfer	✓
4	DTMF RFC	✓
5	Calling Party Number Presentation	✓
6	Calling Party Number Restricted	✓
7	Ring Group	✓
8	Auto Attendant	✓
9	Voice Mail	✓
10	Call Recording	✓
11	Call Forwarding by PSTN	✓
12	Short Codes Dialing	✓
13	Call Conference	✗
14	Simultaneous Ring	✗
15	Non E164 format	✗
16	Call Decline with 603 response	✗

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.

The below issues will be addressed by Google Voice in their upcoming releases.

S. No	Caveats	Description
1	Navigate Google-Voice Mail system	There is no option to navigate the voicemail portal after leaving voicemail. To complete the voice mail recording, you must hang up the phone.
2	Send 486 Busy response	Call waiting cannot be turned off at the moment. Google Voice does not send a 486 Busy response. This is a Google Voice limitation.

3	Session Refresh	Google Voice supports only UPDATE as a session refresh mechanism.
4	Call decline with 603 response	Google Voice does not support call rejection. When a Google Voice user declines a call, Google Voice forwards the call to voicemail.
5	Conference	Google Voice does not support conference.
6	Short code dial	Google Voice does not support the short code dial 0.
7	Call Forwarding	Google Voice does not support call forwarding

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 the successful configuration for Google Voice SIP Link interop involving the Ribbon SBC Edge 1000.

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.

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