# 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.TheSBC 1000is 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. Usethis 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 atWorking 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 visitWorking with Certificates
- Google Workspace and Domain
  - Google Voice Premier license for the users
  - For more details, contactGoogle 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.

ribbon	Welcome to Ribbon SBC 1000
	Users (authorized or unauthorized) have no explicit or implicit expectation of privacy. Any or all user 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 ciminal 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 ribbon Password Copyright 2010-2018 Sonus Networks, Inc. (a Ribbon Communications Company). All Rights Reserved

## 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.

noddin			G Monitor	Tasks	Settings	Diagnostics	System		
A faut	Current Licenses								Febry
Frank All Colours All Baland	Historical Usage								
Cat Routing     Singular Groups				Por	t Licenses				
Linked Signaling Groups									
🖡 🥩 Node Interfaces	Total 2 PortLacense Fous			_	_				
Application Solution Module	Feature	Licensed		Number of Lice	nsed Ports				
Visite and Settings	DS1 Ports	₩.		2					
0x6	FXS Ports	₩.		4					
DSPs									
System Timing				Feat	ire Licenses				
System Companding Law									
Current Licenses	Total 20 Feature License Rows								
License Keys	Feature			Licensed		Total Licenses		Available Licenses	
Sofeare Vanacement	SIP Calls			₩.		Unlimited		Unlimited	
Auth and Directory Services	SIP Registrations			<b>U</b>		Unlimited		Unlimited	
Protocols	DSP Resources			<b>W</b>		Unlimited		Unlimited	
F ≠ CAS	Forking			₩.		Unlimited		Unlimited	
Isourty	SBA			₩.		Unlimited		Unlimited	
<ul> <li>Media</li> <li>Jone Tables</li> </ul>	Active Directory			₩.		Unlimited		Unlimited	
Telephony Mapping Tables	Transcoding					Unlimited		Unlimited	
SNUMAIams     Leoping Configuration	CAS			₩.		Unlimited		Unlimited	
Emergency Services	CDR			₩.		Unlimited		Unlimited	
Notification Manager	OSPF			₩		Unlimited		Unlimited	
	839			<b>V</b>		Unlimited		Unlimited	
	1Psec			6		Not Licensed		Not Licensed	
	RBA			6		Not Licensed		Not Licensed	
	QoE			₩		Unlimited		Unlimited	
	BroadSoft Subscriber Data					Not Licensed		Not Licensed	
	AMR-WB					Not Licensed		Not Licensed	
	Video Passthrough			6		Not Licensed		Not Licensed	
	Additional WS2012R2 ASM Licens			₩.		Unlimited		Unlimited	
	SIP VQ Reporting					Not Licensed		Not Licensed	
	Direct Routing SBA			₩.		Unlimited		Unlimited	

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

For more details on Licenses, refer toWorking 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.

riddon			Monitor	Tasks	Settings	Diagnostics	System	
Q Search	Generate Certificate Signing	Request						
Call Routing	-	Subject Distinguished Name		_				
Signaling Groups	Common Name	rbbniot.com	* Hostname or FQDN					
Linked Signaling Groups	Subject Alternative Name DNS		comma-separated FQDI	v list				
▶	Email Address	[	Ĩ					
Auth and Directory Services								
Protocols	ISO Country Code	India						
▶ Ø SIP	State/Province							
CAS	Locality	e.g.: (	īty					
🕨 💋 Users	Organization	e.g.: (	Company					
🕨 💋 Login Messages	Organizational Unit	e.a.:	Department					
SBC Certificates	Key Length	2048 bits 🗸						
SBC Primary Certificate				_				
SBC Supplementary Certificates								
TLS Brofiles				ОК				
Change Password								
Eibbon Protect Bad Actors								
🕨 🏓 Media								
🕨 🥬 Tone Tables								
Telephony Mapping Tables								
SNMP/Alarms								
Elogging Configuration								
Emergency Services     Motification Manager								

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

്ര						Welcome: we
rioddin		Monitor	Tasks	Settings	Diagnostics	System
Q. Search       Excand All   Collapse All   Reload         ▶ Call Routing         ▶ Signaling Groups         ▶ Inked Signaling Groups         ▶ Inked Signaling Groups         ▶ Mode Interfaces         ▶ System         ▶ Auth and Directory Services         ▶ Protocols         ▶ Protocols         ▶ Login Messages         ▼ SBC Certificates         ■ Generate SBC Edge CSR         ■ SBC Primary Certificates         ■ SBC Supplementary Certificates         ■ Tusted CA Certificates         ■ Tusted CA Certificates         ■ Tusted CA Certificates         ■ Change Passovid	SBC Certificates Index  Generate SBC Edge CS SBC Primary Certificat SBC Supplementary C Trusted CA Certificate	SR te ertificates s				
Ribbon Protect Bad Actors						

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

#### To import an X.509 signed certificate:

- 1. SelectX.509 Signed Certificate from the Importmenu at the top of the page.
- 2. Chose the import mode (Copy and PasteorFile Upload) from theModepull-down menu.
- 3. If you choseFile Upload, use theBrowsebutton to find the file and clickOK.
- 4. If you chooseCopy and Paste, open the file in a text editor, paste the contents into thePaste Base64 Certificatetext field and clickOK.

#### To import a PKCS12 Certificate and Key:

- 1. SelectPKCS12 Certificate and Keyfrom theImportmenu at the top of the page.
- 2. Enter the password used to export the certificate in the Passwordfield.
- 3. Browse for the PKCS certificate and key file and click OK.

Import X.509 Server Certificate	Import X.50	9 Server Certificate
Mode Copy and Paste V	Mode F Select File (	le Upload
Paste Base64 Certificate		
	ок	
Import PKC Password Select File C	2 Server Certificate se File No file chosen Extensions [.pfx or .p12 OK	]*  -

#### **Trusted CA Certificates**

(i)

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 Settingstab, navigate to Security > SBC Certificates > Trusted CA Certificates.

<i>P</i> 3						Welcome:	wel
ribbon		Monitor	Tasks	Settings	Diagnostics	System	
Q Search	SBC Certificates Index						
Expand All   Collapse All   Reload	Generate SBC Edge CS	5R					
Call Routing	SBC Primary Certificat	te					
Signaling Groups	SBC Supplementary Control	ertificates					
Linked Signaling Groups	Trusted CA Certificate	s					
Mode Interfaces							
🕨 📁 System							
Auth and Directory Services							
Protocols							
▶ <mark> </mark> SIP							
CAS							
Security							
Users							
Login Messages							
Generate SBC Edge CSR							
SBC Primary Certificate							
SBC Supplementary Certificates Trusted CA Certificates							
TLS Profiles							
Change Password							
Ribbon Protect Bad Actors							
🕨 📁 Media							
Tone Tables							
Telephony Mapping Tables							
SNMP/Alarms							
Logging Configuration							

This section describes the process of importing Trusted Root CA Certificates using either the File UploadorCopy and Pastemethod.

- 1. To import a Trusted CA Certificate, click the Import Trusted CA Certificate (11) Icon.
- 2. Select eitherCopy and PasteorFile Uploadfrom theModemenu.
- 3. If you chooseFile Upload, use theSelect Filebutton to find the file.
- 4. ClickOK.

Import Trusted CA	Certificate		Import T	rusted CA Certificate	
Import Trusted CA ( Mode	Copy and Paste V		Import I Mode Select File	File Upload V Choose File No file chosen	Extensions (pem, der, cer, ber, p7b) *
	ОК	-			

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 toUnable To Get Local Issuer Certificatefor more information.

#### **Networking Interfaces**

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

#### NavigatetoNode Interfaces > Logical Interfaces.

Q Search	Logical I	nterfaces				
Expand All   Collapse All   Reload	×10	Total 7 LogicalInterface Ro	ws			
🕨 🥩 Call Routing		Interface Name	IPv4 Address	IPv6 Address	Description	Admin State
Signaling Groups	🕨 🗋 🗆	Ethernet 1 IP	10			Enabled
Vode Interfaces	Þ 🗊 🗆	Ethernet 2 IP	192			Enabled
Ports	🕨 🗋 🗆	Loopback 1				Disabled
Logical Interfaces	► 🗀 🗆	Loopback 2				Disabled
Ethernet 2 IP	🕨 🗋 🗆	Loopback 3				Disabled
Loopback 1	Þ 🗊 🗆	Loopback 4				Disabled
Loopback 2	Þ 🗊 🗆	Loopback 5				Disabled
E Loopback 4						
Loopback 5						

#### 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.

Q Search	Logical Interfaces			
Expand All   Collapse All   Reload	🧹   🥝 Total 7 I	ogicalInterface Rows		
🕨 🥖 Call Routing	Interface Name		IPv4 Address	
🕨 🥖 Signaling Groups	Fthernet 1 ID		10	
🥩 Linked Signaling Groups			10	
Node Interfaces     Ports		Identification/S	tatus	
Copical Interfaces     Ethernet 1 IP     Ethernet 2 IP     Loopback 1     Loopback 2	Interface Name Etherr I/F Index 29 Alias Description	net 1 IP		
E Loopback 3	Admin State Enab	led 🗸		
Loopback 4				
Loopback 5				
🕨 📁 Bridge		Networking	0	
Relay Config				
Application Solution Module     Application Solution Module     System     Node-Level Settings     QoE	MAC Address IP Addressing Mode	s 00:10	~	
DSPs		IPv4 Information		
System Timing				
System Companying Law		ACL In None	~	
<ul> <li>Construction</li> <li>Construction</li> <li>Construction</li> <li>Construction</li> </ul>		ACL Out None	~	
Auth and Directory Convince	A	CL Forward None	~	
Auth and Directory Services	IP Ass	ign Method Static	~	
	Prima	ary Address 10		XXXX
Gir     G	Prima	ry Netmask 255.25	5 255 0	****
d Local / Pass-thru Auth Tables	Configure Secondar	v Interface		
🕨 🥑 SIP Profiles	Configure Seconda	y interface Uisable	eu 🗸	
IP Server Tables				

#### 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.

Ked Signaling Groups     Node Interfaces	v 📄 🗆 Ethernet 2 IP 192.
🕨 🥩 Ports	
<ul> <li>Logical Interfaces</li> </ul>	Identification/Status
Ethernet 1 IP	
Ethernet 2 IP	Interface Name Ethernet 2 IP
Loopback 1	I/Findex 30
Loopback 2	Alias
Loopback 3	Description
Loopback 4	Admin State Enabled 🗸
Loopback b	
🕨 💋 Bridge	Hetwelie
Relay Config	Networking
Application Solution Module	
🔻 💋 System	
Node-Level Settings	MAC Address 00:10
III QoE	
C DSPs	
System Timing	IPv4 Information
System Companding Law	
Licensing	ACL In block
🕨 📁 Software Management	ACL Out None Y
Auth and Directory Services	
Protocols	
V SIP	
Local Registrars	Primary Address 192
Local / Pass-thru Auth Tables	Primary Netmask 255.255.0 x.x.x
SIP Profiles	Configure Secondary Interface Disabled
SIP Server Tables	
Irunk Groups	
NAT Qualified Prefix Tables	

## **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 ordefault 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 Settingstab, navigate to Protocols> IP > Static Routes. Click the ticon to add the entries.

ciobon			O Monitor	Tasks	Settings	Diagnostics	System
Q Search	Static IP Rou	te Table					
Expand All   Collapse All   Reload	E ×	Total 5 IP Route Rows					
▶ 📁 Call Routing	Row ID	Destination IP	Mask	Gateway	Adm	inistrative Distance	Primary Key
Signaling Groups	1	172.16.	255.255.255.0	10.54.	1		1
<ul> <li>Metworking Interfaces</li> <li>System</li> </ul>	2	74.125.	255.255.255.0	115.110.	1		2
Auth and Directory Services	3	216.239.	255.255.255.255	115.110.	1		3
Protocols	4	8.8.8.8	255.255.255.255	115.110.	1		4
V 🌽 IP	5	10.70.	255.255.0.0	10.54.	1		5
Static Routes							

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

						weicome
noddin		Monitor	Tasks	Settings	Diagnostics	System
Q Search Expand All   Collapse All   Reload	Media Profiles	Total 4 M	edia Profile Rows	_	_	_
Call Routing     Signaling Groups     Linked Signaling Groups	Voice Codec Profile Fax Codec Profile			Description Default G711A		
Node Interfaces     System     Auth and Directory Services     P      Protocols	G.722 WB     G.729			G722 G729		
<ul> <li>▶ SIP</li> <li>▶ GAS</li> <li>▶ Security</li> <li>▼ Media</li> </ul>						
Media System Configuration  Media Profiles  Default G711A  Default G711u						
G722 G729 G729 G729 G729 G729 G729 G729						
🕨 🭺 Media List						

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 C	odec Profile	
Vo	ice Codec Configurati	on
Description	G729	
Codec	G.729 🗸	
Payload Size	20 🗸	ms
_		_
		UK
Create Voice C	odec Profile	
Create Voice C	odec Profile	
Create Voice C	odec Profile	tion
Create Voice C	odec Profile vice Codec Configurat	tion
Create Voice C	odec Profile bice Codec Configurat G22 G.722	tion
Create Voice C Vo Description Codec Rate	odec Profile bice Codec Configurat G22 G.722 ~ 64000 b/s	tion
Create Voice C Vo Description Codec Rate Payload Size	odec Profile bice Codec Configurat G22 G.722 64000 b/s 20 ms	tion
Create Voice C Vo Description Codec Rate Payload Size	odec Profile	tion

#### **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.

O		Welcome: admin i
noddin	O Monitor Tasks Settings Diagnostics System	
	Transformation	
Q Search	Transionnation	
Expand All   Collapse All   Reload	t X I B Total 6 Transformation Table Rows	
Call Routing	Description Primary Key	
V 🖉 Transformation	▶	
GV_TO_PBX	b G GV TO PSTN 3	
PBX TO GV		
PBX TO PSTN	▶ □ PBX_IO_6V 6	
PSTN_TO_GV	▶ 🔯 🗋 PBX_T0_PSTN 1	
PSTN_TO_PBX	▶	
💋 Time of Day Table		
Call Routing Table		
Call Actions		
Signaling Groups		
Linked Signaling Groups		
Mode Interfaces		
Application Solution Module		
System		
Auth and Directory Services		
CAS		
Security		
🕨 🧀 Media		
🕨 🍺 Tone Tables		
🕨 🥬 Telephony Mapping Tables		
🕨 🧊 SNMP/Alarms		
Logging Configuration		
Emergency Services		
p Notification Manager		

#### **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.

<b>6</b> 3							Procession, and	III I Last Logit. Teo 22, 2023 00	Device
noddin		O Monitor	Tasks	Settings	Diagnostics	System			
Q Search	GV_TO_PBX							February 22, 2	2023 23:14
Expand All   Collapse All   Reload	VIOI+IXI/2 Total 2 Transformat	ion Entry Rows							
Total Routing	Admin Input Field Type	Input Field Value		Output Field Type		Output Field Value	Match Type	Description	Prin
Transformation	🔻 📴 🗌 🎼 Called Address/Number	\+1(9 12(.*))		Called Address/Num	ber	\1	Optional (Match One)	Entry ID 1	1
GV_TO_PBX									
PBX_TO_GV									
PBX_TO_PSTN	Description		_						
PSTN_TO_GV	Admin State Enabled 🗸		_						
PSTN_TO_PBX	Match Type Optional (Match One) V		_						
Time of Day Table									
Call Routing Table									
PSTN TO ENTERPRISE Voice	Input Field	Output Field							
GV_TO_PBX_&_PSTN									
E PBX_TO_GV_&_PSTN	Type Called Address/Number 🗸	Type Called Address/Number	~						
🕨 🥩 Call Actions	Value \+1(9 12(.*))	Value \1							
🕨 🧊 Signaling Groups									
Linked Signaling Groups			_						
P P Node Interfaces									
Approarbon Solution Module			Apply						
Auth and Directory Services									
🕨 🥬 Protocols									
🕨 🍺 SIP	Calling Address/Number	\+1(.*)		Calling Address/Numbe	er	\1	Optional (Match One)	Entry ID 2	2

#### 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**.

		Monitor	Tasks Settings	Diagnostics System	• • • • • • • • • • • • • • • • • • • •	о, амини і клигерун, і ор којкоко кононк 
Cibbon	CV_TO_PSTN       Cold D Transformation Entry       Mathematical State       V I I V       Called Address/Number       Description       Admin State       Final       Units       Total D Transformation Entry       V I I V       V I I V       V I I V       V I I V       V I I V       V I I V       V I I V       V I I V       V I I V       V I V   <	Control Monitor	Tasks Settings Output Field Type Called Address/Number	Diagnostics System	Hatch Type Optional (Hatch One)	February 21, 2023 ( pescription Entry ID 1
Image Signaling Groups       Image Signaling Groups <t< th=""><th>Input Field           Type         Called Address/Number         ✔           Value         [+1](vil (0))         ↓</th><th>Output Field Type Called Address/Number Value 11</th><th>Apply</th><th></th><th></th><th></th></t<>	Input Field           Type         Called Address/Number         ✔           Value         [+1](vil (0))         ↓	Output Field Type Called Address/Number Value 11	Apply			
Fore Tables     Jelephony Mapping Tables     Jelephony Mapping Tables     Jelephony Mapping Tables	Calling Address/Number      Called Address/Number	\+1(\d(10}) (.*)	Calling Address/Number Called Address/Number	u/ 1/	Optional (Match One) Optional (Match One)	Entry ID 2 Entry ID 3

#### Transformation Table Entry for PBX\_TO\_GV

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

ribbon		O Monitor	Tasks Settings	Diagnostics System		
Q. Search	PBX_TO_GV	try Rows				February 21, 202
Call Routing	Admin State Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description
	Called Address/Number  Called Address/Number  Called Address/Number  Imput Field  Type Called Address/Number Value  ()  Called Address/Number Value ()  Calle	(9 2262(*1)) Output Field Type Calica Address/humber  Value +111	Calied Address/Number	+1/1	Optional (Hatch One)	Entry ID 1
Media						
Course startes     Telephony Mapping Tables     ShiMP/Alarms     ShiMP/Alarms     Gengency Services     Notification Manager	Laling Address/Number	(,-')	Caming Address/Number	+1/1	uptional (Match One)	Entry ID 2

#### 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.

Control     Tasks     Settings     Diagnostics     System       Qtexts     PSTM_TO_GV     Tasks     Settings     Diagnostics     System       Qtexts     PSTM_TO_GV     Tasks     Tasks     Settings     Diagnostics     System       C_texts     PSTM_TO_GV     Tasks     Tasks     Tasks     Tasks     Tasks     Tasks       C_texts     PSTM_TO_GV     Tasks     Tasks     Tasks     Tasks     Tasks     Tasks       C_texts     PSTM_TO_GV     Tasks     Tasks     Tasks     Tasks     Tasks     Tasks       C_texts     PSTM_TO_GV     State     Tasks     Tasks     Tasks     Tasks     Tasks       C_texts     State     Tasks     Tasks     Tasks     Tasks     Tasks     Tasks       C_texts     State     State     Tasks     Tasks     Tasks     Tasks     Tasks       C_texts     State     State     Tasks     Tasks     Tasks     Tasks     Tasks       C_texts     State     State     State     Tasks     Tasks     Tasks     Tasks       C_texts     State     State     State     State     State     Tasks     Tasks       State     S	e: admin   Last Login: Feb 21, 2023 00:38:19 De	Welcome: ad							Ô
C Statistic     PSTH_TO_GOV       Excelled   follower A  [follower A ]     Total I Transformation Entry Flow       C < d Bonder		Diagnostics System	Diagnostics	Settings	Monitor Tasks				rioddin
Forty To, PEX     Twe of Dy Take     Cal Rading Groups     Lited Spating Groups     Lited Spating Groups     Lited Spating Groups     Cal Rading Groups     Cal Rading Take     Cal R	February 21, 2023 (7 Geoergition Entry 10 1	Output Field Hatch Value Type +19 52262/1 Optional (Hatch One)	Output Value +19	staut Field Type alled Address/Number	(262(-*) () () () () () () () () () () () () () (	Cartay Nov Ingest Fink (+13) Type Colled Addre Value 137 - 200	Total 1 transformation       Input Field Type       Called Address/Number       Enabled       Input Field       Called Address/Number       V       V	PSTN_TO_GV	Q. Senth.  Reach.   Collect.Al   Selical  Collect.   Collect.Al   Selical  Port Collect.   Selical  Port Collect.
<ul> <li>In a Tables</li> <li>In a Tables</li> <li>In a Tables</li> <li>In a Tables</li> <li>Statistical and the state of the s</li></ul>									<ul> <li>▶ 2 Tone Tables</li> <li>▶ 2 Telephony Mapping Tables</li> <li>▶ 2 SNMP/Alarms</li> <li>▶ 2 Logging Configuration</li> <li>▶ 2 Emergency Services</li> </ul>

#### Transformation Table Entry for PSTN\_TO\_PBX

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

<b>6</b> 3					Welcome: admin   Last Lo
cippoo			Taska	Discussion Sustain	
nooon		O Monitor	Tasks Settings	Diagnostics System	
Q Search	PSTN_TO_PBX				
Current All J. Colleges All J. Deleged	Total 1 Transformation E	Entry Row			
Exterior All 1 Consider All 1 Actual	Admin	Input Field		Output Field	Match
Call Routing	State Input Field Type	Value	Output Field Type	Value	Туре
Transformation	💌 📋 🗋 🖳 Called Address/Number	97 012(.*)	Called Address/Number	97 3012\1	Optional (Match One)
B PBY TO PSTN	Description				
PSTN TO GV	Admin State Enabled V				
PSTN_TO_PBX	Natch Tune Ontional (Match One)				
Time of Day Table	inatch type optional (match one)				
Call Routing Table					
🕨 🥩 Call Actions			_		
Signaling Groups	Input Field	Output Field			
Linked Signaling Groups					
Mode Interfaces	Type Called Address/Number 🗸	Type Called Address/Number 🗸			
Application Solution Module	Value 97: #8012(.*)	Value 97 18012\1			
🕨 🏓 System					
Auth and Directory Services			_		
Protocols					
🕨 🏓 SIP	1				
🕨 🏓 CAS			Apply		
Security					
🕨 🃁 Media					
Tone Tables					
Telephony Mapping Tables					
SNMP/Alarms					
Elogging Configuration					
Emergency Services					
Notification Manager					

## **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 Settingstab, navigate to Call Routing>Call Routing Table. Click the + iconto create a Call Routing Table.

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

	Manifer				Wel	come: webui l
Call Routing Tables	Monitor all Routing Table R Google Chrome vebui90.e.sonusn ole	Tasks ows et.com:8080/cg Fet	Settings	Diagnostics	System	Primary 1 2 3 5

## **Call Routing Table Entry**

PSTN to ENTERPRISE\_Voice:

<b>&amp;</b> 2										Device Name: sbc7
noddin			💿 Mor	itor Tasks	Settings	Diagnostics	System			SBC 1000
(Q, Search	PSTN_TO_ENTERPRISE	Voice							Febru	ary 21, 2023 02:21:49 🔿 🙆
Expand All   Colleges All   Beload	🗸 1 🖉 1 🕂 1 🗙 1 🥂 1 Disp	lay Counters Ti	cal 2 Call Route Entry Rows							
🖲 🌽 Call Rousing	Admin	Priority	Transformation Table	Destinatio	n	First Signaling Group	p	 Description	Fork	Primary Key
🕨 🥩 Transformation		1	PSTN_TO_GV	Normal		(SIP) Google_SG		PSTN_TO_GV	Yes	1
Time of Day Table	1 (B) (B) (B)		DETN TO DEV	Normal		(810) Oncome 08V		OPTN TO DRY		2
Call Rousing Table		-	PSIN_IO_PEX	reormai		(StP) On-prem_PBX		PSIN_IO_PBX	140	4
PSTN TO ENTERPRISE Writer										
GV_TO_PBX_&_PSTN										
PBX_TO_GV_&_PSTN										
🕨 🥩 Call Actions										
Signaling Groups										
Linked Signaling Groups										
🕨 🂋 Node Interfaces										
🕨 💋 Application Solution Module										
🕨 🏓 System										
Auth and Directory Services										
Protocola										
k d Sanata										
k 🖬 Meria										
Tone Tables	5									
Telephony Mapping Tables										
🖡 🂋 SNMPIAlams										
Logging Configuration										
Emergency Services										
🥬 Notification Manager										

#### 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.

Ô										TRACTING BARRET I LAN LO	Device Name
rioddin				O Monitor	Tasks	Settings	Diagnostics	System			SBC
O Search	PSTN_TO_ENTERPRISE_Voi			_							February 21, 2023 02:21:49
Expand All   Collapse All   Reload		iounters Total 2 Cr									
🕶 🌽 Call Routing	Admin Priority Transformation Table				Destination		First Signaling Gro	up	 Description	Fork	Primary
Transformation	- 🗊 🖡	1	PSTN_TO_GV		Normal		(SIP) Google_SG		PSTN_TO_GV	Yes	1
Tal Routing Table	Descrip	tion PSTN TO GV									
ESTN TO ENTERPRISE Voice	Admin s	itate Enabled V									
OV_TO_PEX_6_PSTN	Route Pri	ority 1 💌									
PEX_TO_OV_&_PSTN	Call Pri	ority Normal V									
<ul> <li>Gal Actors</li> <li>Signaling Groups</li> </ul>	Number/Name Transformation 1	Table PSTN_TO_GV	v +								
📁 Linked Signaling Groups	Time or pay Nasth	coon [reone	•		_						
<ul> <li>P Inode Interfaces</li> <li>Acceleration Solution Module</li> </ul>			Destination Information								
🖡 📁 System											
Auth and Directory Services	Destination Type	Normal V	¥ 4								
▶ 💋 SIP	Cause Code Reroutes	None									
🕨 🏓 CAS	Cancel Others upon Forwarding	Disabled ¥									
<ul> <li>Seconty</li> <li>Vedia</li> </ul>	Fork Call	Yes 🗸									
🕨 🧀 Tone Tables		(SIP) Google_SG	A 40								
<ul> <li>Palephony Mapping Tables</li> <li>Control Visions</li> </ul>			Down								
Logging Configuration	Description signaling Groups		Add/Edit								
Emergency Services			* Ramova								
Notification Manager	Enable Maximum Call Duration	Disabled V									
		Media	Quality	of Service	_						
	Audio/Fax Stream Mode	DSP	Quality Metrics Number of C	alls 10 [1.1	207						
	Media Transrodina	Eashied	Quality Metrics Time Before Re	rry 10(7-6	37 min.						
	Media Ust	Default Media List	V +	old 0 5.0	. 100)						
			Enable Min NOS Thresh Enable May, 8/T De	lav Eastlad V							
			Max. B/T De	lav (45535 mm)	10000						
			Enable Max. Jr	ter Enabled ¥							
			Max. 20	ner 3000 mu (	1.3000/						
					_						

#### 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 :

<ul> <li>viðbon</li> </ul>				O Monitor	Tasks Settings	Diagnostics	System			
Q. Search	GV_TO_PBX_&_PSTN								Feb	ruary 21, 2
Expand All   Collapse All   Reload	🧈 I 🥥 I 🕂 I 🗙 I 🥂 I Displ	ay Counters Total 2	Call Route Entry Rows							
Y 🔂 Cal Bryten	Admin	Priority	Transformation Table		Destination	First Signaling Grou	19	Description	Fork	Primary
Ingristormation	state	2	GV TO PSTN		Normal	(TCDN) 001 T1		GV TO BETN	No	Rey
🥩 Time of Day Table		-	Gr_10_r514		Normal	(1904) MU_LT		GV_10_P31N	140	
🔻 🔂 Call Routing Table	▶ 🔝 🗆 🕸	1	GV_TO_PBX		Normal	(SIP) On-prem_PBX		GV_TO_PBX	No	2
CTU T_QUITERREQUEA										

#### 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.

									Device Name: :
noddin			Monitor Tasks	Settings	Diagnostics	System			SBC 1
(Q, Search	GV_TO_PBX_&_PSTN							Febru	ary 21, 2023 02:27:55 🧲
Expand All   Collagae All   Seload	🗸   🖉   🐥   🗶   🥂   Display Counters 🛛 🛛 To	tal 2 Call Route Entry Rows							
🕈 💋 Call Routing	Admin Priority	Transformation Table	Destina	tion	First Signaling Group		Description	Fork	Primary Key
Transformation	v 😭 🖖 2	GV_TO_PSTN	Norma		(ISDN) PRI_T1		GV_TO_PSTN	No	1
Call Rousing Table	Description GV TO PSTN								
Default Route Table	Admin State Foabled								
PSTN_TO_ENTERPRISE_Voice	Boute Priority 2	1							
PBX_TO_GV_&_PSTN	Call Priority Normal	-							
🕨 🥬 Call Actions	Number/Name Transformation Table GV_TO_PSTN	× +							
🕨 💋 Signaling Groups	Time of Day Restriction None	<b>~</b> +							
Linked Signaling Groups									
Application Solution Module		Destination Information							
🕨 🃁 System		7							
Auth and Directory Services	Destination Type Normal V								
k ∰ siP	None None	· ·							
🕨 🥩 CAS	Cased Others uses Exceeding District M								
Security	Early Call No.								
<ul> <li>Media</li> <li>Tone Tables</li> </ul>	000000 001 T1								
🖡 🍺 Telephony Mapping Tables	(30N) PN_11	∩Up							
SNMPIAlams	Destination Signaling Groups	Down +							
<ul> <li>Logging Configuration</li> <li>Emergency Services</li> </ul>		Add/Edit							
p Notification Manager		-							
	Enable Maximum Call Duration Disabled ¥	]							
			Quality of Consists	_					
	media		Quality of Service						
	Audio/Fax Stream Mode DSP	Quality Metric	cs Number of Calls 10	[7.100]					
	Video/Application Stream Mode Disabled	Quality Metrics	Time Before Retry 10	(1-60) min.					
	Media Transcoding Enabled	× N	lin. ASR Threshold 0	96 (D. TOD)					
	Media List Default Media List	- + Enable M	tin MOS Threshold Disabled V	1					
		Enab	ole 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 (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.

ribbon				Monitor	Tasks	Settings	Diagnostics	System			
Q. Search	GV_TO_PBX_&_PSTN									Fel	ruary 21, 2023
Expand All   Collapse All   Reload	🧈 🕗 l 💠 l 🗙 l 🥂   Display Coun	nters Total 2 C	Call Route Entry Rows								
🔻 💋 Cal Routing	Admin Prior	rity Tra	nsformation Table		Destination Type		First Signaling Gro	up	Description	Fork	Primary Key
Transformation	▶ 🛄 🗋 🗤 🔰 2	GV	_TO_PSTN		Normal		(ISDN) PRI_T1		GV_TO_PSTN	No	1
Call Routing Table	* 🛄 🗋 🗤 🔰 1	GV	_TO_PBX		Normal		(SIP) On-prem_	PBX	GV_TO_PBX	No	2
Default Route Table	Description	GV TO PBX									
GV TO PEX & PSTN	Admin State	e Enabled V									
PBX_TO_GV_&_PSTN	Route Priority	y 1 💙									
🕨 📁 Call Actions	Call Priority	y Normal 🗸									
Signaling Groups	Number/Name Transformation Table	e GV_TO_PBX	× +								
<ul> <li>Cirket algrang broups</li> <li>Mode Interfaces</li> </ul>	Time of Day Restriction	n None	¥ 🔸								
Application Solution Module						_					
System Auth and Directory Services		1	Destination Information								
Protocols	Destination Type	Normal									
▶ 💋 SIP	Message Translation Table	None	× +								
P CAS	Cause Code Reroutes	None	<b>*</b> +								
🕨 🏓 Media	Cancel Others upon Forwarding	Disabled 🗸									
🕨 📁 Tone Tables	Fork Call	No 👻									
<ul> <li>Jecephony Mapping Tables</li> <li>SNMP/Alarms</li> </ul>		(SIP) On-prem_PBX	A 100								
Logging Configuration			Down								
Emergency Services     Manual Measure	Destination Signaling Groups		Add/Edit								
			Remove								
	Enable Maximum Call Duration	Disabled 🗸									
						_					
	N	Aedia		Quality of S	ervice						
	Audio/Fax Stream Mode	DSP	V Quality Me	etrics Number of Calls	10 //	a					
	Video/Application Stream Mode D	lisabled	Quality Metr	ics Time Before Retry	10 (1-6)	() min					
	Media Transcoding	Enabled	~	Nin, ASR Threshold	0 8.0	1001					
	Media List	None	✓ + Enab	le Min MOS Threshold	Disabled V						
				inable Max. R/T Delay	Enabled 💙						
				Max. R/T Delay	65535 ms [	.65535)					
				Enable Max. Jitter	Enabled 💙						
				Max. Jitter	3000 ms (	.30007					
						_					

#### PBX\_TO\_GV\_&\_PSTN:

$\bigcirc$										•	Device Name
noddin				Monitor	Tasks	Settings	Diagnostics	System			SBC
Q Search	PBX_TO_GV_&_PST	N								Februs	ry 21, 2023 02:48:47
Expand All   Collapse All   Reload	<101+1×1/¦1	Display Counters	Total 2 Call Route Entry Rours								
T 🖉 Call Routing	Admin	Priority	Transformation Table		Destinat	ion	First Signaling Gro	oup	Description	Fork	Primary
🕨 🥩 Transformation	k In O By	2	PRX TO PSTN		Normal		(ISDN) PRI TI		PRX PSTN	No	1
💋 Time of Day Table							()				
Call Routing Table		1	Pex_10_6v		Normai		(STN) Google_SG		Mex_10_GV	reo.	4
PSTN TO ENTERPRISE Voice											
GV_TO_PBX_&_PSTN											
PBX_TO_GV_&_PSTN											
🕨 🥩 Call Actions											
Signaling Groups											
💋 Linked Signaling Groups											
🕨 🥖 Node Interfaces											
Application Solution Module											
🕨 🏓 System											
Auth and Directory Services											
Protocols											
P P SIP											
CAS											
Security											
k Trans Tables											
k d Telesters Mersion Teles											
F and the second sec											
Lospine Configuration											
Emergency Services											
Notification Manager											

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.

$\bigcirc$													D
noddin				Monitor	Tasks	Settings	Diagnostics	System	1				
	PBX TO GV & PSTN											February	21, 2023.0
d search.	Diseley Con	nters Total 2 (	Call Route Entry Rous										_
Expand All   Collapse All   Keldad	Admin		and an attention Table		Destinatio	1	First Classifier C			Description	For		Primary
Call Housing	State		ansion hadie		Туре		Prisc bighaning G	roup		Description	Call		Кеу
🥖 Time of Day Table	* 🗊 🗤 2	PI	BX_TO_PSTN		Normal		(ISDN) PRI_TI			PBX_PSTN	No		1
🕈 🎾 Call Routing Table	Descriptio	n PBX_PSTN											
Default Route Table     PRIN TO ENTERPRISE Voice	Admin Stat	e Enabled 🗸											
W TO PBX & PSTN	Route Priorit	V 2 ¥											
PBX_TO_GV_&_PSTN	Call Priorit	y Normal 💙											
🕨 📁 Call Actions	Number/Name Transformation Tab	e PBX_TO_PSTN	✓ +										
Signaling Groups	Time of Day Restrictio	n None	¥ +										
Should bighting Groups						_							
Application Solution Module			Destination Information										
System	Destination Tune	Normal											
Protocols	Message Translation Table	None	× .										
🖡 🥩 SIP	Cause Code Bernutes	None	× .										
CAS	Cancel Others upon Enguarding	Disabled ¥											
Security Media	Fork Cal	No.											
🕨 🥩 Tone Tables		ISDND DRI T1											
Telephony Mapping Tables		(10010)110_11	Up										
SNMP/Alarms I continue Configuration	Destination Signaling Groups		Down +										
Emergency Services			Remark										
p Notification Manager													
	Enable Maximum Call Duration	Disabled 🗸											
						_							
		Media		Quality of Se	rvice								
	Audio/Fax Stream Mode	DSP	✓ Quality Met	trics Number of Calls	10 [7.10	y I							
	Video/Application Stream Mode	Disabled	Quality Metri	cs Time Before Retry	10 17-60	min							
	Media Transcoding	Enabled	×	Nin, ASR Threshold	0 % /0.	1001							
	Media List	None	✓ + Enable	Min MOS Threshold	Disabled ¥								
			Er	able Max. R/T Delay	Enabled ¥								
				Max. R/T Delay	65535 ms /1	655351							
				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.

								WED	ome: admin I Last Login: Feb 20,	2023 23 43 24 1 Devi
nocen			O Monito	or Tasks	Settings	Diagnostics	System			
Q. Search	PBX_TO_GV_&_PSTN								Februa	ry 21, 2023 02:
Expand All   Collapse All   Reload	🥪   🥝   🦛   🗙   🥂   Display Co	anters Total 2 Call Route E	ntry Rows							_
T i Call Routing	Admin Pr	iority Transformatic	n Table	Destination		First Signaling Gro	up	 Description	Fork	Primary
Fight Transformation	▶ 🗊 🛛 🗤 2	PBX_TO_PST		Normal		(ISDN) PRI_T1		PBX_PSTN	No	1
Call Routing Table	T 🗊 🗤 🚺	PBX_TO_GV		Normal		(SIP) Google_SG		PBX_TO_GV	No	2
Default Route Table	Description	Dev to cv								
BY TO PBX & PSTN	Admin Sta	the Frahlard V								
PBX_TO_GV_&_PSTN	Route Prior	ity 1								
Call Actions	Call Prior	ity Normal 🗸								
Signaling Groups	Number/Name Transformation Tal	ble PBX_TO_GV	Y +							
Cricked signaling Groups     Mode Interfaces	Time of Day Restricti	on None	<b>*</b> +							
🕨 🥩 Application Solution Module					_					
System		Destinatio	n Information							
<ul> <li>Protocols</li> </ul>		(m								
k 🧋 sin	Destination Type	Normal •	7							
🕨 📁 CAS	Message Translation Table	None	• •							
Becuity	Cause Code Reroutes	None	+							
P 📁 Media	Cancel Others upon Forwarding	Disabled ¥								
<ul> <li>Inter labels</li> <li>Talanhory Manning Tablas</li> </ul>	Fork Call	No ¥								
F SNMP/Alarma		(SIP) Google_SG	*							
Logging Configuration			- Op							
Emergency Services	Destination Signaling Groups		- A44/542							
Notification Manager			Remove							
			*							
	Enable Maximum Call Duration	Disabled ¥								
					_					
		Media	Qualit	ty of Service						
	Audio (Sau Charan Mada	000 14		a 8 10						
	Video (Application Stream Mode	Disabled	Quality Hetrics Number of		0					
	Nedia Transrodion	Frahied Y	Quality Metrics Time Before R	Retry 10 [1-60	min.					
	Martia List	None	Min. ASR Three	shold 0 % (0.	100)					
			Enable Nin NOS Three	shold Disabled ¥						
			Enable Max. R/T D	Delay Enabled 💙						
1			Max. R/T C	Delay 65535 ms (1	.655357					
			Enable Max.	Jitter Enabled 💙						
			Max. ;	Jitter 3000 ms /1	.3000j					

## 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.

						AACIC!	UIIIC. WCD
ribbon		Monitor	Tasks	Settings	Diagnostics	System	
Q. Search Expand All   Collapse All   Reload Call Routing	Media List View	: Media List Rows				Primary Key	
Inked Signaling Groups      Mode Interfaces	Google	ist				1 2	
	On-prem PBX     Description	On-prem PBX Default G711A Default G711u	Lup			4	
<ul> <li>▶ Ø Security</li> <li>▼ Ø Media</li> <li>■ Media System Configuration</li> <li>▶ Ø Media Profiles</li> </ul>	Media Profiles List	6729	Add/E Remov	n idit ivve			
SDES-SRTP Profiles DTLS-SRTP Profiles	SDES-SRTP Profile	None	✓ Associate	ed SIP SG Listen Ports shou	ıld be TLS only. 🕇		
Vedia List	DTLS-SRTP Profile	None	<b>∼</b> +				
Default Media List     Google     On-prem PBX	Media DSCP RTCP Mode	46 RTCP	* [063]				
Figure Tables	Dead Call Detection	Disabled	~				
felephony Mapping Tables     SNMP/Alarms     Logging Configuration	Silence Suppression	Enabled	~				
<ul> <li>Emergency Services</li> <li>Notification Manager</li> </ul>							

## 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 Settingstab, navigate to SIP > SIP Server Tables. Click the + iconto create a new SIP Server Table.

- 1. Provide a name for the SIP Server in the Description field.
- 2. Click OK.

rioddin	Monitor Tasks Settings Diagnostics	System
Q Search	SIP Server Tables Total 3 SIP Server Table Rows	Brimany
Call Routing     Signaling Groups     Linked Signaling Groups	Description     Default SIP Server     Gondle	Key 1
Node Interfaces     System     Auth and Directory Services     Sectorals	On-prem_PBX	4
Incal / Pass-hru Auth Tables	Description On-prem_PBX	
SIP Forfiles     Trunk Groups     NAT Qualified Prefix Tables	Apply	
Remote Authorization Tables     Contact Registrant Table     Message Manipulation     Node-Level SIP Settings     SIP Voice Quality 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 PBX.
- 4. Click OK.

						welcome	: we
ribbon		Monitor	Tasks	Settings	Diagnostics	System	
	A On-prom DRV	_					
Q Search	on prem_PBX						
Expand All   Collapse All   Reload	Create SIP Server 🔻 📘	K   🖉 Total 1	SIP Server Row				
Call Routing	IP/FQDN	Serv	rer Lookup	Port	Protocol	Display Counters	
🕨 🍺 Signaling Groups	DNS-SRV	IP/	FODN	5060	UDP	Counters	
📁 Linked Signaling Groups		,					_
🕨 📁 Node Interfaces		Company Used		Trees			
🕨 🃁 System		Server nost		Trans	port		
Auth and Directory Services	Canvas Las laura ID/CO	DN					
🕨 🍺 Protocols	Server Lookup IP/FQ			Monitor None	<b>~</b>		
V 💋 SIP	Priority 1	~	_				
Local Registrars	Host FQDN/IP 172		*				
Local / Pass-thru Auth Tables	Dent 5050	* (1 65525)	·				
SIP Profiles	Port 3000	. [10555]					
V SIP Server Tables	Protocol UDP	❤ *					
Default SIP Server							
Google	Remote Au	uthorization and Contac	ts				
Ch-prem_PBX							
Trunk Groups	Remote Authorization	Table None	<b>v</b> +				
NAT Qualified Prefix Tables			=:				
Contact Registrant Table	Contact Registrant	Iable None	<b>* *</b>				
Monage Manipulation	Session URI Valid	ation Liberal 🗸					
Inde. Level SIP Settings							
SIP Voice Quality Server							
CAS Security					Apply		
Media							
<ul> <li>Impound</li> <li>Tone Tables</li> </ul>							
Telephony Manning Tables	-						

#### **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 whichCall Routesare selected. They are also the location from whichTone TablesandAction Setsare selected.

From the Settingstab, 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 appropriateIP 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	▼	
5	SIP Channels and Routing	
		Media Information
Action Set Table		
Call Routing Table		Supported Audio/Fax Modes
No. of Channels		Direct
SIP Profile	Default SiP Profile	Supported Video/Application Disabled
SIP Mode	Dask to Pack Hars Accest	Modes
Agent Type	Back-to-Back User Agent	Play Pinghack Auto on 190
Interop Mode	Standard	Tone Table Default Tone Table
Load Balancing	Reved Pabla	Play Congestion
Notify Lyon CAC Profile		
Challenge Request		Early 183 Disable
Outbaued Deeue ID/CODN		SDP Enable
Outbound Proxy IP/PQDN		Music on Hold Disabled
Outbound Proxy Port	5060 [165535]	Multiplexing Disable
No Channel Available Override	34: No Circuit/Channel Available	Media Codec Latch Enable
Call Setup Response Timer	255 [180750] secs	
Call Proceeding Timer	[180] [24750] secs	Mapping Tables
QoE Reporting	Disabled	
Use Register as Keep Alive	Enable	SIP To Q.850 Override Table Default (RFC4497)
Forked Call Answered Too Soon	Disable 🗸	Default (REC4497)
		Q.850 To SIP Override Table
		Pass-thru Peer SIP Response Enable 🗸 🗸
		SIP IP Details
		Teams Local Media Optimization Disable
		Signaling/Media Source IP Ethernet 1 IP (10. 1.144)
		Signaling DSCP 40 * /0.631
		NAT TRAVARSA
		105 Ground Divided M
		ICE Support Disabled
		Static NAT - Outbound
		Outbound NAT Traversal None 🗸
		Static NAT - Inbound
		Detection Disabled V
	Listen Ports	Federated IP/FQDN
+		
Listen Port TCP-5060	Add/Edit	IP/FQDN Netmask/Prefix
	✓ Remove	/ I172.3 .53 255.255.255
Message Manipulation Disabled	<b>v</b>	

# 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.

noddin	Monitor Tasks Settings Diagnostics System
SBC Edge Real-Time Monitor           Short Legand   Repart Hendard           ** SBC 1000 Main Board           ** SBC 1000 Main Board           D SS Module 1           ** Oregoin_SG           1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 22 24           ** Oregoin_SG           1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 22 24 25 26 27 28 29 30           ** Oregoin_DEX           1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 22 24 25 26 27 28 29 20	O DS1 Port Details: Port 7:1 - Personal - Microsoft Edge
	Asserted Alarm None  Physical Data Layer  Port Type T1 Signaling Type (SDN v Dd1 Francing (ESS v Line Colling (SBSS v Trunk Type (Long Haul v Line Buildout 0 v eff

## 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			7			
Admin State	Enabled 🗸			_			
Service Status	Up						
	Chanı	nels and Routing			Port and Pro	otocol	
	Channel Hunting	Most Idle 🔹	•		Port Name	(T1) Port 7:1	*
	Direction	Bidirectional	·		Fractional	No	~
	Tone Table	Default Tone Table	<ul> <li>Ringback *</li> </ul>	•	Switch Variant	NI2	~
	Action Set Table	None	• +		ISDN Side	User	~
	Call Routing Table	PSTN_TO_ENTERPRISE_Voice	• •		Play Ringback	Auto on Alert	~
No Chan	nel Available Override	34: No Circuit/Channel Available	·		Service Msg Capability	Enabled	~
Play Inband Mes	ssage Post-Disconnect	No	•		Stop Far-End T310	Disabled	~
Call S	Setup Response Timer	255 [180750] secs			Indicated Channel	Exclusive	~
						arameters —	
					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	~
	Timeo	ut/Timer Settings					
_		Ū.					
T301 1	80 [1600] secs						
тзо2 1	5 [1255] secs						
тзоз 4	[1255] secs						
тзо5 З	0 [1255] secs						
т308 4	[1255] secs						
Т309 б	[1255] secs						
т310 3	10 [1255] secs						
т313 4	[1255] secs						
T314 4	[1255] secs						
T316 1	20 /12551 secs						
T322 4	[1255] secs						
T3M1/T323 1	20 [1.255] sec						
	[[mz35]3663						

# 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.

82		
nooon	O Monitor	Tasks Settings Diagnostics System
	Node-Level Settings	
Q Search		
Expand All   Collapse All   Reload	Set Date/Time   Backup Conng   Kestore Conng   Clear Dits Cache	
Call Routing	Host Information	Domain Name Service
🕨 📁 Signaling Groups		
💋 Linked Signaling Groups	Host Name shc7 *	Use Primary DNS Ves 🗸
Node Interfaces		
Application Solution Module	Domain Name nlabs.com	Primary Server IP 8.8.8.8 * XXXX 07 XXXXX
Node Level Cattings	System Information	Primary Source Ethernet 2 IP (192. 158) V
OoF	System Description	Use Secondary DNS No 🗸
DSPs	System Location	Enable DNS Service Ves
System Timing		
System Companding Law	System Contact	
Licensing		
🕨 🥖 Software Management	Time Management	Ribbon Application Management Platform (RAMP)
Auth and Directory Services		
Protocols	Time Zone (GMT-6:00) Central (US/Canada)	Connect to RAMP No
🕨 💋 SIP	Network Time Protocol	
CAS		
Security	Use hirr ites	
k d Tana Tablas	NTP Server 10. 0.5 * IPv4/6 Address or FQDN	
International American Tables	NTP Server Authentication Disabled V	
SNMP/Alarms	NTP Server 2	
Logging Configuration	Use NTP Server 2 No Y	
Emergency Services		
📁 Notification Manager		
	System LEDs	DHCP Server
	Deven LED Course	Fachie Durch Carrier Na
	Alarea LED Ambar	
	Alarm LED Amber	
	Ready LED Green	
	Locator LED On Green	
	Country Level Information	
	Country Code None	
		Apply

#### **TLS Profile**

TLS Profiles are used by SIP Signaling Groupswhen the TLS transport type is selected for incoming and outgoing SIP trunks (Listen Ports), and inSIP Server Tableswhen TLS is selected as the Server Host protocol.

From the Settingstab, navigate to Security > TLS Profiles. Click the + iconto 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.

P						vveico	ome: web
ribbon		Monitor	Tasks	Settings	Diagnostics	System	
		_		sounge	Diagnootice		
Q Search	TLS Profile						
Expand All   Collapse All   Reload	🕂   🗙 Total 2 TLS P	rofile Rows					
Call Routing	Description					Primary	y
j     j     signaling Groups						Key	
Linked Signaling Groups	Derault TES Prome					1	
Mode Interfaces	V 📄 🗌 Google_TLS					2	
🕨 🍺 System							
Auth and Directory Services		-					
Protocols		Commo	n Attributes —				
🕨 🃁 SIP	TLS Protocol	TLS 1.0-1.2	•				
🕨 📁 CAS	Mutual Authentication	Enabled	-				
V Security							
🕨 🥟 Users	Handshake Inactivity Timeout	30	secs [130]				
Login Messages	Certificate	SBC Edge Certif	icate	~			
SBC Certificates		Client					
Generate SBC Edge CSR		Client	Attributes				
SBC Primary Certificate				·			
SBC Supplementary Certificates		THE SERVIC DO		20.00000	Up		
Trusted CA Certificates		TLS_ECDHE_RSA	A_WITH_AES_128_C	BC_SHA250	Down		
TLS Profiles	Client Cipher List	TLS ECDHE RS/	WITH AES 256 C	BC SHA384	Add/Edit		
Change Password		TLS_ECDHE_RS/	WITH_3DES_EDE_	CBC_SHA	Remove		
Ribbon Protect Bad Actors		TLS_RSA_WITH	AES_128_CBC_SHA	256 👻			
🕨 📁 Media	Velidete Genera Sopri	Dischlad					
🕨 🃁 Tone Tables	validate Server FQDN	Disabled					
Telephony Mapping Tables		Serve	r Attribute ——				
SNMP/Alarms	Validate client SODM	Disabled					
Logging Configuration	validate Client FQDN	Disabled		•			
Emergency Services							
💋 Notification Manager							

#### **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 Settingstab, navigate to Media > SDES-SRTP Profiles. Click the + iconto 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", acrypto 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. ClickOK.

800						Welcome: we
ribbon		Monitor	Tasks	Settings	Diagnostics	System
Call Routing  Call Routing  Call Routing  Call Routing Table  Call Actions  Signaling Groups  Linked Signaling Groups  Linked Signaling Groups  Auth and Directory Services  Signaling Construction  Media Protecols  Signaling Construction  Media Protecols  Media System Configuration  Media Profiles  Media Profiles  Media List  Tone Tables	SDES-SRTP Profiles Total 1 S Description Coperation Option ( Crypto Suite Master Key Lifetime ( Derivation Rate ( Key Identifier Length	Monitor SDES-SRTP Profile Rov SRTP GV_BYOT Supported AES_CM_128_HMAC_SH Master Never Expires V 0 V	Tasks Crypto Suite AES_CM_128_HP Config	Settings	Diagnostics	System
Filephony Mapping Tables     SNIMP/Alarme						

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.

88						Welcome:
		Monitor	Tala	E ut	Dia series di se	Constant of
		Monton	lasks	Settings	Diagnostics	System
Q Search	Media List View					
Expand All   Collapse All   Reload	🕂 i 🗙 Total	3 Media List Rows				_
Call Routing	Description				P	rimary Key
Signaling Groups	🕨 📄 🗌 Default Media	List			1	
Linked Signaling Groups	▼ □ □ Google				2	
System						
Auth and Directory Services		6			_	
Protocols	Description	Google				
SIP		Default G711u				
Security		G22	Dow			
🔻 🧀 Media	Media Profiles List		Add/E	ait *		
Media System Configuration			Remo	ve		
SDES-SRTP Profiles			-			
DTLS-SRTP Profiles	SDES-SRTP Profile	GV_BYOT	✓ Associate	d SIP SG Listen Ports shou	ld be TLS only. 🕂	
Vedia List	DTLS-SRTP Profile	None	¥ 🕇			
Google	Media DSCP	46	* [063]			
On-prem PBX	RTCP Mode	RTCP	~			
🕨 🃁 Tone Tables	Dead Call Detection	Enabled	~			
Telephony Mapping Tables	Silence Suppression	Enabled	~			
SNMP/Alarms	On-prem PBX				4	
<ul> <li>Energency Services</li> </ul>						
Notification Manager						
Q Search	Media List View					
Expand All   Collapse All   Reload	🕂   🗙 Total 3	Media List Rows				
Call Bouting	Description				Prir	nary Key
Signaling Groups	🕨 💼 🗌 Default Media I	ist			1	
Linked Signaling Groups					2	
<ul> <li>Node Interfaces</li> <li>System</li> </ul>						
Auth and Directory Services	Dead Call Detection	Enabled	~			
Protocols	Silence Suppression	Enabled	~			
SIP CAS						
Security	Gain Co	ontrol		Digit Relay		
V Media	Receive Gain 0	(-14 +61 dB	Digit (DTME) B	elay Type REC 2833	~	
Media System Configuration     dedia Profiles	Transmit Cain 0		Digit Relay Pay	load Type 101	195 1271	
b d SDES-SRTP Profiles		[-14+0] UD	bigit Keldy Fdy		[200727]	
DTLS-SRTP Profiles		Papathroug	h/Tone Detection			
Vedia List		Fassuroug	In Tone Detection		_	
Google	Modem Passthrough	Enabled 🗸				
On-prem PBX	Fax Passthrough	Enabled 🗸				
Tone Tables	CNG Tone Detection	Disabled 🗸				
Telephony Mapping Tables	Env Tone Detection	Easthlad				
Shimir'/Alditits	On-prem PBX				4	
Logging Configuration						
<ul> <li>Configuration</li> <li>Emergency Services</li> </ul>						

#### **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 Settingstab, navigate to SIP > Message Manipulation > Message Rule Table. Click the tion 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. ClickOK.

Ô						Welcome:
noddin		Monitor	Tasks	Settings	Diagnostics	System
Q Search	SIP Message Rule Tabl	le Total 1 SI	P Message Manip	ulation Table Row		
Call Routing	Description		Result Type		Message Type	
Linked Signaling Groups	▼ □ Google_Rule		Optional			
<ul> <li>Mode Interfaces</li> <li>System</li> </ul>	Description	Google_Rule				
Auth and Directory Services	Applicable Messages	Selected Messages	~			
Forces     Forces	Message Selection	invite Cancel Options ACK	Ac	sd/Edit *		
Trunk Groups NAT Qualified Prefix Tables	Table Result Type	Optional	~			
Remote Authorization Tables						
Contact Registrant Table						
Message Rule Tables						
Condition Rule Table						

#### Message Rule Table Entry

Header Rule:

- 1. Click on the Message Rule Table GOOGLE\_RULE.
- 2. From theCreate Ruledrop-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**.

R S							Welcome: we
riddon			Monitor	Tasks	Settings	Diagnostics	System
Q Search	Google_	Rule					
Expand All   Collapse All   Reload	<b>√   ⊘  </b>	Create Rule 🔻 🛛 🗙 🚺	<sup>1</sup> <sub>2</sub>   Test Messag	e Tot	al 2 Message Manipula	tion Rules Rows	
Call Routing		Header Rule	Гуре		Result Type	Des	cription
Signaling Groups	<b>v</b>	Request Line Rule	er Rule		Optional	Goo	gle_header
Linked Signaling Groups     Mode Interfaces	Test Rule	Status Line Rule					
🕨 📁 System		Raw Message Rule					
SIP     Local Registrars	Condit	Description Google_h	eader				
Local / Pass-thru Auth Tables     SiP Profiles     SiP Server Tables     Trunk Groups     Trunk Groups		Admin State Enabled Result Type Optional Header Action Add Header Name X-Google	► ► Pbx-Trunk-Secre	*			
Keri documento ricin fubrico Contact Registrant Table     Message Manipulation     Message Rule Tables     Google Rule	Heade	r Value Add	✓ Add/Edit	394	-		j
Condition Rule Table     Node-Level SIP Settings     SIP Voice Quality Server     CAS							

Q Search	Google_Rule				February 14, 2023 14:24:15 🗘 📀 🕈
Expand All   Collapse All   Reload					
🕨 🃁 Call Routing	Admin State	Rule Type	Result Type	Description	Primary Key
Signaling Groups	- CO V	Header Rule	Optional	Google_header	1
Linked Signaling Groups		Edit Message Field			
System					
Auth and Directory Services					
Protocols	Description	Type of Value Literal			
▼ 🖾 SIP	Condition Expression	Value 39	*		
Dcal Registrars	Admin State				
Local / Pass-thru Auth Tables	Admin State				
F SIP Profiles	Result Type		OK Cancel		
SIP Server Tables	Header Action	2			
Trunk Groups	Header Name	2			
NAT Qualified Pretix Tables					
Context Devictor Tables					
Contact Registrant Table	Header Value Add	✓ Add/Edit 39480655			
Message Manipulation     Message Rule Tables					
Google Rule					
k d Condition Dula Table					
Condition Rule Table					
Node-Level SIP Settings					
SIP Voice Quality Server					

Request Line Rule:

- 1. Click on the Message RuleTable **GOOGLE\_RULE**.
- 2. From theCreate Ruledrop-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.

						Welcome:
						Welcome.
noddin		Monitor	Tasks	Settings	Diagnostics	System
Q Search	Google_Rule					
Expand All   Collapse All   Reload	Create Rule 🔻	🗙   🦯   Test Messag	e Tot	al 2 Message Manipulat	ion Rules Rows	
🕨 🥖 Call Routing	Header Rule	/pe		Result Type	Descr	iption
Signaling Groups	Request Line R	ule Rule		Optional	Googl	e_header
Linked Signaling Groups	- C Status Line Rul	e st Line Pule		Ontional	Pequ	est uri FODN
Mode Interfaces				optional	nequ	est_un_r gon
System Auth and Directory Services	Test Rule Raw Message R	tule	_		_	_
Protocols						
V 🖉 SIP						
💋 Local Registrars	Description	Request_uri_FQDN				
📁 Local / Pass-thru Auth Tables	Condition Expression	Add/Edit				
🕨 💋 SIP Profiles	Admin State	Enabled 🗸				
SIP Server Tables	Result Type	Optional 🗸				
Trunk Groups						
NAT Qualified Prefix Tables						
Contact Registrant Table	Request Line Value	Modify 🖌 🖌 Ad	d/Edit Match: sip	link.telephony.goog	Replace: trunk.sip.voice.go	ogle.c
V Message Rule Tables						
Google_Rule						
🕨 💋 Condition Rule Table						
Node-Level SIP Settings						
SIP Voice Quality Server						

0 noddin		Monitor	osks Settings	Welco Diagnostics System	me: webui   Last Login: First Login   Logout   Help Device Name: webui90 Ribbon SBC 2000
Q. Search Expand All   Collasse All   Reload	Google_Rule	I X I Z   Test Message Rule Type Header Rule Edit Message Field	Total 2 Message Manipulatia Result Type Optional	n Rules Rovs Description Google_header Request_uri_FQDN	February 14, 2023 14:24:15 🗘 🖗 🍳
System     System     System     Local Registrars     SiP SiP Server Tables     Trank Groups     NAT Causified Prefix Tables     Code Registrar Table     Message Rule Tables     Code Registrar Table     Message Rule Tables     SiP SiP Settings     SiP SiP Code Legistrari Table     Message Rule Tables     Societ Registrari Table     Message Rule Tables     Societ Registrari Table     Message Rule Tables     SiP SiP Code Quality Server     SiP SiP Code Quality Server     CaS     Security	Test Rule  Description  Condition Expression  Admin State  Result Type   Request Line Value	Type of Value Regex Match Regex siplinik.tr Replace Regex Trunk.sip	Protector approved deprine y globy (Note: globale com *	Cancel	Apply

#### SIP Profile - GV

From the Settingstab, navigate to SIP> SIP Profiles. Click the + iconto 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**.



Note
 The

The session will always be refreshed by Ribbon SBC Edge 1000 as per the Google Voice requirement.

#### SIP Server Table - GV

From the Settingstab, navigate to SIP > SIP Server Tables. Click the + iconto create a new SIP Server Table.

- 1. Provide a name for the SIP Server in the Description field.
- 2. Click OK.

P							Welco	me: webu
noddin			Monitor	Tasks	Settings	Diagnostics	System	
Q. Search Expand All   Collapse All   Reload		SIP Server Tables	r Table Rows			_	Prima	
Call Routing		Description					Key	
Linked Signaling Groups		Default SIP Server					1	
Vode Interfaces		🔻 📄 🗌 Google					2	
🕨 📁 System								
Auth and Directory Services								
Protocols		Description Google						
V SIP			_					
Local Registrars								
Local / Pass-thru Auth Tables					Appl	v		
SIP Fidnes								
Coogle		On-prem_PBX					4	
Cn-prem_PBX								
📁 Trunk Groups								
📁 NAT Qualified Prefix Tables								
prote Authorization Tables								
💋 Contact Registrant Table								
Vessage Manipulation								
Versage Rule Tables								
U Google_Rule								
Condition Rule Table								
Node-Level SIP Settings								
SIP Voice Quality 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 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 Settingstab, 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 appropriateIP 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 Google_SG	
Service Status Up	
SIP Channels and Routing	
Action Set Table None	Media Information
Call Routing Table GV_TO_PBX_&_PSTN V	DSP ^ Add/Edit
No. of Channels 30 * [1960]	Audio/Fax Modes Proxy Remove *
SIP Profile Google	Supported
SIP Mode Basic Call	Video/Application Disabled Modes
Agent Type Back-to-Back User Agent 🗸	Media List ID Google 💙 🕇
Interop Mode Standard 🗸	Play Ringback Auto on 180 🗸
SIP Server Table Google 💙 🕇	Tone Table Default Tone Table 🗸 🔶
Load Balancing Round Robin 🗸	Play Congestion Tone Disable
Notify Lync CAC Profile Disable	Early 183 Disable 🗸
Challenge Request Disable	Allow Refresh Enable
Outbound Proxy IP/FQDN	Music on Hold Disabled
Outbound Proxy Port 5060 [1.65535]	RTCP Multiplexing Disable
No Channel Available Override 34: No Circuit/Channel Available V	Media Codec Enable V
Call Setup Response Timer 255 [180750] secs	Latch
Call Proceeding Timer 180 [24750] secs	Mapping Tables
QoE Reporting Disabled	
Use Register as Keep Alive Enable	SIP To Q.850 Override Table
Forked Call Answered Too Soon Disable	O 850 To STB Ourserido Tablo Default (RFC4497) ▼
	Pass-thru Peer SIP Response Code Enable
	SIP IP Details
	Teams Local Media Optimization Disable
	Signaling/Media Source IP Ethernet 2 IP (192 158)
	Signaling DSCP 40 * (063)
	NAT Traversal
	ICE Support Disabled
	Static NAT - Outbound
	Outbound NAT Traversal None 🗸
	Static NAT - Inbound
	Detection Disabled
Listen Ports	Federated IP/FQDN
+	🛶   🗙 Total 1 SIP Federated IP Row
Listen Port TCP-5060	IP/FQDN Netmask/Prefix
TLS-5061	iplink.telephony.goog 255.255.255
Message Manipulation Enabled V	
labourd Manager Martin Latin	Outbound Massess Masteriation
incound message manipulation	Outbound Message Manipulation
Up	Google_Rule
Down *	Down *
Add/Edit	Add/Edit
	- Remove
	Apple

# **Google Voice Configuration**

For configuration on Google Voice, visitsupport.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	$\checkmark$
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	X
14	Simultaneous Ring	X
15	Non E164 format	X
16	Call Decline with 603 response	X

#### Legend



## 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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