

**Configuration Guide for  
Google Voice SIP Link Using  
Oracle E-SBC Acme Packet  
3900 SCZ8.4.0**



## Table of Contents

1	Audience	3
1.1	Introduction	3
1.1.1	tekVizion Labs	3
2	SIP Trunking Network Components	4
3	Hardware Components	5
4	Software Requirements	5
5	Features	6
5.1	Features Supported by Google Voice SIP Link	6
5.2	Features Not Supported by Google Voice SIP Link	6
5.3	Features Not Supported by Service Provider	6
5.4	Caveats and Limitations	6
6	Configuration	7
6.1	Configuration Checklist	7
6.2	IP Address Worksheet	8
6.3	Google Voice SIP Link Configuration	9
6.4	Oracle E-SBC Configuration	9
6.4.1	Media Manager	9
6.4.2	Physical Interface	10
6.4.3	Network Interface	13
6.4.4	Codec Policy	16
6.4.5	Translation Rules	17
6.4.6	Session Translation	18
6.4.7	Realm Config	20
6.4.8	Steering Pool	32
6.4.9	SDES Profile	34
6.4.10	Media Sec Policy	35
6.4.11	TLS – Certificate Record	37
6.4.12	TLS – TLS Profile	40
6.4.13	Session Timer	41
6.4.14	SIP Interface	42
6.4.15	Session Agent	53
6.4.16	Local Policy	67
6.4.17	SIP Manipulation	71
6.4.18	Redundancy Configuration	81
6.4.19	Oracle SBC deployed behind NAT	86
7	Oracle E-SBC Running configuration	88
8	Summary of Tests and Results	89

# 1 Audience

---

This document is intended for the SIP Trunk customer's technical staff and Value Added Reseller (VAR) having installation and operational responsibilities.

## 1.1 Introduction

This Configuration Guide describes configuration steps for **Google Voice SIP Link** using **Oracle Enterprise Session Border Controller Acme Packet 3900 SCZ8.4.0**.

### 1.1.1 tekVizion Labs

tekVizion Labs™ is an independent testing and verification facility offered by tekVizion, Inc. tekVizion Labs offers several types of testing services including:

- Remote Testing – provides secure, remote access to certain products in tekVizion Labs for pre-Verification and ad hoc testing.
- Verification Testing – Verification of interoperability performed on-site at tekVizion Labs between two products or in a multi-vendor configuration.
- Product Assessment – independent assessment and verification of product functionality, interface usability, assessment of differentiating features as well as suggestions for added functionality, stress and performance testing, etc.

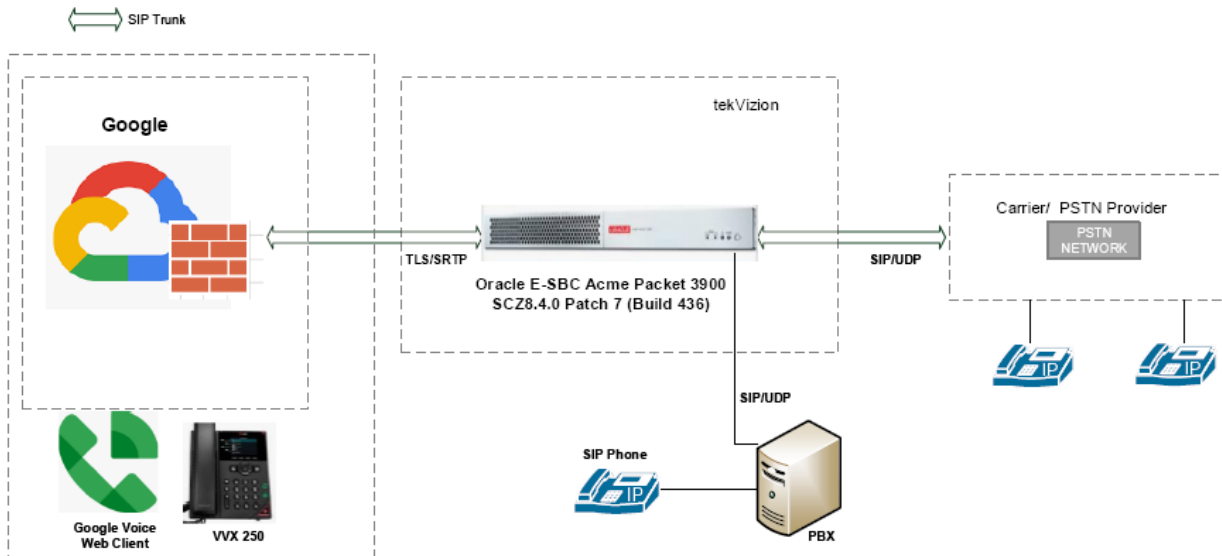
tekVizion is a systems integrator specifically dedicated to the telecommunications industry. Our core services include consulting/solution design, interoperability/Verification testing, integration, custom software development and solution support services. Our services help service providers achieve a smooth transition to packet-voice networks, speeding delivery of integrated services. While we have expertise covering a wide range of technologies, we have extensive experience surrounding our practice areas which include: SIP Trunking, Packet Voice, Service Delivery, and Integrated Services.

The tekVizion team brings together experience from the leading service providers and vendors in telecom. Our unique expertise includes legacy switching services and platforms, and unparalleled product knowledge, interoperability and integration experience on a vast array of VoIP and other next-generation products. We rely on this combined experience to do what we do best: help our clients advance the rollout of services that excite customers and result in new revenues for the bottom line. tekVizion leverages this real-world, multi-vendor integration and test experience and proven processes to offer services to vendors, network operators, enhanced service providers, large enterprises and other professional services firms. tekVizion's headquarters, along with a state-of-the-art test lab and Executive Briefing Center, is located in Plano, Texas.

*For more information on tekVizion and its practice areas, please visit [tekVizion Labs website](#).*

## 2 SIP Trunking Network Components

The network for the SIP Trunk reference configuration is illustrated below and is representative of Google Voice SIP Link with Oracle Enterprise Session Border Controller (E-SBC) Acme Packet 3900 SCZ8.4.0 configuration.



**Figure 1: SIP Trunk Lab Reference Network**

The lab network consists of the following components

- Google Voice SIP Link and Workspace subscriptions
- Oracle E-SBC Acme Packet 3900
- Poly VVX 250 OBI Edition Phone
- onPrem PBX (Asterisk PBX)

### 3 Hardware Components

---

- Oracle E-SBC Acme Packet 3900

### 4 Software Requirements

---

- Oracle E-SBC Acme Packet 3900 SCZ8.4.0 Patch 7 (Build 436)
- Poly VVX 250 OBI Edition V6.4.3.10072
- OnPrem PBX (Asterisk PBX) V13.23.1

## 5 Features

---

### 5.1 Features Supported by Google Voice SIP Link

- Basic calls
- Call Hold and Resume
- Call Transfer
- DTMF RFC 2833
- Calling Party Number Presentation
- Calling Party Number Restricted
- Ring Group
- Auto Attendant
- Voicemail

### 5.2 Features Not Supported by Google Voice SIP Link

- Linked Phone Numbers
- Call Forward
- Short Code calls (e.g. 411)
- Non E164 format

### 5.3 Features Not Supported by Service Provider

- STIR-Shaken

### 5.4 Caveats and Limitations

Call disconnects before answer	When Google Voice (GV) user hangs up the incoming call from PSTN, CANCEL message is not sent from GV user. PSTN user is forwarded to voicemail.
Call Waiting	When a second call is made from PSTN B to GV User A, PSTN B hears ringback and web client does not see call waiting indication. The behavior is same for iOS and Android phones.  Call waiting is supported by a multiline Desk phones like VVX150, 250, 350 & 450.
Handling 486 response	This is same as Call waiting.
Session Refresh	Google Voice supports only UPDATE as a session refresh mechanism

## 6 Configuration

---

### 6.1 Configuration Checklist

Below are the steps that are required to configure Oracle E-SBC.

**Table 1 – Oracle E-SBC Configuration Steps**

Step	Description	Reference
Step 1	Media Manager	<a href="#">Section 6.4.1</a>
Step 2	Physical Interface	<a href="#">Section 6.4.2</a>
Step 3	Network Interface	<a href="#">Section 6.4.3</a>
Step 4	Codec Policy	<a href="#">Section 6.4.4</a>
Step 5	Translation Rules	<a href="#">Section 6.4.5</a>
Step 6	Session Translation	<a href="#">Section 6.4.6</a>
Step 7	Realm Config	<a href="#">Section 6.4.7</a>
Step 8	Steering Pool	<a href="#">Section 6.4.8</a>
Step 9	SDES Profile	<a href="#">Section 6.4.9</a>
Step 10	Media Sec Policy	<a href="#">Section 6.4.10</a>
Step 11	TLS – Certificate Record	<a href="#">Section 6.4.11</a>
Step 12	TLS – TLS Profile	<a href="#">Section 6.4.12</a>
Step 13	Session Timer	<a href="#">Section 6.4.13</a>
Step 14	SIP Interface	<a href="#">Section 6.4.14</a>
Step 15	Session Agent	<a href="#">Section 6.4.15</a>
Step 16	Local Policy	<a href="#">Section 6.4.16</a>
Step 17	SIP Manipulation	<a href="#">Section 6.4.17</a>
Step 18	Redundancy Configuration	<a href="#">Section 6.4.18</a>
Step 19	Oracle SBC deployed Behind NAT	<a href="#">Section 6.4.19</a>

## 6.2 IP Address Worksheet

The specific values listed in the table below and in subsequent sections are used in the lab configuration described in this document, and are for **illustrative purposes only**.

**Table 3 - IP Address Worksheet**

Component	IP Address
<b>Google Voice SIP Link</b>	
Signaling	FQDN: Siplink.telephony.goog IP: 216.239.X.X
Media	74.125.X.X
<b>OnPrem PBX</b>	
LAN IP Address	172.16.29.53
<b>Oracle E-SBC</b>	
LAN IP Address	10.80.11.21
WAN IP Address	192.65.X.X



## 6.3 Google Voice SIP Link Configuration

Below link can be referred to configure Google Voice SIP Link.

[support.google.com/a?p=siplink](https://support.google.com/a?p=siplink)

## 6.4 Oracle E-SBC Configuration

The following is the example configuration of Oracle E-SBC for Google Voice SIP Link.

### 6.4.1 Media Manager

Media-Manager handles the media stack required for SIP sessions on the E-SBC. Media Manager is configured as shown below.

Navigate to **Configuration > media-manager > media-manager**.

The screenshot shows the Oracle E-SBC configuration interface. On the left, a navigation menu is visible with 'media-manager' selected. The main panel is titled 'Modify Media Manager' and contains the following configuration options:

Parameter	Value	Range
State	<input checked="" type="checkbox"/> enable	
Flow Time Limit	86400	( Range: 0..4294967295 )
Initial Guard Timer	300	( Range: 0..4294967295 )
Subsq Guard Timer	300	( Range: 0..4294967295 )
TCP Flow Time Limit	86400	( Range: 0..4294967295 )
TCP Initial Guard Timer	300	( Range: 0..4294967295 )
TCP Subsq Guard Timer	300	( Range: 0..4294967295 )
Hint Rtcp	<input type="checkbox"/> enable	
Algd Log Level	NOTICE	
Mbcd Log Level	NOTICE	

**Figure 2: Media Manager**

Configuration View Configuration Q

- media-manager
  - codec-policy
  - media-manager**
  - media-policy
  - realm-config
  - steering-pool
- security ▶
- session-router ▶
- system ▶
- media-manager
  - codec-policy
  - media-manager**
  - media-policy
  - realm-config
  - steering-pool
- security ▶
- session-router ▶
- system ▶

### Modify Media Manager

Options audio-allow-asymmetric-pt ✕

Red Max Trans	<input type="text" value="10000"/>	( Range: 0..50000 )
Red Sync Start Time	<input type="text" value="5000"/>	( Range: 0..4294967295 )
Red Sync Comp Time	<input type="text" value="1000"/>	( Range: 0..4294967295 )
Media Policing	<input checked="" type="checkbox"/> enable	
Max Arp Rate	<input type="text" value="10"/>	( Range: 0..100 )
Max Signaling Packets	<input type="text" value="0"/>	( Range: 0..4294967295 )
Max Untrusted Signaling	<input type="text" value="1"/>	( Range: 0..100 )
Min Untrusted Signaling	<input type="text" value="1"/>	( Range: 0..100 )

### Modify Media Manager

Tolerance Window	<input type="text" value="30"/>	( Range: 0..4294967295 )
Untrusted Drop Threshold	<input type="text" value="0"/>	( Range: 0..100 )
Trusted Drop Threshold	<input type="text" value="0"/>	( Range: 0..100 )
Acl Monitor Window	<input type="text" value="30"/>	( Range: 5..3600 )
Trap On Demote To Deny	<input type="checkbox"/> enable	
Trap On Demote To Untrusted	<input type="checkbox"/> enable	
Syslog On Demote To Deny	<input type="checkbox"/> enable	
Syslog On Demote To Untrusted	<input type="checkbox"/> enable	
Anonymous Sdp	<input type="checkbox"/> enable	
Reactive Transcoding	<input type="checkbox"/> enable	
Translate Non Rfc2833 Event	<input type="checkbox"/> enable	
Xcode Fax Max Rate	<input type="text" value="14400"/>	

**Figure 3: Media Manager Cont.**

## 6.4.2 Physical Interface

Navigate to **Configuration > system > phy-interface**.

Configure Physical interface towards Google Voice, OnPrem PBX and PSTN Gateway as shown below.

The interface designated towards Google Voice is named as s0p0 (Slot 0, port 0).

Configuration View Configuration Q

- sip-manipulation
- sip-monitoring
- translation-rules
- system**
- fraud-protection
- host-route
- http-client
- http-server
- network-interface
- ntp-config
- phy-interface**
- redundancy-config

### Modify Phy Interface

Name	s0p0
Operation Type	Media
Port	0 (Range: 0..5)
Slot	0 (Range: 0..2)
Virtual Mac	
Admin State	<input checked="" type="checkbox"/> enable
Auto Negotiation	<input checked="" type="checkbox"/> enable
Duplex Mode	FULL
Speed	100
Wancom Health Score	50 (Range: 0..100)

Figure 4: Physical Interface towards Google Voice

The interface designated towards PSTN Gateway and Onprem PBX are named as s1p0 (Slot 1, port 0).

**Configuration** View Configuration Q

- sip-manipulation
- sip-monitoring
- translation-rules
- system ▼
- fraud-protection
- host-route
- http-client
- http-server
- network-interface
- phy-interface**
- redundancy-config

### Modify Phy Interface

Name	s1p0
Operation Type	Media ▼
Port	0 ( Range: 0..5 )
Slot	1 ( Range: 0..2 )
Virtual Mac	
Admin State	<input checked="" type="checkbox"/> enable
Auto Negotiation	<input checked="" type="checkbox"/> enable
Duplex Mode	FULL ▼
Speed	100 ▼
Wancom Health Score	50 ( Range: 0..100 )

**Figure 5: Physical Interface towards PSTN Gateway and OnPrem PBX**

### 6.4.3 Network Interface

Navigate to **Configuration > system > network-interface**.

Configure Network interface towards Google Voice, OnPrem PBX and PSTN Gateway as shown below.

The screenshot displays the 'Modify Network Interface' configuration page. On the left, a navigation sidebar lists various system components, with 'system' and 'network-interface' highlighted. The main configuration area contains the following fields:

Field	Value
Name	s0p0
Sub Port Id	0 (Range: 0..4095)
Description	
Hostname	sbc3.tekvizionlabs.com
IP Address	192.65.
Pri Utility Addr	
Sec Utility Addr	
Netmask	255.255.255.128
Gateway	192.65.

**Figure 6: Network Interface towards Google Voice**

Note: If SBC is placed behind the NAT, please refer the NAT configuration under [Section 6.4.19](#)

Configuration View Configuration

- sip-manipulation
- sip-monitoring
- translation-rules
- system
- fraud-protection
- host-route
- http-client
- http-server
- network-interface**
- ntp-config
- phy-interface
- redundancy-config
- snmp-community
- spl-config
- media-manager
- security
- session-router
- system
- fraud-protection
- host-route
- http-client
- http-server
- network-interface**
- ntp-config
- phy-interface

### Modify Network Interface


**▲ Gw Heartbeat**

State	<input checked="" type="checkbox"/> enable	
Heartbeat	<input type="text" value="10"/>	( Range: 0..65535 )
Retry Count	<input type="text" value="3"/>	( Range: 0..65535 )
Retry Timeout	<input type="text" value="3"/>	( Range: 1..65535 )
Health Score	<input type="text" value="0"/>	( Range: 0..100 )
DNS IP Primary	<input type="text" value="8.8.8.8"/>	
DNS IP Backup1	<input type="text"/>	
DNS IP Backup2	<input type="text"/>	
DNS Domain	<input type="text" value="tekvizionlabs.com"/>	
DNS Timeout	<input type="text" value="11"/>	( Range: 0..4294967295 )
DNS Max Ttl	<input type="text" value="86400"/>	( Range: 30..2073600 )
Signaling Mtu	<input type="text" value="0"/>	( Range: 0,576..4096 )

### Modify Network Interface

HIP IP List	<input type="text"/>
ICMP Address	<input type="text"/>
SSH Address	<input type="text"/>

Tunnel Config



No tunnel config to display. Please add or upload tunnel config.

**Figure 7: Network Interface towards Google Voice Cont.**

Configure Network interface towards Google Voice, OnPrem PBX and PSTN Gateway as shown below

Configuration View Configuration

- sip-manipulation
- sip-monitoring
- translation-rules
- system
  - fraud-protection
  - host-route
  - http-client
  - http-server
  - network-interface**
  - ntp-config
  - phy-interface
  - redundancy-config
  - snmp-community
- media-manager
- security
- session-router
- system
  - fraud-protection
  - host-route
  - http-client
  - http-server
  - network-interface**
  - ntp-config
  - phy-interface
- media-manager
- security
- session-router
- system
  - fraud-protection

### Modify Network Interface

Name	slp0	
Sub Port Id	0	( Range: 0..4095 )
Description	<input type="text"/>	
Hostname	<input type="text"/>	
IP Address	10.80.11.21	
Pri Utility Addr	<input type="text"/>	
Sec Utility Addr	<input type="text"/>	
Netmask	255.255.255.0	
Gateway	10.80.11.1	

### Modify Network Interface

▲ Gw Heartbeat

State  enable

Heartbeat	10	( Range: 0..65535 )
Retry Count	3	( Range: 0..65535 )
Retry Timeout	3	( Range: 1..65535 )
Health Score	0	( Range: 0..100 )

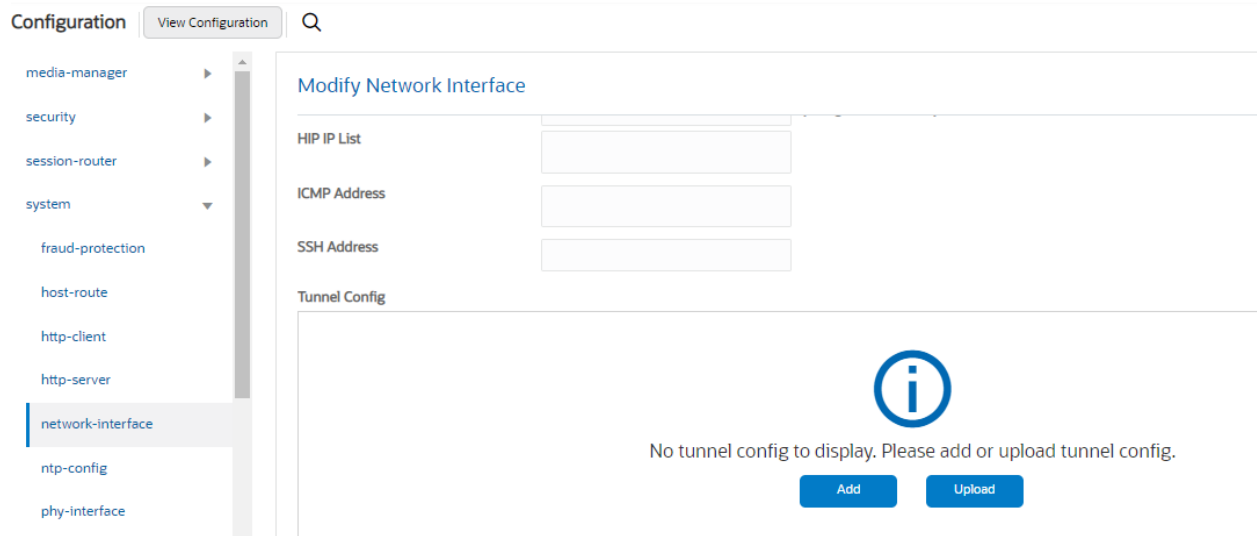
▶ Bfd Config

DNS IP Primary	<input type="text"/>
DNS IP Backup1	<input type="text"/>
DNS IP Backup2	<input type="text"/>

### Modify Network Interface

DNS Domain	<input type="text"/>	
DNS Timeout	11	( Range: 0..4294967295 )
DNS Max Ttl	86400	( Range: 30..2073600 )
Signaling Mtu	0	( Range: 0,576..4096 )

**Figure 8: Network Interface towards PSTN Gateway**

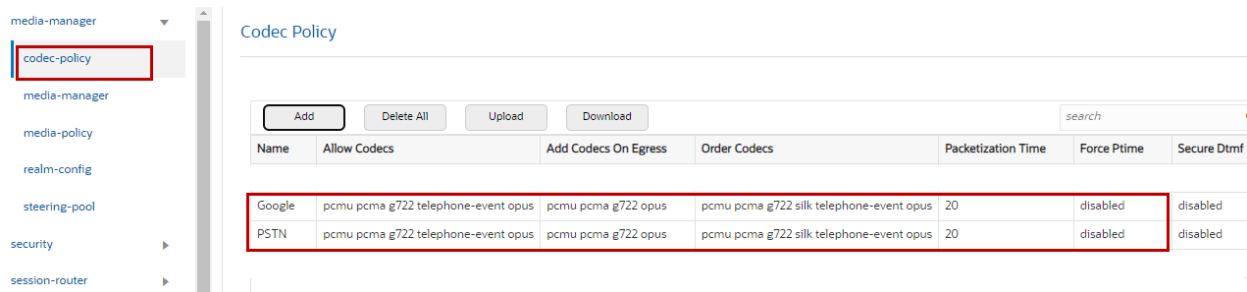


**Figure 9: Network Interface towards PSTN Gateway Cont.**

Note: ICMP IP and HIP IP addresses needs to disabled in production environment.

#### 6.4.4 Codec Policy

Navigate to **Configuration > media-manager > codec-policy** and configure codec policy for PSTN and Google Voice as shown below.



**Figure 10: Codec Policy for Google Voice and PSTN Gateway**



### 6.4.5 Translation Rules

Navigate to **Configuration > session-router > translation-rules** and **configure translation rules** for PSTN Gateway and Google Voice as shown below.

Translation rule is created to send E.164 number format towards Google Voice.

Note: Google Voice supports only E.164 number format and hence this translation rule is created. The example shown here is valid for US only. For other countries, the appropriate E164 format translation rule needs to be created.

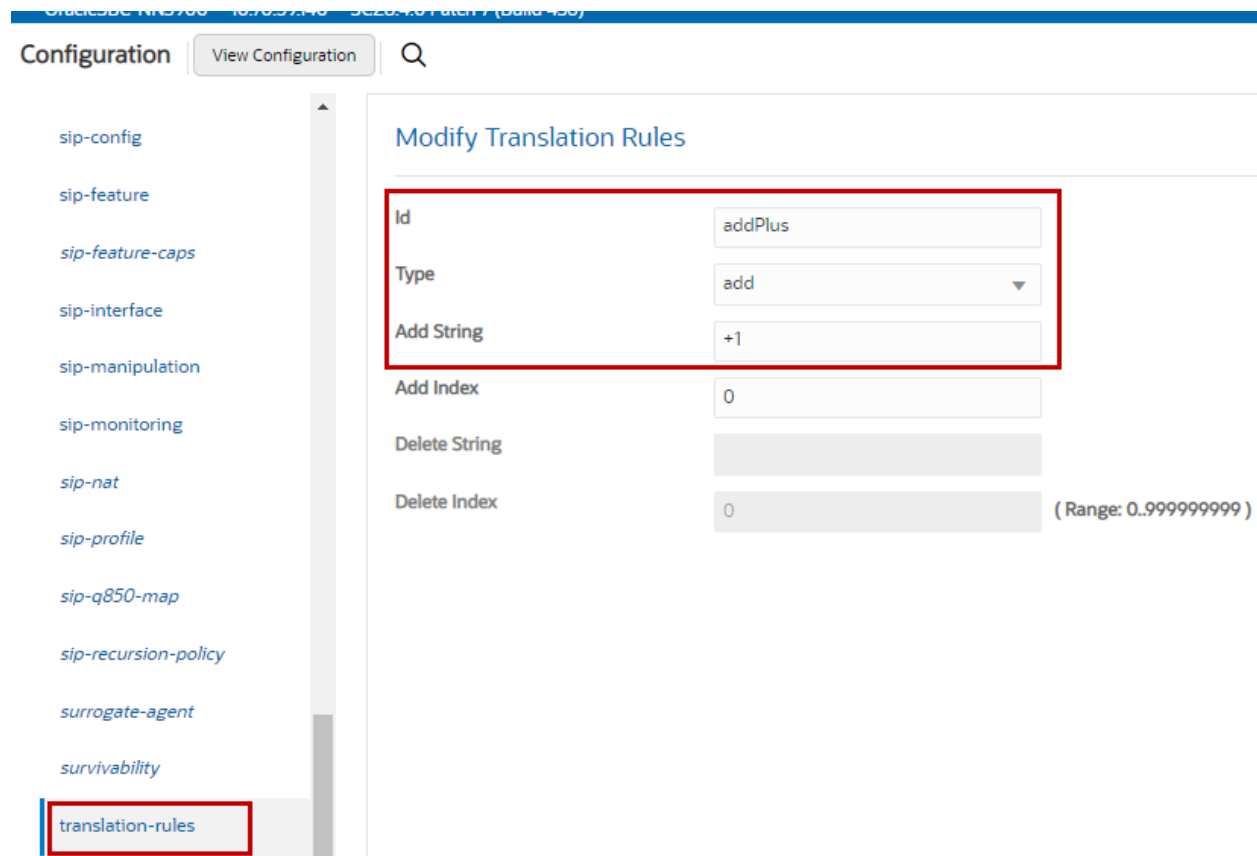


Figure 11: Translation Rule to add send E.164 towards Google Voice

Translation rule is created to send non-E.164 number format towards PSTN Gateway

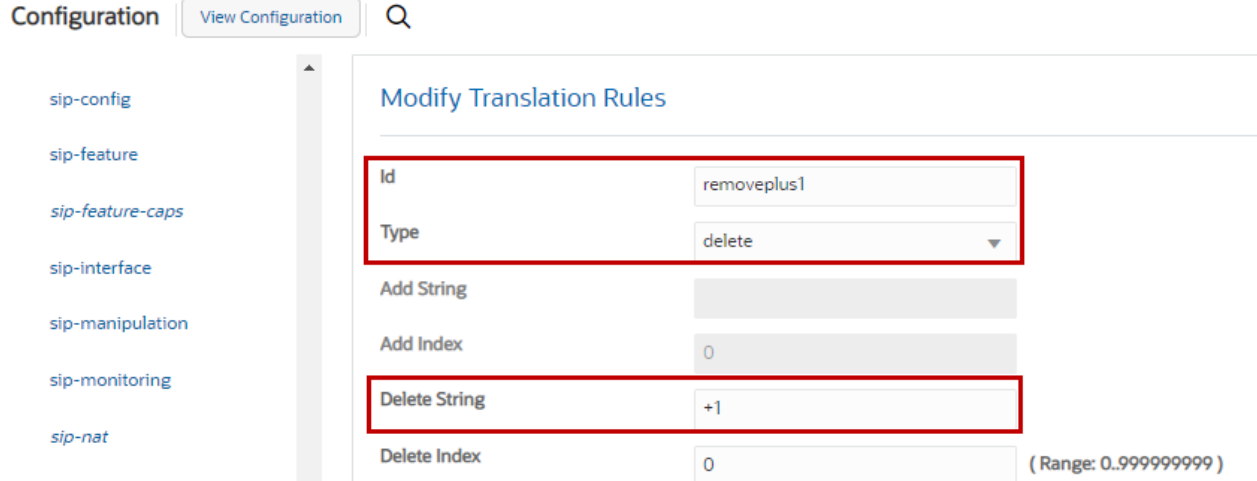


Figure 12: Translation to send non E.164 towards PSTN Gateway

### 6.4.6 Session Translation

Navigate to **Configuration > session-router > session-translation**. The translation rules configured in Section 6.4.5 is mapped for PSTN Gateway and Google Voice is shown below.

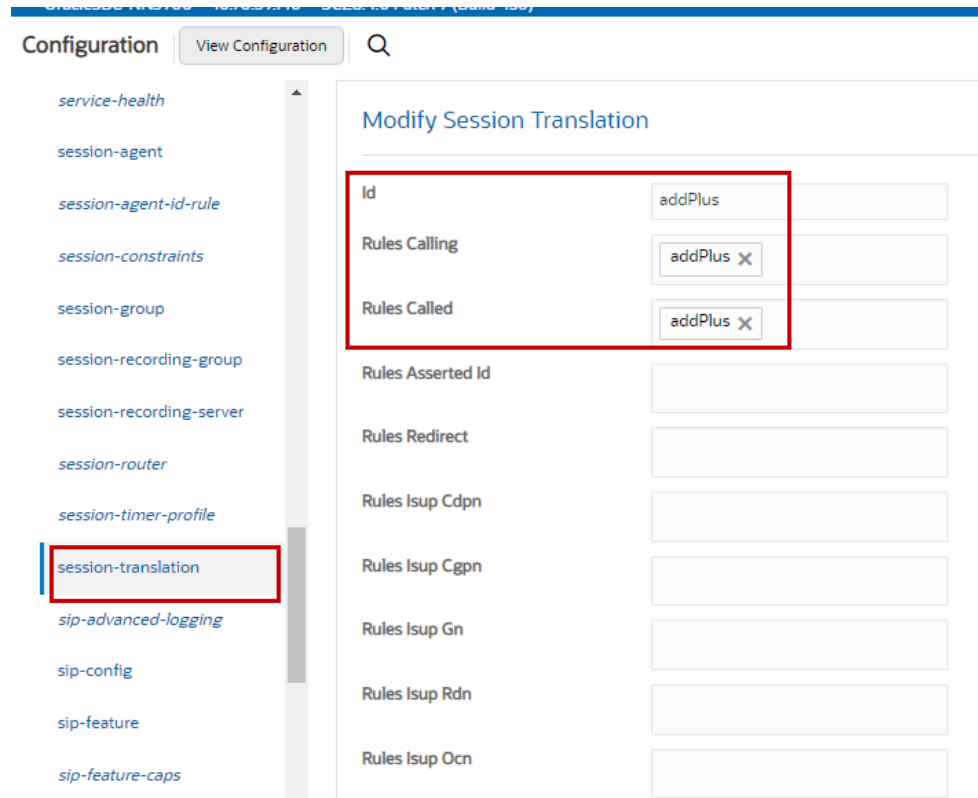


Figure 13: Session Translation towards Google Voice

- session-agent
- session-agent-id-rule
- session-constraints
- session-group
- session-recording-group
- session-recording-server
- session-router
- session-timer-profile
- session-translation**
- sip-advanced-logging
- sip-config
- sip-feature
- sip-feature-caps
- sip-interface

### Modify Session Translation

Id	removeE164
Rules Calling	removeplus1 ✕
Rules Called	removeplus1 ✕ international ✕
Rules Asserted Id	removeplus1 ✕
Rules Redirect	
Rules Isup Cdpn	
Rules Isup Cgpn	
Rules Isup Gn	
Rules Isup Rdn	
Rules Isup Ocn	

Figure 14: Session Translation towards PSTN Gateway

## 6.4.7 Realm Config

Navigate to **Configuration > media-manager > realm-config**.

Realm Config towards Google Voice, OnPrem PBX and PSTN Gateway are shown below.

The screenshot displays the 'Modify Realm Config' interface. On the left, a navigation menu is visible with the following items: media-manager, codec-policy, media-manager, media-policy, realm-config, steering-pool, security, session-router, and system. The 'media-manager' and 'realm-config' items are highlighted with red boxes. The main content area is titled 'Modify Realm Config' and contains the following configuration fields:

- Identifier:** Google
- Description:** (Empty text area)
- Addr Prefix:** 0.0.0.0
- Network Interfaces:** s0p0:0.4
- Media Realm List:** (Empty text area)
- Mm In Realm:**  enable
- Mm In Network:**  enable
- Mm Same Ip:**  enable

**Figure 15: Realm Config towards Google Voice**

The image displays two screenshots of a configuration interface for 'Modify Realm Config'. The left sidebar shows a navigation menu with 'realm-config' selected. The top screenshot shows the following configuration items:

- QoS Enable:  enable
- Max Bandwidth: 0 (Range: 0..999999999)
- Max Priority Bandwidth: 0 (Range: 0..999999999)
- Parent Realm: [Dropdown]
- DNS Realm: [Dropdown]
- Media Policy: [Dropdown]
- Media Sec Policy: SRTP (highlighted with a red box)
- RTCP Mux:  enable
- Ice Profile: [Dropdown]

The bottom screenshot shows the following configuration items:

- SDP Inactive Only:  enable
- DTLS Srtp Profile: [Dropdown]
- Srtp Msm Passthrough:  enable
- Class Profile: [Dropdown]
- In Translationid: [Dropdown]
- Out Translationid: addPlus (highlighted with a red box)
- In Manipulationid: [Dropdown]
- Out Manipulationid: [Dropdown]
- Average Rate Limit: 0 (Range: 0..4294967295)
- Access Control Trust Level: none

Figure 16: Realm Config towards Google Voice cont.

Configuration View Configuration

- media-manager
  - codec-policy
  - media-manager
  - media-policy
  - realm-config**
  - steering-pool
- security
- session-router
- system

---

media-manager
 

- codec-policy
- media-manager
- media-policy
- realm-config**
- steering-pool

security

session-router

system

### Modify Realm Config

Maximum Signal Threshold	<input type="text" value="0"/>	( Range: 0..4294967295 )
Untrusted Signal Threshold	<input type="text" value="0"/>	( Range: 0..4294967295 )
Nat Trust Threshold	<input type="text" value="0"/>	( Range: 0..65535 )
Max Endpoints Per Nat	<input type="text" value="0"/>	( Range: 0..65535 )
Nat Invalid Message Threshold	<input type="text" value="0"/>	( Range: 0..65535 )
Wait Time For Invalid Register	<input type="text" value="0"/>	( Range: 0,4..300 )
Deny Period	<input type="text" value="30"/>	( Range: 0..4294967295 )
Session Max Life Limit	<input type="text" value="0"/>	
Untrust Cac Failure Threshold	<input type="text" value="0"/>	( Range: 0..4294967295 )
Subscription Id Type	<input type="text" value="END_USER_NONE"/>	
Trunk Context	<input type="text"/>	
Early Media Allow	<input type="text"/>	

---

### Modify Realm Config

Restricted Latching	<input type="text" value="none"/>
Options	<input type="text"/>
SPL Options	<input type="text"/>
Delay Media Update	<input type="checkbox"/> enable
Refer Call Transfer	<input type="text" value="disabled"/>
Hold Refer Reininvite	<input type="checkbox"/> enable
Refer Notify Provisional	<input type="text" value="none"/>
Dyn Refer Term	<input type="checkbox"/> enable
Codec Policy	<input type="text" value="Google"/>

Figure 17: Realm Config towards Google Voice Cont.

Configuration View Configuration

- media-manager
  - codec-policy
  - media-manager
  - media-policy
  - realm-config**
  - steering-pool
- security
- session-router
- system

---

### Modify Realm Config

Codec ManIP In Realm	<input type="checkbox"/>	enable
Codec ManIP In Network	<input checked="" type="checkbox"/>	enable
RTCP Policy	<input type="text"/>	
Constraint Name	<input type="text"/>	
Session Recording Server	<input type="text"/>	
Session Recording Required	<input type="checkbox"/>	enable
SIP Profile	<input type="text"/>	
Flow Time Limit	<input type="text" value="-1"/>	( Range: -1..2147483647 )
Initial Guard Timer	<input type="text" value="-1"/>	( Range: -1..2147483647 )
Subsq Guard Timer	<input type="text" value="-1"/>	( Range: -1..2147483647 )
TCP Flow Time Limit	<input type="text" value="-1"/>	( Range: -1..2147483647 )

---

### Modify Realm Config

TCP Initial Guard Timer	<input type="text" value="-1"/>	( Range: -1..2147483647 )
TCP Subsq Guard Timer	<input type="text" value="-1"/>	( Range: -1..2147483647 )
SIP Isup Profile	<input type="text"/>	
QoS Constraint	<input type="text"/>	
TCP Media Profile	<input type="text"/>	
Monitoring Filters	<input type="text"/>	
Node Functionality	<input type="text"/>	
Default Location String	<input type="text"/>	
Alt Family Realm	<input type="text"/>	
Pref Addr Type	<input type="text" value="none"/>	

Figure 18: Realm Config towards Google Voice Cont.

session-router	▶	Ringback Trigger	<input type="text" value="none"/>
system	▶	Ringback File	<input type="text"/>
		Merge Early Dialogs	<input type="checkbox"/> enable
		User Site	<input type="text"/>
		Srvcc Trfo	<input type="text"/>

Show All

Figure 19: Realm Config towards Google Voice Cont.

Configuration

- media-manager
- codecs-policy
- media-manager
- media-policy
- realm-config**
- steering-pool
- security
- session-router
- system

### Modify Realm Config

Identifier	<input type="text" value="onprem"/>
Description	<input type="text" value="onprem"/>
Addr Prefix	<input type="text" value="0.0.0.0"/>
Network Interfaces	<input type="text" value="s1p0:0.4"/>
Media Realm List	<input type="text"/>
Mm In Realm	<input checked="" type="checkbox"/> enable
Mm In Network	<input checked="" type="checkbox"/> enable
Mm Same Ip	<input checked="" type="checkbox"/> enable
QoS Enable	<input type="checkbox"/> enable

Figure 20: Realm Config towards OnPrem PBX



- media-manager ▼
- codec-policy
- media-manager
- media-policy
- realm-config
- steering-pool
- security ▶
- session-router ▼
- access-control
- media-manager ▼
- codec-policy
- media-manager
- media-policy
- realm-config
- steering-pool
- security ▶
- session-router ▼
- access-control
- account-config
- filter-config
- ldap-config
- local-policy

### Modify Realm Config

Max Bandwidth	<input type="text" value="0"/>	( Range: 0..999999999 )
Max Priority Bandwidth	<input type="text" value="0"/>	( Range: 0..999999999 )
Parent Realm	<input type="text"/>	▼
DNS Realm	<input type="text"/>	▼
Media Policy	<input type="text"/>	▼
Media Sec Policy	<input type="text" value="RTP"/>	▼
RTCP Mux	<input type="checkbox"/> enable	
Ice Profile	<input type="text"/>	▼

---

### Modify Realm Config

SDP Inactive Only	<input type="checkbox"/> enable	
DTLS Srtplib Profile	<input type="text"/>	▼
Srtplib Msm Passthrough	<input type="checkbox"/> enable	
Class Profile	<input type="text"/>	▼
In Translationid	<input type="text"/>	▼
Out Translationid	<input type="text" value="removeE164"/>	▼
In Manipulationid	<input type="text"/>	▼
Out Manipulationid	<input type="text"/>	▼
Average Rate Limit	<input type="text" value="0"/>	( Range: 0..4294967295 )
Access Control Trust Level	<input type="text" value="none"/>	▼
Invalid Signal Threshold	<input type="text" value="0"/>	( Range: 0..4294967295 )
Maximum Signal Threshold	<input type="text" value="0"/>	( Range: 0..4294967295 )

**Figure 21: Realm Config towards OnPrem PBX Cont.**

Configuration View Configuration

- media-manager
  - codec-policy
  - media-manager
  - media-policy
  - realm-config**
  - steering-pool
- security ▶
- session-router ▶
- system ▶

### Modify Realm Config

Untrusted Signal Threshold	<input type="text" value="0"/>	( Range: 0..4294967295 )
Nat Trust Threshold	<input type="text" value="0"/>	( Range: 0..65535 )
Max Endpoints Per Nat	<input type="text" value="0"/>	( Range: 0..65535 )
Nat Invalid Message Threshold	<input type="text" value="0"/>	( Range: 0..65535 )
Wait Time For Invalid Register	<input type="text" value="0"/>	( Range: 0,4..300 )
Deny Period	<input type="text" value="30"/>	( Range: 0..4294967295 )
Session Max Life Limit	<input type="text" value="0"/>	
Untrust Cac Failure Threshold	<input type="text" value="0"/>	( Range: 0..4294967295 )
Subscription Id Type	<input type="text" value="END_USER_NONE"/>	
Trunk Context	<input type="text"/>	
Early Media Allow	<input type="text"/>	
Enforcement Profile	<input type="text"/>	

Configuration View Configuration

- media-manager
  - codec-policy
  - media-manager
  - media-policy
  - realm-config**
  - steering-pool
- security ▶
- session-router ▶
- system ▶

### Modify Realm Config

Additional Prefixes	<input type="text"/>
Restricted Latching	<input type="text" value="none"/>
Options	<input type="text"/>
SPL Options	<input type="text"/>
Delay Media Update	<input type="checkbox"/> enable
Refer Call Transfer	<input type="text" value="disabled"/>
Hold Refer Reinvite	<input type="checkbox"/> enable
Refer Notify Provisional	<input type="text" value="none"/>
Dyn Refer Term	<input type="checkbox"/> enable
Codec Policy	<input type="text"/>

**Figure 22: Realm Config towards OnPrem PBX Cont.**

Configuration View Configuration

- media-manager ▾
- codec-policy
- media-manager
- media-policy
- realm-config**
- steering-pool
- security ▶
- session-router ▶
- system ▶

### Modify Realm Config

Codec ManIP In Realm	<input type="checkbox"/>	enable
Codec ManIP In Network	<input checked="" type="checkbox"/>	enable
RTCP Policy	<input type="text"/>	
Constraint Name	<input type="text"/>	
Session Recording Server	<input type="text"/>	
Session Recording Required	<input type="checkbox"/>	enable
SIP Profile	<input type="text"/>	
Flow Time Limit	<input type="text" value="-1"/>	( Range: -1..2147483647 )
Initial Guard Timer	<input type="text" value="-1"/>	( Range: -1..2147483647 )
Subsq Guard Timer	<input type="text" value="-1"/>	( Range: -1..2147483647 )
TCP Flow Time Limit	<input type="text" value="-1"/>	( Range: -1..2147483647 )

Configuration View Configuration

- media-manager ▾
- codec-policy
- media-manager
- media-policy
- realm-config**
- steering-pool
- security ▶
- session-router ▶
- system ▶

### Modify Realm Config

TCP Initial Guard Timer	<input type="text" value="-1"/>	( Range: -1..2147483647 )
TCP Subsq Guard Timer	<input type="text" value="-1"/>	( Range: -1..2147483647 )
SIP Isup Profile	<input type="text"/>	
QoS Constraint	<input type="text"/>	
TCP Media Profile	<input type="text"/>	
Monitoring Filters	<input type="text"/>	
Node Functionality	<input type="text"/>	
Default Location String	<input type="text"/>	
Alt Family Realm	<input type="text"/>	
Pref Addr Type	<input type="text" value="none"/>	
Sm Icsi Match For Invite	<input type="text"/>	

**Figure 23: Realm Config towards OnPrem PBX Cont.**

steering-pool

security ▶

session-router ▶

system ▶

Show All

Sm Icsi Match For Message

Ringback Trigger: none

Ringback File

Merge Early Dialogs:  enable

User Site

Srvcc Trfo

OK Back

Figure 24: Realm Config towards OnPrem PBX Cont.

Configuration View Configuration 🔍

media-manager ▼

codec-policy

media-manager

media-policy

realm-config

steering-pool

security ▶

session-router ▶

system ▶

### Modify Realm Config

Identifier: PSTNGW

Description: PSTNGW

Addr Prefix: 0.0.0.0

Network Interfaces: s1p0:0.4 ✕

Media Realm List

Mm In Realm:  enable

Mm In Network:  enable

Mm Same Ip:  enable

QoS Enable:  enable

Figure 25: Realm Config towards PSTN Gateway

The image displays two screenshots of a web-based configuration interface for a realm. The left sidebar contains a tree view with categories like 'media-manager', 'security', and 'session-router'. The main content area is titled 'Modify Realm Config'.

**Top Screenshot:** Shows configuration options including Max Bandwidth (0), Max Priority Bandwidth (0), Parent Realm, DNS Realm, Media Policy, **Media Sec Policy (RTP)**, RTCP Mux (enable), and Ice Profile.

**Bottom Screenshot:** Shows configuration options including SDP Inactive Only (enable), DTLS Srtp Profile, Srtp Msm Passthrough (enable), Class Profile, In Translationid, **Out Translationid (removeE164)**, In Manipulationid, Out Manipulationid, Average Rate Limit (0), Access Control Trust Level (none), Invalid Signal Threshold (0), and Maximum Signal Threshold (n).

**Figure 26: Realm Config towards PSTN Gateway Cont.**

Configuration View Configuration Q

- media-manager
  - codec-policy
  - media-manager
  - media-policy
  - realm-config**
  - steering-pool
- security
- session-router
- system

---

### Modify Realm Config

Untrusted Signal Threshold	0	( Range: 0..4294967295 )
Nat Trust Threshold	0	( Range: 0..65535 )
Max Endpoints Per Nat	0	( Range: 0..65535 )
Nat Invalid Message Threshold	0	( Range: 0..65535 )
Wait Time For Invalid Register	0	( Range: 0,4..300 )
Deny Period	30	( Range: 0..4294967295 )
Session Max Life Limit	0	
Untrust Cac Failure Threshold	0	( Range: 0..4294967295 )
Subscription Id Type	END_USER_NONE	
Trunk Context		
Early Media Allow		
Enforcement Profile		

---

### Modify Realm Config

Restricted Latching	none
Options	
SPL Options	
Delay Media Update	<input type="checkbox"/> enable
Refer Call Transfer	disabled
Hold Refer Reinvite	<input type="checkbox"/> enable
Refer Notify Provisional	none
Dyn Refer Term	<input type="checkbox"/> enable
<b>Codec Policy</b>	<b>PSTN</b>

**Figure 27: Realm Config towards PSTN Gateway Cont.**

Note: Codec Policy towards PSTN has codecs which are supported by PSTN.

Configuration View Configuration

- media-manager ▾
- codec-policy
- media-manager
- media-policy
- realm-config**
- steering-pool
- security ▶
- session-router ▶
- system ▶

### Modify Realm Config

Codec ManIP In Realm	<input type="checkbox"/>	enable
Codec ManIP In Network	<input checked="" type="checkbox"/>	enable
RTCP Policy	<input type="text"/>	▾
Constraint Name	<input type="text"/>	▾
Session Recording Server	<input type="text"/>	
Session Recording Required	<input type="checkbox"/>	enable
SIP Profile	<input type="text"/>	▾
Flow Time Limit	<input type="text" value="-1"/>	( Range: -1..2147483647 )
Initial Guard Timer	<input type="text" value="-1"/>	( Range: -1..2147483647 )
Subsq Guard Timer	<input type="text" value="-1"/>	( Range: -1..2147483647 )
TCP Flow Time Limit	<input type="text" value="-1"/>	( Range: -1..2147483647 )

Configuration View Configuration

- media-manager ▾
- codec-policy
- media-manager
- media-policy
- realm-config**
- steering-pool
- security ▶
- session-router ▶
- system ▶

### Modify Realm Config

TCP Initial Guard Timer	<input type="text" value="-1"/>	( Range: -1..2147483647 )
TCP Subsq Guard Timer	<input type="text" value="-1"/>	( Range: -1..2147483647 )
SIP Isup Profile	<input type="text"/>	▾
QoS Constraint	<input type="text"/>	▾
TCP Media Profile	<input type="text"/>	▾
Monitoring Filters	<input type="text"/>	
Node Functionality	<input type="text"/>	▾
Default Location String	<input type="text"/>	
Alt Family Realm	<input type="text"/>	▾
Pref Addr Type	<input type="text" value="none"/>	▾
Sm Icsi Match For Invite	<input type="text"/>	

Show All

**Figure 28: Realm Config towards PSTN Gateway Cont.**

## 6.4.8 Steering Pool

Navigate to **Configuration > media-manager > steering-pool**.

Steering pool allows configuration to assign IP address, ports and a realm

Steering Pool Config towards Google Voice, OnPrem PBX and PSTN Gateway are shown below.

The screenshot shows the 'Modify Steering Pool' configuration page. The left sidebar has a menu with 'media-manager' expanded, and 'steering-pool' selected. The main content area contains the following configuration fields:

IP Address	10.80.11.21	
Start Port	20000	( Range: 0,1..65535 )
End Port	39999	( Range: 0,1..65535 )
Realm ID	PSTNGW	▼
Network Interface	s1p0:0.4	▼

**Figure 29: Steering Pool towards PSTN Gateway**

The screenshot shows the 'Modify Steering Pool' configuration page. The left sidebar has a menu with 'media-manager' expanded, and 'steering-pool' selected. The main content area contains the following configuration fields:

IP Address	192.65.	
Start Port	20000	( Range: 0,1..65535 )
End Port	39999	( Range: 0,1..65535 )
Realm ID	Google	▼
Network Interface	s0p0:0.4	▼

**Figure 30: Steering Pool towards Google Voice**



media-manager ▼

codec-policy

media-manager

media-policy

realm-config

**steering-pool**

security ▶

### Modify Steering Pool

IP Address	10.80.11.21	
Start Port	50000	( Range: 0,1..65535 )
End Port	59999	( Range: 0,1..65535 )
Realm ID	onprem	▼
Network Interface	s1p0:0.4	▼

**Figure 31: Steering Pool towards OnPrem PBX**

### 6.4.9 SDES Profile

Navigate to **Configuration > Security > media-security > sdes-profile** and configure SDES profile as shown below.

The screenshot shows a web configuration interface for SDES profiles. On the left is a navigation menu with items like 'security-association', 'security-policy', 'local-accounts', 'media-security', 'dtls-srtp-profile', 'media-sec-policy', 'sdes-profile', 'sipura-profile', 'password-policy', 'security-config', 'ssh-config', 'ssh-key', 'tls-global', 'ssh-key', 'tls-global', and 'tls-profile'. The 'media-security' and 'sdes-profile' items are highlighted with red boxes. The main content area is titled 'Modify Sdes Profile' and contains the following configuration fields:

Name	SDES
Crypto List	AES_CM_128_HMAC_SHA1_80
Srtp Auth	<input checked="" type="checkbox"/> enable
Srtp Encrypt	<input checked="" type="checkbox"/> enable
SrTCP Encrypt	<input checked="" type="checkbox"/> enable
Mki	<input type="checkbox"/> enable
Egress Offer Format	same-as-ingress
Use Ingress Session Params	
Options	
Key	
Salt	
Srtp Rekey On Re Invite	<input type="checkbox"/> enable
Lifetime	0 ( Range: 0,20..48 )

**Figure 32: SDES Profile for TLS**

### 6.4.10 Media Sec Policy

Navigate to **Configuration > security > media-security > media-sec-policy** and configure media security policy as shown below.

The screenshot shows a configuration page titled "Modify Media Sec Policy". On the left is a navigation menu with items like "local-accounts", "media-security", "dtls-srtp-profile", "media-sec-policy" (highlighted with a red box), "sdes-profile", "sipura-profile", "password-policy", "security-config", "ssh-config", "ssh-key", "tls-global", and "tls-profile". The main content area has the following settings:

- Name:** RTP (highlighted with a red box)
- Pass Through:**  enable
- Options:** [Empty text box]
- Inbound:**
  - Profile:** [Empty dropdown]
  - Mode:** rtp
  - Protocol:** none
  - Hide Egress Media Update:**  enable
- Outbound:**
  - Profile:** [Empty dropdown]
  - Mode:** rtp

**Figure 33: Media Security Policy for RTP**

SDES profile created in Section 6.4.9 is associated with Media Security Policy for SRTP is shown below.

The screenshot shows a configuration page titled "Modify Media Sec Policy" for a policy named "SRTP". The left sidebar lists various configuration categories, with "media-sec-policy" selected. The main content area is divided into sections for "Inbound" and "Outbound" settings. Red boxes highlight the "Name" field, the "Inbound" settings, and the "Outbound" settings.

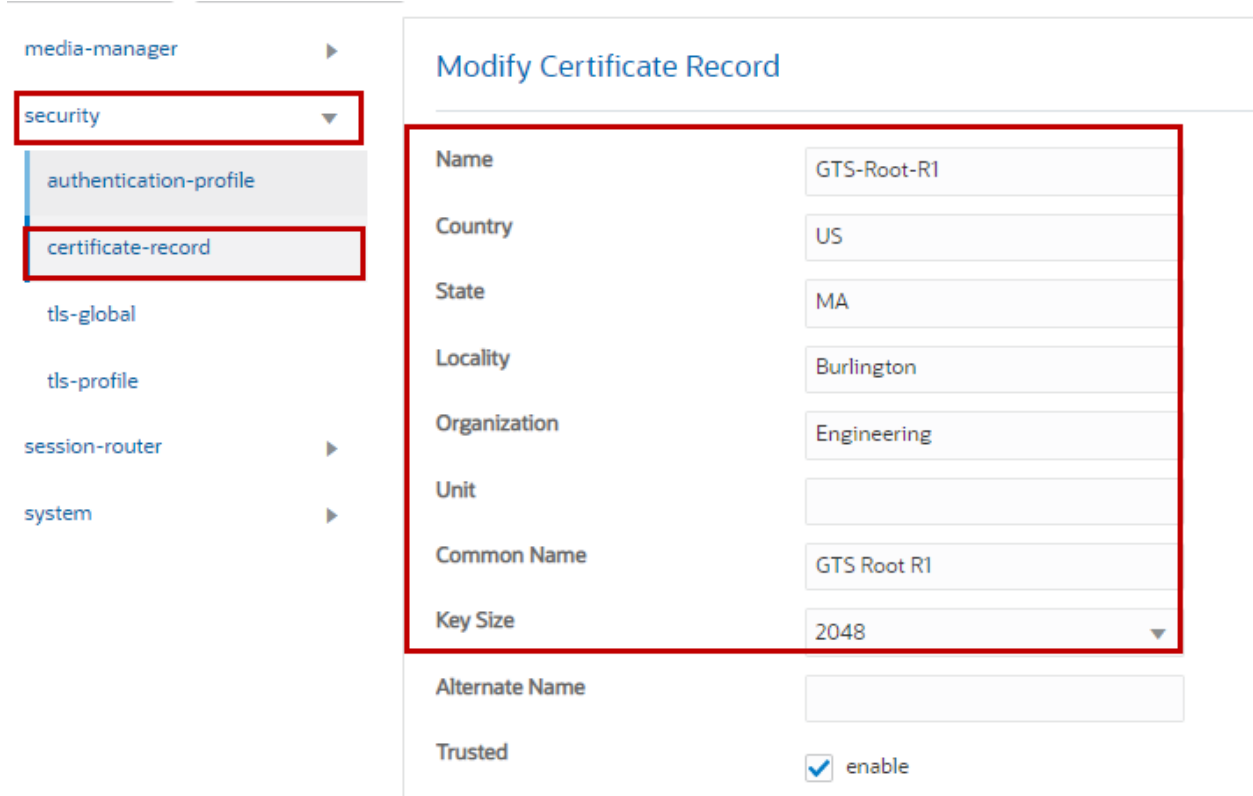
Section	Field	Value
General	Name	SRTP
	Pass Through	<input type="checkbox"/> enable
Inbound	Profile	SDES
	Mode	srtp
	Protocol	sdes
	Hide Egress Media Update	<input type="checkbox"/> enable
Outbound	Profile	SDES
	Mode	srtp
	Protocol	sdes

Figure 34: Media Security Policy for SRTP

### 6.4.11 TLS – Certificate Record

Certificate Record are configuration elements on Oracle SBC which captures information for a TLS certificate such as common-name, key-size etc. Navigate to **Configuration > security > certificate-record**.

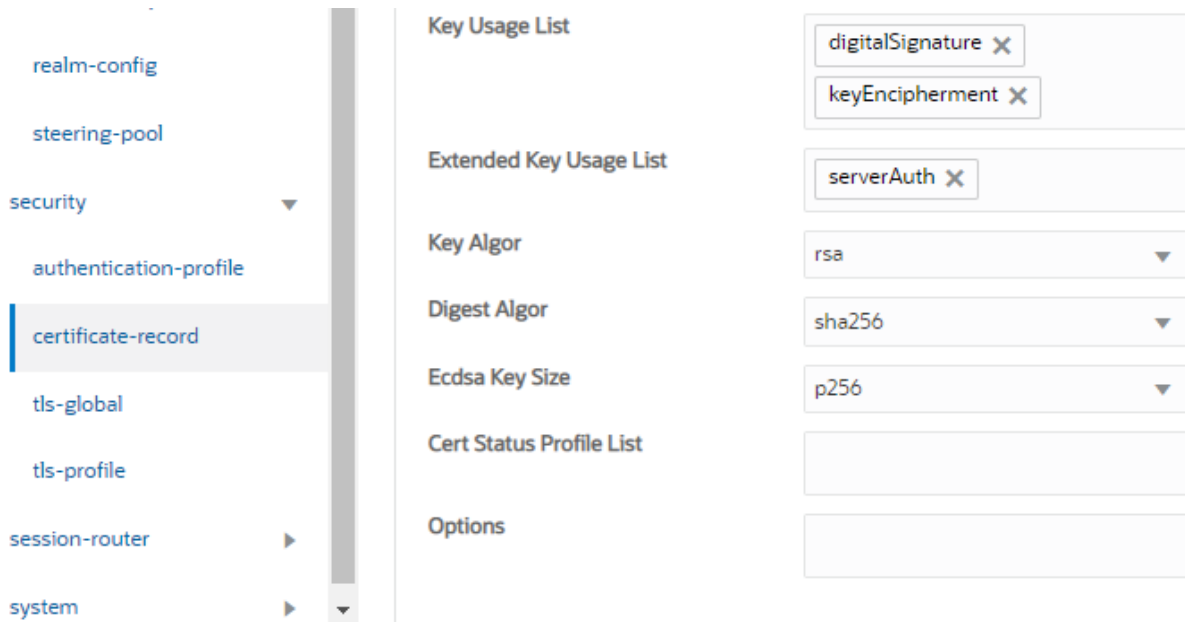
Create a certificate record for Google Voice as shown below.



The screenshot displays the 'Modify Certificate Record' configuration page. On the left sidebar, the 'security' menu item is highlighted with a red box, and the 'certificate-record' sub-item is also highlighted with a red box. The main content area shows the following fields:

Name	GTS-Root-R1
Country	US
State	MA
Locality	Burlington
Organization	Engineering
Unit	
Common Name	GTS Root R1
Key Size	2048
Alternate Name	
Trusted	<input checked="" type="checkbox"/> enable

**Figure 35: Create Certificate Record for Google Voice ROOT CA**



**Figure 36: Create Certificate Record for Google Voice Root CA Cont.**

Create a certificate record for Oracle E-SBC as shown below.

1. Select the Certificate record and Click **Generate icon** to generate CSR.
2. Get the CSR signed and click **Import** to import the signed certificate.

**Note**

Refer to Google Voice SIP Link documentation for other compatible CAs

media-manager  
codecs-policy  
media-manager  
media-policy  
realm-config  
steering-pool  
security  
authentication-profile  
certificate-record  
tls-global  
tls-profile  
session-router

### Modify Certificate Record

Name	sbc3
Country	US
State	Texas
Locality	Plano
Organization	tekvizionLabs
Unit	Tekvizion
Common Name	sbc3.tekvizionlabs.com
Key Size	2048
Alternate Name	
Trusted	<input checked="" type="checkbox"/> enable

Figure 37: Create Certificate Record for Oracle E-SBC

**Figure 38: Create Certificate Record for Oracle E-SBC Cont.**

Similarly create other certificate records for Google Voice and SBC Root CAs and import the certificates as shown below. The following certificate-records are required on the Oracle SBC to connect with Google Voice

Certificate Record

Name	Country	State	Locality	Organization	Unit	Common Name
GTS-Root-R1	US	MA	Burlington	Engineering		GTS Root R1
GoDaddyClass2CertificationAuthority	US	Texas	Plano	Tekvizion	Tekvizion	Go Daddy Class 2 Certification Authority
GoDaddyRootCertificate	US	Texas	Plano	tekvizionLabs	Tekvizion	Go Daddy Root Certificate Authority - G2
GoDaddySecureCertificateG2	US	Texas	Plano	Tekvizion	Tekvizion	Go Daddy Secure Certificate Authority - G2
sbc3	US	Texas	Plano	tekvizionLabs	Tekvizion	sbc3.tekvizionlabs.com

**Figure 39: Certificate Records**



## 6.4.12 TLS – TLS Profile

A TLS profile configuration on the SBC allows for specific certificates to be assigned. Navigate to **Configuration > security > tls-profile**.

Create a TLS profile for Google Voice as shown below.

The screenshot displays the 'Modify TLS Profile' configuration page. The left sidebar shows a navigation menu with 'tls-profile' highlighted. The main content area is titled 'Modify TLS Profile' and contains the following fields:

- Name:** TLSSGBYOC
- End Entity Certificate:** sbc3
- Trusted Ca Certificates:** GoDaddyRootCertificate, GoDaddyClass2CertificationAuthority, GoDaddySecureCertificateG2, GTS-Root-R1
- Cipher List:** DEFAULT
- Verify Depth:** 10 (Range: 0..10)
- Mutual Authenticate:**  enable
- TLS Version:** t12sv12
- Options:** (empty text box)
- Cert Status Check:**  enable
- Cert Status Profile List:** (empty text box)
- Ignore Dead Responder:**  enable
- Allow Self Signed Cert:**  enable

At the bottom of the form are 'OK' and 'Back' buttons. A 'Show All' toggle is located at the bottom left of the sidebar area.

**Figure 40: TLS Profile**

### 6.4.13 Session Timer

Navigate to **Configuration > session-router > session-timer-profile**.

Configure session timer for Google Voice as shown below. This profile is used in Section 6.4.14 SIP Interface for SBC to send UPDATE message as a session-refresher towards Google Voice.

The screenshot shows a configuration page titled "Modify Session Timer Profile". On the left, a navigation menu lists several configuration sections: "session-router", "session-timer-profile" (highlighted with a red box), "session-translation", "sip-advanced-logging", "sip-config", "sip-feature", "sip-feature-caps", and "sip-interface". The main content area contains the following fields:

Name	SessionTimer
Session Expires	900 ( Range: 64..999999999 )
Min Se	90 ( Range: 64..999999999 )
Force Reinvite	<input type="checkbox"/> enable
Request Refresher	uac
Response Refresher	uas

**Figure 41: Session Timer**

## 6.4.14 SIP Interface

Navigate to **Configuration > session-router > sip-interface**.

Create SIP interface towards Google Voice, PSTN Gateway and OnPrem PBX as shown below.

The screenshot displays the 'Modify SIP Interface' configuration page. The left sidebar shows the navigation menu with 'sip-interface' selected. The main configuration area is split into two panels. The top panel includes the following fields:

- State:**  enable
- Realm ID:** onprem
- Description:** (empty text area)

The 'SIP Ports' table below contains the following data:

Action	Sel...	Address	Port	Transport Protocol	TLS Profile	Allow Anonymous	Multi Home Addr
⋮	<input type="checkbox"/>	10.80.11.21	5067	UDP		all	

The bottom panel lists various SIP parameters:

- Initial Inv Trans Expire:** 0 (Range: 0..999999999)
- Session Max Life Limit:** 0
- Proxy Mode:** (dropdown menu)
- Redirect Action:** (dropdown menu)
- Nat Traversal:** none (dropdown menu)
- Nat Interval:** 30 (Range: 0..4294967295)
- TCP Nat Interval:** 90 (Range: 0..4294967295)
- Registration Caching:**  enable
- Min Reg Expire:** 300 (Range: 0..999999999)
- Registration Interval:** 3600 (Range: 0..4294967295)
- Route To Registrar:**  enable
- Secured Network:**  enable

**Figure 42: SIP Interface for OnPrem PBX**

- [sip-advanced-logging](#)
- [sip-config](#)
- [sip-feature](#)
- [sip-feature-caps](#)
- [sip-interface](#)
- [sip-manipulation](#)
- [sip-monitoring](#)
- [sip-nat](#)
- [sip-profile](#)
- [sip-q850-map](#)
- [sip-recursion-policy](#)
- [surrogate-agent](#)

### Modify SIP Interface

---

Uri Fqdn Domain	<input type="text"/>	
Options	<input type="text"/>	
SPL Options	<input type="text"/>	
Trust Mode	<input type="text" value="all"/>	
Max Nat Interval	<input type="text" value="3600"/>	( Range: 0..4294967295 )
Stop Recurse	<input type="text" value="401,407"/>	
Port Map Start	<input type="text" value="0"/>	( Range: 0,1025..65535 )
Port Map End	<input type="text" value="0"/>	( Range: 0,1025..65535 )
In Manipulationid	<input type="text"/>	
Out Manipulationid	<input type="text"/>	
SIP Atcf Feature	<input type="checkbox"/> enable	

---

### Modify SIP Interface

Rfc2833 Payload	<input type="text" value="101"/>	( Range: 96..127 )
Rfc2833 Mode	<input type="text" value="transparent"/>	
Response Map	<input type="text"/>	
Local Response Map	<input type="text"/>	
Sec Agree Feature	<input type="checkbox"/> enable	
Enforcement Profile	<input type="text"/>	
TCP Keepalive	<input type="text" value="none"/>	
Add SDP Invite	<input type="text" value="disabled"/>	
Add SDP In Msg	<input type="text"/>	
P Early Media Header	<input type="text" value="disabled"/>	
P Early Media Direction	<input type="text"/>	

**Figure 43: SIP Interface for OnPrem PBX Cont.**

- [sip-advanced-logging](#)
- [sip-config](#)
- [sip-feature](#)
- [sip-feature-caps](#)
- [sip-interface](#)
- [sip-manipulation](#)
- [sip-monitoring](#)
- [sip-nat](#)
- [sip-profile](#)
- [sip-q850-map](#)
- [sip-recursion-policy](#)

### Modify SIP Interface

---

Add SDP Profiles

Add SDP Profiles In Msg

SIP Profile

SIP Isup Profile

TCP Conn Dereg  ( Range: 0..999999999 )

Kpml Interworking  enable

Kpml2833 lwf On Hairpin  enable

Msrp Delay Egress Bye  enable

Send 380 Response

Pcsf Restoration

- [sip-advanced-logging](#)
- [sip-config](#)
- [sip-feature](#)
- [sip-feature-caps](#)
- [sip-interface](#)
- [sip-manipulation](#)
- [sip-monitoring](#)
- [sip-nat](#)
- [sip-profile](#)
- [sip-q850-map](#)
- [sip-recursion-policy](#)
- [surrogate-agent](#)

### Modify SIP Interface

---

Session Timer Profile

Session Recording Server

Session Recording Required  enable

Service Tag

Reg Cache Route  enable

Diversion Info Mapping Mode

Atcf Icsi Match

SIP Recursion Policy

Asymmetric Preconditions  enable

Asymmetric Preconditions Mode

Sm Icsi Match For Invite

**Figure 44: SIP Interface for OnPrem PBX Cont.**

**Figure 45: SIP Interface for OnPrem PBX Cont.**

TLS Profile is configured with TLSGBYOC configured in Section 6.4.12

Modify SIP Interface Show

State  enable

Realm ID

Description

SIP Ports

Add					
Address	Port	Transport Protocol	TLS Profile	Allow Anonymous	Multi Home Addr
192.65. [redacted]	5061	TLS	TLSGBYOC	agents-only	

**Figure 46: SIP Interface for Google Voice**

- session-constraints*
- session-group
- session-recording-group
- session-recording-server
- session-router
- session-timer-profile
- session-translation
- sip-advanced-logging
- sip-config
- sip-feature
- sip-feature-caps
- sip-interface

### Modify SIP Interface

Initial Inv Trans Expire	<input type="text" value="0"/>	( Range: 0..999999999 )
Session Max Life Limit	<input type="text" value="0"/>	
Proxy Mode	<input type="text"/>	▼
Redirect Action	<input type="text"/>	▼
Nat Traversal	<input type="text" value="none"/>	▼
Nat Interval	<input type="text" value="30"/>	( Range: 0..4294967295 )
TCP Nat Interval	<input type="text" value="90"/>	( Range: 0..4294967295 )
Registration Caching	<input type="checkbox"/> enable	
Min Reg Expire	<input type="text" value="300"/>	( Range: 0..999999999 )
Registration Interval	<input type="text" value="3600"/>	( Range: 0..4294967295 )
Route To Registrar	<input type="checkbox"/> enable	
Secured Network	<input type="checkbox"/> enable	

- session-agent-id-rule
- session-constraints*
- session-group
- session-recording-group
- session-recording-server
- session-router
- session-timer-profile
- session-translation
- sip-advanced-logging
- sip-config
- sip-feature
- sip-feature-caps

### Modify SIP Interface

Uri Fqdn Domain	<input type="text"/>	
Options	<input type="text"/>	
SPL Options	<input type="text"/>	
Trust Mode	<input type="text" value="all"/>	▼
Max Nat Interval	<input type="text" value="3600"/>	( Range: 0..4294967295 )
Stop Recurse	<input type="text" value="401,407"/>	
Port Map Start	<input type="text" value="0"/>	( Range: 0,1025..65535 )
Port Map End	<input type="text" value="0"/>	( Range: 0,1025..65535 )
In Manipulationid	<input type="text"/>	▼
Out Manipulationid	<input type="text"/>	▼
SIP Atcf Feature	<input type="checkbox"/> enable	

**Figure 47: SIP Interface for Google Voice Cont.**

- [session-constraints](#)
- [session-group](#)
- [session-recording-group](#)
- [session-recording-server](#)
- [session-router](#)
- [session-timer-profile](#)
- [session-translation](#)
- [sip-advanced-logging](#)
- [sip-config](#)
- [sip-feature](#)
- [sip-feature-caps](#)
- [sip-interface](#)

### Modify SIP Interface

---

Rfc2833 Payload	<input type="text" value="101"/>	( Range: 96..127 )
Rfc2833 Mode	<input type="text" value="transparent"/>	
Response Map	<input type="text"/>	
Local Response Map	<input type="text"/>	
Sec Agree Feature	<input type="checkbox"/> enable	
Enforcement Profile	<input type="text"/>	
TCP Keepalive	<input type="text" value="none"/>	
Add SDP Invite	<input type="text" value="disabled"/>	
Add SDP In Msg	<input type="text"/>	
P Early Media Header	<input type="text" value="disabled"/>	
P Early Media Direction	<input type="text"/>	

- [session-group](#)
- [session-recording-group](#)
- [session-recording-ser...](#)
- [session-translation](#)
- [sip-config](#)
- [sip-feature](#)
- [sip-interface](#)
- [sip-manipulation](#)
- [sip-monitoring](#)
- [sti-server](#)
- [translation-rules](#)

### Modify SIP Interface

---

Add SDP Profiles	<input type="text"/>	
Add SDP Profiles In Msg	<input type="text"/>	
SIP Profile	<input type="text"/>	
SIP Isup Profile	<input type="text"/>	
TCP Conn Dereg	<input type="text" value="0"/>	( Range: 0..999999999 )
Kpml Interworking	<input type="checkbox"/> enable	
Kpml2833 lwf On Hairpin	<input type="checkbox"/> enable	
Msrp Delay Egress Bye	<input type="checkbox"/> enable	
Send 380 Response	<input type="text"/>	

**Figure 48: SIP Interface for Google Voice Cont.**



*session-agent-id-rule*

*session-constraints*

*session-group*

*session-recording-group*

*session-recording-server*

*session-router*

*session-timer-profile*

*session-translation*

*sip-advanced-logging*

*sip-config*

*sip-feature*

*sip-feature-caps*

*session-recording-server*

*session-router*

*session-timer-profile*

*session-translation*

*sip-advanced-logging*

*sip-config*

*sip-feature*

*sip-feature-caps*

**sip-interface**

### Modify SIP Interface

Session Timer Profile

Session Recording Server

Session Recording Required  enable

Service Tag

Reg Cache Route  enable

Diversion Info Mapping Mode

Atcf Icsi Match

SIP Recursion Policy

Asymmetric Preconditions  enable

Asymmetric Preconditions Mode

Sm Icsi Match For Invite

Sm Icsi Match For Message

S8hr Profile

Ringback Trigger

Ringback File

Npli Profile

Hist To Div For Cause 380

User Agent

**Figure 49: SIP Interface for Google Voice Cont.**

local-policy  
local-routing-config  
media-profile  
session-agent  
session-group  
session-recording-group  
session-recording-server  
session-translation  
sip-config  
sip-feature  
sip-interface

### Modify SIP Interface

State  enable  
 Realm ID PSTNGW  
 Description To trunk

SIP Ports

Action	Sel...	Address	Port	Transport Protocol	TLS Profile	Allow Anonymous	Multi Home Addr
:	<input type="checkbox"/>	10.80.11.21	5060	UDP		agents-only	

Configuration View Configuration

- sip-advanced-logging
- sip-config
- sip-feature
- sip-feature-caps
- sip-interface**
- sip-manipulation
- sip-monitoring
- sip-nat
- sip-profile
- sip-q850-map
- sip-recursion-policy
- surrogate-agent

### Modify SIP Interface

Initial Inv Trans Expire 0 (Range: 0..999999999)  
 Session Max Life Limit 0  
 Proxy Mode  
 Redirect Action  
 Nat Traversal none  
 Nat Interval 30 (Range: 0..4294967295)  
 TCP Nat Interval 90 (Range: 0..4294967295)  
 Registration Caching  enable  
 Min Reg Expire 300 (Range: 0..999999999)  
 Registration Interval 3600 (Range: 0..4294967295)  
 Route To Registrar  enable  
 Secured Network  enable

Figure 50: SIP Interface for PSTN Gateway

Configuration View Configuration Q

- sip-advanced-logging*
- sip-config
- sip-feature
- sip-feature-caps*
- sip-interface**
- sip-manipulation
- sip-monitoring
- sip-nat*
- sip-profile*
- sip-q850-map*
- sip-recursion-policy*
- surrogate-agent*

---

### Modify SIP Interface

Uri Fqdn Domain	<input type="text"/>	
Options	<input type="text"/>	
SPL Options	<input type="text"/>	
Trust Mode	all	▼
Max Nat Interval	3600	( Range: 0..4294967295 )
Stop Recurse	401,407	
Port Map Start	0	( Range: 0,1025..65535 )
Port Map End	0	( Range: 0,1025..65535 )
In Manipulationid	<input type="text"/>	▼
<b>Out Manipulationid</b>	<b>TowardsPSTN</b>	<b>▼</b>
SIP Atcf Feature	<input type="checkbox"/> enable	

---

### Modify SIP Interface

Rfc2833 Payload	101	( Range: 96..127 )
Rfc2833 Mode	transparent	▼
Response Map	<input type="text"/>	▼
Local Response Map	<input type="text"/>	▼
Sec Agree Feature	<input type="checkbox"/> enable	
Enforcement Profile	<input type="text"/>	▼
TCP Keepalive	none	▼
Add SDP Invite	disabled	▼
Add SDP In Msg	<input type="text"/>	
P Early Media Header	disabled	▼
P Early Media Direction	<input type="text"/>	

**Figure 51: SIP Interface for PSTN Gateway Cont.**

- [sip-advanced-logging](#)
- [sip-config](#)
- [sip-feature](#)
- [sip-feature-caps](#)
- [sip-interface](#)
- [sip-manipulation](#)
- [sip-monitoring](#)
- [sip-nat](#)
- [sip-profile](#)
- [sip-q850-map](#)
- [sip-recursion-policy](#)
- [surrogate-agent](#)

### Modify SIP Interface

---

Add SDP Profiles

Add SDP Profiles In Msg

SIP Profile

SIP Isup Profile

TCP Conn Dereg  ( Range: 0..999999999 )

Kpml Interworking  enable

Kpml2833 lwf On Hairpin  enable

Msrp Delay Egress Bye  enable

Send 380 Response

Pcscf Restoration

Session Timer Profile

---

### Modify SIP Interface

Session Recording Server

Session Recording Required  enable

Service Tag

Reg Cache Route  enable

Diversion Info Mapping Mode

Atcf Icsi Match

SIP Recursion Policy

Asymmetric Preconditions  enable

Asymmetric Preconditions Mode

Sm Icsi Match For Invite

**Figure 52: SIP Interface for PSTN Gateway Cont.**

<b>sip-interface</b>		
sip-manipulation		
sip-monitoring		
<i>sip-nat</i>		
<i>sip-profile</i>		
<i>sip-q850-map</i>		
<i>sip-recursion-policy</i>		
<i>surrogate-agent</i>		
	Sm Icsi Match For Message	
	S8hr Profile	
	Ringback Trigger	none
	Ringback File	
	Npli Profile	
	Hist To Div For Cause 380	inherit
	User Agent	

**Figure 53: SIP Interface for PSTN Gateway Cont.**

## 6.4.15 Session Agent

Session-agents are config elements which are trusted agents which can send/receive traffic from the SBC with direct access to trusted data path. Navigate to **Configuration > session-router > session-agent**.

Configure Session Agent for Google Voice, OnPrem PBX and PSTN Gateway as shown below.

The screenshot displays the 'Modify Session Agent' configuration page. The left sidebar shows the navigation menu with 'session-agent' selected. The main content area is divided into two sections. The top section, titled 'Modify Session Agent', contains the following fields:

- Hostname: siplink.telephony.goog
- IP Address: (empty)
- Port: 5672 (Range: 0,1025..65535)
- State:  enable
- App Protocol: SIP
- App Type: (empty)
- Transport Method: StaticTLS
- Realm ID: Google
- Egress Realm ID: (empty)
- Description: (empty)

The bottom section, also titled 'Modify Session Agent', is for the 'Match Identifier'. It features a large information icon and the text: 'No match identifier to display. Please add.' Below this is an 'Add' button.

At the bottom of the page, there are additional configuration fields:

- Associated Agents: (empty)
- Constraints:  enable
- Max Sessions: 0 (Range: 0..999999999)
- Max Inbound Sessions: 0 (Range: 0..999999999)

Figure 54: Session Agent for Google Voice

Configuration View Configuration Q

- media-manager
- security
- session-router
  - access-control
  - account-config
  - filter-config
  - ldap-config
  - local-policy
  - local-routing-config
  - media-profile
  - session-agent
  - session-group

### Modify Session Agent

Max Outbound Sessions	0	( Range: 0..999999999 )
Max Burst Rate	0	( Range: 0..999999999 )
Max Inbound Burst Rate	0	( Range: 0..999999999 )
Max Outbound Burst Rate	0	( Range: 0..999999999 )
Max Sustain Rate	0	( Range: 0..999999999 )
Max Inbound Sustain Rate	0	( Range: 0..999999999 )
Max Outbound Sustain Rate	0	( Range: 0..999999999 )
Min Asr	0	( Range: 0..100 )
Cac Trap Threshold	0	( Range: 0..99 )
Session Max Life Limit	0	
Time To Resume	0	( Range: 0..999999999 )
In Service Period	0	( Range: 0..999999999 )

- rph-policy
- rph-profile
- service-health
- session-agent
- session-agent-id-rule
- session-constraints
- session-group
- session-recording-group
- session-recording-ser...
- session-router
- session-timer-profile

### Modify Session Agent

Burst Rate Window	0	( Range: 0..999999999 )
Sustain Rate Window	0	( Range: 0..999999999 )
Proxy Mode		
Redirect Action		
Loose Routing	<input checked="" type="checkbox"/> enable	
Response Map		
Ping Method	OPTIONS	
Ping Interval	30	( Range: 0..4294967295 )
Ping Send Mode	keep-alive	
Ping All Addresses	<input type="checkbox"/> enable	

Figure 55: Session Agent for Google Voice Cont.

Configuration View Configuration Q

- media-manager ▶
- security ▶
- session-router ▼
  - access-control
  - account-config
  - filter-config
  - ldap-config
  - local-policy
  - local-routing-config
  - media-profile
  - session-agent**
  - session-group

### Modify Session Agent

Options	<input type="text"/>
SPL Options	<input type="text"/>
Media Profiles	<input type="text"/>
In Translationid	<input type="text"/> ▼
Out Translationid	<input type="text"/> ▼
Trust Me	<input type="checkbox"/> enable
Local Response Map	<input type="text"/> ▼
Ping Response	<input checked="" type="checkbox"/> enable
In Manipulationid	<input type="text"/> ▼
Out Manipulationid	GoogleManipulation ▼

Configuration View Configuration Q

- media-manager ▶
- security ▶
- session-router ▼
  - access-control
  - account-config
  - filter-config
  - ldap-config
  - local-policy
  - local-routing-config
  - media-profile
  - session-agent**
  - session-group

### Modify Session Agent

Manipulation String	<input type="text"/>
Manipulation Pattern	<input type="text"/>
Trunk Group	<input type="text"/>
Max Register Sustain Rate	<input type="text" value="0"/> ( Range: 0..999999999 )
Invalidate Registrations	<input type="checkbox"/> enable
Rfc2833 Mode	none ▼
Rfc2833 Payload	<input type="text" value="0"/> ( Range: 0,96..127 )
Codec Policy	<input type="text"/> ▼
Refer Call Transfer	enabled ▼
Refer Notify Provisional	none ▼
Reuse Connections	NONE ▼

**Figure 56: Session Agent for Google Voice Cont.**



- filter-config
- ldap-config
- local-policy
- local-routing-config
- media-profile
- session-agent
- session-group
- session-recording-group
- session-recording-server
- session-translation

### Modify Session Agent Show

TCP Keepalive enabled ▼

TCP Reconn Interval 00 (Range: 0,2..300)

Max Register Burst Rate 0 (Range: 0..999999999)

Rate Constraints

Method	Max Inbound Burst Rate	Max Outbound Burst Rate	Max Inbound Sustain Rate	Max Outbound Sustain Rate
No data to display.				

---

**Configuration** View Configuration Q

- media-manager ▶
- security ▶
- session-router ▼
- access-control
- account-config
- filter-config
- ldap-config
- local-policy
- local-routing-config
- media-profile
- session-agent
- session-group

### Modify Session Agent

SIP Isup Profile ▼

Kpm1 Interworking inherit ▼

Kpm2833 lwf On Hairpin inherit ▼

Precedence 0 (Range: 0..4294967295)

Monitoring Filters [Empty Field]

Auth Attribute

No auth attributes to display. Please add.

**Figure 57: Session Agent for Google Voice Cont.**

session-router ▼

- access-control
- account-config
- filter-config
- ldap-config
- local-policy
- local-routing-config
- session-agent
- session-group
- session-recording-group ▼

Show All

Session Recording Server

Session Recording Required  enable

Hold Refer Reinvite  enable

Send TCP Fin  enable

SIP Recursion Policy

Sm Icsi Match For Invite

Sm Icsi Match For Message

Ringback Trigger

Ringback File

**Figure 58: Session Agent for Google Voice Cont.**

Configuration View Configuration

- access-control
- account-config
- filter-config
- ldap-config
- local-policy
- local-routing-config
- media-profile
- session-agent**
- session-group
- session-recording-group
- session-recording-server
- session-translation

### Modify Session Agent


Hostname	<input type="text" value="172.16.29.53"/>	
IP Address	<input type="text" value="172.16.29.53"/>	
Port	<input type="text" value="5060"/>	( Range: 0,1025..65535 )
State	<input checked="" type="checkbox"/> enable	
App Protocol	<input type="text" value="SIP"/>	
App Type	<input type="text"/>	
Transport Method	<input type="text" value="UDP"/>	
Realm ID	<input type="text" value="onprem"/>	
Egress Realm ID	<input type="text"/>	
Description	<input type="text"/>	

Configuration View Configuration

- access-control
- account-config
- filter-config
- ldap-config**
- local-policy
- local-routing-config
- media-profile
- session-agent**
- session-group
- session-recording-group
- session-recording-server
- session-translation

### Modify Session Agent

Match Identifier



No match identifier to display. Please add.

Associated Agents

Constraints  enable

Max Sessions  ( Range: 0..999999999 )

**Figure 59: Session Agent for OnPrem PBX**

account-config  
 filter-config  
 ldap-config  
 local-policy  
 local-routing-config  
 media-profile  
 session-agent  
 session-group  
 session-recording-group  
 session-recording-server  
 session-translation  
 rph-policy  
 rph-profile  
 service-health  
 session-agent  
 session-agent-id-rule  
 session-constraints  
 session-group  
 session-recording-group  
 session-recording-ser...  
 session-router

Max Inbound Sessions	0	( Range: 0..999999999 )
Max Outbound Sessions	0	( Range: 0..999999999 )
Max Burst Rate	0	( Range: 0..999999999 )
Max Inbound Burst Rate	0	( Range: 0..999999999 )
Max Outbound Burst Rate	0	( Range: 0..999999999 )
Max Sustain Rate	0	( Range: 0..999999999 )
Max Inbound Sustain Rate	0	( Range: 0..999999999 )
Max Outbound Sustain Rate	0	( Range: 0..999999999 )
Min Asr	0	( Range: 0..100 )
Cac Trap Threshold	0	( Range: 0..99 )
Session Max Life Limit	0	
Time To Resume	0	( Range: 0..999999999 )
In Service Period	0	( Range: 0..999999999 )

### Modify Session Agent

	0	( Range: 0..999999999 )
Burst Rate Window	0	( Range: 0..999999999 )
Sustain Rate Window	0	( Range: 0..999999999 )
Proxy Mode		
Redirect Action		
Loose Routing	<input checked="" type="checkbox"/> enable	
Response Map		
Ping Method	OPTIONS	
Ping Interval	30	( Range: 0..4294967295 )
Ping Send Mode	keep-alive	
Ping All Addresses	<input type="checkbox"/> enable	

Figure 60: Session Agent for OnPrem PBX Cont.

- access-control
- account-config
- filter-config
- ldap-config
- local-policy
- local-routing-config
- media-profile
- session-agent
- session-group
- session-recording-group
- session-recording-server
- session-translation

### Modify Session Agent

---

Options	<input type="text"/>
SPL Options	<input type="text"/>
Media Profiles	<input type="text"/>
In Translationid	<input style="border-bottom: 1px solid #ccc;" type="text"/> ▼
Out Translationid	<input style="border-bottom: 1px solid #ccc;" type="text"/> ▼
Trust Me	<input type="checkbox"/> enable
Local Response Map	<input style="border-bottom: 1px solid #ccc;" type="text"/> ▼
Ping Response	<input type="checkbox"/> enable
In Manipulationid	<input style="border-bottom: 1px solid #ccc;" type="text"/> ▼
Out Manipulationid	<input style="border-bottom: 1px solid #ccc;" type="text"/> ▼

---

### Modify Session Agent

Manipulation String	<input type="text"/>
Manipulation Pattern	<input type="text"/>
Trunk Group	<input type="text"/>
Max Register Sustain Rate	<input type="text" value="0"/> ( Range: 0..999999999 )
Invalidate Registrations	<input type="checkbox"/> enable
Rfc2833 Mode	<input type="text" value="none"/> ▼
Rfc2833 Payload	<input type="text" value="0"/> ( Range: 0,96..127 )
Codec Policy	<input style="border-bottom: 1px solid #ccc;" type="text"/> ▼
Refer Call Transfer	<input type="text" value="disabled"/> ▼
Refer Notify Provisional	<input type="text" value="none"/> ▼
Reuse Connections	<input type="text" value="NONE"/> ▼

**Figure 61: Session Agent for OnPrem PBX Cont.**

- access-control
- account-config
- filter-config
- ldap-config
- local-policy
- local-routing-config
- media-profile
- session-agent
- session-group
- session-recording-group
- session-recording-server
- session-translation

### Modify Session Agent


---

TCP Keepalive

TCP Reconn Interval  ( Range: 0,2..300 )

Max Register Burst Rate  ( Range: 0.999999999 )

Rate Constraints



No rate constraints to display. Please add.

---

SIP Profile

---

### Modify Session Agent

---

SIP Isup Profile


Kpml Interworking

Kpml2833 lwf On Hairpin

Precedence  ( Range: 0.4294967295 )

Monitoring Filters

Auth Attribute



No auth attributes to display. Please add.

**Figure 62: Session Agent for OnPrem PBX Cont.**

filter-config

ldap-config

local-policy

local-routing-config

media-profile

session-agent

session-group

session-recording-group

session-recording-server

session-translation

Show All

Session Recording Server

Session Recording Required  enable

Hold Refer Reinvite  enable

Send TCP Fin  enable

SIP Recursion Policy

Sm Icsi Match For Invite

Sm Icsi Match For Message

Ringback Trigger none

Ringback File

OK Back

Figure 63: Session Agent for OnPrem PBX Cont.

Configuration View Configuration

access-control

account-config

filter-config

ldap-config

local-policy

local-routing-config

media-profile

session-agent

session-group

session-recording-group

session-recording-server

session-translation

Modify Session Agent

Hostname 10.64.1.72

IP Address 10.64.1.72

Port 5060 (Range: 0,1025..65535)

State  enable

App Protocol SIP

App Type

Transport Method UDP

Realm ID PSTNGW

Egress Realm ID


Description

Figure 64: Session Agent for PSTN Gateway

- access-control
- account-config
- filter-config
- ldap-config
- local-policy
- local-routing-config
- media-profile
- session-agent
- session-group
- session-recording-group
- session-recording-server
- session-translation

### Modify Session Agent

**Match Identifier**



No match identifier to display. Please add.

Add

**Associated Agents**

**Constraints**  enable

**Max Sessions**  ( Range: 0..999999999 )

**Max Inbound Sessions**  ( Range: 0..999999999 )

---

### Modify Session Agent

Max Outbound Sessions	<input type="text" value="0"/>	( Range: 0..999999999 )
Max Burst Rate	<input type="text" value="0"/>	( Range: 0..999999999 )
Max Inbound Burst Rate	<input type="text" value="0"/>	( Range: 0..999999999 )
Max Outbound Burst Rate	<input type="text" value="0"/>	( Range: 0..999999999 )
Max Sustain Rate	<input type="text" value="0"/>	( Range: 0..999999999 )
Max Inbound Sustain Rate	<input type="text" value="0"/>	( Range: 0..999999999 )
Max Outbound Sustain Rate	<input type="text" value="0"/>	( Range: 0..999999999 )
Min Asr	<input type="text" value="0"/>	( Range: 0..100 )
Cac Trap Threshold	<input type="text" value="0"/>	( Range: 0..99 )
Session Max Life Limit	<input type="text" value="0"/>	
Time To Resume	<input type="text" value="0"/>	( Range: 0..999999999 )
In Service Period	<input type="text" value="0"/>	( Range: 0..999999999 )
Burst Rate Window	<input type="text" value="0"/>	( Range: 0..999999999 )

**Figure 65: Session Agent for PSTN Gateway Cont.**



**Modify Session Agent**

Sustain Rate Window: 0 ( Range: 0..999999999 )

Proxy Mode: [Dropdown]

Redirect Action: [Dropdown]

Loose Routing:  enable

Response Map: [Dropdown]

Ping Method: OPTIONS

Ping Interval: 30 ( Range: 0..4294967295 )

Ping Send Mode: keep-alive [Dropdown]

Ping All Addresses:  enable

Ping In Service Response Codes: [Text Field]

---

**Modify Session Agent**

SPL Options: [Text Field]

Media Profiles: [Text Field]

In Translationid: [Dropdown]

Out Translationid: [Dropdown]

Trust Me:  enable

Local Response Map: [Dropdown]

Ping Response:  enable

In Manipulationid: [Dropdown]

Out Manipulationid: [Dropdown]

Manipulation String: [Text Field]

Manipulation Pattern: [Text Field]

**Figure 66: Session Agent for PSTN Gateway Cont.**

- access-control
- account-config
- filter-config
- ldap-config
- local-policy
- local-routing-config
- media-profile
- session-agent
- session-group
- session-recording-group
- session-recording-server
- session-translation

### Modify Session Agent

---

Trunk Group

Max Register Sustain Rate  ( Range: 0..999999999 )

Invalidate Registrations  enable

Rfc2833 Mode

Rfc2833 Payload  ( Range: 0,96..127 )

Codec Policy

Refer Call Transfer

Refer Notify Provisional

Reuse Connections

TCP Keepalive


TCP Reconn Interval  ( Range: 0,2..300 )

---

### Modify Session Agent

Max Register Burst Rate  ( Range: 0..999999999 )

Rate Constraints



No rate constraints to display. Please add.

[Add](#)

SIP Profile

SIP Isup Profile

Kpml Interworking

**Figure 67: Session Agent for PSTN Gateway Cont.**

- access-control
- account-config
- filter-config
- ldap-config
- local-policy
- local-routing-config
- media-profile
- session-agent
- session-group
- session-recording-group
- session-recording-server

- filter-config
- ldap-config
- local-policy
- local-routing-config
- media-profile
- session-agent
- session-group
- session-recording-group
- session-recording-server
- session-translation

### Modify Session Agent


---

Kpml2833 lwf On Hairpin inherit ▼

Precedence 0 ( Range: 0..4294967295 )

Monitoring Filters

Auth Attribute



No auth attributes to display. Please add.

Add

---

Session Recording Server

Session Recording Required  enable

Hold Refer Reinvite  enable

Send TCP Fin  enable

SIP Recursion Policy  ▼

Sm Icsi Match For Invite

Sm Icsi Match For Message

Ringback Trigger none ▼

Ringback File

**Figure 68: Session Agent for PSTN Gateway Cont.**

### 6.4.16 Local Policy

Local policy config allows for the SBC to route calls from one end of the network to the other based on routing criteria. Navigate to **Configuration > session-router > local-policy**.

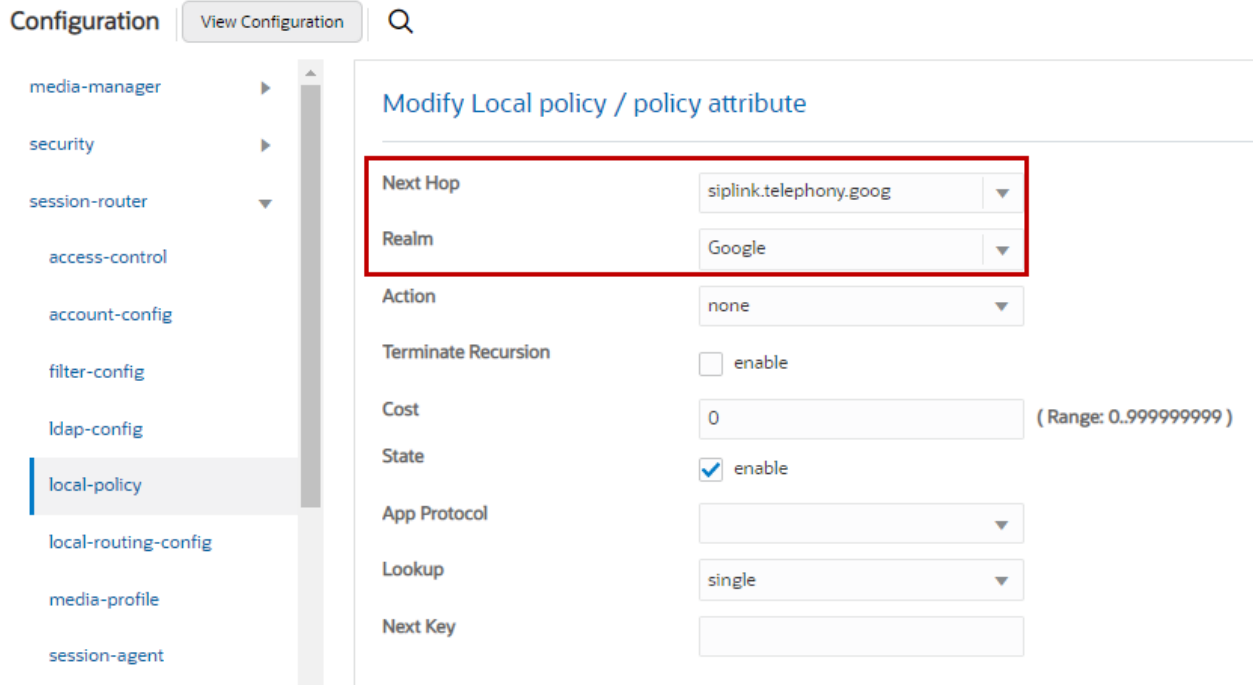
Configure local policy for Google Voice, OnPrem PBX and PSTN Gateway as shown below.

Below Local Policy is used to route calls from PSTN and OnPrem PBX towards Google Voice.

The screenshot displays the 'Modify Local Policy' configuration page. On the left, a navigation sidebar lists various configuration options, with 'session-router' and 'local-policy' highlighted by red boxes. The main configuration area is titled 'Modify Local Policy' and contains the following fields:

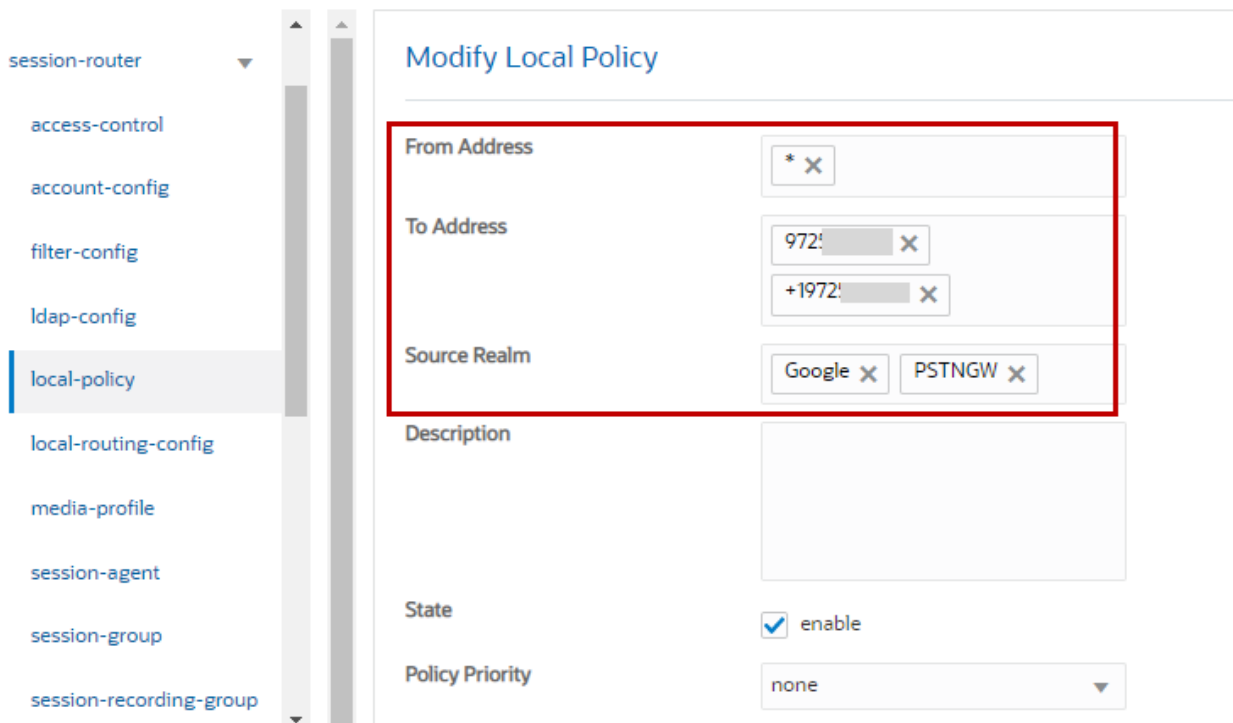
- From Address:** A text input field containing the asterisk (\*) character, with a clear button (X).
- To Address:** A text input field containing the number '972.', with a clear button (X).
- Source Realm:** Three buttons labeled 'cucm', 'onprem', and 'PSTNGW', each with a clear button (X).
- Description:** A text area containing the text 'PSTNGW'.
- State:** A checkbox that is checked, with the label 'enable'.
- Policy Priority:** A dropdown menu currently set to 'none'.

**Figure 69: Local Policy towards Google Voice**



**Figure 70: Local Policy towards Google Voice Cont.**

Below Local Policy is used to route calls from Google Voice and PSTN Gateway towards OnPrem PBX.



**Figure 71: Local Policy towards OnPrem PBX**

media-manager  
security  
session-router  
access-control  
account-config  
filter-config  
ldap-config  
local-policy  
local-routing-config  
media-profile  
session-agent

### Modify Local policy / policy attribute

Next Hop	172.16.29.53
Realm	onprem
Action	none
Terminate Recursion	<input type="checkbox"/> enable
Cost	0 ( Range: 0..999999999 )
State	<input checked="" type="checkbox"/> enable
App Protocol	
Lookup	single
Next Key	

**Figure 72: Local Policy towards OnPrem PBX Cont.**

Below Local Policy is used to route calls from Google Voice and OnPrem PBX towards PSTN Gateway.

session-router  
access-control  
account-config  
filter-config  
ldap-config  
local-policy  
local-routing-config  
media-profile  
session-agent  
session-group  
session-recording-group

### Modify Local Policy

From Address	* X
To Address	0119199 X 214 X 800 X +1214 X +1800 X +1866 X +1888 X +9199 X
Source Realm	Google X onprem X
Description	
State	<input checked="" type="checkbox"/> enable

**Figure 73: Local Policy towards PSTN Gateway**

media-manager ▶  
security ▶  
session-router ▼  
access-control  
account-config  
filter-config  
ldap-config  
local-policy  
local-routing-config  
media-profile  
session-agent

### Modify Local policy / policy attribute

Next Hop	10.64.172	▼
Realm	PSTNGW	▼
Action	none	▼
Terminate Recursion	<input type="checkbox"/> enable	
Cost	0	( Range: 0..999999999 )
State	<input checked="" type="checkbox"/> enable	
App Protocol		▼
Lookup	single	▼
Next Key		

**Figure 74: Local Policy towards PSTN Gateway Cont.**

## 6.4.17 SIP Manipulation

Navigate to **Configuration > session-router > sip-manipulation**.

Configure SIP manipulation towards Google Voice and PSTN Gateway as shown below.

### SIP manipulation towards Google Voice

Configuration View Configuration

- local-routing-config
- media-profile
- session-agent
- session-group
- session-recording-group
- session-recording-server
- session-translation
- sip-config
- sip-feature
- sip-interface
- sip-manipulation**
- sip-monitoring

#### Modify SIP Manipulation

Name: GoogleManipulation  
Description: Manipulations on google side

Split Headers:

Join Headers:

CfgRules

Action	Sel...	Name	Element Type
:	<input type="checkbox"/>	changeReqUri	header-rule
:	<input type="checkbox"/>	changeFromIP	header-rule

**Figure 75: SIP Manipulation towards Google Voice**



Below header rule is created to change Request-URI towards Google Voice to “trunk.sip.voice.google.com”. Msg type is set to out-of-dialog indicates that this rule applies only to out-of-dialog request

The screenshot shows the 'Modify Sip manipulation / header rule' configuration page. The left sidebar lists various configuration categories, with 'sip-manipulation' selected. The main area contains the following fields:

- Name:** changeReqUri
- Header Name:** Request-URI
- Action:** manipulate
- Comparison Type:** pattern-rule
- Msg Type:** out-of-dialog
- Methods:** INVITE, OPTIONS
- Match Value:** (empty)
- New Value:** (empty)

Below these fields is a table for 'CfgRules' with the following entry:

Action	Sel...	Name	Element Type
:	<input type="checkbox"/>	ReqURI	element-rule

**Figure 76: SIP Manipulation towards Google Voice to change Request-URI**

The screenshot shows the 'Modify Sip manipulation / header rule / element rule' configuration page. The left sidebar lists various configuration categories, with 'media-profile' selected. The main area contains the following fields:

- Name:** ReqURI
- Parameter Name:** (empty)
- Type:** uri-host
- Action:** replace
- Match Val Type:** any
- Comparison Type:** case-insensitive
- Match Value:** (empty)
- New Value:** "trunk.sip.voice.google.com"

**Figure 77: SIP Manipulation towards Google Voice to change Request-URI Cont.**

Below header rule is created to change the host part of FROM header to SBC IP.

local-routing-config  
media-profile  
session-agent  
session-group  
session-recording-group  
session-recording-server  
session-translation  
sip-config  
sip-feature  
sip-interface  
sip-manipulation  
sip-monitoring  
translation-rules  
svstem

Modify Sip manipulation / header rule

Name: changeFromIP  
Header Name: FROM  
Action: manipulate  
Comparison Type: pattern-rule  
Msg Type: request  
Methods: INVITE x

Match Value:   
New Value:

CfgRules

Action	Sel...	Name	Element Type
:	<input type="checkbox"/>	changelP	element-rule

**Figure 78: SIP Manipulation towards Google Voice to change host part of FROM IP**

local-routing-config  
media-profile  
session-agent  
session-group  
session-recording-group  
session-recording-server  
session-translation  
sip-config  
sip-feature  
sip-interface

Modify Sip manipulation / header rule / element rule

Name: changeIP  
Parameter Name:

Type: uri-host  
Action: replace  
Match Val Type: any  
Comparison Type: pattern-rule  
Match Value:   
New Value: \$LOCAL\_IP

**Figure 79: SIP Manipulation towards Google Voice to change host part of FROM IP Cont.**

Below header rule is created to change the host part of TO header to Google Voice IP.

The screenshot shows the 'Modify Sip manipulation / header rule' configuration page. A sidebar on the left lists various configuration categories, with 'sip-manipulation' selected. The main form contains the following fields:

- Name: changeToIP
- Header Name: TO
- Action: manipulate
- Comparison Type: pattern-rule
- Msg Type: request
- Methods: INVITE
- Match Value: (empty)
- New Value: (empty)

Below the form is a 'CfgRules' table with the following entry:

Action	Sel...	Name	Element Type
:	<input type="checkbox"/>	changIP	element-rule

Figure 80: SIP Manipulation towards Google Voice to change host part of TO IP

The screenshot shows the 'Modify Sip manipulation / header rule / element rule' configuration page. The sidebar on the left is the same as in Figure 80. The main form contains the following fields:

- Name: changIP
- Parameter Name: (empty)
- Type: uri-host
- Action: replace
- Match Val Type: any
- Comparison Type: pattern-rule
- Match Value: (empty)
- New Value: \$REMOTE\_IP

Figure 81: SIP Manipulation towards Google Voice to change host part of TO IP Cont.

Below header rule is created to add header “X-Google-Pbx-Trunk-Secret-Key” with key value generated on the Google Voice admin console during SIP trunk creation.

The screenshot shows a web interface for configuring SIP manipulation. On the left is a navigation menu with items: sip-config, sip-feature, sip-interface, sip-manipulation (highlighted), sip-monitoring, sti-server, translation-rules, system, fraud-protection, host-route, and http-client. The main area is titled "Modify Sip manipulation / header rule". The configuration form includes the following fields:

Name	AddXGoogleheader
Header Name	X-Google-Pbx-Trunk-Secret-Key
Action	add
Comparison Type	case-insensitive
Msg Type	request
Methods	INVITE X OPTIONS X
Match Value	
New Value	"ffd57 [redacted]

At the bottom of the form, there is a label "CfgRules".

**Figure 82: SIP Manipulation towards Google Voice to add X-Google-Pbx-Trunk-Secret-Key Header**

Below header rule is created to add P-Asserted-Identity header. This rule sets P-Asserted-Identity to E.164 number format.

The screenshot shows the configuration for a SIP manipulation rule. The left sidebar lists various configuration categories, with 'sip-manipulation' selected. The main panel is titled 'Modify Sip manipulation / header rule'. The configuration fields are as follows:

Name	storepai
Header Name	P-Asserted-Identity
Action	store
Comparison Type	case-insensitive
Msg Type	request
Methods	INVITE
Match Value	< sip: (.*) (@ .*)
New Value	

New Value \$storepai.\$1+"1"+\$storepai.\$2+"@sbc3.tekvizionlabs.com"+">" adds +1 to the PAI header

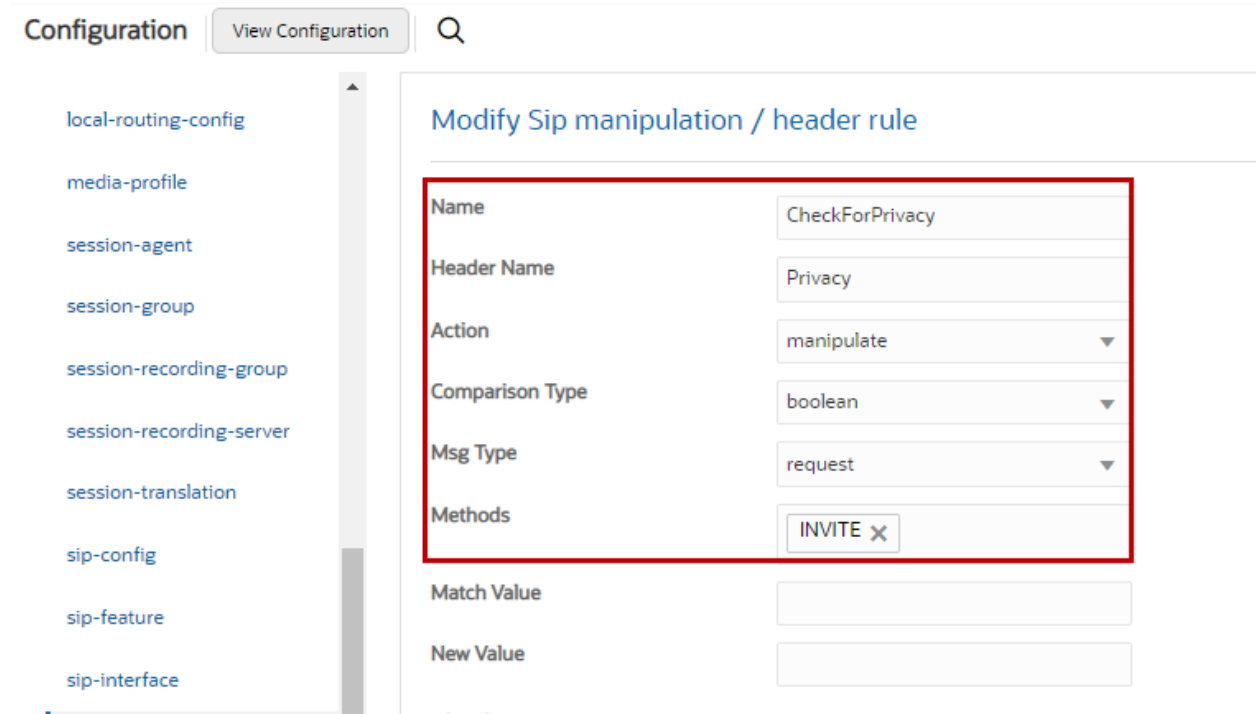
The screenshot shows the configuration for a SIP manipulation rule. The left sidebar lists various configuration categories, with 'sip-manipulation' selected. The main panel is titled 'Modify Sip manipulation / header rule'. The configuration fields are as follows:

Name	addPAI
Header Name	P-Asserted-Identity
Action	add
Comparison Type	case-sensitive
Msg Type	request
Methods	INVITE
Match Value	
New Value	\$storepai.\$1+"1"+\$storepai.\$2+"@sbc3.tekvizionlabs.com">

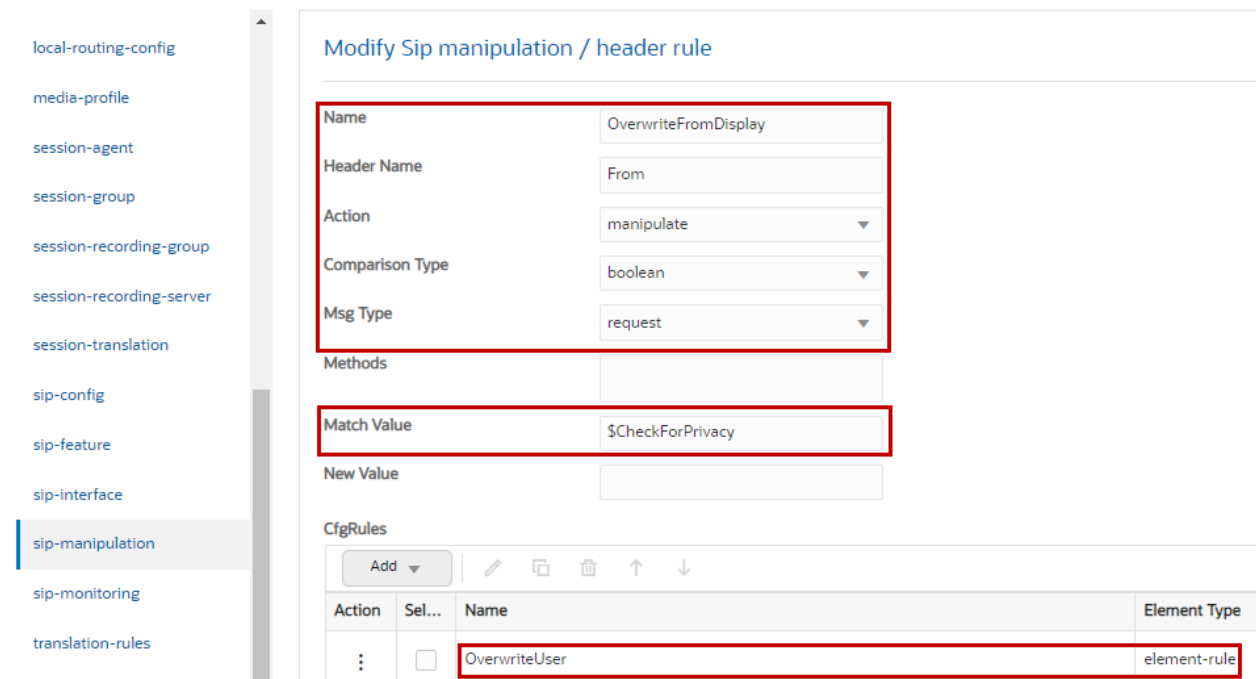
**Figure 83: SIP Manipulation towards Google Voice to add P-Asserted-Identity with E.164 format**



Below header rule is created to check the Privacy header from PSTN Gateway and overwrite the URI-USER part of FROM header with “anonymous” for anonymous calls.



**Figure 84: SIP Manipulation towards Google Voice to check for Privacy header**



**Figure 85: SIP Manipulation towards Google Voice to overwrite User part of FROM header to “anonymous”**

session-recording-server

session-translation

sip-config

sip-feature

sip-interface

**sip-manipulation**

sip-monitoring

translation-rules

system

fraud-protection

---

### Modify Sip manipulation / header rule / element rule

Name	OverwriteUser
Parameter Name	
Type	uri-user
Action	replace
Match Val Type	any
Comparison Type	case-sensitive
Match Value	
New Value	anonymous

**Figure 86: SIP Manipulation towards Google Voice to overwrite User part of FROM header to “anonymous” Cont.**



## SIP manipulation towards PSTN Gateway

The screenshot shows the 'Modify SIP Manipulation' configuration page. The left sidebar contains a menu with the following items: *sip-feature-caps*, *sip-interface*, *sip-manipulation* (selected), *sip-monitoring*, *sip-nat*, *sip-profile*, *sip-q850-map*, and *sip-recursion-policy*. The main content area is titled 'Modify SIP Manipulation' and contains the following fields:

Name	TowardsPSTN
Description	TowardsPSTN
Split Headers	
Join Headers	
CfgRules	

Figure 87: SIP Manipulation towards PSTN Gateway

Below header rule is created to change host part of FROM header with SBC IP.

The screenshot shows the 'Modify Sip manipulation / header rule' configuration page. The left sidebar contains a menu with the following items: *sip-feature-caps*, *sip-interface*, *sip-manipulation* (selected), *sip-monitoring*, *sip-nat*, *sip-profile*, *sip-q850-map*, *sip-recursion-policy*, *surrogate-agent*, and *survivability*. The main content area is titled 'Modify Sip manipulation / header rule' and contains the following fields:

Name	changeFromIP
Header Name	From
Action	manipulate
Comparison Type	case-sensitive
Msg Type	request
Methods	INVITE X
Match Value	
New Value	
CfgRules	

Figure 88: SIP Manipulation towards PSTN Gateway to change FROM header with SBC IP

session-recording-server  
 session-translation  
 sip-config  
 sip-feature  
 sip-interface  
 sip-manipulation  
 sip-monitoring  
 translation-rules  
 system  
 fraud-protection

### Modify Sip manipulation / header rule / element rule

Name	changeFromIP
Parameter Name	
Type	uri-host
Action	replace
Match Val Type	any
Comparison Type	case-sensitive
Match Value	
New Value	\$LOCAL_IP

**Figure 89: SIP Manipulation towards PSTN Gateway to change FROM header with SBC IP Cont.**

Below header rule is created to change host part of TO header with PSTN Gateway IP.

sip-feature-caps  
 sip-interface  
 sip-manipulation  
 sip-monitoring  
 sip-nat  
 sip-profile  
 sip-q850-map  
 sip-recursion-policy  
 surrogate-agent

### Modify Sip manipulation / header rule

Name	changeToIP
Header Name	To
Action	manipulate
Comparison Type	case-sensitive
Msg Type	request
Methods	INVITE
Match Value	
New Value	

**Figure 90: SIP Manipulation towards PSTN Gateway to change TO header**

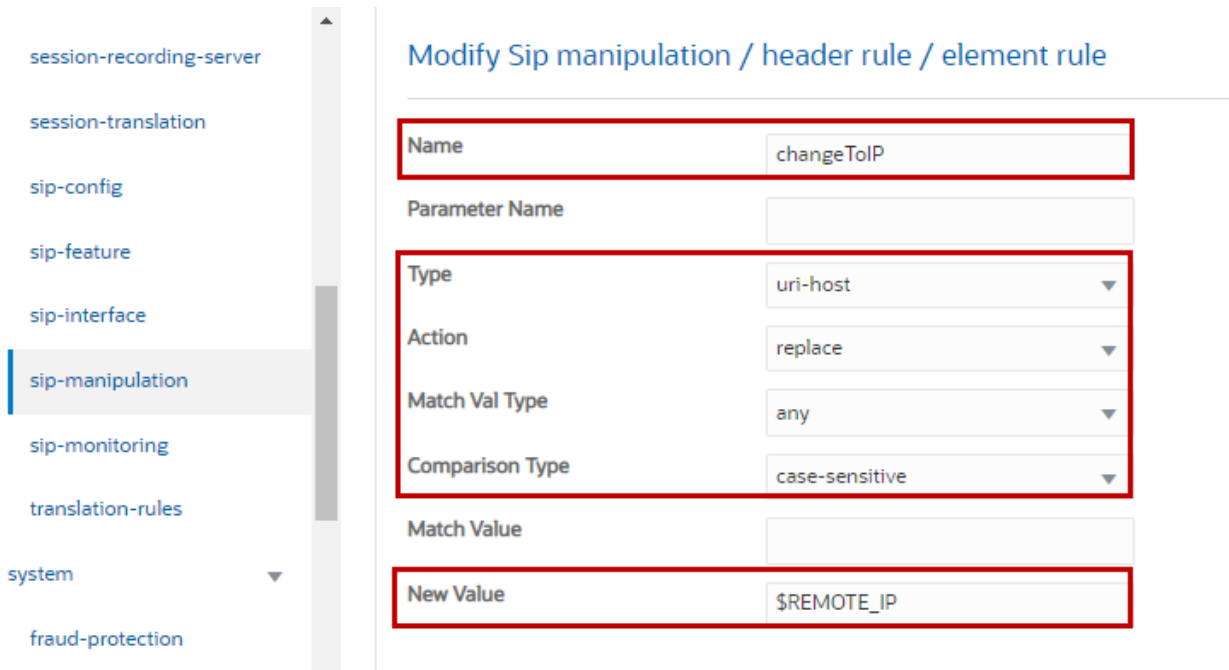


Figure 91: SIP Manipulation towards PSTN Gateway to change TO header Cont.

#### 6.4.18 Redundancy Configuration

In addition to the above configurations, the below configurations are done for redundancy.

#### HA Topology Diagram

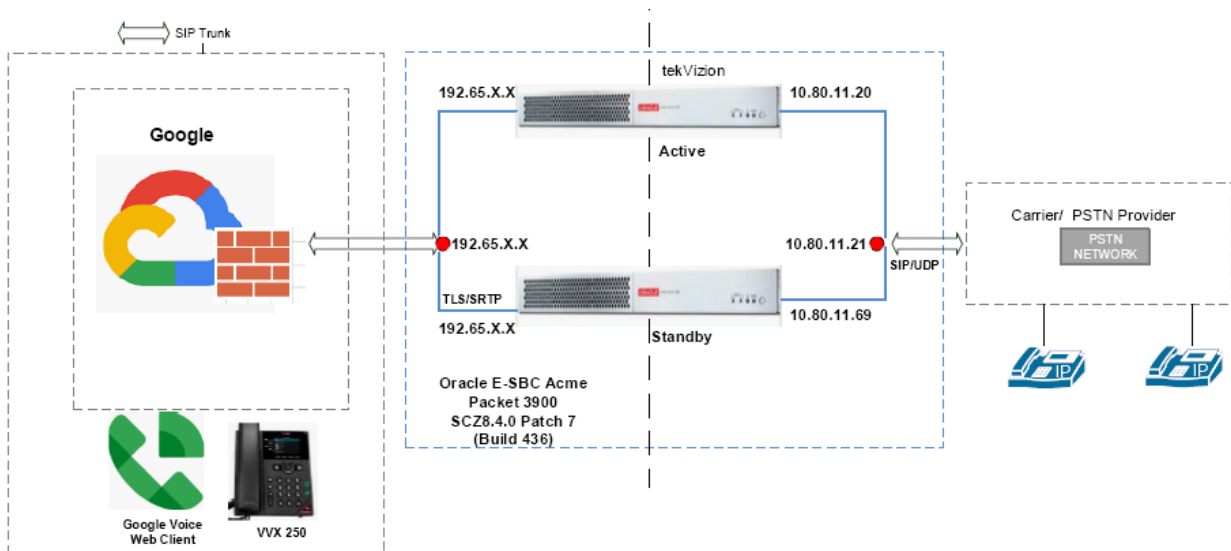


Figure 92: Topology Diagram – HA Pair



## Create Physical Interface

Navigate to **System > phy-interface**. Wancom1 interface is configured for High Availability (HA).

The screenshot shows the 'Modify Phy Interface' configuration page for the 'wancom1' interface. The left sidebar contains a navigation menu with 'system' and 'phy-interface' highlighted with red boxes. The main configuration area includes the following fields:

Name	wancom1
Operation Type	Control
Port	1 (Range: 0..5)
Slot	0 (Range: 0..2)
Virtual Mac	
Admin State	<input checked="" type="checkbox"/> enable
Auto Negotiation	<input checked="" type="checkbox"/> enable
Duplex Mode	
Speed	
Wancom Health Score	8 (Range: 0..100)

**Figure 93: Create phy-interface – Wancom1**

## Modify Network Interface

Navigate to **System > network-interface**. Primary and Secondary utility IP addresses of Oracle SBC 1 and SBC 2 for Wancom1, SOP0 and S1P0 are configured as shown below.

The image shows three stacked screenshots of the 'Modify Network Interface' configuration page for the interface 'wancom1'. The left sidebar shows the navigation menu with 'system' and 'network-interface' highlighted. The main content area is divided into sections for basic interface configuration, Gw Heartbeat, and DNS settings.

Field	Value	Range
Name	wancom1	
Sub Port Id	0	( Range: 0..4095 )
Description		
Hostname		
IP Address		
Pri Utility Addr	10.80.11.82	
Sec Utility Addr	10.80.11.83	
Netmask	255.255.255.0	
Gateway		
State	<input type="checkbox"/> enable	
Heartbeat	0	( Range: 0..65535 )
Retry Count	0	( Range: 0..65535 )
Retry Timeout	1	( Range: 1..65535 )
Health Score	0	( Range: 0..100 )
DNS IP Primary		
DNS IP Backup1		
DNS IP Backup2		
DNS Domain		
DNS Timeout	11	( Range: 0..4294967295 )
DNS Max Ttl	86400	( Range: 30..2073600 )
Signaling Mtu	0	( Range: 0,576..4096 )
HIP IP List		
ICMP Address		
SSH Address		

**Figure 94: Modify Network-Interface for Wancom1**

system

- fraud-protection
- host-route
- http-client
- http-server
- network-interface
- ntp-config
- phy-interface
- redundancy-config
- snmp-community
- spl-config

### Modify Network Interface

Name	s0p0	
Sub Port Id	0	( Range: 0..4095 )
Description		
Hostname	sbc3.tekvizionlabs.com	
IP Address	192.65.128.1	
Pri Utility Addr	192.65.128.2	
Sec Utility Addr	192.65.128.3	
Netmask	255.255.255.128	

**Figure 95: Modify Network-Interface of S0P0**

system

- fraud-protection
- host-route
- http-client
- http-server
- network-interface
- ntp-config
- phy-interface
- redundancy-config
- snmp-community
- spl-config

### Modify Network Interface

Name	s1p0	
Sub Port Id	0	( Range: 0..4095 )
Description		
Hostname		
IP Address	10.80.11.21	
Pri Utility Addr	10.80.11.20	
Sec Utility Addr	10.80.11.69	
Netmask	255.255.255.0	

**Figure 96: Modify Network-Interface of S1P0**

## Create Redundancy Config

The Primary and Secondary SBC's are configured as shown below. The IP address used here are the addresses of Wancom1 assigned to both SBC.

Navigate to **System > redundancy-config** to configure the peers.

Configuration View Configuration Q

system

fraud-protection

host-route

http-client

http-server

network-interface

ntp-config

phy-interface

**redundancy-config**

snmp-community

spl-config

system-config

### Modify Redundancy Config

State  enable

Log Level INFO

Becoming Standby Time 180000 (Range: 5..2147483647)

Becoming Active Time 100 (Range: 5..2147483647)

Media If Peercheck Time 0 (Range: 0..500)

Peers

Action	Sel...	Name	State	Type
:	<input type="checkbox"/>	sbc01	enabled	Primary
:	<input type="checkbox"/>	sbc02	enabled	Secondary

Figure 97: Create Redundancy configuration for SBC1 and SBC2

### Modify Redundancy config / peer

Name sbc01

State  enable

Type Primary

Destinations

Action	Sel...	Address	Network Interface
:	<input type="checkbox"/>	10.80.11.82:9090	wancom1:0

Figure 98: Configure SBC1 as Primary



## Modify Redundancy config / peer

Name

State  enable

Type

Destinations

Action	Sel...	Address	Network Interface
:	<input type="checkbox"/>	10.80.11.83:9090	wancom1:0

**Figure 99: Configure SBC2 as Secondary**

The CLI command **acquire-config** is used to acquire configuration from the Primary SBC. This is executed from Secondary SBC.

### 6.4.19 Oracle SBC deployed Behind NAT

The Support for SBC Behind NAT SPL plug-in changes information in SIP messages to hide the end point located inside the Private network.

The specific information that the Support for SBC Behind NAT SPL plug-in changes depends on the direction of the call, for example, from the NAT device to the SBC or from the SBC to the NAT device.

Configure the Support for SBC Behind NAT SPL plug-in for each SIP interface that is connected to a NAT device. One Public-Private address pair is required for each SIP interface that uses the SPL plug-in, as follows.

- The Private IP address must be the same IP as configured on both the SIP Interface and Steering Pool.
- The Public IP address must be the Public IP address of the NAT device.

Here is an example configuration with SBC Behind NAT SPL config.

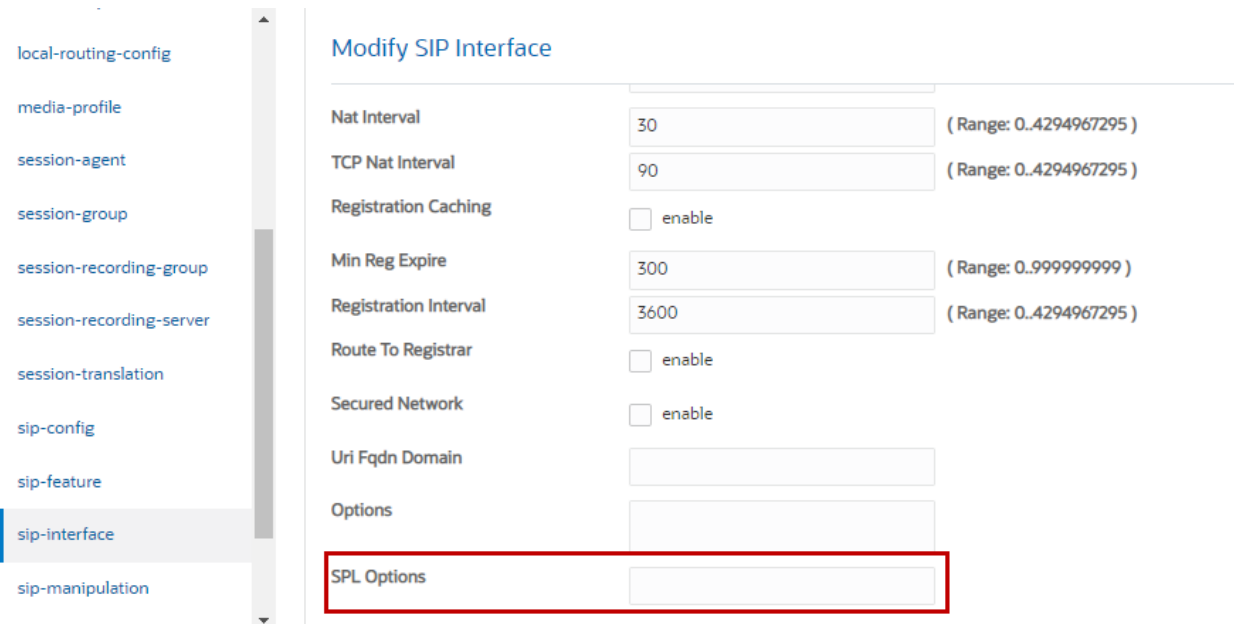
The SPL is applied to the Google Voice SIP Link side SIP interface.

Navigate to session-router > sip-interface.

Navigate via ACLI: config t > session-router > sip-interface.

HeaderNatPublicSipIfIp is the Public Interface IP.

HeaderNatPrivateSipIfIp is the Private IP.



**Figure 100: Oracle SBC Behind NAT – SPL Options configuration**

SPL Options is set to: HeaderNatPublicSipIfIp= <Public IP of Google Voice SIP Link Interface>, HeaderNatPrivateSipIfIp = <Private IP of Google Voice SIP Link Interface>

The SPL Options needs to be applied to every SIP Interface on the SBC that is connected through a NAT.

## 7 Oracle E-SBC Running configuration

---

Attached the Oracle E-SBC running configuration.



Orade\_RunningCo  
nfig.txt

## 8 Summary of Tests and Results

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
<b>Inbound</b>					
1.0	BYOT Phone Number > Auto Attendant > Press DTMF	Place a call from External network to a BYOT phone number assigned to an Auto Attendant. Navigate the Voice User Interface by entering DTMF input.	Should be able to hear the Auto Attendant and DTMF navigation should be successful.	PASSED	
1.2	BYOT Phone Number > User Termination > Web Client	Place a call from External network to a BYOT phone number assigned to an Google Voice WebClient. Ensure 2 way voice.	The Web client should ring and upon answering the call 2 way audio conversation should be successful.	PASSED	
1.3	BYOT Phone Number > User Termination > Android Client/iOS Client	Place a call from External network to a BYOT phone number assigned to an Google Voice Android Client. Ensure 2 way voice.	The mobile client should ring and upon answering the call 2 way audio conversation should be successful.	PASSED	
1.4	BYOT Phone Number > User Termination > Deskphone	Place a call from External network to a BYOT phone number assigned to an Google Voice Deskphone. Ensure 2 way voice.	The deskphone should ring and upon answering the call 2 way audio conversation should be successful.	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
1.5	PBX Extension Number > PBX User > BYOT Phone Number > User Termination	Place a call from External network to a PBX phone number assigned to an PBX user. PBX user then transfers the call to BYOT phone number for User Termination. Ensure 2 way voice.	The GV client should ring and upon answering the call 2 way audio conversation should be successful.	PASSED	
1.6	Long Call Duration: BYOT Phone Number > User Termination	Place a call from External network to a BYOT phone number assigned to an Google Voice WebClient leave the call up for a long duration greater than 30 mins. Ensure 2 way voice.	The client endpoint should ring and upon answering the call 2 way audio conversation should be successful for greater than 30mins. Audio should not drop or get cutout	PASSED	
1.7	BYOT Phone Number > Auto Attendant > User Termination	Place a call from External network to a BYOT phone number assigned to an Google Voice Auto Attendant. Press DTMF <> to transfer call to Google Voice User for termination. Ensure 2 way voice.	Tester should be able to navigate the Auto Attendant and select the DTMF option for endpoint (user). The client endpoint should ring and upon answering the call 2 way audio	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
			conversati on should be successful.		
1.8	BYOT Phone Number > Auto Attendant > User Termination > Voicemail	Place a call from External network to a BYOT phone number assigned to an Google Voice Auto Attendant. Press DTMF <> to transfer call to Google Voice User for termination. Let the call go to Voicemail and leave a message.	Tester should be able to navigate the Auto Attendant and select the DTMF option for endpoint (user). The client endpoint should let the call pass to voicemail and tester should leave a VM. Tester should be able to retrieve the message	PASSED	
1.9	BYOT Phone Number > Auto Attendant > PBX Phone Number > User Termination > PBX User	Place a call from External network to a BYOT phone number assigned to an Google Voice Auto Attendant. Press DTMF <> to transfer call to PBX User for termination. Ensure 2 way voice.	The PBX Extension should be able to answer the call and 2 way conversati on should be possible. The PBX extension should be	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
			able to perform functions like Hold Transfer Conference		
1.10	BYOT Phone Number > Auto Attendant > PBX Phone Number > User Termination > PBX User > Voicemail	Place a call from External network to a BYOT phone number assigned to an Google Voice Auto Attendant. Press DTMF <> to transfer call to PBX User for termination. Let the call go to Voicemail and leave a message.	The PBX voicemail system should answer and the GV user should be able to successfully leave a voicemail as well as able to exercise the various dtmf options in the voicemail menu	PASSED	
1.11	PBX Phone Number > PBX User > transfer to BYOT Phone Number > Auto Attendant > User Termination)	Place a call from External network to a PBX phone number assigned to an PBX User. Answer & transfer call to Google Voice Auto Attendant for termination. Navigate the interface with DTMF. Press option to terminate to a user and ensure 2 way voice.	The GV BYOT User should be able to answer the call and 2 way conversation should be possible. The GV BYOT User should be able to perform functions like	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
			Hold Transfer Conference		
1.12	PBX Phone Number > PBX User > transfer to BYOT Phone Number > Auto Attendant > User Termination > Voicemail	Place a call from External network to a PBX phone number assigned to an PBX User. Answer & transfer call to Google Voice Auto Attendant for termination. Navigate the interface with DTMF. Press option to terminate to a user and let the call go to Voicemail	The GV voicemail system should answer and the PBX user should be able to successfully leave a voicemail as well as able to exercise the various dtmf options in the voicemail menu	PASSED	
1.13	BYOT Phone Number > Ring Group > User Termination > Voicemail	Place a call from External network to a BYOT phone number assigned to an Google Voice RingGroup. Call should ring on a Google Voice User for termination. Let the call go to Voicemail. Leave a message.	The client endpoint should let the call pass to voicemail and tester should leave a VM. Tester should be able to retrieve the message	PASSED	
1.14	PBX Phone Number > PBX User > transfer to BYOT Phone Number > Ring Group >	Place a call from External network to a PBX phone number assigned to an PBX User. Transfer call to the BYOT Ring Group and	The GV BYOT User should be able to answer the call and 2 way	PASSED	



ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
	User Termination	should terminate on a Google Voice user. Ensure 2 way voice.	conversations should be possible. The GV BYOT User should be able to perform functions like Hold Transfer Conference		
1.15	PBX Phone Number > PBX User > transfer to BYOT Phone Number > Ring Group > User Termination > Voicemail	Place a call from External network to a PBX phone number assigned to an PBX User. Transfer call to the BYOT Ring Group and should terminate on a Google Voice user. Let the call go to Voicemail. Leave a message.	The GV voicemail system should answer and the PBX user should be able to successfully leave a voicemail as well as able to exercise the various dtmf options in the voicemail menu	PASSED	
1.17	BYOT Phone Number > Auto Attendant > Ring Group > User Termination	Place a call from External network to a BYOT phone number assigned to an Google Voice Auto Attendant. Press DTMF <> to transfer call to Google Voice Ring Group which will	Tester should be able to navigate the Auto Attendant and select the DTMF option for endpoint (user). The	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		<p>terminate call on a GV user Client. Ensure 2 way voice.</p>	<p>client endpoint should ring and upon answering the call 2 way audio conversation should be successful.</p>		
1.18	<p>BYOT Phone Number &gt; Auto Attendant &gt; Ring Group &gt; User Termination &gt; Voicemail</p>	<p>Place a call from External network to a BYOT phone number assigned to an Google Voice Auto Attendant. Press DTMF &lt;&gt; to transfer call to Google Voice Ring Group which will terminate call on a GV user Client. Let the call go to Voicemail and leave a message.</p>	<p>Tester should be able to navigate the Auto Attendant and select the DTMF option for endpoint (user). The client endpoint should let the call pass to voicemail and tester should leave a VM. Tester should be able to retrieve the message</p>	PASSED	
1.19	<p>PBX Phone Number &gt; PBX User &gt; transfer to BYOT Phone Number &gt; Auto Attendant &gt; Ring Group &gt;</p>	<p>Place a call from External network to a PBX phone number assigned to an PBX User. PBX User will transfer the call to a Google Voice Auto Attendant. Press DTMF &lt;&gt; to transfer call to</p>	<p>The GV BYOT User should be able to answer the call and 2 way conversation should be</p>	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
	User Termination	Google Voice Ring Group which will terminate call on a GV user Client. Ensure 2 way voice.	possible. The GV BYOT User should be able to perform functions like Hold Transfer Conference		
1.20	PBX Phone Number > PBX User > transfer to BYOT Phone Number > Auto Attendant > Ring Group > User Termination >	Place a call from External network to a PBX phone number assigned to an PBX User. PBX User will transfer the call to a Google Voice Auto Attendant. Press DTMF <> to transfer call to Google Voice Ring Group which will terminate call on a GV user Client. Let the call go to Voicemail and leave a message.	The GV voicemail system should answer and the PBX user should be able to successfully leave a voicemail as well as able to exercise the various dtmf options in the voicemail menu	PASSED	
<b>Outbound</b>					
1.21	User Originated > BYOT Trunk Termination	GV endpoint user will place a call to External no. <> via BYOT trunk	The call should be successful with 2 way audio.	PASSED	
1.22	Short Code call > BYOT Trunk Termination	GV user to dial a short code to test outbound call via short codes via BYOT trunk	The call should be successful with 2 way audio.	NOT SUPPORTED	Short code dialing is not supported by Google Voice. Google updated that this feature is

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
					under work in progress.
1.23	Long call: User Originated > BYOT Trunk Termination	GV user will place a call to External no. <> via BYOT trunk. ensure call is up for greater and 30mins	The call should be successful with 2 way audio. Ensure that the call does not drop and is terminated only when either GV endpoint or External no. disconnects the call.	PASSED	
1.24	GV User Originated > BYOT Trunk Termination > Destination Voicemail	GV user will place a call to External no. <> via BYOT trunk. Let call go to Voicemail and leave a message	The call should be successful with a successful voicemail to the external phone number	PASSED	
1.25	GV User Originated > BYOT Trunk Termination > Destination Auto Attendant	GV user will place a call to External Auto Attendant <> via BYOT trunk. Navigate the Menu via DTMF and ensure response.	The call should be successful. DTMF input should be accepted.	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
1.26	GV User Originated > BYOT Trunk Termination > PBX Auto Attendant	GV user will place a call to PBX Auto Attendant <> via BYOT trunk. Navigate the Menu via DTMF and ensure response.	The GV User should be able to navigate the PBX auto attendant via DTMF and should also be able to hear the prompts clearly.	PASSED	
1.27	GV User Originated > BYOT Trunk Termination > PBX User	GV user will place a call to PBX User<> via BYOT trunk. Ensure 2 way audio.	The GV User should be able to converse with the PBX user.	PASSED	
1.28	GV User Originated > BYOT Trunk Termination > PBX User > Voicemail	GV user will place a call to PBX User<> via BYOT trunk. Leave a voicemail.	The GV User should be able to leave a voicemail with the PBX voicemail and navigate the IVR of the Voicemail.	PASSED	
<b>SIP Options</b>					

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
2.1	SIP OPTIONS	SBC send SIP options every 60 seconds	Verify SBC sends SIP OPTIONS every 60 seconds and responded with 200 OK	PASSED	
<b>Inbound Call – Calling or Called Party Disconnects</b>					
2.2	Inbound call: Calling Party disconnects the call.	PSTN Calls GV user A GV user A answers the call. Allow the call to be connected more than 32 seconds Calling party disconnects the call	Verify Call is established with bidirectional audio Verify call is disconnected properly	PASSED	
2.3	Inbound call: Called Party disconnects the call.	PSTN Calls GV user A GV user A answers the call. Allow the call to be connected more than 32 seconds Called party disconnects the call	Verify Call is established with bidirectional audio Verify call is disconnected properly	PASSED	
<b>Inbound Call – Calling or Called Party Disconnects</b>					
2.4	Outbound call: Calling Party disconnects the call.	GV user A Calls PSTN A PSTN A answers the call. Allow the call to be connected more than 32 seconds Calling party disconnects the call	Verify Call is established with bidirectional audio Verify call is disconnected properly	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
2.5	Outbound call: Called Party disconnects the call.	GV user A Calls PSTN A PSTN A answers the call. Allow the call to be connected more than 32 seconds Called party disconnects the call	Verify Call is established with bidirectional audio Verify call is disconnected properly	PASSED	
<b>Terminate the call before answer</b>					
2.6	Terminate the call before answer	GV user A Calls PSTN A. GV user A hangs up before answer	Verify Call Cancelling is handled properly Verify CANCEL 200 OK 487 ACK	FAILED	When GV user hangs up the call, CANCEL message is not sent. Google updated this is a known issue to be fixed.
<b>Caller ID Restriction</b>					
2.7	Outbound Caller ID Restricted	GV user A Calls PSTN A PSTN A answers the call. Allow the call to be connected more than 32 seconds Calling party disconnects the call	Verify SIP header Privacy:id Verify caller ID appears as restricted in PSTN	PASSED	
2.8	Receiving anonymous inbound calls	PSTN A with restricted Caller ID calls GV user A GV user A answers the call Bidirectional Audio is present GV user A hangs up the call	Caller ID received on PBX user as anonymous	PASSED	
<b>Early Media</b>					

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
2.9	Handling Early media	GV user A Calls PSTN A with Early media. GV user A hangs up before answer	Verify early media is heard on GV user A Verify CANCEL 200 OK 487 ACK	FAILED	When GV user hangs up the call, CANCEL message is not sent. Google updated that this is a known issue to be fixed.
<b>Long duration</b>					
2.10	Long duration call - Inbound call- 1 hour max	PSTN Calls GV user A GV user A answers the call. Allow the call to be connected for 60 mins	Call is connected and bidirectional audio for more than 60 mins  Bidirectional audio is present after session audit or session refresh invite from the DUT.  Call is properly disconnected when either party disconnects the call	PASSED	
2.11	Long duration call- Outbound Call- 1 hour max	GV user A Calls PSTN A PSTN A answers the call. Allow the call to be connected for 60 mins	Call is connected and bidirectional audio for more than 60 mins  Bidirectional audio is	PASSED	



ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
			<p>present after session audit or session refresh invite from the DUT.</p> <p>Call is properly disconnected when either party disconnects the call</p>		
2.12	Long duration hold and resume (wait until session audit\session refresh occurs from DUT)	<p>PSTN Calls GV user A. GV user A answer the call</p> <p>GV user A places the call on hold for 30 mins or until session audit occurs. GV user A resumes the call.</p> <p>Bidirectional Audio is present after resuming the call</p>	<p>Call is connected and bidirectional audio before hold. MOH is heard during hold. If applicable.</p> <p>MOH is present after session audit that occurred during hold.</p> <p>Bidirectional audio is present after resume. Call is properly disconnected when</p>	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
			either party disconnects the call.		
2.13	Long duration hold and resume ( wait until session audit/session refresh occurs from DUT)	GV user A Calls PSTN A. PSTN A answers the call GV user A places the call on hold for 30 mins or until session audit occurs. Extension A resumes the call. Bidirectional Audio is present after resuming the call	Call is connected and bidirectional audio before hold. MOH is heard during hold. If applicable.  MOH is present after session audit that occurred during hold.  Bidirectional audio is present after resume. Call is properly disconnected when either party disconnects the call	PASSED	
<b>Terminate during Hold</b>					

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
2.14	Terminate the call during hold	PSTN Calls GV user A. GV user A answers the call GV user A places the call on hold GV user A hangs up during hold	Call is properly disconnected when either party disconnects the call	PASSED	
<b>Do Not Disturb</b>					
2.15	User is on DND	PSTN A calls GV user A PBX sends SIP Error response code for DND	Verify SIP error response is handled properly	PASSED	
<b>Simultaneous Ring</b>					
2.16	Simultaneous ring	PSTN A calls GV user A GV user sets simultaneous ring to PSTN B	PSTN B answers the call Bidirectional Audio is present	NOT SUPPORTED	Google Voice does not support Simultaneous ring.
<b>Call Forward</b>					
2.17	Call Forward	PSTN A calls GV user A GV user forwards the call to another GV User B	Verify caller ID of PSTN A is displayed properly on GV User B	NOT SUPPORTED	Google Voice does not support Call Forward. Google voice client does not have option to select GV user to forward the call.
2.18	Call forward	PSTN A calls GV user A GV user forwards the call to another PSTN number	Verify caller ID of PSTNA is displayed properly on the PSTN phone Verify Diversion header is sent	NOT SUPPORTED	Google Voice does not support Call Forward. When GV user forwards the call to PSTN 2, The forwarded call to PSTN2 always routes via Google Voice trunk and

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
					not via Oracle E-SBC.
<b>Toll Free</b>					
2.19	Calls to 800 numbers (Toll Free Numbers)	GV user A Calls TOLL FREE number starts with 800	Verify Call is established with bidirectional audio Verify call is disconnected properly	PASSED	
2.20	Calls to 877 numbers (Toll Free Numbers)	GV user A Calls TOLL FREE number starts with 877	Verify Call is established with bidirectional audio Verify call is disconnected properly	PASSED	
2.21	Calls to 866 numbers (Toll Free Numbers)	GV user A Calls TOLL FREE number starts with 866	Verify Call is established with bidirectional audio Verify call is disconnected properly	PASSED	
2.22	Calls to 888 numbers (Toll Free Numbers)	GV user A Calls TOLL FREE number starts with 888	Verify Call is established with bidirectional audio Verify call is disconnected	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
			ed properly		
2.23	Call to 5551212	GV user A Calls 5551212	Verify Call is established with bidirectional audio Verify call is disconnected properly	NOT SUPPORTED	Google Voice does not support short-code dialing. Google updated that this feature is under work in progress.
2.24	Calls to 0 operator assistance.	GV user A Calls 0 Operator answers the call	Verify Call is established with bidirectional audio Verify call is disconnected properly	NOT SUPPORTED	Google Voice does not support Operator code dialing. Google updated that this feature is under work in progress.
<b>Handling E164 and Non E164</b>					
2.25	Handling E164 and non E164 format	GV user A Calls PSTN A with E164 format PSTN A answer the call. Allow the call to be connected more than 32 seconds Called party disconnects the call Repeat the same with non E164 format	Verify Call is established with bidirectional audio Verify call is disconnected properly	NOT SUPPORTED	Google Voice does not support Non E164. Google updated that this feature is under work in progress.
<b>International Dialing</b>					

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
2.26	International dialing	GV user A Calls international number PSTN A answer the call. Allow the call to be connected more than 32 seconds Called party disconnects the call	Verify Call is established with bidirectional audio Verify call is disconnected properly	PASSED	
<b>Caller ID</b>					
2.27	Verifying caller ID	GV user A Calls PSTN A PSTN A answer the call. Allow the call to be connected more than 32 seconds Calling party disconnects the call	Verify Caller ID is displayed properly on the PSTN phone Verify sip header Privacy: none	PASSED	
<b>Call Waiting</b>					
2.28	Call waiting	GV user A Calls PSTN A PSTN A answer the call. Allow the call to be connected PSTN B makes an incoming call to GV user A, GV user A receives call waiting indication GV user A answers PSTN B PSTN A Placed on hold and hears MOH PSTN B hangs up the call GV user A resumes PSTN A.  GV user A hangs up the call.	Verify Call is established with bidirectional audio Verify Call waiting indication Verify MOH on PSTNA during hold Verify call is disconnected properly	NOT SUPPORTED	When a second call is made from PSTN B to GV User A, GV user does not see call waiting notification. PSTN B hears ringback. Google updated that this is a known behavior as per design. The behavior is same for iOS and Android phones.

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
<b>Handling 486 response</b>					
2.29	Handling 486 response	PSTN A calls a Busy Extension	Verify Busy tone is heard Verify 486 busy in signaling if present.	NOT SUPPORTED	This is same as Call waiting. Google updated that this is a known behavior as per design. The behavior is same for iOS and Android phones.
<b>Handling Error Codes, 486, 4XX</b>					
2.30	Handling Error Codes, 486, 4XX	GV user A calls a Busy PSTN Number	Verify Busy tone is heard Verify 486 busy in Signaling if present.	PASSED	
<b>Handling Error Codes 603 Decline</b>					
2.31	Handling Error codes 603 decline	PSTN Calls GV user A, GV user A rejects the incoming call	Verify DUT handle calls rejection	FAILED	When GV user rejects the incoming call, CANCEL message is not sent from GV user. Google updated that this is a known issue to be fixed.
<b>Codec</b>					
2.32	GV user A calls PSTN A using G711 U law Codec	GV user A Calls PSTN A PSTN A answers the call. Allow the call to be connected more than 32 seconds Calling party disconnects the call	Verify Call is established with bidirectional audio and G711 U law is negotiated Verify call is disconnect	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
			ed properly		
2.33	GV user A calls PSTN A using G711 A law Codec	GV user A Calls PSTN A PSTN A answers the call. Allow the call to be connected more than 32 seconds Calling party disconnects the call	Verify Call is established with bidirectional audio and G711 A law is negotiated Verify call is disconnected properly	PASSED	
<b>Calls to 911, 411 and 511</b>					
2.34	GV user Calls 911	GV user A Calls 911 911 operator answers the call.	Verify Call is connected and bidirectional audio present Verify Proper Caller ID displayed at operator end	NOT SUPPORTED	Google updated that 911 calls routes via Google Voice trunk and not via Oracle E-SBC. This feature is under work in progress.
2.35	GV user Calls 411	GV user A Calls 411 Call is connected and bidirectional audio is present	Verify Call is established with bidirectional audio Verify call is disconnected properly	NOT SUPPORTED	Google Voice does not support 3-digit dialing. Google updated that this feature is under work in progress.



ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
2.36	GV user Calls 511	GV user A Calls 511 Call is connected and bidirectional audio is present	Verify Call is established with bidirectional audio Verify call is disconnected properly	NOT SUPPORTED	Google Voice does not support 3-digit dialing. Google updated that this feature is under work in progress.
2.37	GV user Calls XXX (short code varies depend on region-outside of US)	GV user A Calls XXX Call is connected and bidirectional audio is present	Verify Call is established with bidirectional audio Verify call is disconnected properly	NOT TESTED	Test case is already executed with other short codes e.g. 411, 511 since US carrier is used for this setup.
<b>Stir-Shaken</b>					
2.38	Stir-Shaken: inbound call-SPAM Number	PSTN Calls GV user A, GV user A rejects the incoming call	Verify Call is rejected by the SBC	NOT TESTED	Stir-Shaken is not supported by the Service Provider used in this setup.
2.39	Stir-Shaken: inbound call-Anonymous inbound	PSTN Calls GV user A, GV user A rejects the incoming call	Verify Call is rejected by the SBC	NOT TESTED	Stir-Shaken is not supported by the Service Provider used in this setup.
2.40	Stir-Shaken: inbound call-Verified Number	PSTN Calls GV user A, GV user A answers the calls.	Verify identification header is present Verify bidirectional audio is present.	NOT TESTED	Stir-Shaken is not supported by the Service Provider used in this setup.
<b>Looped Calls</b>					
2.41	Looped Calls	BYOT Phone Number > Auto Attendant > BYOT User Termination 1 transfer	Ensure calls that may get looped	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		to PBX Auto Attendant > PBX Auto Attendant > BYOT Ring Group > BYOT User Termination 2	with BYOT and any customer PBX are not impacted		
2.42	Codec	BYOT Phone Number> Auto Attendant (mulaw) > BYOT User Termination Web Client (opus) > Transfer to BYOT Deskphone (G.722) > Outbound call to PBX (mulaw)	Media should connect and call must progress with 2 way audio	PASSED	
2.43	Third party IVR	3rd Party IVR calling BYOT Phone Number >User Termination > Press DTMF	Should be able to hear the external IVR and DTMF navigation should be successful. Ensure Telephony -events being passed for DTMF interaction . Test DTMF from a called party perspective.	PASSED	
2.44	Glare Condition	GV user A Calls PSTN A. PSTN A answers the call GV user A and PSTN A places the call on hold at the same time. Bidirectional Audio is present after	Call should connect successfully with bidirectional media before and after the	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		resuming the call Creating a glare condition to occur over the BYOT trunk	hold events. Check Re-Invites.		
2.45	HA	Unplug the LAN side cable of Primary SBC	Unplug the LAN side cable of Primary SBC, verify both incoming and outgoing calls work through secondary SBC	PASSED	
2.46	HA	Plug the LAN cable back for Primary SBC, verify the incoming/outgoing call going through the Secondary SBC. (Note: Can also be executed by shutting/unshutting the interfaces)	Plug the LAN cable back for Primary SBC, verify the incoming/outgoing call going through the Secondary SBC	PASSED	
2.47	HA	Unplug the WAN side cable of Primary SBC, verify both incoming and outgoing calls work through secondary SBC (Note: Can also be executed by shutting/unshutting the interfaces)	Unplug the WAN side cable of Primary SBC, verify both incoming and outgoing calls work through secondary SBC	PASSED	
2.48	HA	Plug the WAN cable back for Primary SBC, verify the incoming/outgoing	Plug the WAN cable back for Primary	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		call going through the Secondary SBC (Note: Can also be executed by shutting/unshutting the interfaces)	SBC, verify the incoming/ outgoing call going through the Secondary SBC		
2.49	HA	Shutdown Primary SBC and verify the traffic goes through via secondary SBC.	Shutdown Primary SBC and verify the traffic go through via secondary SBC.	PASSED	
2.50	HA	Bring up the Primary SBC and verify the traffic is going through the secondary SBC	Bring up the Primary SBC and verify the traffic is going through the secondary SBC	PASSED	
2.51	HA	PSTN GV calls GV user A and the call is established. Switchover SBC.	Verify if the call is still established and no disconnect during failover	PASSED	
2.52	HA	PSTN GV calls GV user A and the call is established. Switchover SBC. Keep the call exists for more than 30 minutes and disconnect	Verify if the call is still established after long duration and no disconnect	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
			during failover.		