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



Table of Contents

1	Audie	nce	3
1	l.1 In	troduction	3
	1.1.1	tekVizion Labs	3
2	SIP Tr	unking Network Components	4
3	Hardv	vare Components	5
4	Softw	are Requirements	5
5	Featu	res	6
5	5.1 Fe	eatures Supported by Google Voice SIP Link	6
5	5.2 Fe	eatures Not Supported by Google Voice SIP Link	6
5	5.3 Fe	eatures Not Supported by Service Provider	6
5	5.4 C	aveats and Limitations	6
6	Confi	guration	7
e	5.1 C	onfiguration Checklist	7
e	5.2 IP	Address Worksheet	8
e	5.3 G	oogle Voice SIP Link Configuration	9
e	5.4 O	racle E-SBC Configuration	9
	6.4.1	Media Manager	9
	6.4.2	Physical Interface	10
	6.4.3	Network Interface	13
	0.4.4 4 4 E	Codec Policy Translation Bulan	10 17
	0.4.5	Section Translation	17
	64.0	Poolm Config	10
	619	Steering Pool	20
	6/0	SDES Profile	32
	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	Oracl	e E-SBC Running configuration	88
8	Summ	nary 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 helps 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.

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

Table 1 – Oracle E-SBC Configuration Steps

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**.

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

Table 3 - IP Address Worksheet

6.3 Google Voice SIP Link Configuration

Below link can be referred to configure Google Voice SIP Link. <u>support.google.com/a?p=siplink</u>

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.

0	•	· ·	J		
Configuration	Configuration	Q			
media-manager	•	Modify Media Manage	r		
codec-policy	_				
media-manager		State	🖌 enable		
media-policy		Flow Time Limit	86400		(Range: 04294967295)
		Initial Guard Timer	300		(Range: 04294967295)
realm-config		Subsq Guard Timer	300		(Range: 04294967295)
steering-pool		TCP Flow Time Limit	86400		(Range: 04294967295)
security	•	TCP Initial Guard Timer	300		(Range: 04294967295)
session-router	•	TCP Subsq Guard Timer	300		(Range: 04294967295)
system	•	Hnt Rtcp	enable		
		Algd Log Level	NOTICE	•	
		Mbcd Log Level	NOTICE		

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

Figure 2: Media Manager

Configuration	View Configuration	Q		
media-manager	•	Modify Media Manager		
codec-policy		Options	audio-allow-asymmetric-pt 🗙	
media-manager		Red May Trans		
media-policy		Red Max Hans	10000	(Range: 050000)
realm-config		Red Sync Start Time	5000	(Range: 04294967295)
steering pool		Red Sync Comp Time	1000	(Range: 04294967295)
steering-poor		Media Policing	✓ enable	
security	►	Max Arp Rate	10	(Range: 0100)
session-router	•	Max Signaling Packets	0	(Range: 04294967295)
system	►	Max Untrusted Signaling	1	(Range: 0100)
		Min Untrusted Signaling	1	(Range: 0100)
media-manager	•	Modify Media Manager		
codec-policy		Tolerance Window	30	(Range: 04294967295)
media-manager		Untrusted Drop Threshold	0	(Range: 0100)
media-policy		Trusted Drop Threshold	0	(Range: 0100)
realm-config		Acl Monitor Window	30	(Range: 53600)
Comp		Trap On Demote To Deny	enable	
steering-pool		Trap On Demote To Untrusted		
security	►	hap on benote to ontrasted	enable	
session-router	•	Syslog On Demote To Deny	enable	
custom		Syslog On Demote To Untrusted	enable	
system	F	Anonymous Sdp	enable	
		Reactive Transcoding	enable	
		Translate Non Rfc2833 Event	enable	
		Xcode Fax Max Rate	14400	v

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	1	Modify Phy Interface	e		
translation-rules	i	Name	s0p0		
system	•	Operation Type	Media	•	
fraud-protection		Port	0		(Range: 05)
host-route		Slot	0		(Range: 02)
http-client		Virtual Mac			
http-server		Admin State	enable		
network-interfac	e	Auto Negotiation	🖌 enable		
ntp-config		Duplex Mode	FULL	•	
phy-interface		Speed	100	•	
redundancy-con	fig	Wancom Health Score	50		(Range: 0100)

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	Modify Phy Interface		
sip-monitoring			•
translation-rules	Name	s1p0	
system 🔻	Operation Type	Media 🔻	
fraud-protection	Port	0	(Range: 05)
host-route	Slot	1	(Range: 02)
http-client	Virtual Mac		
http-server	Admin State	🖌 enable	
network-interface	Auto Negotiation	✓ enable	
ntp-config	Duplex Mode	FULL	
phy-interface	Speed	100 💌	
redundancy-config	Wancom Health Score	50	(Range: 0100)

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.

security session-router system fraud-protection host-route http-client http-server IP Address IP: Utility Addr phy-interface redundancy-config Gateway IP2.65	media-manager	•	Modify Network Interface			
session-router Name system Sub Port Id o (Range: 04095) fraud-protection host-route http-client http-server Inetwork-interface phy-interface phy-interface redundancy-config Sateway sopo sopo <td>security</td> <td>· •</td> <td></td> <td></td> <td></td>	security	· •				
system fraud-protection host-route http-client http-server network-interface phy-interface phy-interface redundancy-config Gateway Sub Port Id 0 0 (Range: 0.4095) (Range: 0.4095)	session-router	•	Name	s0p0	•	
fraud-protection host-route http-client http-server network-interface ntp-config phy-interface phy-interface ntp-config Sec Utility Addr Netmask 255.255.255.128	system	-	Sub Port Id	0	(Range: 04095)	
host-routeHostnamesbc3.tekvizionlabs.comhttp-clientHostnamesbc3.tekvizionlabs.comhttp-serverIP Address192.65.1network-interfacePri Utility Addrntp-configSec Utility Addrphy-interfaceNetmask255.255.128redundancy-configGateway192.65.1	fraud-protection	-	Description			
http-client http-server network-interface ntp-config phy-interface phy-interface Netmask 255.255.128	host-route					
http-server IP Address 192.65.1 network-interface Pri Utility Addr ntp-config Sec Utility Addr phy-interface Netmask redundancy-config Gateway	http-client		Hostname	sbc3.tekvizionlabs.com	1	
network-interface Pri Utility Addr ntp-config Sec Utility Addr phy-interface Netmask redundancy-config Gateway	http-server		IP Address	192.65.		
ntp-config Sec Utility Addr phy-interface Netmask redundancy-config Gateway	network-interface		Pri Utility Addr		-	
phy-interface Netmask redundancy-config Gateway	ntp-config		Sec Utility Addr			
redundancy-config Gateway	phy-interface		Netmask	255.255.255.128		
192.03.	redundancy-config		Gateway	192.65.		

Figure 6: Network Interface towards Google Voice

Note: If SBC is placed behind the NAT, please refer the NAT configuration under <u>Section 6.4.19</u>

Configuration	View Config	uration	Q				
sip-manipulation	1	•	Modify Network	Interface			
translation-rules			Gw Heartbeat				
system	•		State		🗸 enable		
fraud-protection			Heartbeat		10		(Range: 065535)
host-route			Retry Count		3		(Range: 065535)
http-client			Retry Timeout		3		(Range: 165535)
Lu			Health Score		0		(Range: 0100)
nπp-server			DNS IP Primary		8.8.8.8		
network-interfac	e		DNS IP Backup1				•
ntp-config			DNS IP Backup2				
phy-interface			DNS Domain		tekvizionlabs.con	n	
redundancy-conf	fig		DNS Timeout		11		(Range: 04294967295)
snmp-communit	y		DNS Max Ttl		86400		(Range: 302073600)
spl-config			Signaling Mtu		0		(Range: 0,5764096)
media-manager	•	Modi	fy Network Interface				
security	•	HIP IP L	ist				
session-router							
system	•	ICMP A	ddress				
fraud-protection		SSH Ad	dress				
host-route		Tunnel	Config				
http-client							
http-server						(i)	
network-interface					la turne d'anne f		
ntp-config				I	vo tunnel config	to display. Please ad	a or upioaa tunnel config.
phy-interface							



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

Configuration	View Configuration	Q			
sip-manipulation	n 🔺	Modify Network Interfac	ce		
translation-rules	5	Name	s1p0	•	
system	•	Sub Port Id	0		(Range: 04095)
fraud-protection	1	Description			
host-route					
http-client		Hostname			
http-server		IP Address	10.80.11.21		
network-interfac	ce 👘	Pri Utility Addr			
ntp-config		Sec Utility Addr			
phy-interface		Netmask	255.255.255.0		
redundancy-con	fig	Gateway	10.80.11.1		
snmp-communi	ty				•
media-manager	•	Modify Network Interface			
security	•	4 Gw Hoarthoat			
session-router	• •	State			
system	.	State	✓ enable		
fraud protection		Heartbeat	10	(Ran	ge: 065535)
hadd-protection		Retry Count	3	(Ran	ge: 065535)
host-route		Retry Timeout	3	(Ran	ge: 165535)
http-client		Health Score	0	(Ran	ge: 0100)
http-server		Bfd Config			
network-interface	e	DNS IP Primary			
ntp-config		DNS IP Backup1			
phy-interface		DNS IP Backup2			
media-manager	•	Modify Network Interface			
security	•	DNS Domain			
session-router	•	DNS Timeout			
system	•		Π	(Ran	ge: 04294967295)
			86400	(Ran	ge: 302073600)
traud-protection	1	Signaling Mtu	0	(Ran	ge: 0,5764096)

Figure 8: Network Interface towards PSTN Gateway

Property of tekVizionLabs Page 15

Configuration	View Configuration	Q	
media-manager	*	Modify Network Interface	
security	•	HIP IP List	-
session-router	•	ICM D Address	
system	•	ICMP Address	
fraud-protectior		SSH Address	
host-route	_	Tunnel Config	
http-client			
http-server		L L L	
nto-config	.e	No tunnel config to display. Please	add or upload tunnel config.
phy-interface		Add	Upload

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.

media-manager codec-policy	•		Codec Poli	су					
media-manager			Add	Delete All Upload	Download			search	(
media-policy realm-config			Name	Allow Codecs	Add Codecs On Egress	Order Codecs	Packetization Time	Force Ptime	Secure Dtmf
steering-pool			Google	pcmu pcma g722 telephone-event opus	pcmu pcma g722 opus	pcmu pcma g722 silk telephone-event opus	20	disabled	disabled
security	Þ		PSTN	pcmu pcma g722 telephone-event opus	pcmu pcma g722 opus	pcmu pcma g722 silk telephone-event opus	20	disabled	disabled
session-router	Þ	ι.							

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.

Configuration View Configuration	Q		
sip-config	Modify Translation Rules		
sip-feature	Id	addPlus	1
sip-feature-caps	Туре	add 🔹	
sip-interface	Add String	+1	
sip-manipulation	Add Index	0	1
sip-monitoring	Delete String		
sip-nat	Delete Index	0	(Range: 0999999999)
sip-profile			
sip-q850-map			
sip-recursion-policy			
surrogate-agent			
survivability			
translation-rules			

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

Configuration	ew Configuration	Q		
sip-config	•	Modify Translation Rules		
sip-feature		Id	removeplus1	
sip-feature-caps		Туре	delete 💌	
sip-interface		Add String		-
sip-manipulation		Add Index	0	
sip-monitoring		Delete String	+1	
sip-nat		Delete Index	0	(Range: 0999999999)

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.

Configuration View Configuration			
service-health	Modify Session Translation		
session-agent			
session-agent-id-rule	ld	addPlus	
session-constraints	Rules Calling	addPlus 🗙	
session-group	Rules Called	addPlus 🗙	
session-recording-group	Rules Asserted Id		
session-recording-server	Rules Redirect		
session-router			
session-timer-profile	Rules Isup Cdpn		
session-translation	Rules Isup Cgpn		
sip-advanced-logging	Rules Isup Gn		
sip-config	Rules Isup Rdn		
sip-feature	Dalas Java Ora		
sip-feature-caps	Rules Isup Ocn		

Figure 13: Session Translation towards Google Voice

Configuration View Configuration	Q	
session-agent	Modify Session Trans	slation
session-agent-id-rule	ld	removeE164
session-constraints	Rules Calling	removeplus1 🗙
session-group	Rules Called	removeplus1 🗙
session-recording-group		international 🗙
session-recording-server	Rules Asserted Id	removeplus1 🗙
session-router	Rules Redirect	
session-timer-profile		
session-translation	Rules Isup Cdpn	
sip-advanced-logging	Rules Isup Cgpn	
sip-config	Rules Isup Gn	
sip-feature	Rules Isup Rdn	
sip-feature-caps	Dulas laura Ora	
sip-interface	Rules Isup Och	

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.

media-manager	•	Modify Realm Config		
codec-policy		Identifier	Carala	
media-manager		Description	Google	
media-policy		Description		
realm-config				
steering-pool		Addr Prefix	0.0.0.0	
security	►	Network Interfaces	s0p0:0.4 🗙	
session-router	►	Media Realm List		
system	►			
		Mm In Realm	🗸 enable	
		Mm In Network	🖌 enable	
		Mm Same Ip	enable	

Figure 15: Realm Config towards Google Voice

media-manager	•	Modify Realm Config		
codec-policy		QoS Enable	enable	
media-manager		Max Bandwidth		
media-policy		Max Priority Bandwidth	0	(Range: 0.999999999)
realm-config		Parent Realm	0	(Range: 0999999999)
steering-pool				•
steering-poor		DNS Realm		•
security	•	Media Policy		•
session-router	-	Media Sec Policy	SRTP	•
access-control		RTCP Mux	enable	
account-config		Ice Profile		•
media-manager	•	Modify Realm Config		
codec-policy		SDP Inactive Only	enable	
media-manager		DTLS Srtp Profile		·
media-policy		Srtp Msm Passthrough	enable	
realm-config		Class Profile		
steering-pool		In Translationid		
security	•	Out Translationid		
session-router	•		addPlus	
system		in Manipulationid		r
-,	F	Out Manipulationid		·
		Average Rate Limit	0	(Range: 04294967295)
		Access Control Trust Level	none 🔹	·

Figure 16: Realm Config towards Google Voice cont.

Configuration	View Configuration	Q					
media-manager	•	Modify Realm Config					
codec-policy		Maximum Signal Threshold	0			(Range: 0	4294967295)
media-manager		Untrusted Signal Threshold	0			(Range: 0	4294967295)
media-policy		Nat Trust Threshold	0			(Range: 0	65535)
realm-config		Max Endpoints Per Nat	0			(Range: 0	65535)
steering-pool		Nat Invalid Message Threshold	0			(Range: 0	65535)
security		Wait Time For Invalid Register	0			(Range: 0,4	300)
security		Deny Period	30			(Range: 0	4294967295)
session-router	•	Session Max Life Limit	0				
system	•	Untrust Cac Failure Threshold	0			(Range: 0	4294967295)
		Subscription Id Type	END_U	SER_NONE	•		
		Trunk Context					
		Early Media Allow			•		
media-manager	· · · ·	Modify Realm Conf	ig				
codec-policy							
media-manag	ger	Restricted Latching		none		•	
media-policy		Options					
realm-config		SPL Options					
steering-pool		Delay Media Update		enable			
security	•	Refer Call Transfer		disabled		•	
session-router	•	Hold Refer Reinvite		enable			
system	• •	Refer Notify Provisional		none		•	
		Dyn Refer Term		enable			
		Codec Policy		Google		•	

Figure 17: Realm Config towards Google Voice Cont.

Configuration	View Configuration	Q			
media-manager	•	Modify Realm Config			
codec-policy		Codec ManIP In Realm	enable		
media-manager		Codec ManIP In Network	✓ enable		
realm-config		RTCP Policy		•	
steering-pool		Constraint Name		•	
security	•	Session Recording Server			
session-router	•	Session Recording Required	enable		
system	•	SIP Profile		•	
		Flow Time Limit	-1		(Range: -12147483647)
		Initial Guard Timer	-1		(Range: -12147483647)
		Subsq Guard Timer	-1		(Range: -12147483647)
		TCP Flow Time Limit	-1		(Range: -12147483647)

codec-policy TCP Initial Guard Timer -1 (Range: -1.2147483647) media-manager TCP Subsq Guard Timer -1 (Range: -1.2147483647) media-policy SIP Isup Profile I I realm-config QoS Constraint I I steering-pool TCP Media Profile I I security Monitoring Filters I I Node Functionality Default Location String I I Alt Family Realm I I I Pref Addr Type none I I	media-manager		Modify Realm Config			
media-manager TCP Subsq Guard Timer media-policy TCP Subsq Guard Timer realm-config QoS Constraint steering-pool TCP Media Profile security Monitoring Filters session-router Node Functionality opfault Location String Alt Family Realm Pref Addr Type	codec-policy	- 1	TCP Initial Guard Timer			(
media-policy SIP Isup Profile realm-config QoS Constraint steering-pool TCP Media Profile security Monitoring Filters session-router Node Functionality pefault Location String Alt Family Realm Pref Addr Type	media-manager	- 1	TCP Subso Guard Timer	-1		(Range: -12147483647)
realm-config QoS Constraint steering-pool TCP Media Profile security Monitoring Filters session-router Node Functionality system Default Location String Alt Family Realm Pref Addr Type none	media-policy	- 1	SIP Isun Profile	-1		(Range: -12147483647)
ream-coning steering-pool security security Monitoring Filters session-router Node Functionality Default Location String Alt Family Realm Pref Addr Type	realm-config	- 1			•	
steering-pool TCP Media Profile security Monitoring Filters session-router Node Functionality system Default Location String Alt Family Realm Pref Addr Type none	ream-coning	- 1	QoS Constraint		•	
security Monitoring Filters Monitoring Filters Mode Functionality Node Functionality Default Location String Alt Family Realm Pref Addr Type none	steering-pool	- 1	TCP Media Profile		•	
session-router Node Functionality Node Functionality Default Location String Alt Family Realm Pref Addr Type none	security	•	Monitoring Filters			
system Default Location String Alt Family Realm Pref Addr Type none	session-router	•	Node Functionality			
Alt Family Realm Pref Addr Type none	system	•	Defects Leasting Chine		•	
Alt Family Realm Pref Addr Type none		- 1	Default Location String			
Pref Addr Type none		- 1	Alt Family Realm		•	
			Pref Addr Type	none	•	



session-router	•	Ringback Trigger	none 🔻	,
system	•	Ringback File		
		Merge Early Dialogs	enable	
		User Site		
		Srvcc Trfo		
Show All		ОК	Back	



Configuration	View Configuration	Q		
media-manager	Ŧ	Modify Realm Config		
codec-policy				
media-manager		Identifier	onprem	
media-policy		Description	onprem	
realm-config				
steering-pool		Addr Prefix	0.0.0.0	
security	•	Network Interfaces	s1p0:0.4 🗙	
session-router	•	Media Realm List		
system	•			
		Mm in Realm	🖌 enable	
		Mm In Network	🖌 enable	
		Mm Same Ip	✓ enable	
		QoS Enable	enable	

Figure 20: Realm Config towards OnPrem PBX

media-manager	•	^	Modify Realm Config			
codec-policy			Max Bandwidth	0	(Range: 0999999999)
media-manager			Max Priority Bandwidth	0	(Range: 0999999999)
media-policy			Parent Realm			
realm-config		۰.	DNS Realm		•	
steering-pool			Media Policy		•	
security	►		Media Sec Policy	RTP	T	7
session-router	•		RTCP Mux	enable		_
access-control			Ice Profile		•	
media-manager	•	^	Modify Realm Config			
codec-policy			SDP Inactive Only	enable		
media-manager			DTLS Srtp Profile			
media-policy			Srtp Msm Passthrough			
realm-config			Class Profile	enable	_	
steering-pool			In Translationid		•	
security	►		Out Translationid		•	
session-router				removeE164	•	
Session router	•		In Manipulationid		•	
access-control			Out Manipulationid		•	
account-config			Average Rate Limit	0		(Range: 04294967295)
filter-config			Access Control Trust Level	none	•	
ldap-config			Invalid Signal Threshold	0		(Range: 04294967295)
local-policy			Maximum Signal Threshold	n		(Range: 04294967295)

Figure 21: Realm Config towards OnPrem PBX Cont.

Configuration	View Configuration	Q				
media-manager	•	Modify Realm Config				
codec-policy		Untrusted Signal Threshold	0		(Range: 04294967	295)
media-manager		Nat Trust Threshold	0		(Range: 065535)	
media-policy		Max Endpoints Per Nat	0		(Range: 065535)	
realm-config		Nat Invalid Message Threshold	0		(Range: 065535)	
steering-pool		Wait Time For Invalid Register	0		(Range: 0,4300)	
security	•	Deny Period	30		(Range: 04294967	295)
session-router	•	Session Max Life Limit	0			
system	•	Untrust Cac Failure Threshold	0		(Range: 04294967	295)
		Subscription Id Type	END_USER_	NONE		
		Trunk Context				
		Early Media Allow		•		
		Enforcement Profile		•		
Configuration	View Configurat	ion Q				
media-manager	r v	Modify Realm Con	ifig			
codec-policy	ner	Additional Prefixes				
media-policy	P.,	Restricted Latching		none	•	
realm-config		Options				
steering-pool	I	SPL Options				
security	•	Delay Media Update		enable		
session-router	•	Refer Call Transfer		disabled	•	
system	Þ	Hold Refer Reinvite		enable		
		Refer Notify Provisional		none	•	
		Dyn Refer Term		enable		
		Codec Policy			•	

Figure 22: Realm Config towards OnPrem PBX Cont.

Property of tekVizionLabs Page 26

Configuration	View Configuration	Q		
media-manager	•	Modify Realm Config		
codec-policy		Codec ManIP In Realm	enable	
media-manager		Codec ManIP In Network	enable	
media-policy		RTCP Policy	· · · · · · · · · · · · · · · · · · ·	
realm-config		Constraint Name		
steering-pool		Session Recording Server		
security	•	Session Recording Required	enable	
session-router	•	SIP Profile	•	
system	,	Flow Time Limit	-1	(Range: -12147483647)
		Initial Guard Timer	-1	(Range: -12147483647)
		Subsq Guard Timer	-1	(Range: -12147483647)
		TCP Flow Time Limit	-1	(Range: -12147483647)
Configuration	View Configuration	Q		
media-manager	•	Modify Realm Config		
codec-policy		TCP Initial Guard Timer	-1	(Range: -12147483647)
media-manager		TCP Subsq Guard Timer	-1	(Range: -12147483647)
media-policy		SIP Isup Profile		
realm-config		QoS Constraint		
steering-pool		TCP Media Profile		
security	•	Monitoring Filters		
session-router	•	Node Functionality		
system	•	Default Location String		
		Alt Family Realm		
		Pref Addr Type	none 🗸	

Figure 23: Realm Config towards OnPrem PBX Cont.

Property of tekVizionLabs Page 27

steering-pool			
security	•	Sm Icsi Match For Message	
session-router	•	Ringback Trigger	none 🔻
system	•	Ringback File	
		Merge Early Dialogs	enable
		User Site	
		Srvcc Trfo	
Show All		ОКВ	lack
Show All			

Figure 24: Realm Config towards OnPrem PBX Cont.

Configuration	View Configuration	Q	
media-manager	~	Modify Realm Config	
codec-policy			
media-manager		Identifier	PSTNGW
media-policy		Description	PSTNGW
realm-config			
steering-pool		Addr Prefix	0.0.0.0
security	•	Network Interfaces	s1p0:0.4 🗙
session-router	•	Media Realm List	
system	•		
		Mm In Realm	🖌 enable
		Mm In Network	✓ enable
		Mm Same Ip	✓ enable
		QoS Enable	enable

Figure 25: Realm Config towards PSTN Gateway

media-manager	•	Modify Realm Config			
codec-policy		Max Bandwidth	0		(Range: 0.,999999999)
media-manager		Max Priority Bandwidth	0		(Range: 0999999999)
media-policy		Parent Realm			
realm-config		DNS Realm			-
steering-pool		Media Policy			•
security	•	Media Sec Policy	RTP		-
session-router	•	RTCP Mux	enable		
access-control		Ice Profile			*
media-manager	•	Modify Realm Config			
codec-policy	- 1	SDP Inactive Only	enable		
media-manager	- 1	DTLS Srtp Profile		•	
media-policy		Srtp Msm Passthrough	enable		
realm-config		Class Profile		•	
steering-pool		In Translationid		•	
security	•	Out Translationid	removeE164	•	
session-router	•	In Manipulationid		•	
access-control		Out Manipulationid		•	
account-config		Average Rate Limit	0	1)	Range: 04294967295)
filter-config		Access Control Trust Level	none	•	-
ldap-config		Invalid Signal Threshold	0	(F	Range: 04294967295)
local-policy		Maximum Signal Threshold	n	()	Range: 0.,4294967295)

Figure 26: Realm Config towards PSTN Gateway Cont.

Configuration	View Configuration	Q				
media-manager	~	Modify Realm Config				
codec-policy			U		(Kange: U4294	40/245)
media-manager		Untrusted Signal Threshold	0		(Range: 04294	967295)
media-policy		Nat Trust Threshold	0		(Range: 06553	5)
media-policy		Max Endpoints Per Nat	0		(Range: 06553	5)
realm-config		Nat Invalid Message Threshold	0		(Range: 06553	5)
steering-pool		Wait Time For Invalid Register	0		(Range: 0,4300))
security	•	Deny Period	30		(Range: 04294	967295)
session-router	•	Session Max Life Limit	0			
system	•	Untrust Cac Failure Threshold	0		(Range: 04294	967295)
		Subscription Id Type	END_USER_NONE	•		
		Trunk Context				
		Early Media Allow				
		Enforcement Profile				
media-manager	•	Modify Realm Confi	2			
codec-policy						
media-manage	r i i	Restricted Latching	none		•	
media-policy		Options				
realm-config		SPL Options				
steering-pool		Delay Media Update	enable			
security	•	Refer Call Transfer	disabled		•	
session-router	•	Hold Refer Reinvite	enable			
system	Þ	Refer Notify Provisional	none		•	
		Dyn Refer Term	enable			
		Codec Policy	PSTN		-	

Figure 27: Realm Config towards PSTN Gateway Cont.

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

Configuration	View Configuration	Q		
media-manager	•	Modify Realm Config		
codec-policy media-manager		Codec ManIP In Realm	enable	
media-policy		Codec ManIP In Network	🖌 enable	
realm-config		RTCP Policy	•	
steering-pool		Constraint Name	•	
security	•	Session Recording Server		
session-router	•	Session Recording Required	enable	
system	•	SIP Profile	▼	
		Flow Time Limit	-1	(Range: -12147483647)
		Initial Guard Timer	-1	(Range: -12147483647)
		Subsq Guard Timer	-1	(Range: -12147483647)
		TCP Flow Time Limit	-1	(Range: -12147483647)
Configuration	View Configuration	Q		
media-manager	•	Modify Realm Config		
codec-policy		TCP Initial Guard Timer	-1	(Range: -12147483647)
media-manager		TCP Subsq Guard Timer	-1	(Range: -12147483647)
media-policy		SIP Isup Profile		
realm-config		QoS Constraint		
steering-pool		TCP Media Profile		
security	۱.	Monitoring Filters		
session-router	•	Node Functionality		
system	•	Default Location String		
		Alt Family Realm		
		Pref Addr Type	none 🔹	
		Sm Icsi Match For Invite		
Show All		ОКВ	ack	

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.

media-manager v	Modify Steering Pool			
,				
media-manager	IP Address	10.80.11.21		
media-policy	Start Port	20000		(Range: 0,165535)
	End Port	39999		(Range: 0,165535)
realm-config	Bealm ID			
	Realm ID	PSTNGW	•	
steering-pool	Network Interface	s1p0:0.4		
security 🕨			1	

Figure 29: Steering Pool towards PSTN Gateway

media-manager	•	Modify Steering Pool			
codec-policy					
media-manager		IP Address	192.65.		
media-policy		Start Port	20000	(Range: 0,165535)	
realm-config		End Port	39999	(Range: 0,165535)	
ream-comig		Realm ID	Google	v	
steering-pool		Network Interface	s0p0:0.4	•	
security	►				
session-router	•				

Figure 30: Steering Pool towards Google Voice

media-manager	•	Modify Steering Pool			
codec-policy					
media-manager		IP Address	10.80.11.21		
media-policy		Start Port	50000		(Range: 0,165535)
realm-config		End Port	59999		(Range: 0,165535)
- Coming		Realm ID	onprem	-	
steering-pool				· ·	
Steering poor		Network Interface	s1p0:0.4	•	
security	•				

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.

Configuration View Configuration	recv	Q X
security-association	Modify Sdes Profile	
security-policy		
local-accounts	Name	SDES
media-security 🔻	Crypto List	AES_CM_128_HMAC_SHA1_80 🗙
dtls-srtp-profile	Srtp Auth	✓ enable
media-sec-policy	Srtp Encrypt	✓ enable
sdes-profile	SrTCP Encrypt	✓ enable
sipura-profile	Mki	enable
password-policy	Egress Offer Format	same-as-ingress
security-config	Use Ingress Session Params	
ssh-config	Options	
ssh-key	Key	
tls-global	Salt	
a 14	Salt	
ssh-key		
tls-global	Srtp Rekey On Re Invite	enable
tls-profile	Lifetime	0 (Range: 0,2048)

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.

Configuration View Configuration	Q		
local-accounts	Modify Media Sec Policy		
media-security 👻	Name	RTP	
dtls-srtp-profile	Pass Through	enable	
media-sec-policy	Options		
sdes-profile			
sipura-profile	⊿ Inbound		
	Profile	•	
password-policy	Mode	rtp	•
security-config	Protocol	none	•
ssh-config	Hide Egress Media Update	enable	
ssh-key	(Outbound		
tis-giodai	Profile	v	
tls-profile	Mode	rtp	•
	Durate1		

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.

Configuration	/iew Configuration	recv	Q X	
ike	▶ ▲	Modify Media Sec Policy		
ipsec	•			
local-accounts		Name	SRTP	
media-security	-	Pass Through	enable	
dtls-srtp-profile	- 11	Options		
media-sec-policy	·	Inbound		
sdes-profile		Profile	SDES 💌	
sipura-profile		Mode	srtp 💌	
password-policy		Protocol	sdes 🔹	
patenena peney		Hide Egress Media Update enable		
security-config				
ssh-config		▲ Outbound		
ssh-kev		Profile	SDES	
		Mode	srtp 💌	
tls-global		Protocol	sdes 💌	
tls-profile				

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**.

media-manager	•	Modify Certificate Record					
security	•						
authentication-profile		Name	GTS-Root-R1				
certificate-record		Country	US				
tls-global		State	МА				
tls-profile		Locality	Burlington				
session-router	•	Organization	Engineering				
system	►	Unit					
		Common Name	GTS Root R1				
		Key Size	2048 💌				
		Alternate Name					
		Trusted	✓ enable				

Create a certificate record for Google Voice as shown below.

Figure 35: Create Certificate Record for Google Voice ROOT CA

realm-config		L	Key Usage List	digitalSignature 🗙	
steering-pool			Extended Key Usage List		
security	•			serverAuth 🗙	
authentication-profile			Key Algor	rsa	•
certificate-record			Digest Algor	sha256	•
tls-global		τ.	Ecdsa Key Size	p256	•
tls-profile		L	Cert Status Profile List		
session-router	►	L	Options		
system	►	-			

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	•	Modify Certificate Record				
codec-policy						
media-manager		Name	sbc3			
media-policy		Country	US			
realm-config		State	Texas			
steering-pool		Locality	Plano			
security	•	Organization	tekvizionLabs			
authentication-profile		Unit	Tekvizion			
certificate-record		Common Name	sbc3.tekvizionlabs.com			
tls-global		Key Size	2048 💌			
tls-profile		Alternate Name				
session-router	•	Trusted	✓ enable			

Figure 37: Create Certificate Record for Oracle E-SBC

media-manager	•	Modify Certifi	cate Record	
codec-policy		Alternate Name		
media-manager		Trusted		
media-policy		Key Lisago List	✓ enable	
realm-config		Key Usage List	digitalSignature 🗙	
steering-pool		Extended Key Usage	e List	
security	•		serverAuth 🗙 client	Auth 🗙
authentication-profile		Key Algor	rsa	•
certificate-record		Digest Algor	sha256	•
tls-global		Ecdsa Key Size	p256	•
tls-profile		Cert Status Profile L	ist	
		Options		

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

ertificate Record								
Add Delete All	Upload D	ownload				search		0
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 Certifica	tion 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.

media-manager	•	Modify TLS Profile	
security	•	Name	TLSCRVOC
authentication-profile		End Entity Cortificato	115GBF0C
certificate-record		End Entity Certificate	sbc3
the clobal		Trusted Ca Certificates	GoDaddyRootCertificate 🗙
us-giobai			GoDaddyClass2CertificationAuthority
tls-profile			×
session-router	•		GoDaddySecureCertificateG2 🗙
system			GTS-Root-R1 🗙
		Cipher List	DEFAULT ×
		Vorify Donth	
		Verity Deput	10 (Range: 0.10)
		Mutual Authenticate	✓ enable
		TLS Version	tlev12 =
steering-pool		Ontions	VISVIE V
security	•	options	
session-router	•	Cert Status Check	enable
system	•	Cert Status Profile List	
		Ignore Dead Responder	
		Allow Self Signed Cert	
			enable
Show All		ОК	Back

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.

Configuration View Configuration	Q		
session-router	Modify Session Timer Pro		
session-timer-profile	Name		
session-translation	Hame	SessionTimer	
sis advanted baseline	Session Expires	900	(Range: 64999999999)
sip-advanced-logging	Min Se	90	(Range: 64999999999)
sip-config	Force Reinvite	enable	
sip-feature	Request Refresher	uac 💌	
sip-feature-caps	Response Refresher	uas 🔻	
sip-interface			

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.

Configuration View Configuration	Q									Discard 😧 Verify
session-group	Modify	/ SIP I	nterface							Show Configurati
session-recording-group	State					1				
session-recording-server	Dealer ID			✓ enable						
session-translation	Realm ID			onprem	•					
sip-config	Descripti	on								
cin_feature										
	SID Dorte									
sip-intenace		17	G @							
sip-manipulation	Action	Sel	Address	Por	t	Transport Protocol	TLS Profile	Allow Anon	iymous	Multi Home Addrs
sip-monitoring			10.80.11.21	506	57	UDP		all	1	
translation-rules	· ·	_								
sip-advanced-logging			Modify	y SIP Int	terface					
sip-config			Initial Inv	Trans Exp	ire	0				(Range: 0999999999)
sip-feature			Session N	Aax Life Li	Life Limit 0					
sip-feature-caps			Proxy Mo	de					•	
sip-interface			Redirect	Action				,	•	
sip-manipulation			Nat Trave	ersal		none		,	•	
sip-monitoring			Nat Interv	val		30				(Range: 04294967295)
	TCP Nat		TCP Nat I	Interval		90				(Range: 04294967295)
sip-nat			Registrat	ion Cachin	g	enable				
sip-profile			Min Reg I	Expire		300				(Range: 0999999999)
sip-q850-map			Registrat	ion Interva	1	3600				(Range: 04294967295)
sip-recursion-policy			Route To	Registrar		enable				
surrogate-agent			Secured 1	Network		enable				



sip-advanced-logging	Modify SIP Interface		
sip-config	Uri Fqdn Domain		
sip-feature	Options		
sip-feature-caps	SPL Options		
sip-interface	Trust Mode	all	•
sip-manipulation	Max Nat Interval	3600	(Range: 04294967295)
sip-monitoring	Stop Recurse	401,407	
sip-nat	Port Map Start	0	(Range: 0,102565535)
sip-profile	Port Map End	0	(Range: 0,102565535)
sip-q850-map	In Manipulationid		•
sip-recursion-policy	Out Manipulationid		•
surrogate-agent	SIP Atcf Feature	enable	
sip-advanced-logging	Modify SIP Interface		
sip-config	Rfc2833 Payload	101	(Range: 96127)
sip-feature	Rfc2833 Mode	transparent	•
sip-feature-caps	Response Map		•
sip-interface	Local Response Map		•
sip-manipulation	Sec Agree Feature	enable	
sip-monitoring	Enforcement Profile		•
sip-nat	TCP Keepalive	none	•
sip-profile	Add SDP Invite	disabled	•
sip-q850-map	Add SDP In Msg		
sip-recursion-policy	P Early Media Header	disabled	•

Figure 43: SIP Interface for OnPrem PBX Cont.

sip-advanced-logging	Modify SIP Interface	
sip-config	Add SDP Profiles	
sip-feature	Add SDP Profiles In Msg	
sip-feature-caps	CID Destile	
sip-interface	SIP Profile	
sip-manipulation	SiP isup Profile	
sip-monitoring	TCP Conn Dereg 0	(Range: 0999999999)
sip-nat	Komi2923 luf On Hairnin	
sip-profile	Mero Dolay Egrees Buo	
sip-q850-map	enable	
sip-recursion-policy	Desef Posteration	
	No dife CID latente en	
sip-advanced-logging	Modity SIP Interface	
sip-config	Session Timer Profile	•
sip-feature	Session Recording Server	
sip-feature-caps	Session Recording Required enable	
sip-interface	Service Tag	
sip-manipulation	Reg Cache Route enable	
sip-monitoring	Diversion Info Mapping Mode none	•
sip-nat	Atcf Icsi Match	
sip-profile	SIP Recursion Policy	•
sip-q850-map	Asymmetric Preconditions enable	
sip-recursion-policy	Asymmetric Preconditions Mode send-with-nodelay	•
surrogate-agent	Sm Icsi Match For Invite	

Figure 44: SIP Interface for OnPrem PBX Cont.

sip-interface	Sm Icsi Match For Message		
sip-manipulation			
sip-monitoring	S8hr Profile		•
sip-nat	Ringback Trigger	none	•
sio-profile	Ringback File		
cia a ⁹⁵⁰ man	Npli Profile		•
sip-qoso-map	Hist To Div For Cause 380	inherit	
sip-recursion-policy	User Agent		
surrogate-agent			

Figure 45: SIP Interface for OnPrem PBX Cont.

TLS Profile is configured with TLSGBYOC configured in Section 6.4.12

Modify SI	P Interfac	e							Show
State		~	enable	٦					
Realm ID		G	oogle	•					
Description		G	oogle						
SIP Ports									
Add									
Address	1	Port	Transport Protocol		TLS Profile	Allow Anonym	ous	Multi Home Addrs	
192.65.		5061	TLS		TLSGBYOC	agents-only			

Figure 46: SIP Interface for Google Voice

session-constraints	Modify SIP Interface		
	Initial Inv Trans Expire	0	(Range: 0999999999)
session-group	Session Max Life Limit	0	
session-recording-group	Proxy Mode		
session-recording-server	Redirect Action		
session-router	Nat Traversal	none 🗸	
session-timer-profile	Nat Interval	30	(Range: 04294967295)
session-translation	TCP Nat Interval	90	(Range: 04294967295)
sip-advanced-logging	Registration Caching	enable	
sip-config	Min Reg Expire	300	(Range: 0999999999)
sip-feature	Registration Interval	3600	(Range: 04294967295)
sip-feature-caps	Route To Registrar	enable	
sip-interface	Secured Network	enable	

session-agent-id-rule	*	Modify SID Interface		
session-constraints		Modify SIP Interface		
session-group		Uri Fqdn Domain		
Session Broop		Options		
session-recording-group				
session-recording-server		SPL Options		
session-router		Trust Mode	all	•
session-timer-profile		Max Nat Interval	3600	(Range: 04294967295)
session-translation		Stop Recurse	401,407	
sip-advanced-logging		Port Map Start	0	(Range: 0,102565535)
		Port Map End	0	(Range: 0,102565535)
sip-config		In Manipulationid		•
sip-feature		Out Manipulationid		-
<i>sip-feature-caps</i>		SIP Atcf Feature	enable	

Figure 47: SIP Interface for Google Voice Cont.

	Modify SIP Interface				
session-constraints	Rfc2833 Payload	101			(Range: 96127)
session-group	Rfc2833 Mode	transparent		•	
session-recording-group	Response Map				
session-recording-server	Local Response Map				
session-router	Sec Agree Feature			•	
session-timer-profile	Enforcement Profile	enable			
session-translation	TCD Keeneliker			•	
sip-advanced-logging	ICP Keepalive	none		•	
sip-config	Add SDP Invite	disabled		•	
sip-feature	Add SDP In Msg				
sin-feature-cans	P Early Media Header	disabled		•	
	P Early Media Direction				
sip-interface					
session-group	Modify SIP Interface				
session-recording-group	Add SDP Profiles				
session-recording-ser	Add SDP Profiles In Msg				
session-translation	SIP Profile				
sip-config	SIP Isup Profile		•		
sip-feature	TCP Conn Dereg	0	•	(Panger O	000000000)
sip-interface	Kpml Interworking	0 enable		(Kalige, O.	
sip-manipulation	Kpml2833 lwf On Hairpin	Kpml2833 lwf On Hairpin enable			
sip-monitoring	Msrp Delay Egress Bye	enable			
sti-server	Send 380 Response				
uansiation-rules					

Figure 48: SIP Interface for Google Voice Cont.

session-agent-id-rule	Modify SIP Interface		
session-constraints	Sector Tree Partie		
session-group	Session Timer Profile	SessionTimer 🔻	
session-recording-group	Session Recording Server		
session-recording-server	Session Recording Required	enable	
session-router	Service Tag		
session-timer-profile	Reg Cache Route	enable	
session-translation	Diversion Info Mapping Mode	none 🔻	
sip-advanced-logging	Atcf Icsi Match		
sip-config	SIP Recursion Policy	•	
sip-feature	Asymmetric Preconditions	enable	
sip-feature-caps	Asymmetric Preconditions Mode	send-with-nodelay 🔹	
session-recording-server	Sm Icsi Match For Invite		
session-router	Sm Icsi Match For Message		
session-timer-profile	S8hr Profile		•
session-translation	Ringback Trigger	none	•
sip-advanced-logging	Ringback File		
sip-config	Npli Profile		•
sip-feature	Hist To Div For Cause 380	inherit	•
sip-feature-caps	User Agent		
sip-interface			

Figure 49: SIP Interface for Google Voice Cont.

local-policy	Modify SIP Inter	rface						Show Cor
local-routing-config	State							
media-profile	Dealer ID	✓ enab	le					
session-agent	Description	PSTNGV	V					
session-group	Description	To trunk						
session-recording-group								
session-recording-server	SIP Ports							
session-translation	D. / G	Ē						
sis scafin	Action Sel Ac	ddress	Port	Transport Protocol	TLS Profile	Allow Anony	mous	Multi Home Addrs
sip-feature	: 🗌 10	.80.11.21	5060	UDP		agents-only		
sip-interface								
Configuration Vie	w Configuration	Q						
sip-advanced-logging	7	Modify SI	P Interface					
sip-config		Initial Inv Tran	s Expire	0			(Range: C)9999999999)
sip-feature		Session Max L	ife Limit	0				
sip-feature-caps		Proxy Mode				•		
sip-interface		Redirect Actio	n			•		
sip-manipulation		Nat Traversal		none		•		
sip-monitoring		Nat Interval		30			(Range: C)4294967295)
		TCP Nat Interv	al	90			(Range: 0)4294967295)
sip-nat		Registration C	aching	enable				
sip-profile		Min Reg Expire	e	300			(Range: 0)9999999999)
sip-q850-map		Registration In	iterval	3600			(Range: C)4294967295)
sip-recursion-policy		Route To Regis	strar	enable				
surrogate-agent		Secured Netwo	ork	enable				

Figure 50: SIP Interface for PSTN Gateway

Configuration View Configuration	Q			
sip-advanced-logging	Modify SIP Interface			
sip-config	Uri Fqdn Domain			
sip-feature	Options			
sip-feature-caps	SPL Options			
sip-interface	Trust Mode	all	•	
sip-manipulation	Max Nat Interval	3600		(Range: 04294967295)
sip-monitoring	Stop Recurse	401,407		
sip-nat	Port Map Start	0		(Range: 0,102565535)
sip-profile	Port Map End	0		(Range: 0,102565535)
sip-q850-map	In Manipulationid		-	
sip-recursion-policy	Out Manipulationid	TowardsPSTN	•	
surrogate-agent	SIP Atcf Feature	enable		
sip-advanced-logging	Modify SIP Interface			
sip-config	Rfc2833 Payload	101		(Range: 96127)
sip-feature	Rfc2833 Mode	transparent		•
sip-feature-caps	Response Map			•
sin_interface	Local Response Map			•
sprinterrace	Sec Agree Feature	enable		
sip-manipulation	Enforcement Profile			-
sip-monitoring	TCP Keepalive			
sip-nat		none		•
sip-profile	Add SDP Invite	disabled		•
sip-q850-map	Add SDP In Msg			
sip-recursion-policy	P Early Media Header	disabled		•
sprecusion policy	P Early Media Direction			
surrogate-agent				

Figure 51: SIP Interface for PSTN Gateway Cont.

sip-advanced-logging	Modify SIP Interface	
sip-config	Add SDP Profiles	
sip-feature	Add SDP Profiles In Msg	
sip-feature-caps	SIP Profile	
sip-interface	SIP Isup Profile	Ť
sip-manipulation	TCP Conn Dereg	(Range: 0.00000009)
sip-monitoring	Kpml Interworking	enable
sip-nat	Kpml2833 lwf On Hairpin	enable
sip-profile	Msrp Delay Egress Bye	enable
sip-q850-map	Send 380 Response	
sip-recursion-policy	Pcscf Restoration	
surrogate-agent	Session Timer Profile	v
sip-advanced-logging	Modify SIP Interface	e
sip-config	Session Recording Server	
sip-feature	Session Recording Required	d enable
sip-feature-caps	Service Tag	
sip-interface	Reg Cache Route	enable
sip-manipulation	Diversion Info Mapping Mo	de none 💌
sip-monitoring	Atcf Icsi Match	
sip-nat	SIP Recursion Policy	•
sip-profile	Asymmetric Preconditions	enable
sip-q850-map	Asymmetric Preconditions	Mode send-with-nodelay 💌
sip-recursion-policy	Sm Icsi Match For Invite	
surrogate-agent		

Figure 52: SIP Interface for PSTN Gateway Cont.

sin-interface		
spintenace	Sm Icsi Match For Message	
sip-manipulation		
sip-monitoring	S8hr Profile	•
sip-nat	Ringback Trigger	none 💌
sip-profile	Ringback File	
	Npli Profile	•
sip-q850-map	Hist To Div For Cause 380	
sip-recursion-policy	hist to big for cause 500	inherit 🔻
	User Agent	
surrogate-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.

Configuration	View Configu	uration	Q				
media-manager	•	^	Modify Sessior	n Agent			
security	•						
session-router	•		Hostname		siplink.telephon	y.goog	
access-control			IP Address				
account-config			Port		5672		(Range: 0,102565535)
filter-config			State		✓ enable		
Idap-config			App Protocol		SIP	▼	
local-policy		Ι.,	Арр Туре			•	
local-routing-co	nfig		Transport Method		StaticTLS	•	
media-profile			Realm ID		Google	•	
session-agent			Egress Realm ID			•	
session-group			Description				
session-recordir	ng-group						
media-manager	•	Modify	y Session Agent				
security	•	Match Ide	entifier				
session-router							
access-control						\bigcirc	
account-config						U	
ldap-config					No ma	itch identifier to displa	ay. Please add.
local-policy							
local-routing-config		Associate	ed Agents				
media-profile		_					
session-agent		Constrain	sions	enable			
session-group		Max Inbo	ound Sessions	0		(kange: 0999999999) (Range: 0999999999)	
			_				

Figure 54: Session Agent for Google Voice

Configuration View Config	guration Q		
media-manager	Modify Session Agent		
security >	Max Outbound Sessions	0	(Paraget 0, 000000000)
session-router 🔹 🔻	Max Burst Rate	0	(Range, 0
access-control	Max Inbound Burst Rate	0	(Range: 0.99999999)
account-config	Max Outbound Burst Rate	0	(Range: 0.99999999)
	Max Suctain Pate	0	(Range: 0.999999999)
filter-config	Max Sustain Rate	0	(Range: 0999999999)
ldap-config	Max Indound Sustain Rate	0	(Range: 0.999999999)
local-policy	Max Outbound Sustain Rate	0	(Range: 0999999999)
	Min Asr	0	(Range: 0.100)
local-routing-config	Cac Trap Threshold	0	(Range: 099)
media-profile	Session Max Life Limit	0	
session-agent	Time To Resume	0	(Range: 0999999999)
session-group	In Service Period	0	(Range: 0999999999)
rph-policy	Modify Session Agent	t	
rph-profile	Burst Rate Window	0	(Range: 0999999999)
	Sustain Rate Window	0	(Range: 0999999999)
service-neaith	Proxy Mode		~
session-agent	Redirect Action		-
session-agent-id-rule	Loose Routing	enable	· ·
session-constraints	Response Map		•
session-group	Ping Method	OPTIONS	
session-recording-group	Ping Interval	20	(Pange: 0, 4204047205)
session-recording-ser	Ping Send Mode	keep-alive	(Nange, 0+27+70/275)
session-router	Ping All Addresses	enable	
session-timer-profile			

Figure 55: Session Agent for Google Voice Cont.

Configuration	View Configuration	Q				
media-manager	•	Modify Session Age	nt			
security	•	Options				
session-router	•	SPL Options				
access-control		Media Profiles				
account-config						
filter-config		In Translationid				•
ldap-config		Out Translationid				•
local-policy		Trust Me		enable		
local-routing-co	onfig	Local Response Map				•
media-profile		Ping Response		🗸 enable		
session-agent		In Manipulationid				•
session-group		Out Manipulationid		GoogleManipulation	n	•
Configuration	View Configuration					
media-manager	► Î Mo	odify Session Agent				
security	► Mar	anulation String				
session-router	•					
access-control	Mar	hipulation Pattern				
account-config	Trur	nk Group				
filter-config	Max	Register Sustain Rate	0			(Range: 0999999999)
ldap-config	Inva	lidate Registrations	enable			
local-policy	Rfc2	2833 Mode	none		•	
local-routing-confi	Rfc2	2833 Payload	0			(Range: 0,96127)
media-profile	Cod	lec Policy			•	
session-agent	Refe	er Call Transfer	enabled		•	
session-group	Refe	er Notify Provisional	none		•	
	Reu	se Connections	NONE		•	

Figure 56: Session Agent for Google Voice Cont.

filter-config	Modify Sessio	on Agent							Show
ldap-config	, 								
local-policy	TCP Keepalive		enabled	•					
local-routing-config	TCP Reconn Interval		60		(Range: 0,2300)				
media-profile	Max Register Burst Rate 0		0		(Range: 09999999	999)			
session-agent	Rate Constraints								
session-group	Add								
session-recording-group	Method	Max Inbound Bur	st Rate	Max Outbound Bur	st Rate	Max Inboun	d Sustain Rate	Max Outbound Sustain Rate	
session-recording-server					No data to disp	lay.			
session-translation									
Configuration View Config	guration Q								
media-manager	Mod	dify Session	n Agent						
security	SIP Is	up Profile				•			
session-router 🔹	Kpml	Interworking		inherit					
access-control	Kpml	2833 lwf On Ha	irpin	inherit					
account-config	Prece	dence		0			(Range: 04294967)	295)	
filter-config	Monit	toring Filters							
ldap-config	Auth	Attribute							
local-policy									
local-routing-config									
media-profile							(i)	
session-agent						No au	th attributes to c	lisplay. Please add	
session-group						He du	Add		
								_	

Figure 57: Session Agent for Google Voice Cont.

session-router 🗸	Session Recording Server		
access-control			
account-config	Session Recording Required	enable	
film and a	Hold Refer Reinvite	enable	
niter-config	Send TCP Fin		
ldap-config	CID Desursion Deline	enable	
local-policy	SIP Recursion Policy		•
local-routing-config	Sm Icsi Match For Invite		
	Sm Icsi Match For Message		
media-profile	on test materix of message		
session-agent	Ringback Trigger	none	•
session-group	Ringback File		
session-recording-group			
	ОК	Back	
Show All		DOCK	

Figure 58: Session Agent for Google Voice Cont.

Configuration View Configuration	on Q		
access-control	Modify Session Ag	ent	
account-config	Hostname	172.16.29.53	
filter-config	IP Address	172162053	
ldap-config	Port	5040	(Dan see 04025 (5575)
local-policy	State		(Range: 0,102505555)
local-routing-config	App Protocol	enable	
media-profile		216	V
session-agent	Арр Туре		•
Session agent	Transport Method	UDP	
session-group	Realm ID	onprem	•
session-recording-group	Egress Realm ID		•
session-recording-server	Description		
session-translation			
Configuration View Configuration	Q		
access-control	Modify Session Agent		
account-config	Match Identifier		
filter-config			
ldap-config			\frown
local-policy			
local-routing-config		No mate	h identifier to display. Please add.
media-profile			Add
session-agent			
session-group	Associated Agents		
session-recording-group	Constraints		
session-recording-server	Max Sessions	enable	Bangar 0,00000000)
session-translation		0	Kalike: O''AAAAAAAA

Figure 59: Session Agent for OnPrem PBX

account-config	Max Inbound Sessions	0	(Range: 0999999999)
filter-config	Max Outbound Sessions	0	(Range: 0999999999)
Idan-config	Max Burst Rate	0	(Range: 0999999999)
idap-comg	Max Inbound Burst Rate	0	(Range: 0999999999)
local-policy	Max Outbound Burst Rate	0	(Range: 0999999999)
local-routing-config	Max Sustain Rate	0	(Range: 0999999999)
media-profile	Max Inbound Sustain Rate	0	(Range: 0999999999)
session-agent	Max Outbound Sustain Rate	0	(Range: 0999999999)
	Min Asr	0	(Range: 0100)
session-group	Cac Trap Threshold	0	(Range: 099)
session-recording-group	Session Max Life Limit	0	
session-recording-server	Time To Resume	0	(Paper 0, 00000000)
session-translation	In Service Period	0	(Range: 0.000000000)
	Modify Session Agent	0	(Kauße: 0
rph-policy		0	(Nange. 0
rph-profile	Burst Rate Window	0	(Range: 0999999999)
service-health	Sustain Rate Window	0	(Range: 0999999999)
	Proxy Mode		-
session-agent	Pedirect Action		¥
session-agent-id-rule	Redirect Action		•
session-constraints	Loose Routing	🗸 enable	
session-group	Response Map		•
Seen Breek	Ping Method	OPTIONS	
session-recording-group	Ping Interval		
session-recording-ser	Ding Send Mode	30	(Range: 04294967295)
session-router	Fing Send Mode	keep-alive	•
	Ping All Addresses	enable	

Figure 60: Session Agent for OnPrem PBX Cont.

access-control	Modify Session Age	ent	
account-config	Options		
filter-config	SPL Options		
ldap-config	Media Profiles		
local-policy			
local-routing-config	In Translationid		•
media-profile	Out Translationid		•
session-agent	Trust Me	enable	
session-group	Local Response Map		•
session-recording-group	Ping Response	enable	
session-recording-server	In Manipulationid		•
session-translation	Out Manipulationid		•
access-control	Modify Session Agent		
account-config	Manipulation String		
filter-config	Manipulation Pattern		
ldap-config	Trunk Group		
local-policy	Max Register Sustain Rate	0	(Range: 0999999999)
local-routing-config	Invalidate Registrations	enable	
media-profile	Rfc2833 Mode	none 💌	
session-agent	Rfc2833 Payload	0	(Range: 0,96127)
session-group	Codec Policy	•	
session-recording-group	Refer Call Transfer	disabled 🔹	
session-recording-server	Refer Notify Provisional	none 💌	
session-translation	Reuse Connections	NONE	

Figure 61: Session Agent for OnPrem PBX Cont.

access-control	Modify Session Agent			
account-config	TCP Keepalive	none	•	
filter-config	TCP Reconn Interval	0		(Range: 0,2300)
ldap-config	Max Register Burst Rate	0		(Range: 099999999)
local-policy	Rate Constraints			
local-routing-config				
media-profile				\frown
session-agent				
session-group			No rat	e constraints to display. Please add.
session-recording-group				Add
session-recording-server				
session-translation	SIP Profile		•	
	Modify Session Agent			
access-control				
account-config	SIP Isup Profile		•	
filter-config	Kpml Interworking	inherit	•	
ldap-config	Kpml2833 lwf On Hairpin	inherit	•	
local-policy	Precedence	0		(Range: 04294967295)
local-routing-config	Monitoring Filters			
media-profile	Auth Attribute			
session-agent				
session-group				
session-recording-group				\cup
session-recording-server			No a	uth attributes to display. Please add.
session-translation				Add
-				



filter-config	Session Recording Server	
ldap-config	Session Recording Required	enable
local-policy	Hold Refer Reinvite	enable
local-routing-config	Send TCP Fin	enable
media-profile	SIP Recursion Policy	•
session-agent	Sm Icsi Match For Invite	
session-group	Sm Icsi Match For Message	
session-recording-group	Sincs Match of Message	
session-recording-server	Ringback Trigger	none 🔻
session-translation	Ringback File	
	ОК	Back
Show All		

Figure 63: Session Agent for OnPrem PBX Cont.

Configuration View Configuration	Q			
access-control	Modify Session Agent			
account-config	Hostname	10.64.1.72		
filter-config	IP Address	10.64.1.72		
ldap-config	Port	5060		(Range: 0,102565535)
local-policy	State	✓ enable		
local-routing-config	App Protocol	SIP	•	
media-profile	Арр Туре		•	
session-agent	Transport Method	UDP	•]
session-group	Realm ID	PSTNGW	•	
session-recording-group	Egress Realm ID		•	•
session-recording-server	Description			
session-translation				

Figure 64: Session Agent for PSTN Gateway

	access-control	Modify Session Agent				
	account-config	Match Identifier				
ľ	filter-config					
	ldap-config					
	local-policy					
	local-routing-config			Noma	tch identifier to display. P	lease add.
	media-profile				Add	
	session-agent					
	session-group	Associated Agents				
	session-recording-group	Constraints				
	session-recording-server	Lonstraints	enat	ble		
	session-translation	Max Sessions	0		(Range: 0999999999)	
	*					
	access-control	Modify Session Agen	nt			
	account-config	Max Outbound Sessions		0	(Range: 09999	99999)
	filter-config	Max Burst Rate		0	(Range: 09999	99999)
	Idan config	Max Inbound Burst Rate		0	(Range: 09999	99999)
	loap-conng	Max Outbound Burst Rate		0	(Range: 09999	99999)
	local-policy	Max Sustain Rate		0	(Range: 09999	99999)
	local-routing-config	Max Inbound Sustain Rate		0	(Range: 09999	99999)
	media-profile	Max Outbound Sustain Rate		0	(Range: 09999	99999)
	session-agent	Min Asr		0	(Range: 0100)	
	session-group	Cac Trap Threshold		0	(Range: 099)	
		Session Max Life Limit		0		
	session-recording-group	Time To Resume		0	(Range: 09999	99999)
	session-recording-server	In Service Period		0	(Range: 09999	99999)
	session-translation	Burst Rate Window		0	(Range: 09999	99999)

Figure 65: Session Agent for PSTN Gateway Cont.

response-map	Modify Sossian Agent		
rph-policy	Sustain Date Window		
rph-profile	Sustain Rate window	0	(Range: 0999999999)
service-health	Proxy Mode		•
	Redirect Action		•
session-agent	Loose Routing	✓ enable	
session-agent-id-rule	Response Map		•
session-constraints	Ping Method	OPTIONS	
session-group	Ping Interval		(
session-recording-group	Ping Send Mode	30	(Kange: 04294907295)
session-recording-ser	Ding All Addresses	keep-alive	•
	Ping All Addresses	enable	
session-router	Ping In Service Response Codes		
session-timer-profile 🔹			
access-control	Modify Session	Agent	
account-config	SPL Options		
filter-config	Media Profiles		
ldap-config	In Translationid		-
local-policy	Out Translationid		•
local-routing-config	Trust Me	enable	
media-profile	Local Response Map		•
session-agent	Ping Response	✓ enable	
session-group	In Manipulationid		•
session-recording-group	Out Manipulationid		-
session-recording-server	Manipulation String		
session-translation	Manipulation Pattern		

Figure 66: Session Agent for PSTN Gateway Cont.

	▲		
access-control	Modify Session Agent		
account-config	Trunk Group		
filter-config	Max Register Sustain Rate	0	(Range: 0999999999)
ldap-config	Invalidate Registrations	enable	
local-policy	Rfc2833 Mode	none	•
local-routing-config	Rfc2833 Payload	0	(Range: 0,96127)
media-profile	Codec Policy		•
session-agent	Refer Call Transfer	disabled	•
session-group	Refer Notify Provisional	none	•
session-recording-group	Reuse Connections	NONE	•
session-recording-server	TCP Keepalive	none	•
session-translation	TCP Reconn Interval	0	(Range: 0,2300)
access-control	Modify Session Agent		
account-config	Max Register Burst Rate 0		(Range: 099999999)
filter-config	Rate Constraints		
ldap-config			
local-policy			\bigcirc
local-routing-config			\mathbf{U}
media-profile		No ra	te constraints to display. Please add.
session-agent			Add
session-group			
session-recording-group	SIP Profile	•	
session-recording-server	SIP Isup Profile	v	
session-translation	Kpml Interworking inh	erit 💌	

Figure 67: Session Agent for PSTN Gateway Cont.

access-control	Modi	ify Session Agent				
account-config	Kpml28	833 lwf On Hairpin	inherit			
filter-config	Preced	ence	0		(Range: 04294967295)	
ldap-config	Monito	oring Filters				
local-policy	Auth A	ttribute				
local-routing-config						
media-profile					\bigcirc	
session-agent					(i)	
session-group				No au	th attributes to display	y. Please add
session-recording-group					Add	
session-recording-server						
filter-config		Session Recording	Server			
ldap-config		Session Recording	Required	enable		
local-policy		Hold Refer Reinvite	9	enable		
local-routing-config		Send TCP Fin		enable		
media-profile		SIP Recursion Polic	y			•
session-agent		Sm Icsi Match For I	nvite			
session-group	а.	Sm losi Match For I	Mossago			
session-recording-group		Sinconvacanton	Hessage			
session-recording-server		Ringback Trigger		none		•
session-translation		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.

session-router	Modify Local Polic	у
access-control	From Address	* **
account-config		
filter-config	To Address	972 🗙
ldap-config	Source Realm	cucm 🗙 onprem 🗙
local-policy		PSTNGW 🗙
local-routing-config	Description	PSTNGW
media-profile		
session-agent		
session-group	State	🖌 enable
session-recording-group	Policy Priority	none 🔻

Figure 69: Local Policy towards Google Voice

Configuration	View Configuration	Q				
media-manager	•	Modify Local policy / policy attribute				
security	•			_		
session-router	-	Next Hop	siplink.telephony.goog	v		
access-control		Realm	Google	•		
account-config		Action	none	•		
filter-config		Terminate Recursion	enable			
ldap-config		Cost	0	(Range: 0999999999)		
local-policy		State	✓ enable			
local-routing-con	fig	App Protocol		•		
media-profile		Соокир	single	•		
session-agent		Next Key				

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.

session-router 🔹		Modify Local Policy	
access-control		From Address	*•
account-config			
filter-config	ш	To Address	972: ×
ldap-config			
local-policy		Source Realm	Google 🗙 PSTNGW 🗙
local-routing-config		Description	
media-profile			
session-agent			
session-group	ы.	State	✓ enable
session-recording-group		Policy Priority	none 🔻

media-manager	•	Modify Local policy	y / policy attribute		
security	•				
session-router	•	Next Hop	172.16.29.53	•	
access-control		Realm	onprem	•	
account-config		Action	none	•	
filter-config		Terminate Recursion	enable		
ldap-config		Cost	0	((Range: 0999999999)
local-policy		State	🗸 enable		
		App Protocol		•	
local-routing-config		Lookup			
media-profile			single	•	
session-agent		Next Key			

Figure 71: Local Policy towards OnPrem PBX

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 🔻	Modify Local Policy	
access-control	From Address	* X
account-config	To Address	0119199 × 214 ×
filter-config		800 x +1214 x +1800 x
ldap-config		+1866 × +1888 ×
local-policy		+9199 ×
local-routing-config	Source Realm	Google 🗙 Onprem 🗙
media-profile	Description	
session-agent		
session-group	State	
session-recording-group	State	✓ enable



media-manager	×	•	Modify Local policy / policy attribute				
security	•					-	
session-router	•		Next Hop	10.64.1.72	•		
access-control			Realm	PSTNGW	•		
account-config			Action	none	•	_	
filter-config			Terminate Recursion	enable			
ldap-config			Cost	0		(Range: 0999999999))	
local-policy			State	✓ enable			
local-routing-config			App Protocol		•		
media-profile			Lookup	single	•		
session-agent			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	Q					
local-routing-co	nfig	Modify	SIP N	lanipulation			
media-profile		Name			GoogleManipulation		
session-agent	session-agent		n		Maninulations on google side		
session-group				Hamparatons on BooBie side			
session-recordin	ng-group						
session-recordin	ng-server	Split Head	ers				
session-translati	ion	Join Head	ers				
sip-config		CfraDules					
sip-feature		Add	•	/ 6 0	\uparrow \downarrow		
sip-interface		Action	Sel	Name			Element Type
sip-manipulation	n	:		changeReqUri			header-rule
sip-monitoring		:		changeFromIP			header-rule
	Fig	gure 7	5: SI	P Manipula	ation towards Goog	le 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

local-routing-config	Modify Sip manipulation /	header rule		
media-profile	Name	changeReqUri		
session-agent	Header Name	Request-URI		
session-group	Action	manipulate	•	
session-recording-group	Comparison Type	pattern-rule	•	
session-recording-server	Msg Туре	out-of-dialog	•	
session-translation	Methods			
sin-feature	Match Value			
sip-interface	New Value			
sip-manipulation	CfgRules			
sip-monitoring	Action Sel Name	ΤΨ		Element Type
translation-rules	ReqURI			element-rule

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

local-routing-config	Modify Sip mar	nipulation / header rule / element rule
media-profile	Name	Real IRI
session-agent	Parameter Name	Regord
session-group	Type	
session-recording-group	Action	uri-host 💌
session-recording-server	Match Val Type	replace 🔻
session-translation	Comparison Time	any 🔻
sip-config	Comparison Type	case-insensitive 🔻
sip-feature	Match Value	
sip-interface	New Value	"trunk.sip.voice.google.com"

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

local-routing-config	Modify Sip manipulation /	' header rule	
media-profile	Name	changeFromIP	
session-agent	Header Name	FROM	
session-group	Action	manipulate 💌	
session-recording-group	Comparison Type	pattern-rule 💌	
session-recording-server	Msg Туре	request 💌	
session-translation	Methods	INVITE ×	
sip-config	Match Value		
sip-feature	New Value		
sip-interface	CfeRules		
sip-manipulation	Add v C i	ī ↑ ↓	
sip-monitoring	Action Sel Name		Element Type
translation-rules	: changelP		element-rule
svstem			

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

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

local-routing-config	Modify Sip manipulati	Modify Sip manipulation / header rule / element rule			
media-profile	Name	changelP			
session-agent	Parameter Name				
session-group	Туре	uri-host	•		
session-recording-group	Action	replace	•		
session-recording-server	Match Val Type	anv	•		
session-translation	Comparison Type	pattern-rule	•		
sip-config	Match Value				
sip-feature	New Value	\$LOCAL_IP			
sip-interface		_			

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

1 1 1 1	Madif. Ciamania dation	(
local-routing-config	Modify Sip manipulation /	r neader rule		
media-profile	Name	changeToIP		
session-agent	Header Name	то		
session-group	Action	manipulate	•	
session-recording-group	Comparison Type	pattern-rule	•	
session-recording-server	Msg Type	request	•	
session-translation	Methods			
sip-config	Match Value			
sip-feature	Match Value			
sip-interface	New Value			
sip-manipulation	CfgRules			
sip-monitoring	Action Sel Name			Element Type
translation-rules	: changelP			element-rule

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

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

local-routing-config	Modify Sip manipulat	Modify Sip manipulation / header rule / element rule			
media-profile	Name	changelP			
session-agent	Parameter Name				
session-group	Туре	uri-host	•		
session-recording-group	Action	replace	•		
session-recording-server	Match Val Type	any	-		
session-translation	Comparison Type	pattern-rule	-		
sip-config	Match Value				
sip-feature	New Value	\$REMOTE_IP			
sip-interface					

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.

sip-config	•	Modify Sip manipulation ,	/ header rule
sip-feature			
sip-interface		Name	AddXGoogleheader
cin manipulation		Header Name	X-Google-Pbx-Trunk-Secret-Key
sip-manipulation		Action	add 💌
sip-monitoring		Comparison Type	case-insensitive
sti-server		Msg Type	
translation-rules			request 💌
system	-	Methods	INVITE X OPTIONS X
fraud protection		Match Value	
naud-protection		New Value	"ffd57
host-route			1007
http-client		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.

session-recording-group	Modify Sip manipulation ,	/ header rule
session-recording-ser	Name	storepai
session-translation	Header Name	P-Asserted-Identity
sip-config	Action	store 💌
sip-feature	Comparison Type	case-insensitive 🔻
sip-interface	Msg Type	request 💌
sip-manipulation	Methods	INVITE X
sip-monitoring	Match Value	
sti-server	New Value	(<sip:)(.*)(@.*)< td=""></sip:)(.*)(@.*)<>
translation-rules		
system	CfgRules	

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

local-routing-config	Modify Sip manipulation / header rule			
media-profile	Name	addPAI		
session-agent	Header Name	P-Asserted-Identity		
session-group	Action	add 💌		
session-recording-group	Comparison Type	case-sensitive 🔻		
session-recording-server	Msg Type	request 💌		
sin-config	Methods	INVITE ×		
sip-feature	Match Value			
sip-interface	New Value	\$storepai.\$1+"+1"+\$storepai.\$2+"@sb		

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.

Configuration View Configuration	Q		
local-routing-config	Modify Sip manipulation /	header rule	
media-profile	Name	CheckForPrivacy	
session-agent	Header Name	Privacy	
session-group	Action	manipulate 💌	
session-recording-group	Comparison Type	boolean 👻	
session-translation	Msg Type	request 💌	
sip-config	Methods	INVITE X	
sip-feature	Match Value		
sip-interface	New Value		

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

▲ local-routing-config	Modify Sip manipulation /	header rule	
media-profile	Name	OverwriteFromDisplay	
session-agent	Header Name	From	
session-group	Action	manipulate 💌	
session-recording-group	Comparison Type	boolean 💌	
session-recording-server	Msg Туре	request 💌	
session-translation	Methods		I
sip-feature	Match Value	\$CheckForPrivacy	
sip-interface	New Value		
sip-manipulation	CfgRules		
sip-monitoring	Add V C 1	$\uparrow \downarrow$	Element Ture
translation-rules	OverwriteUser		element-rule

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

session-recording-server	Modify Sip manipul	ation / header rule / elem	ent rule
session-translation	Name	OverwriteUser	
sip-config	Parameter Name		
sip-feature	Туре	uri-user	
sip-interface	Action	replace	
sip-manipulation	Match Val Type	201	-
sip-monitoring	Comparison Type		•
translation-rules	Match Value	case-sensitive	•
/stem 🔻	New Value		
fraud-protection	Them value	anonymous	

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

SIP manipulation towards PSTN Gateway

sip-feature-caps	Modify SIP Manipu	lation	
sip-interface	Name	TowardsPSTN	
sip-manipulation	Description	TowardsPSTN	
sip-monitoring			
sip-nat	Split Headers		
sip-profile			
sip-q850-map	Join Headers		
sip-recursion-policy	CfgRules		

Figure 87: SIP Manipulation towards PSTN Gateway

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

	Modify Sip manipula	tion / header rule	
sip-feature-caps		,	
sip-interface	Name	changeFromIP	
sip-manipulation	Header Name	From	
sip-monitoring	Action	manipulate	•
sip-nat	Comparison Type	case-sensitive	•
sip-profile	Msg Type	request	T
sip-q850-map	Methods		
sip-recursion-policy	Match Value		
surrogate-agent	New Value		
survivability	CfgRules		

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

session-recording-server	Modify Sip mani	Modify Sip manipulation / header rule / element rule			
session-translation	Name	changeFromIP			
sip-config	Parameter Name				
ip-feature	Туре	uri-bost			
ip-interface	Action	replace			
ip-manipulation	Match Val Type	теріасе			
p-monitoring	Comparison Type	any			
anslation-rules	Match Value	case-sensitive			
tem 🔻	New Value	RECORNED			
raud-protection		SLUCAL_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	Modify Sip manipulatio	n / header rule
sip-interface	Name	changeToIP
sip-manipulation	Header Name	То
sip-monitoring	Action	manipulate 🔹
sip-nat	Comparison Type	case-sensitive 🔹
sip-profile	Msg Type	request 💌
sip-q850-map	Methods	
sip-recursion-policy	Match Value	
surrogate-agent	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



Create Physical Interface

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

steering-pool	•	Modify Phy Interface		
security	•			•
session-router	•	Name	wancom1	
system	•	Operation Type	Control 🔹	
fraud-protection		Port	1	(Range: 05)
host-route		Slot	0	(Range: 02)
		Virtual Mac		
http-client		Admin State	✓ enable	
http-server		Auto Negotiation	🖌 enable	
network-interface	- 11	Duplex Mode		
ntp-config		Speed		
phy-interface		Wancom Health Score	8	(Range: 0.100)
- redundancy-config	-		-	(

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, SOPO and S1PO are configured as shown below.

system 🔻	Modify Netwo	ork Interface				
fraud-protection	Name					
host-route	Sub Port Id		wancom		•	(Dames: 0, 4005)
http-client	Description		0			(Range: 04095)
http-server						
network-interface						
ntp-config	Hostname					
phy-interface	IP Address					
redundancy-config	Pri Utility Addr		10.80.11.82			
snmp-community	Sec Utility Addr		10.80.11.83			
spl-config	Netmask		255.255.255	.0		
system 💌	Modify Network Inte	erface				
fraud-protection	Gateway					
host-route	Gw Heartbeat					
http-client	State	enable	e			
http-server	Heartbeat	0		(R	ange: 06553	35)
network-interface	Retry Count	0		(R	ange: 06553	35)
ntp-config	Retry Timeout Health Score	1		(R	ange: 16553	5)
phy-interface	DNS IP Primary	0		(K	ange. 0100	
redundancy-config	DNS IP Backup1					
snmp-community	DNS IP Backup2					
spl-config	Modify Notwork Interface					
fraud-protection	DNS Domain					
host-route	DNS Timeout	11		(Range: 0429496)	7295)	
http-client	DNS Max Ttl	86400		(Range: 30207360	00)	
http-server	Signaling Mtu	0		(Range: 0,576409	6)	
network-interface	HIP IP List					
ntp-config	ICMP Address					
phy-interface	SSH Address					

Figure 94: Modify Network-Interface for Wancom1

system 💌	Modify Network Int	Modify Network Interface					
fraud-protection	Name	s0p0	v				
host-route	Sub Port Id	0	(Range: 04095)				
http-client	Description						
http-server							
network-interface							
ntp-config	Hostname	sbc3.tekvizionlabs.com					
phy-interface	IP Address	192.65.					
redundancy-config	Pri Utility Addr	192.65.					
snmp-community	Sec Utility Addr	192.65.					
spl-config	Netmask	255.255.255.128					

Figure 95: Modify Network-Interface of SOPO

system 🔻	Modify Network Interface			
fraud-protection	Name	s1p0	-	
host-route	Sub Port Id	0		(Range: 04095)
http-client	Description			
http-server				
network-interface				
ntp-config	Hostname			
phy-interface	IP Address	10.80.11.21		
redundancy-config	Pri Utility Addr	10.80.11.20		
snmp-community	Sec Utility Addr	10.80.11.69		
spl-config	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.

Configuration View Configuration	on Q						
system 💌	Modify	Redu	ndancy Con	fig			
fraud-protection							
host-route	State			✓ enable			
http-client	Log Level			INFO	•		
http-server	Becoming	Standby	Time	180000		(Range: 52147483647)	
natwork interface	Becoming	Active T	ime	100		(Range: 52147483647)	
network-interface	Media If Pe	ercheck	Time	0		(Range: 0500)	
ntp-config	Peers						
phy-interface	D,	/ [È 🗇				
redundancy-config	Action	Sel	Name		State		Туре
snmp-community	:		sbc01		enabled		Primary
spl-config	:		sbc02		enabled		Secondary
system config							

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



Modify Redundancy config / peer						
Name			sbc01			
State			✓ enable			
Туре			Primary 🔻			
Destinatio	ons					
D;	/ T	- III				
Action	Sel	Address		Network Interface		
:		10.80.11.82:9090		wancom1:0		

Figure 98: Configure SBC1 as Primary

Modify Redundancy config / peer

Name			sbc02		
State			🖌 enable		
Туре			Secondary	•	
Destinatio	ns				
D:	/ T				
Action	Sel	Address			Network Interface
:		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.

local-routing-config	Modify SIP Interface		
media-profile	Nat Interval	30	(Range: 04294967295)
session-agent	TCP Nat Interval	90	(Range: 04294967295)
session-group	Registration Caching	enable	
session-recording-group	Min Reg Expire	300	(Range: 0999999999)
session-recording-server	Registration Interval	3600	(Range: 04294967295)
session-translation	Route To Registrar	enable	
sip-config	Secured Network	enable	
sip-feature	Uri Fqdn Domain		
sip-interface	Options		
sip-manipulation	SPL Options		

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.



8 Summary of Tests and Results

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
Inbou	nd				
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 conversati on 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 conversati on 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 conversati on 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 conversati on 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 conversati on 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 Conferenc e		
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 successfull y 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 conversati on 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 Conferenc e		
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 successfull y 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.	conversati on should be possible. The GV BYOT User should be able to perform functions like Hold Transfer Conferenc e		
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 successfull y 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
		terminate call on a GV user Client. Ensure 2 way voice.	client endpoint should ring and upon answering the call 2 way audio conversati on should be successful.		
1.18	BYOT Phone Number > Auto Attendant > Ring Group > 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 Ring Group which will terminate call on a GV user Client. 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	
1.19	PBX Phone Number > PBX User > transfer to BYOT Phone Number > Auto Attendant > Ring Group >	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	The GV BYOT User should be able to answer the call and 2 way conversati on should be	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 Conferenc e		
1.20	PBX Phone Number > PBX User > transfer to BYOT Phone Number > Auto Attandant > 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 successfull y leave a voicemail as well as able to exercise the various dtmf options in the voicemail menu	PASSED	
		Ou	tbound		
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. disconnect s 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	

1.26 GV User Originated > BYOT Trunk 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 Menu 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 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 clearly. PASSED 1.28 GV User Originated > BYOT Trunk Termination > PBX User > Voicemail 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 and navigate the IVR of the Voicemail PASSED	ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
1.27 GV User Originated > BYOT Trunk Termination > PBX User GV user will place a call to PBX User The GV User PASSED 1.28 GV User Originated > BYOT Trunk Termination > PBX User GV user will place a call to PBX User The GV way audio. PASSED 1.28 GV User Originated > BYOT Trunk Termination > PBX User > Voicemail GV user will place a call to PBX User The GV User PASSED 1.28 GV User Voicemail GV user will place a call to PBX User The GV User PASSED 1.28 GV User Voicemail GV user will place a call to PBX User The GV User PASSED BYOT Trunk Termination > PBX User > Voicemail GV user will place a call to PBX User The GV User PASSED Voicemail BYOT trunk. Leave a voicemail Voicemail able to leave a voicemail and navigate the IVR of the Voicemail. PASSED	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	
1.28 GV User GV user will place a The GV PASSED Originated > BYOT Trunk BYOT trunk. Leave a User should be Termination > PBX User > voicemail. leave a voicemail Voicemail voicemail. leave a voicemail with the PBX voicemail and and navigate the IVR of the Voicemail. Voicemail. voicemail. voicemail	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	

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	
		Inbound Call – Calling o	or Called Party	y Disconnects	
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 establishe d with bidirection al audio Verify call is disconnect ed 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 establishe d with bidirection al audio Verify call is disconnect ed properly	PASSED	
		Inbound Call – Calling o	or Called Party	y Disconnects	
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 establishe d with bidirection al audio Verify call is disconnect ed 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 establishe d with bidirection al audio Verify call is disconnect ed properly	PASSED	
		Terminate the	call before ar	nswer	
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.
		Caller ID	Restriction		
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 anonymou s	PASSED	
		Earl	y Media		

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.
		Long	duration		
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 bidirection al audio for more than 60 mins Bidirection al audio is present after session audit or session refresh invite from the DUT. Call is properly disconnect ed when either party disconnect s 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 bidirection al audio for more than 60 mins Bidirection al audio is	PASSED	

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

2.13 Long duration hold and resume (wait until session audit/session refresh occurs from DUT) GV user A Calls PSTN A. Call is connected and GV user A places the call on hold for 30 mins or until session audit occurs. Extension A resumes the call. PASSED connected and bidirection al audio before hold. DUT) GV user A Calls PSTN Answers the call on hold for 30 mins or until session audit occurs. Extension A resumes the call. Call is connected and bidirection al audio before hold. MOH is present after resuming the call MOH is present after session audit that occurred during hold. MOH is present after resume. Bidirection al audio is present after resume.	ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
2.13 Long duration hold and resume (wait until session audit/session refresh occurs from DUT) GV user A Calls PSTN A. Call is connected and bidirection al audio before hold. PASSED Dutto GV user A Calls PSTN A. Call is connected and bidirection al audio before hold. PASSED Dutto GV user A places the call on hold for 30 audit occurs. Extension A resumes the call. Bidirectional Audio is present after resuming the call MOH is heard during hold. If applicable. MOH is present after resuming the call MOH is present after session audit that occurred during hold. Bidirection al audio is present after resume. Bidirection al audio is present after				either party disconnect s the call.		
Call is properly disconnect ed when either party disconnect s 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 bidirection al audio before hold. MOH is heard during hold. If applicable. MOH is present after session audit that occurred during hold. Bidirection al audio is present after resume. Call is properly disconnect ed when either party disconnect s the call	PASSED	

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 disconnect ed when either party disconnect s the call	PASSED	
Do No	t Disturb				
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	
Simult	aneous Ring				
2.16	Simultaneous ring	PSTN A calls GV user A GV user sets simultaneous ring to PSTN B	PSTN B answers the call Bidirection al Audio is present	NOT SUPPORTED	Google Voice does not support Simultaneous ring.
Call Fo	orward				
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.
Toll Fr	ee				
2.19	Calls to 800 numbers (Toll Free Numbers)	GV user A Calls TOLL FREE number starts with 800	Verify Call is establishe d with bidirection al audio Verify call is disconnect ed properly	PASSED	
2.20	Calls to 877 numbers (Toll Free Numbers)	GV user A Calls TOLL FREE number starts with 877	Verify Call is establishe d with bidirection al audio Verify call is disconnect ed properly	PASSED	
2.21	Calls to 866 numbers (Toll Free Numbers)	GV user A Calls TOLL FREE number starts with 866	Verify Call is establishe d with bidirection al audio Verify call is disconnect ed properly	PASSED	
2.22	Calls to 888 numbers (Toll Free Numbers)	GV user A Calls TOLL FREE number starts with 888	Verify Call is establishe d with bidirection al audio Verify call is disconnect	PASSED	
ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
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			ed properly		
2.23	Call to 5551212	GV user A Calls 5551212	Verify Call is establishe d with bidirection al audio Verify call is disconnect ed 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 establishe d with bidirection al audio Verify call is disconnect ed properly	NOT SUPPORTED	Google Voice does not support Operator code dialing. Google updated that this feature is under work in progress.
Handli	ing E164 and Nor	n E164			
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 establishe d with bidirection al audio Verify call is disconnect ed properly	NOT SUPPORTED	Google Voice does not support Non E164. Google updated that this feature is under work in progress.
Intern	ational Dialing				

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 establishe d with bidirection al audio Verify call is disconnect ed properly	PASSED	
Caller	ID				
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	
Call W	aiting				
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 establishe d with bidirection al audio Verify Call waiting indication Verify MOH on PSTNA during hold Verify call is disconnect ed 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			
Handli	Handling 486 response							
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.			
Handli	ing Error Codes,	486, 4XX						
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				
Handli	ing Error Codes	603 Decline						
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.			
Codec	:							
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 establishe d with bidirection al 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 establishe d with bidirection al audio and G711 A law is negotiated Verify call is disconnect ed properly	PASSED	
Calls t	o 911, 411 and 51	1			
2.34	GV user Calls 911	GV user A Calls 911 911 operator answers the call.	Verify Call is connected and bidirection al 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 establishe d with bidirection al audio Verify call is disconnect ed properly	NOT SUPPORTED	Google Voice does not support 3-digit dialing. Google updated that this feature is under work in progress.

Property of tekVizionLabs Page 112

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 establishe d with bidirection al audio Verify call is disconnect ed 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 establishe d with bidirection al audio Verify call is disconnect ed properly	NOT TESTED	Test case is already executed with other short codes e.g. 411, 511 since US carrier is used for this setup.
Stir-Sl	haken				
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 identificati on header is present Verify bidirection al audio is present.	NOT TESTED	Stir-Shaken is not supported by the Service Provider used in this setup.
Loope	d Calls				
2.41	Looped Calls	BYOT Phone Number > Auto Attendant > BYOT User Termination 1 transfer	Ensure calls that may get looped	PASSED	

Property of tekVizionLabs Page 113

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 perspectiv e.	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 successfull y with bidirection al 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	НА	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/o utgoing 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	НА	Plug the WAN cable back for Primary SBC, verify the incoming/outgoing	Plug the WAN cable back for Primary	PASSED	

Property of tekVizionLabs Page 115

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/o utgoing 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	НА	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.		