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

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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.

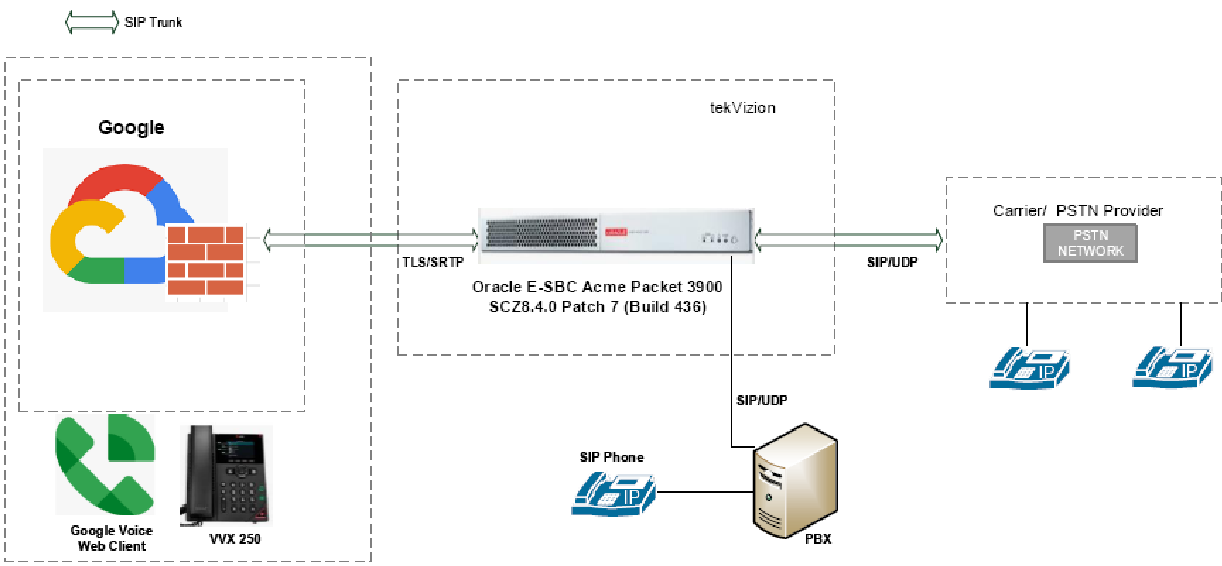


Figure 1: SIP Trunk Lab Reference Network

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.

Table 1 – Oracle E-SBC Configuration Steps

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

6.2 IP Address Worksheet

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

Table 3 - IP Address Worksheet

Component	IP Address
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

6.3 Google Voice SIP Link Configuration

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

6.4 Oracle E-SBC Configuration

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

6.4.1 Media Manager

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

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

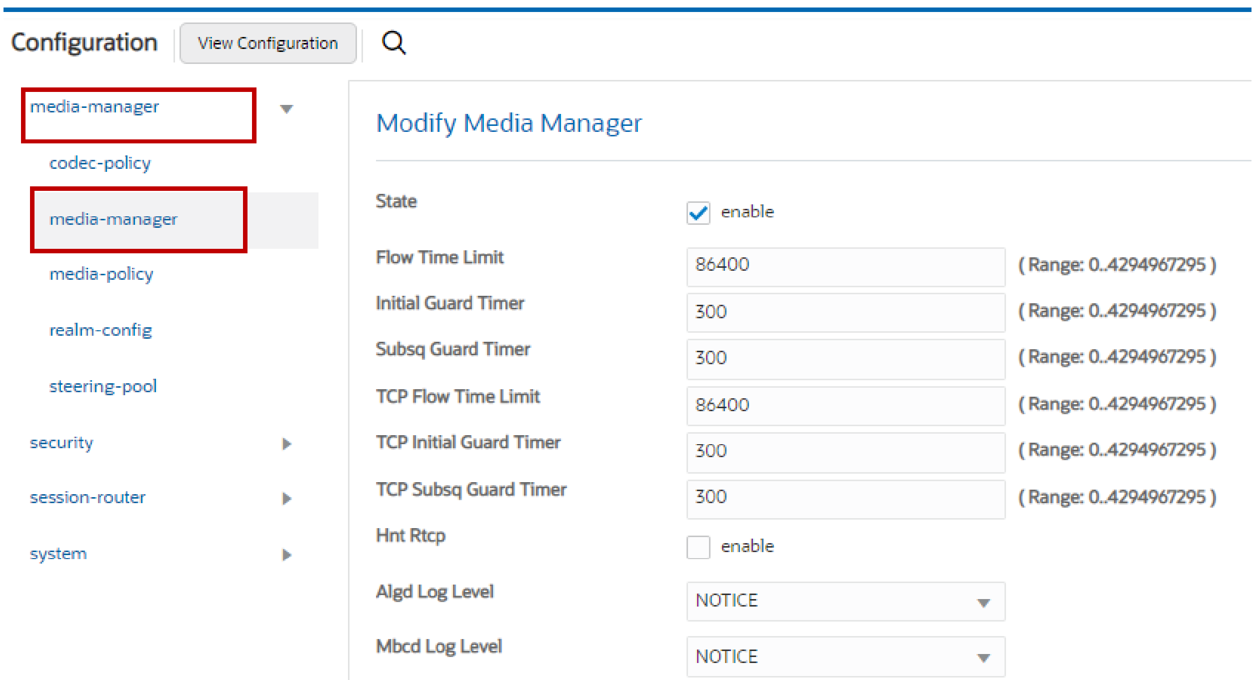


Figure 2: Media Manager

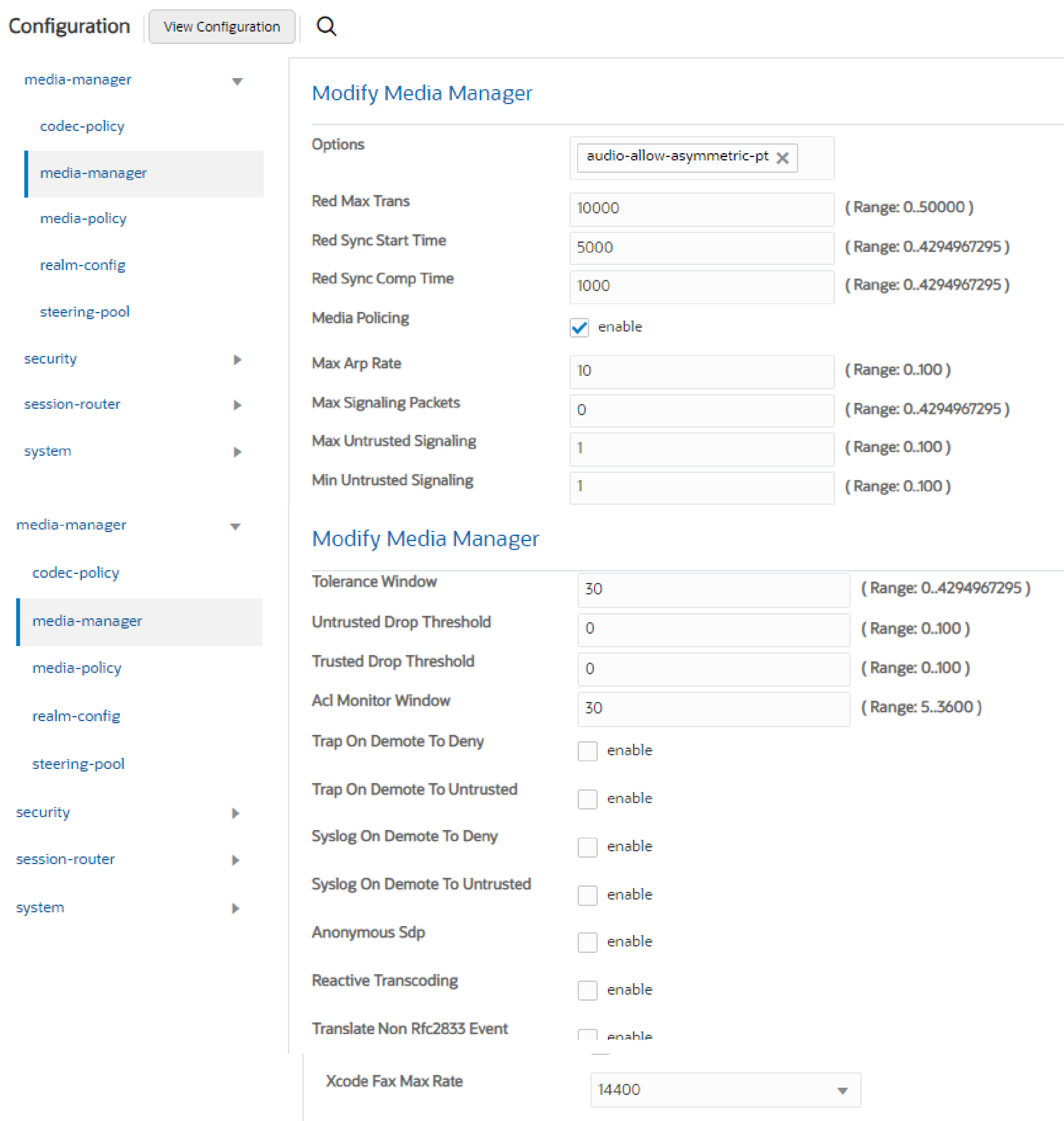


Figure 3: Media Manager Cont.

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).

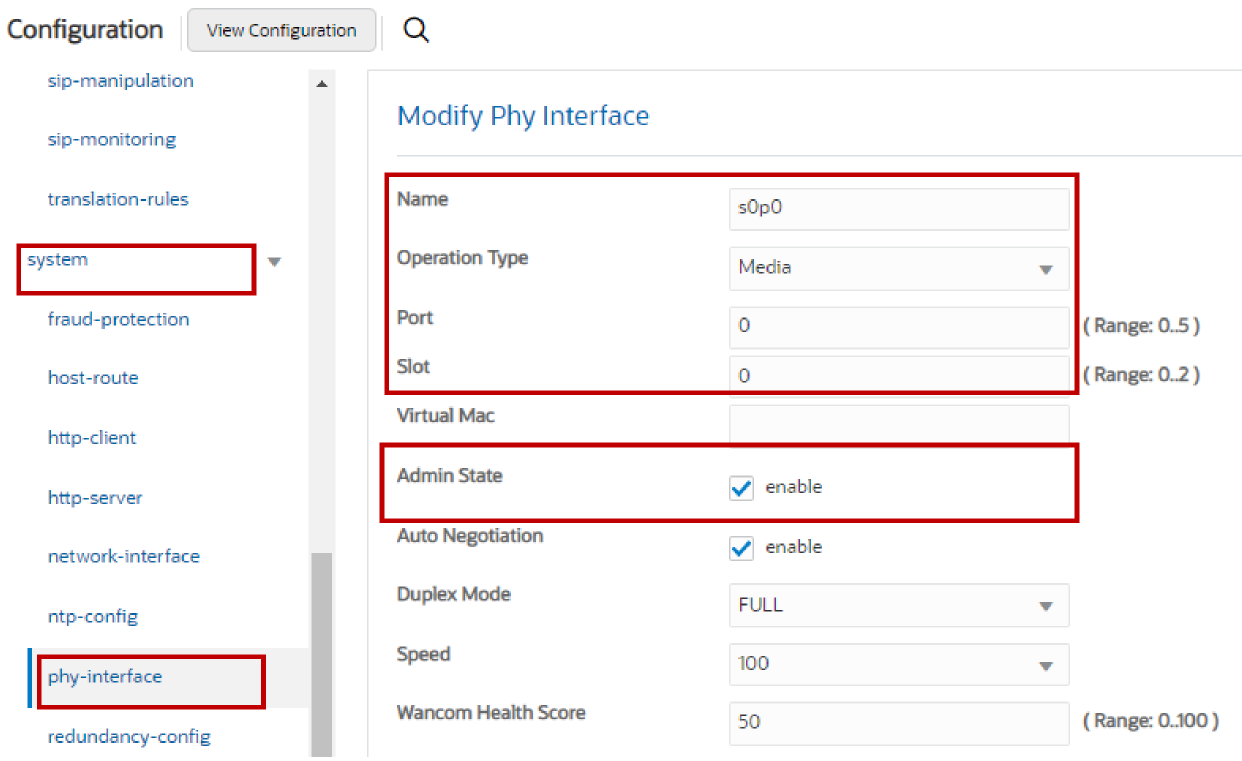


Figure 4: Physical Interface towards Google Voice

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

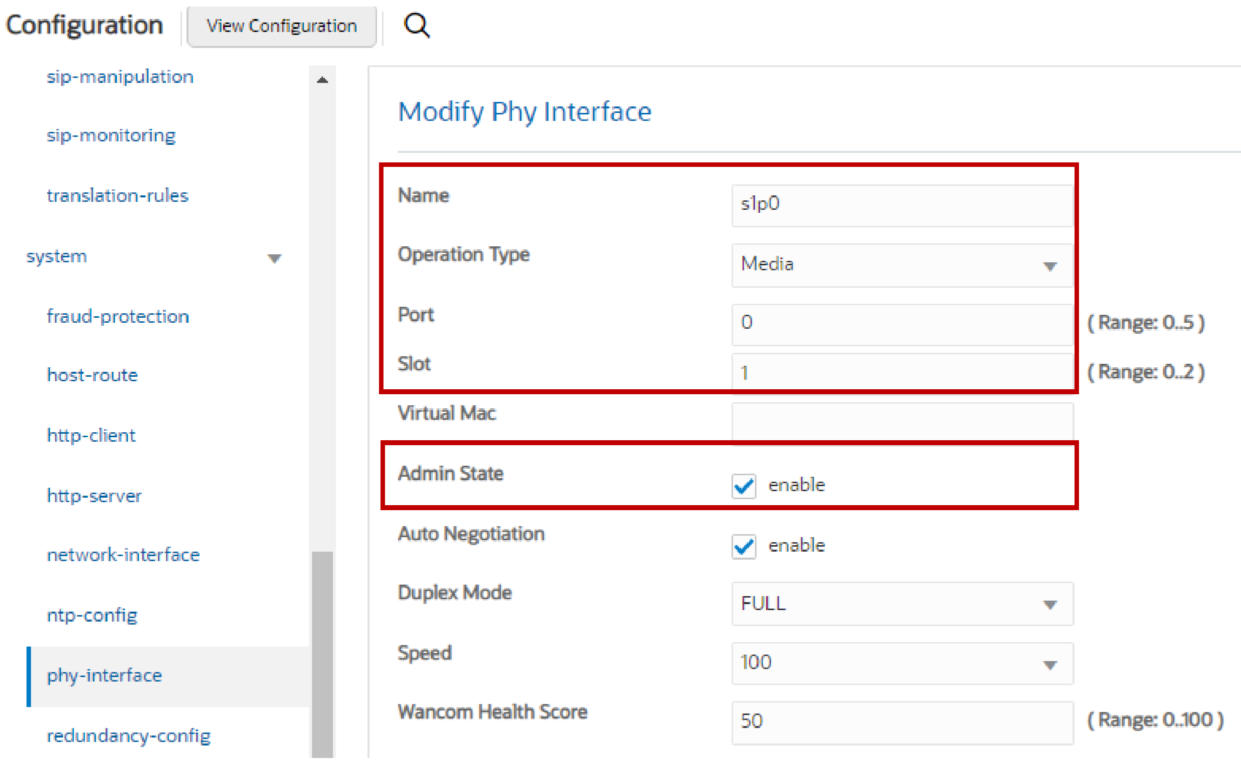


Figure 5: Physical Interface towards PSTN Gateway and OnPrem PBX

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.

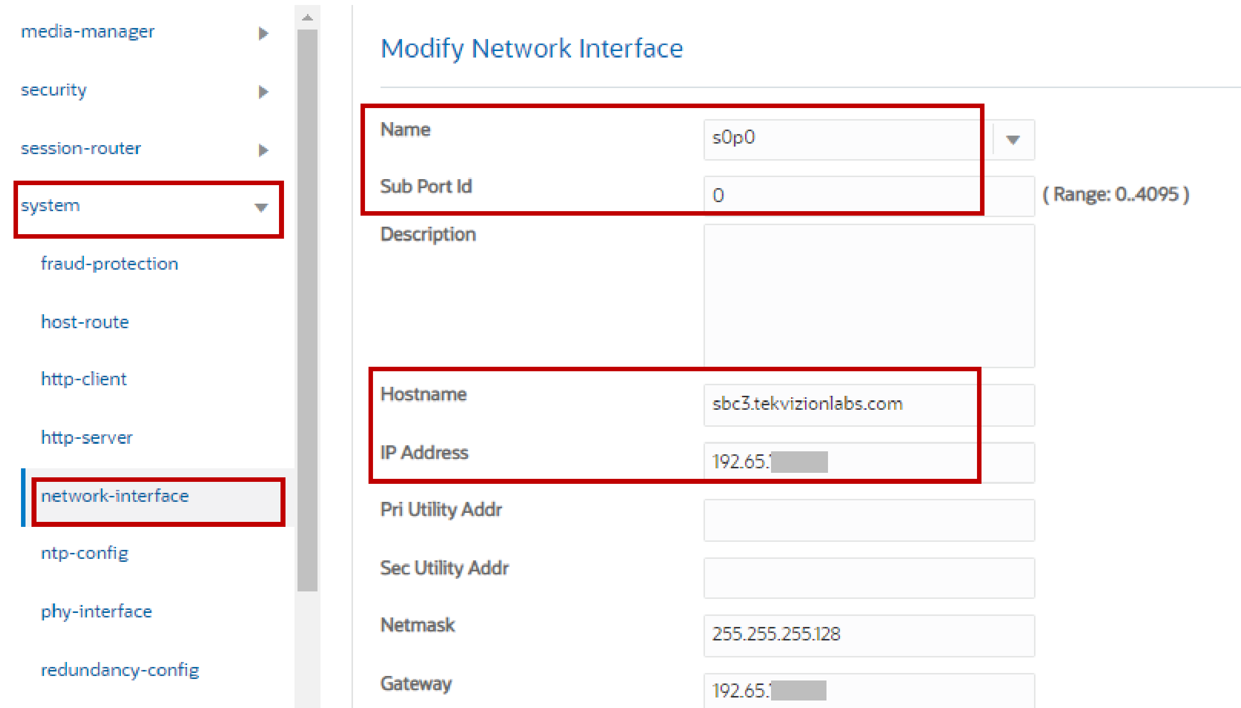


Figure 6: Network Interface towards Google Voice

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

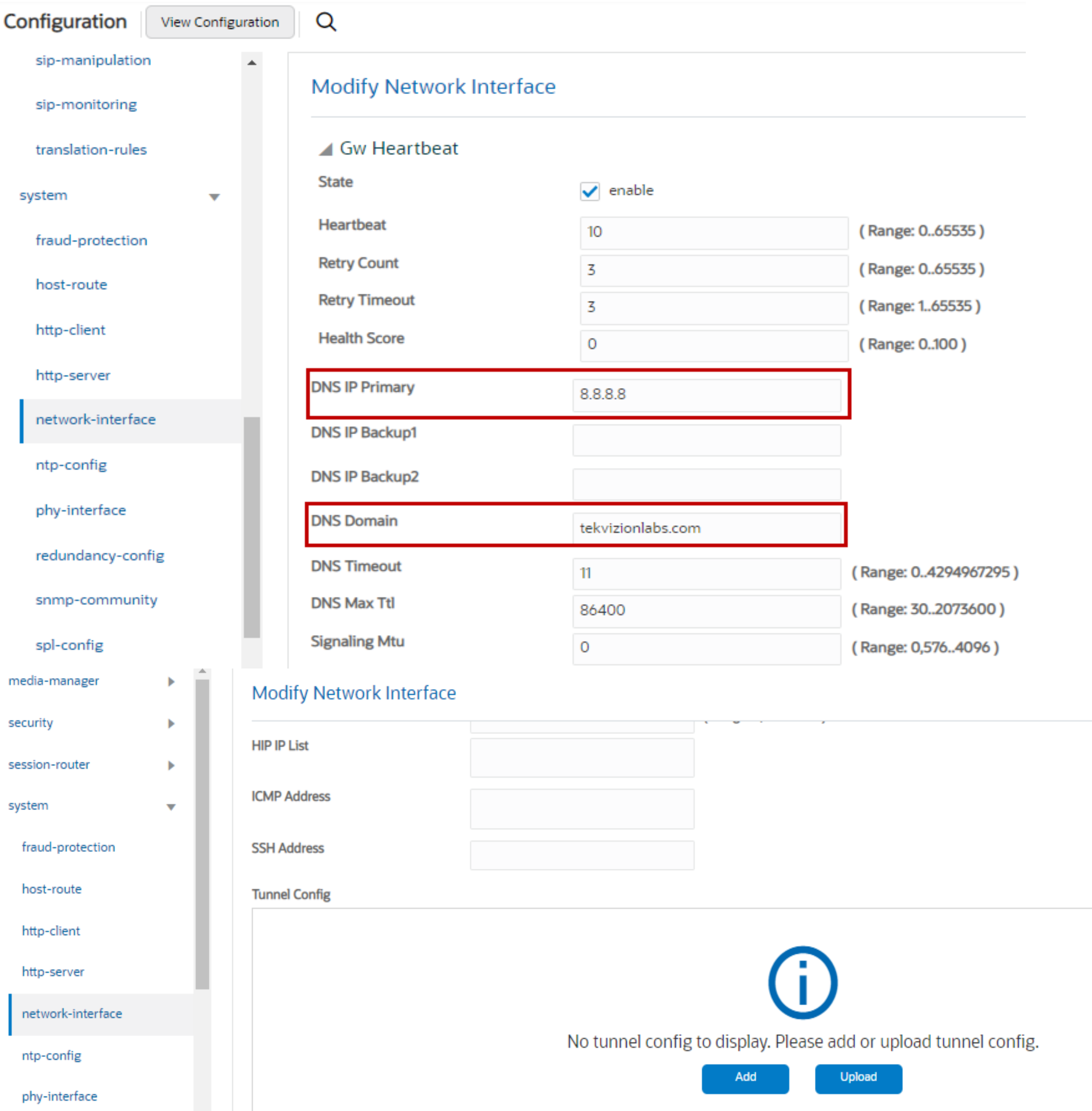


Figure 7: Network Interface towards Google Voice Cont.

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

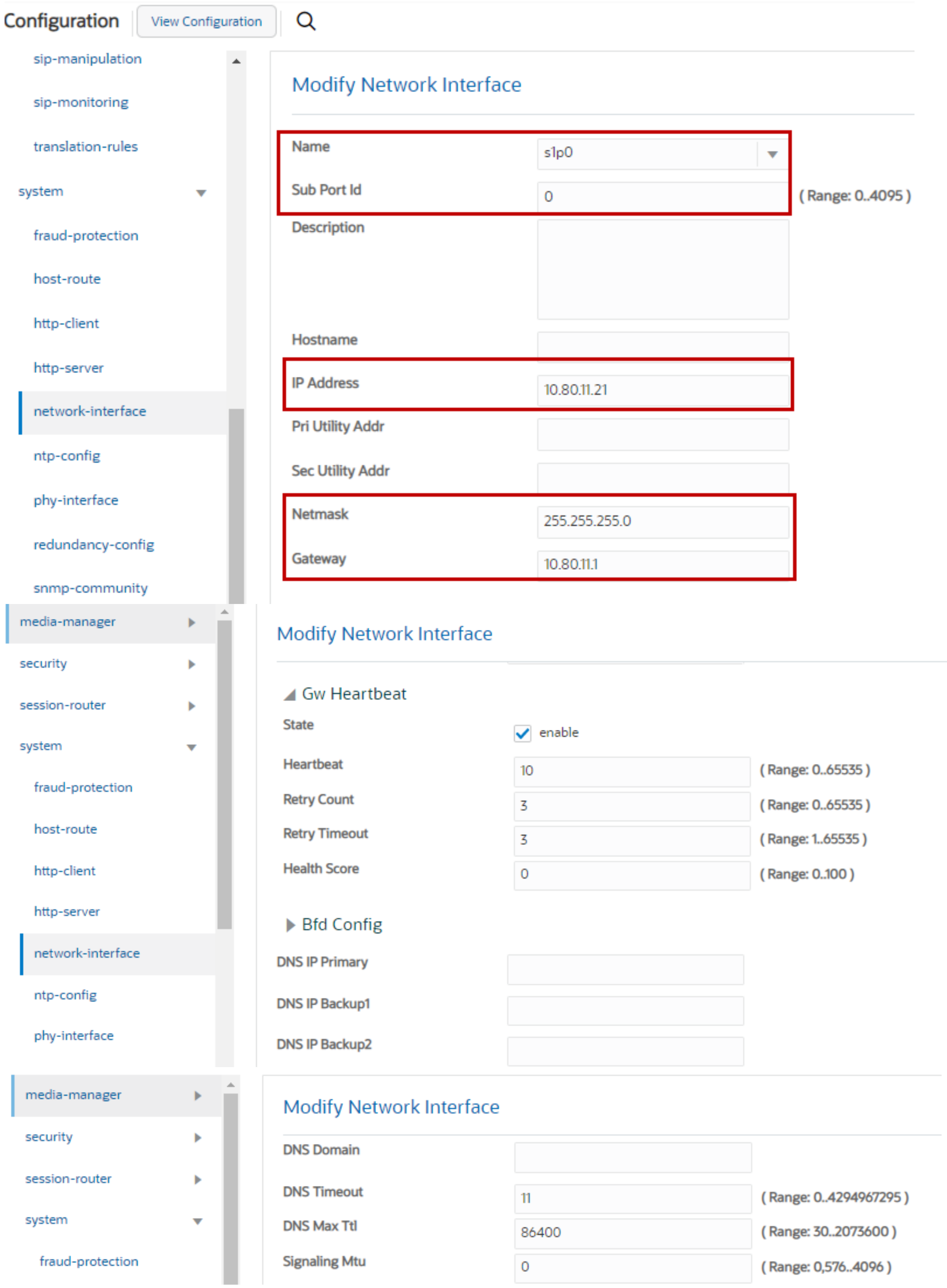


Figure 8: Network Interface towards PSTN Gateway

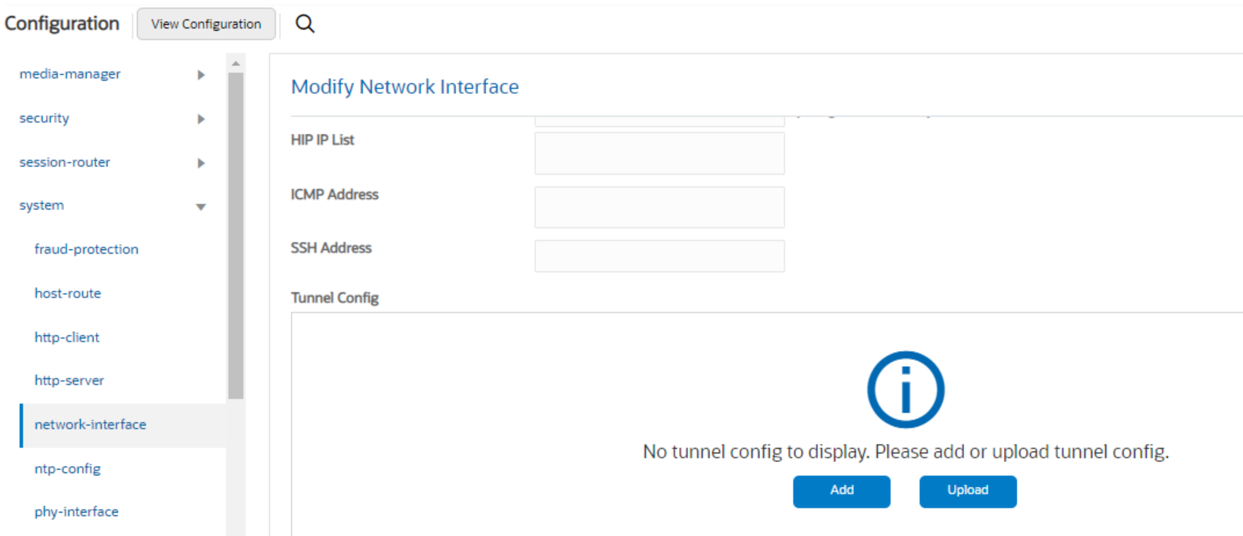


Figure 9: Network Interface towards PSTN Gateway Cont.

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

6.4.4 Codec Policy

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

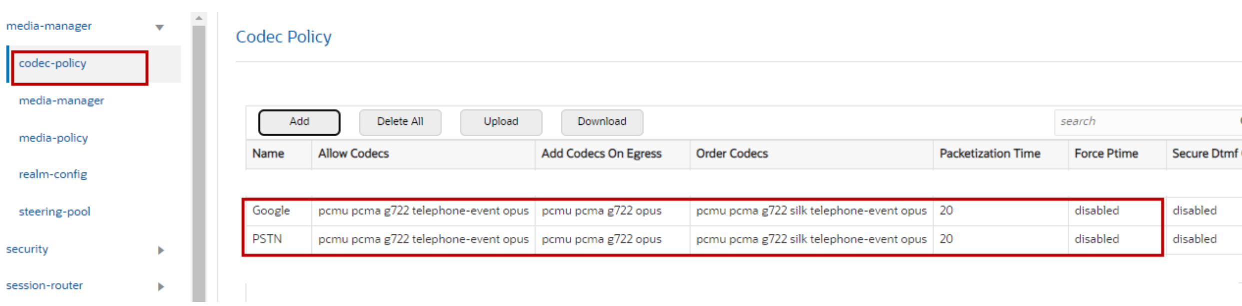


Figure 10: Codec Policy for Google Voice and PSTN Gateway

6.4.5 Translation Rules

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

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

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

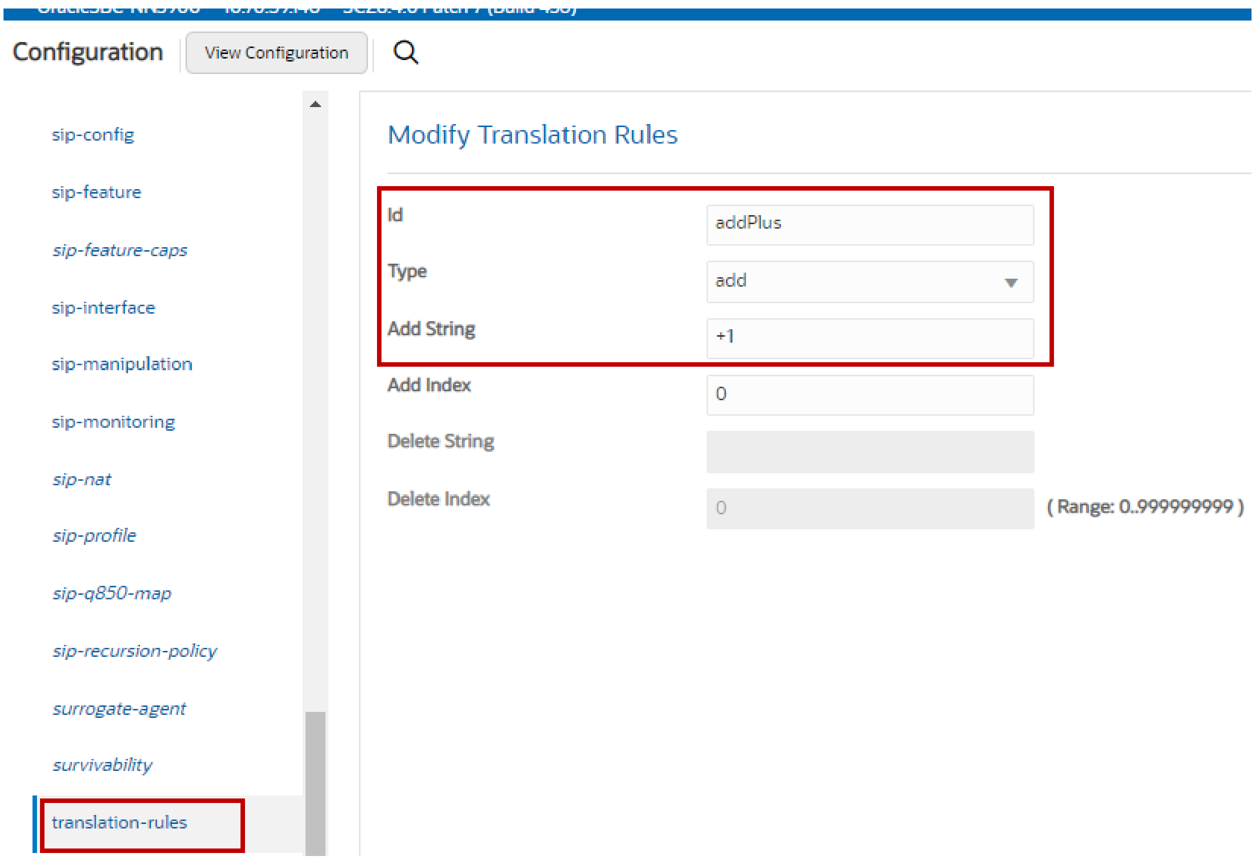


Figure 11: Translation Rule to add send E.164 towards Google Voice

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

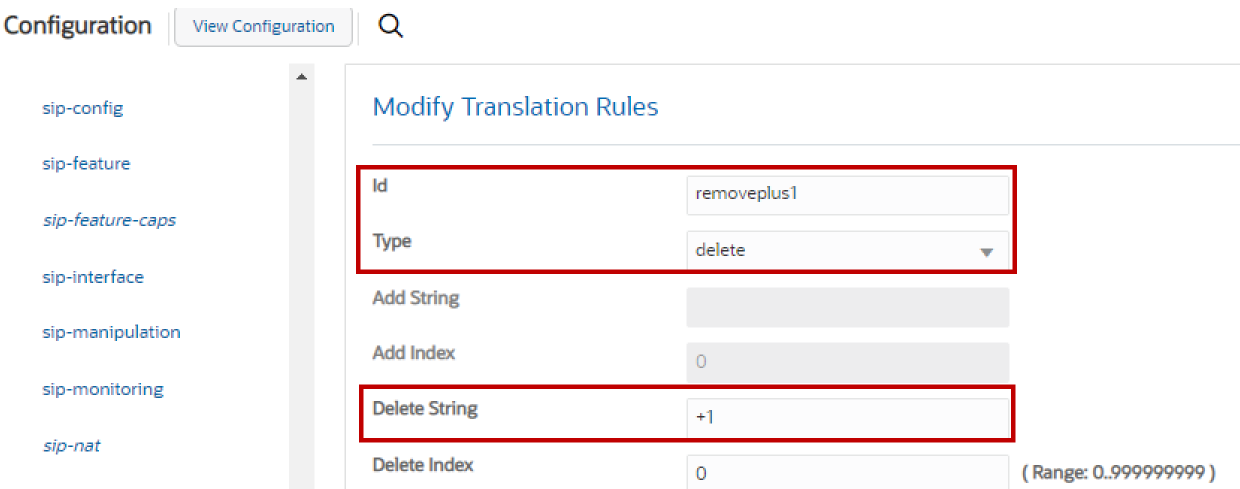


Figure 12: Translation to send non E.164 towards PSTN Gateway

6.4.6 Session Translation

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

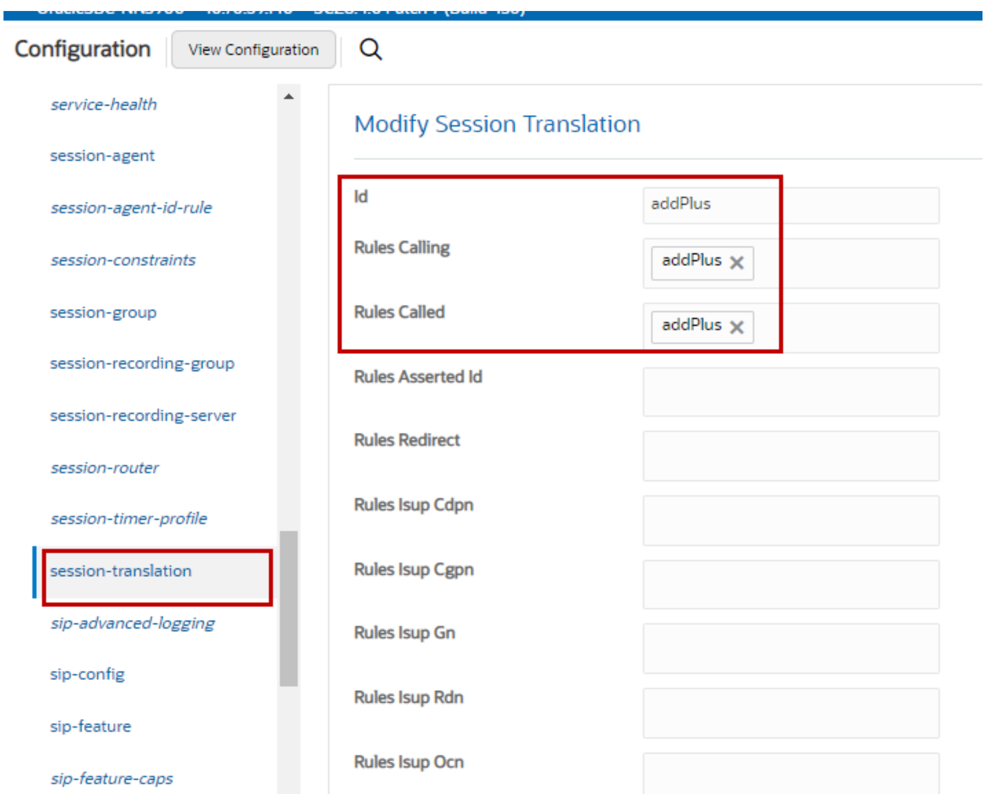


Figure 13: Session Translation towards Google Voice

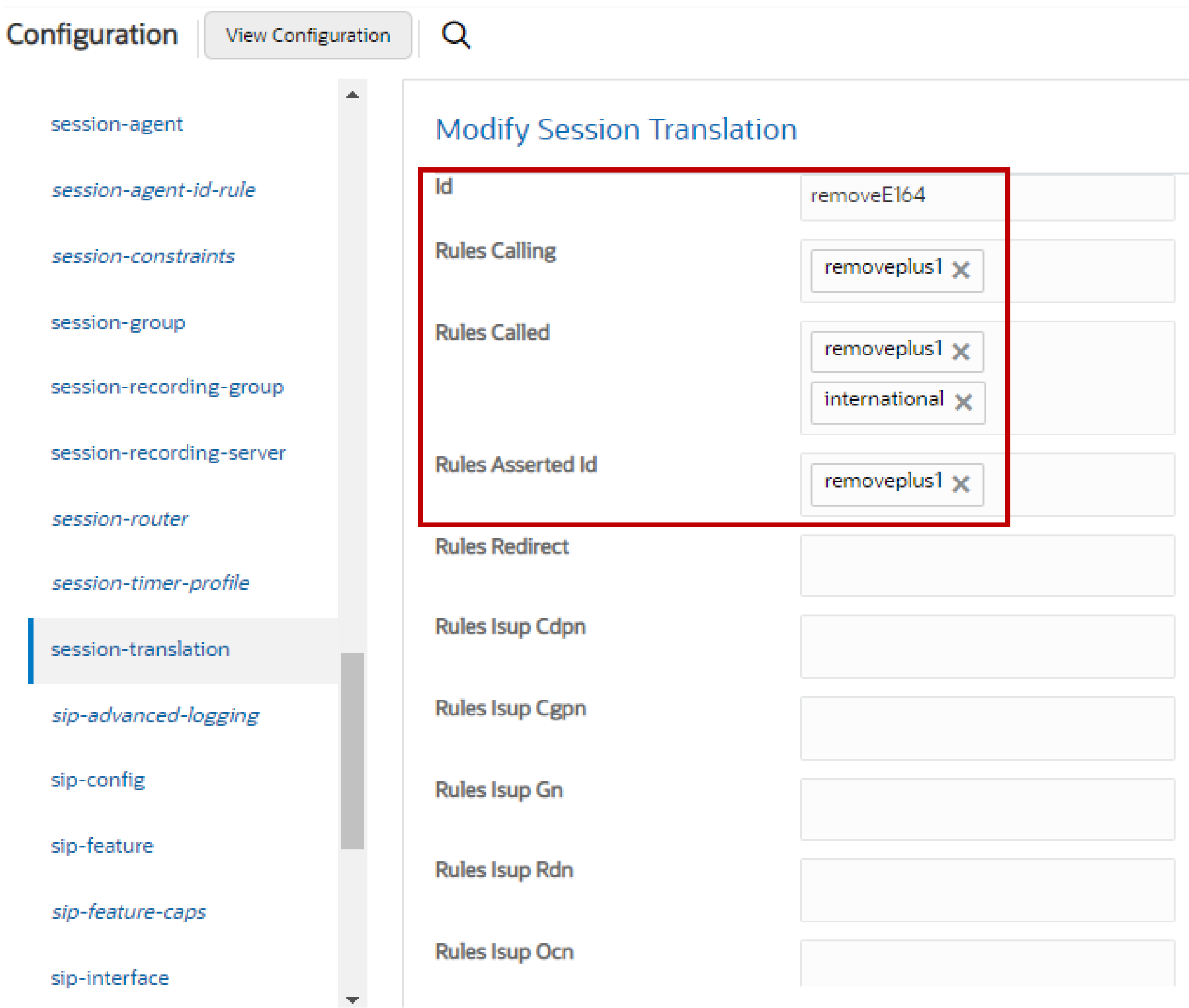


Figure 14: Session Translation towards PSTN Gateway

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.

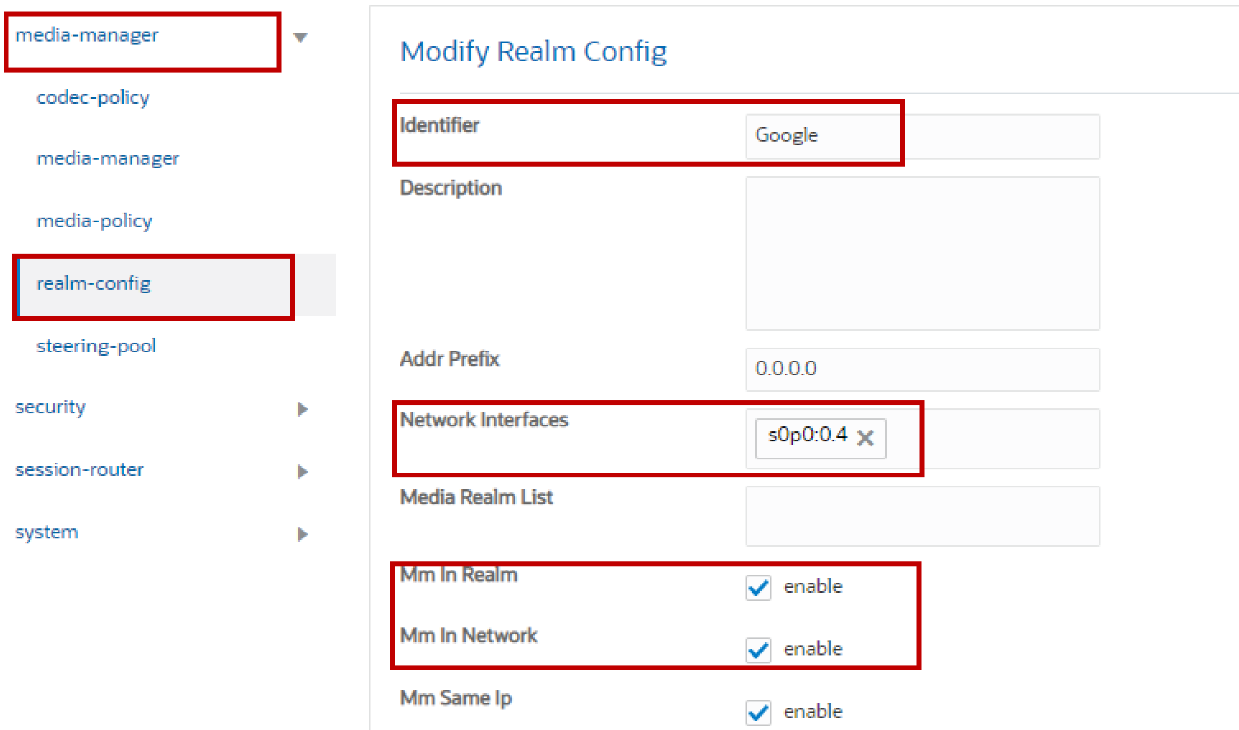


Figure 15: Realm Config towards Google Voice

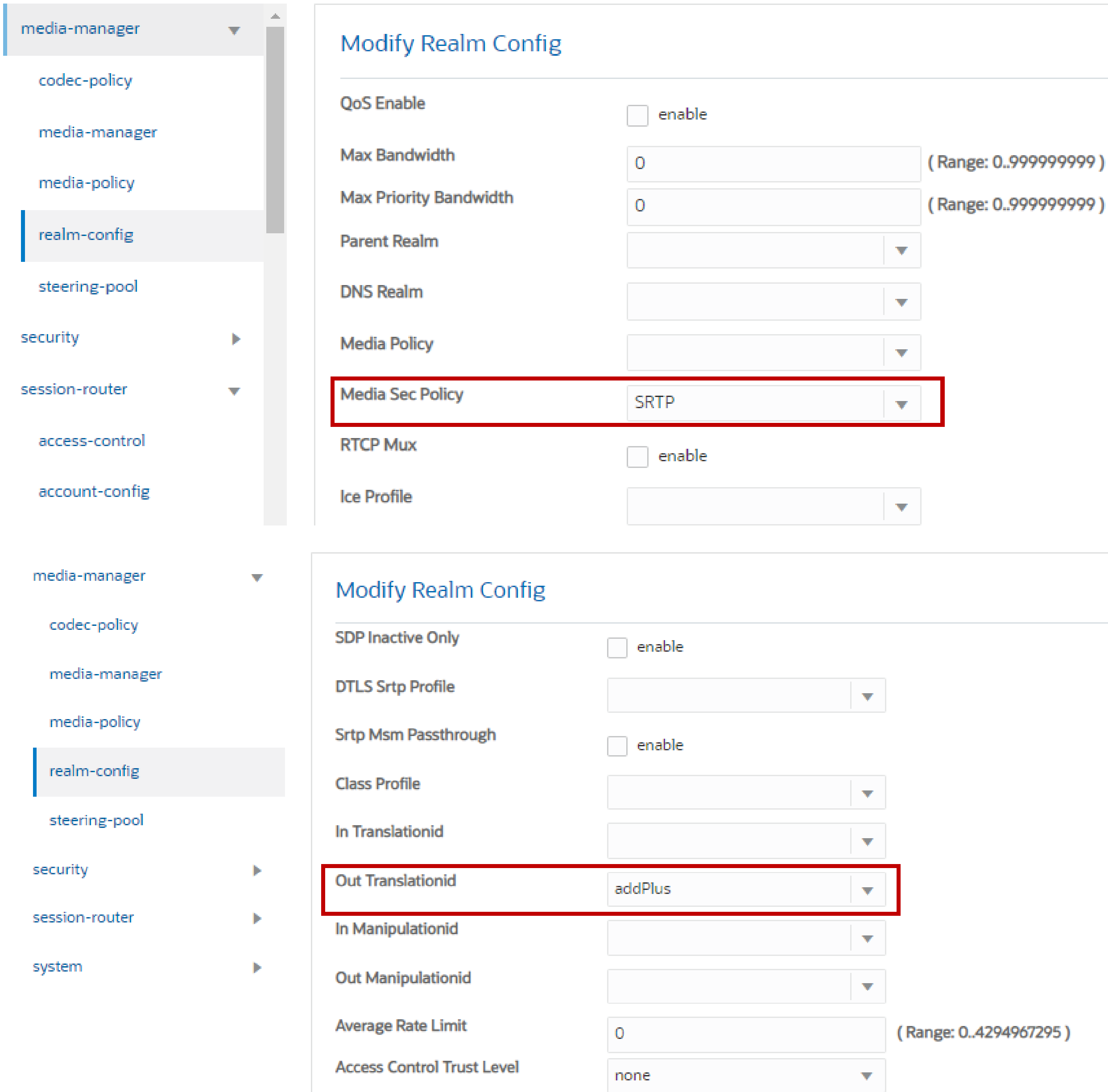


Figure 16: Realm Config towards Google Voice cont.

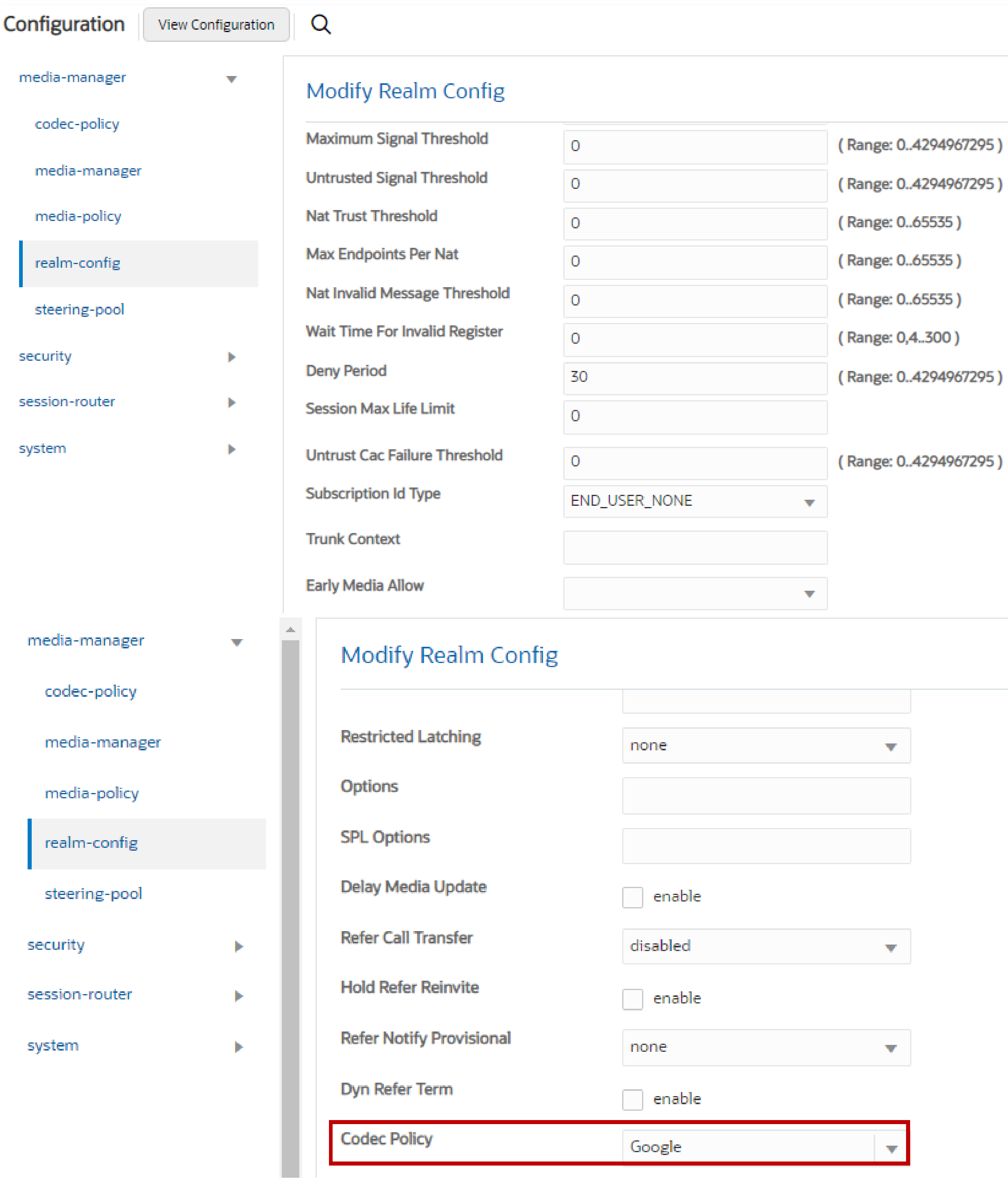


Figure 17: Realm Config towards Google Voice Cont.

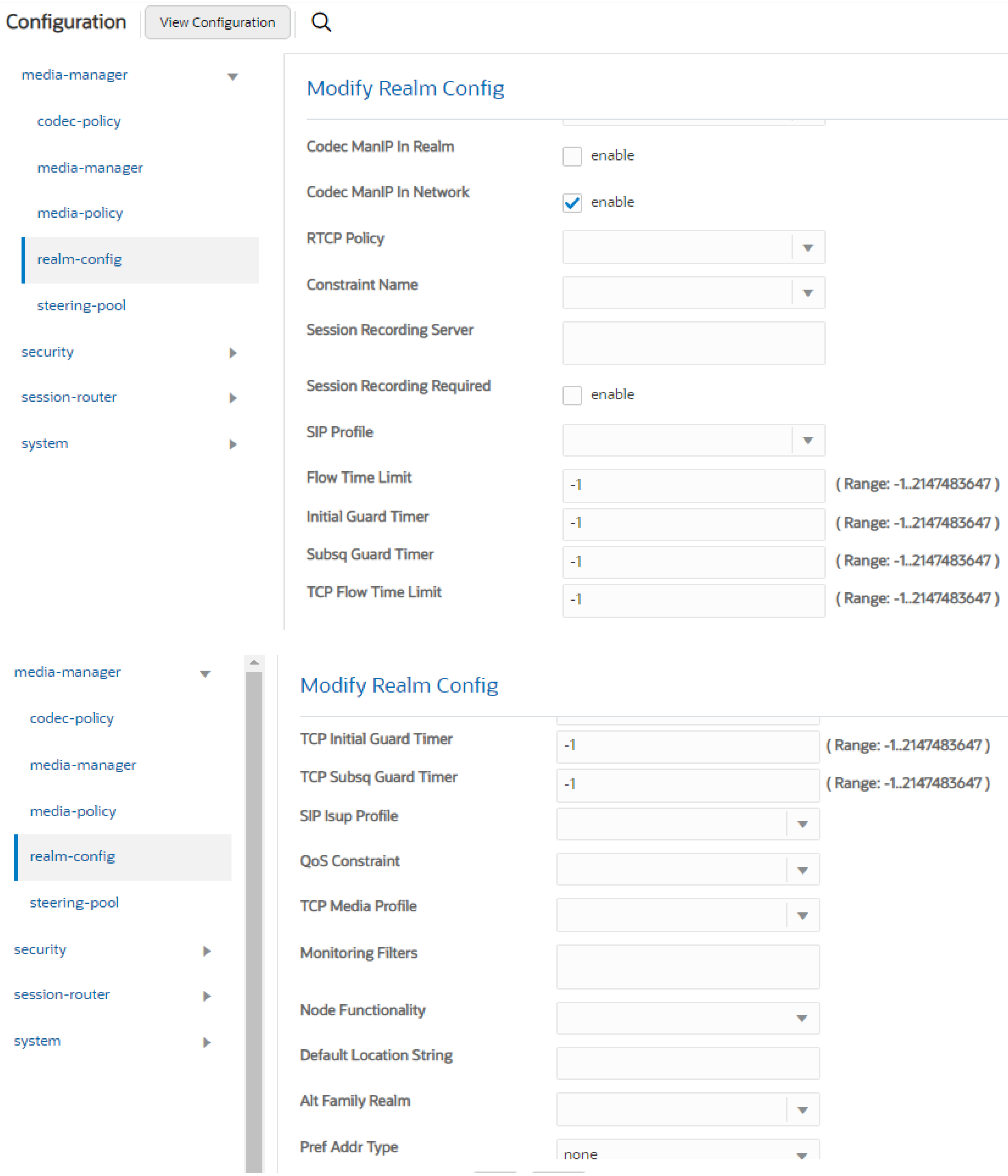


Figure 18: Realm Config towards Google Voice Cont.

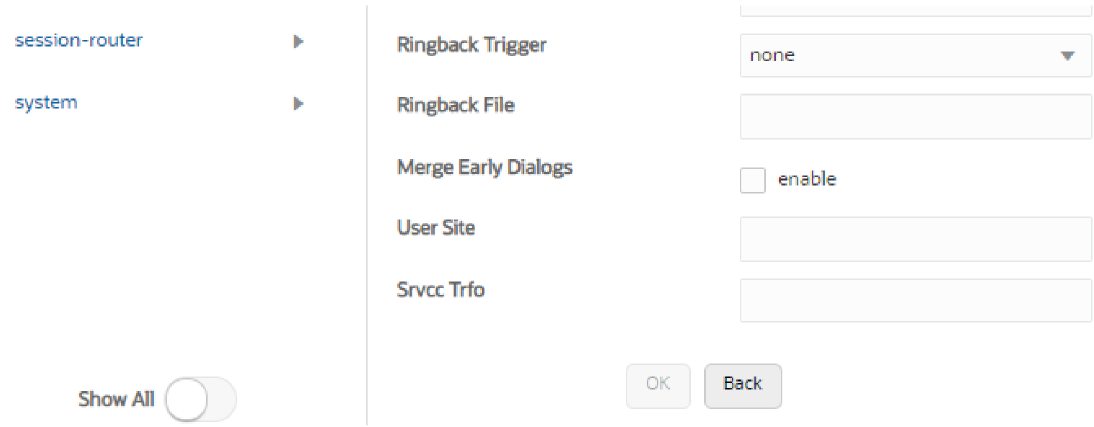


Figure 19: Realm Config towards Google Voice Cont.

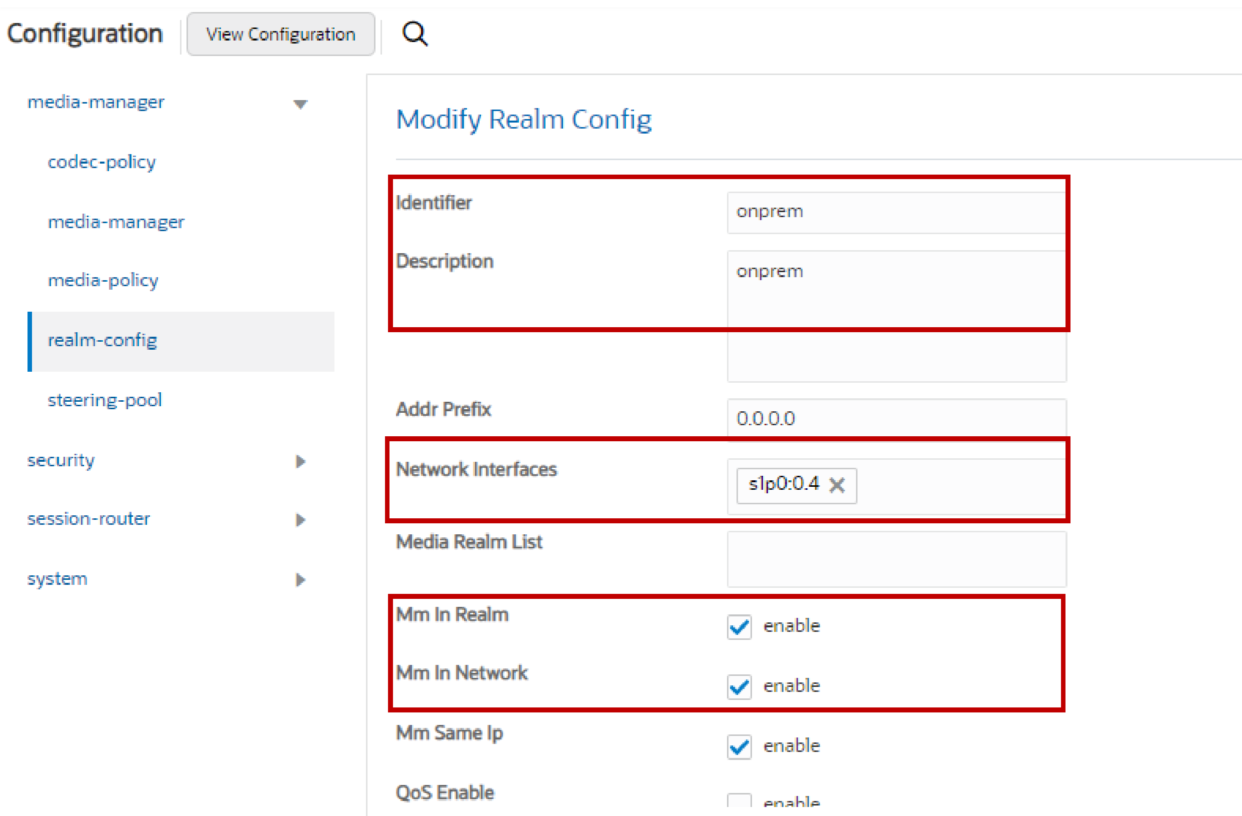


Figure 20: Realm Config towards OnPrem PBX

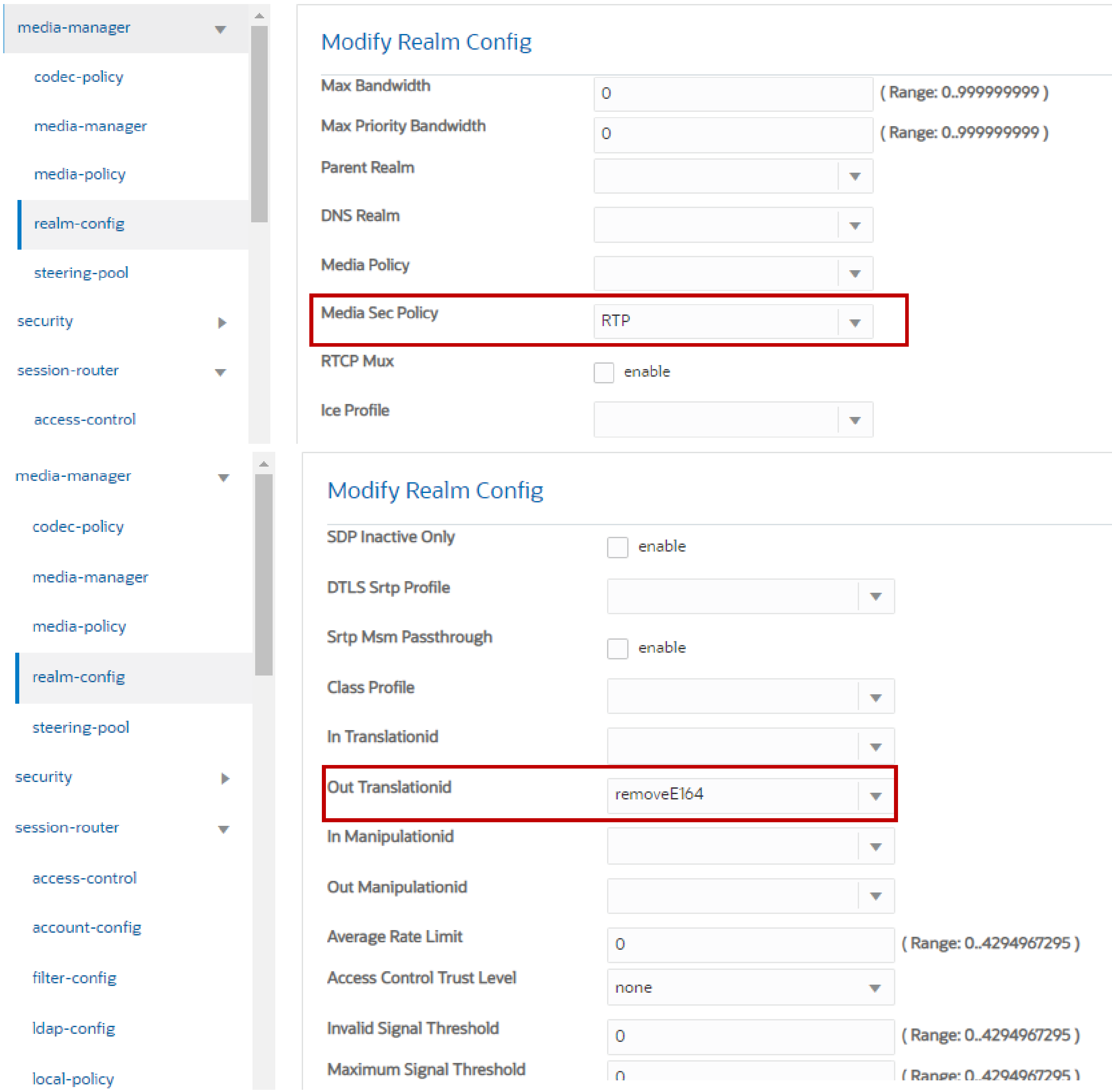
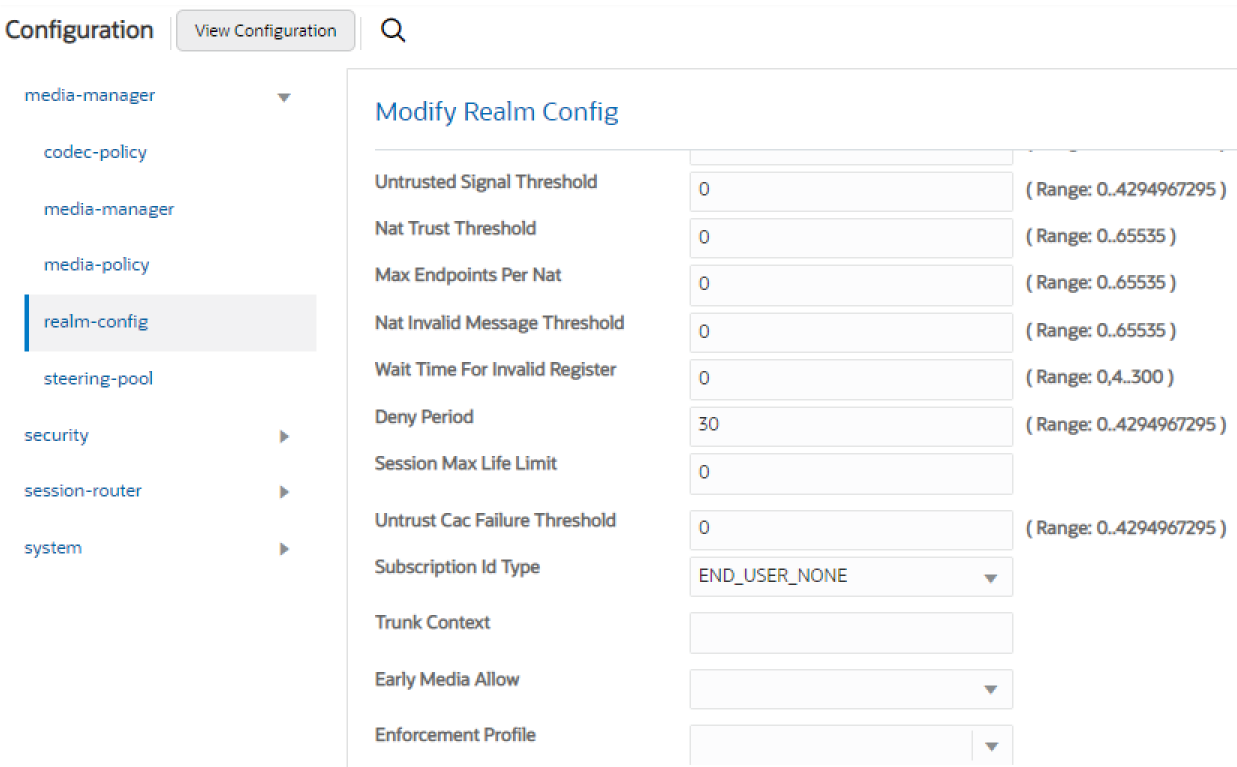


Figure 21: Realm Config towards OnPrem PBX Cont.



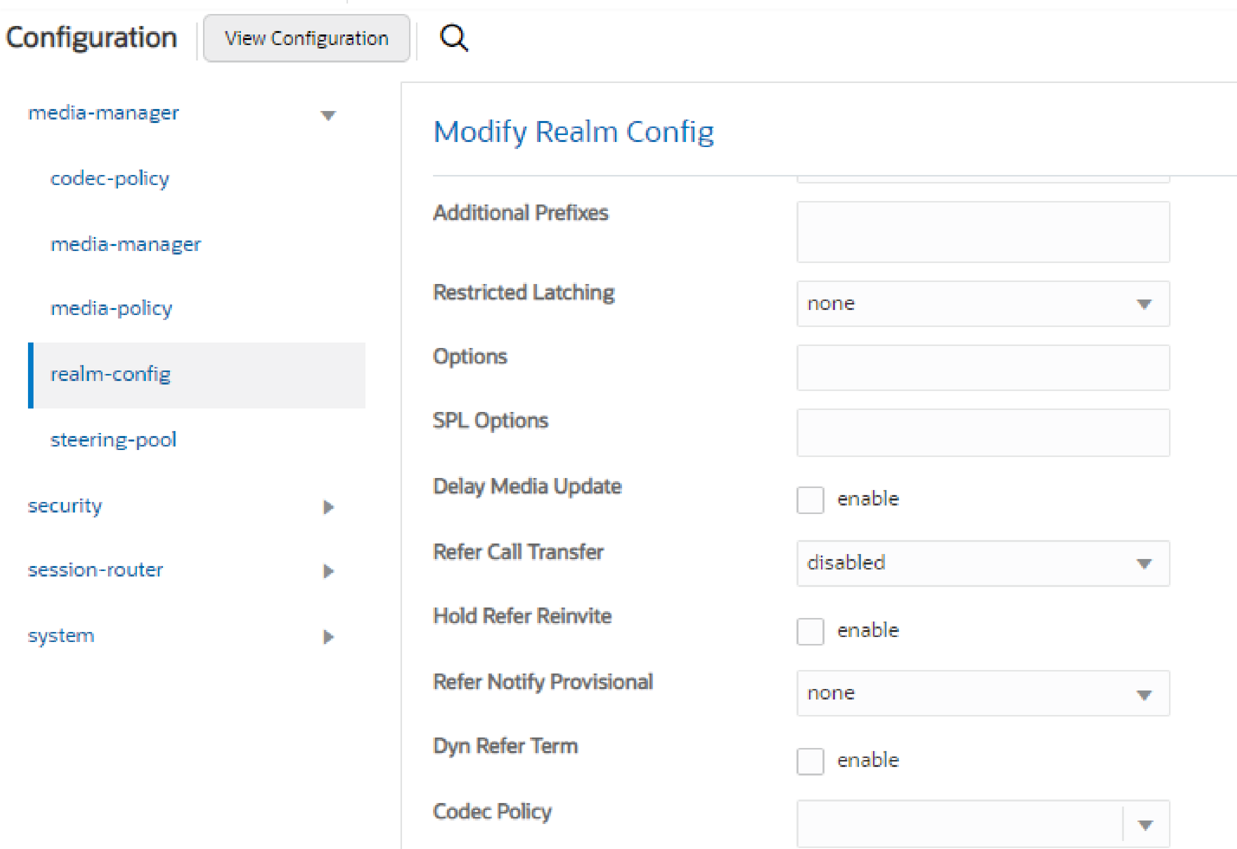
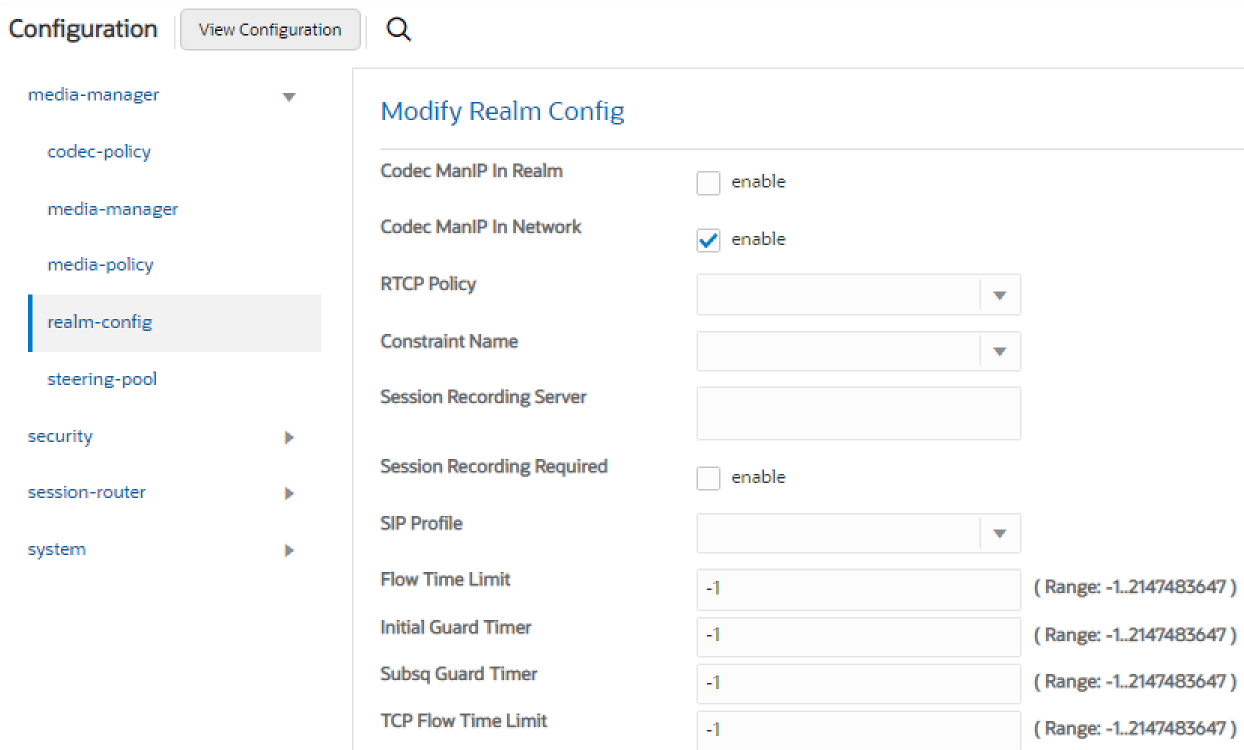


Figure 22: Realm Config towards OnPrem PBX Cont.



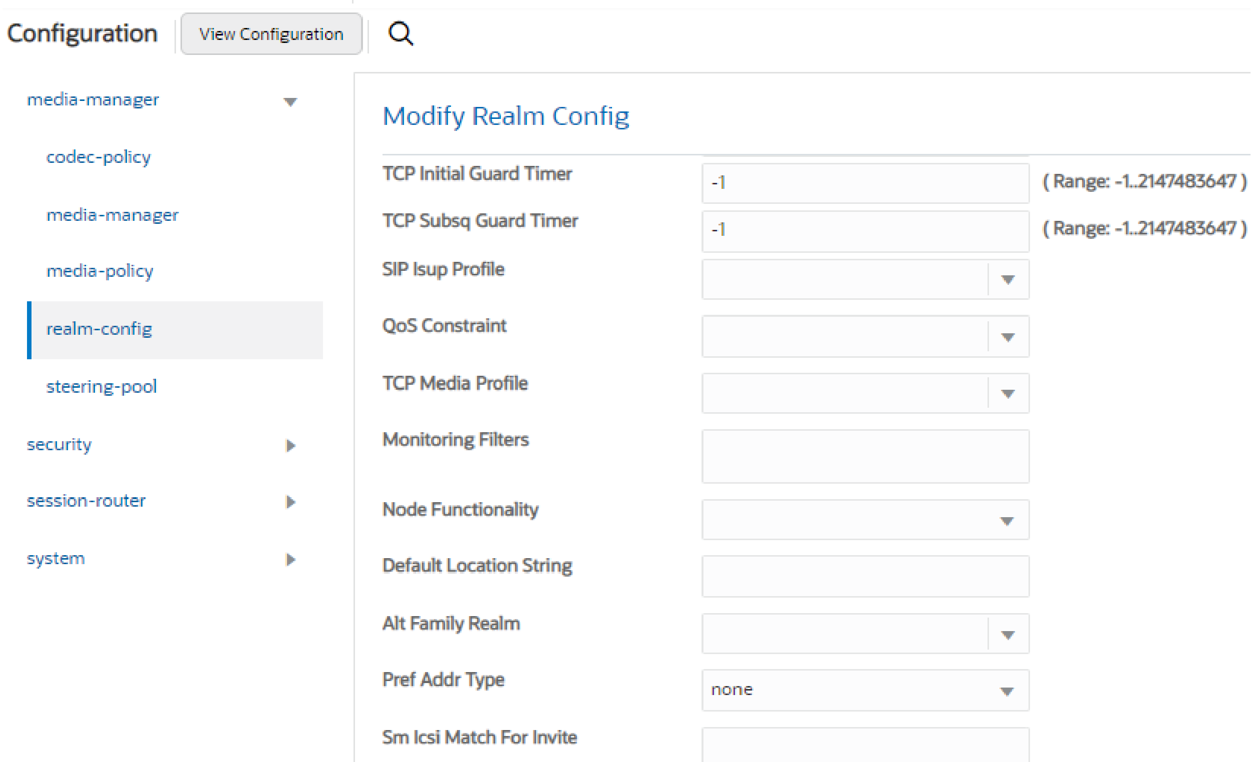


Figure 23: Realm Config towards OnPrem PBX Cont.

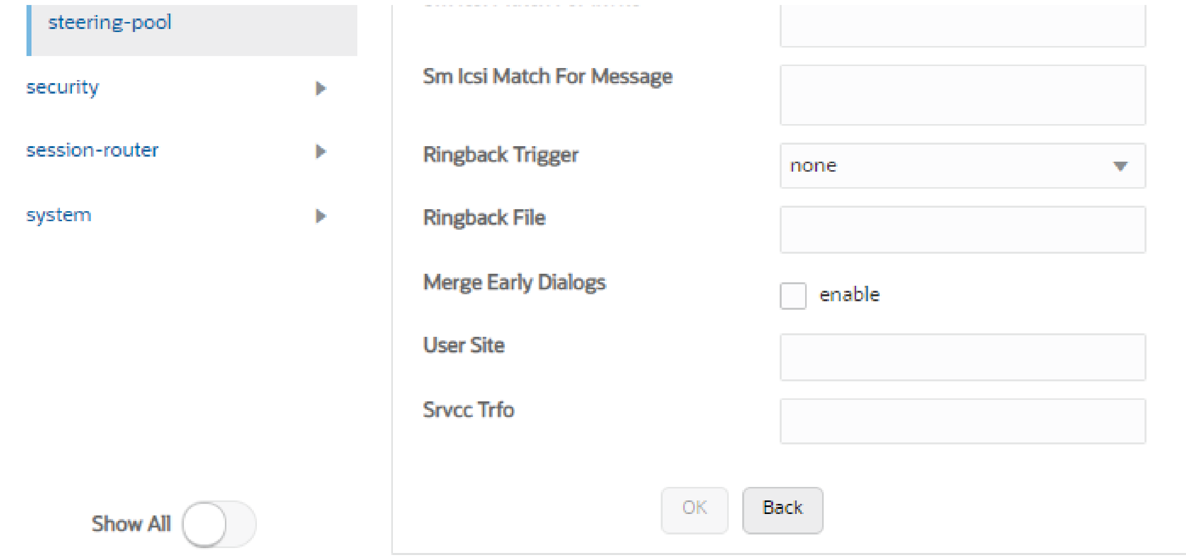


Figure 24: Realm Config towards OnPrem PBX Cont.

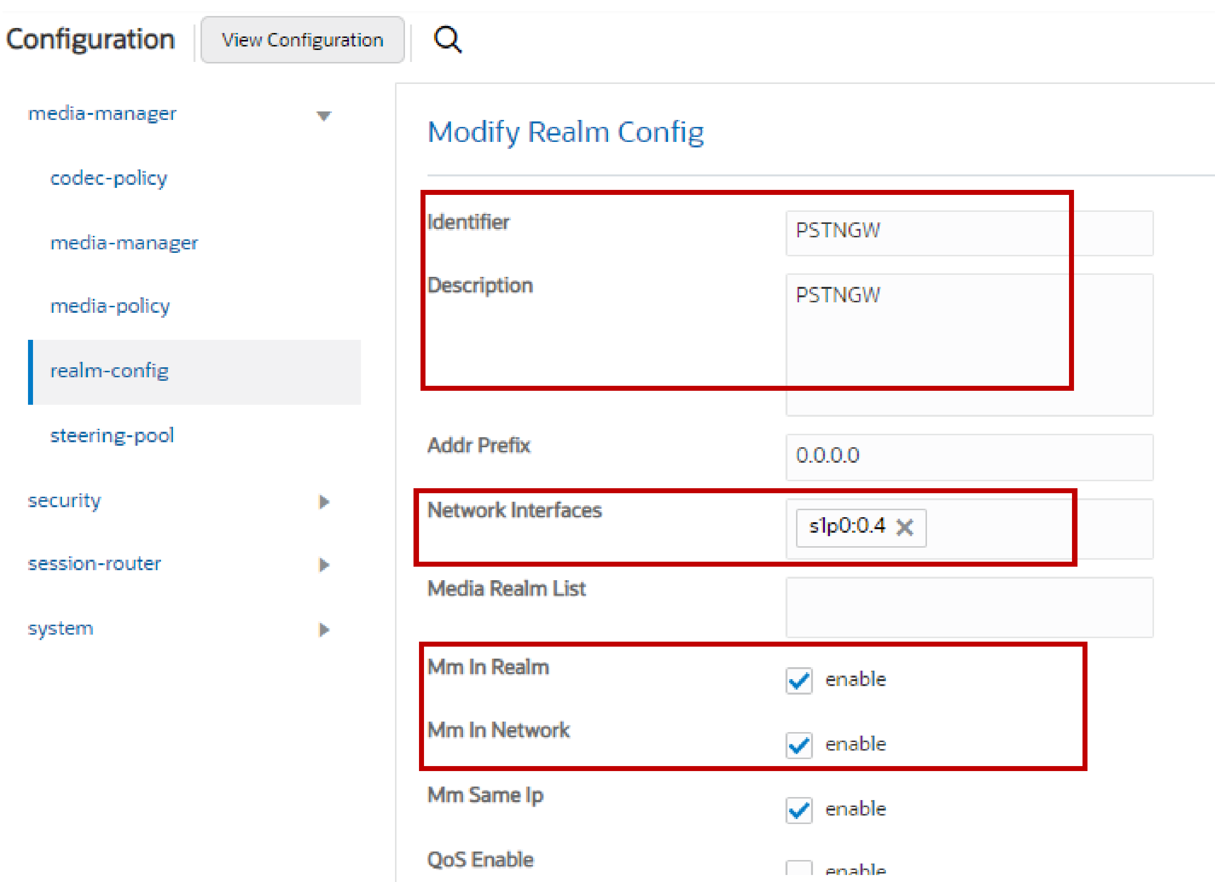


Figure 25: Realm Config towards PSTN Gateway

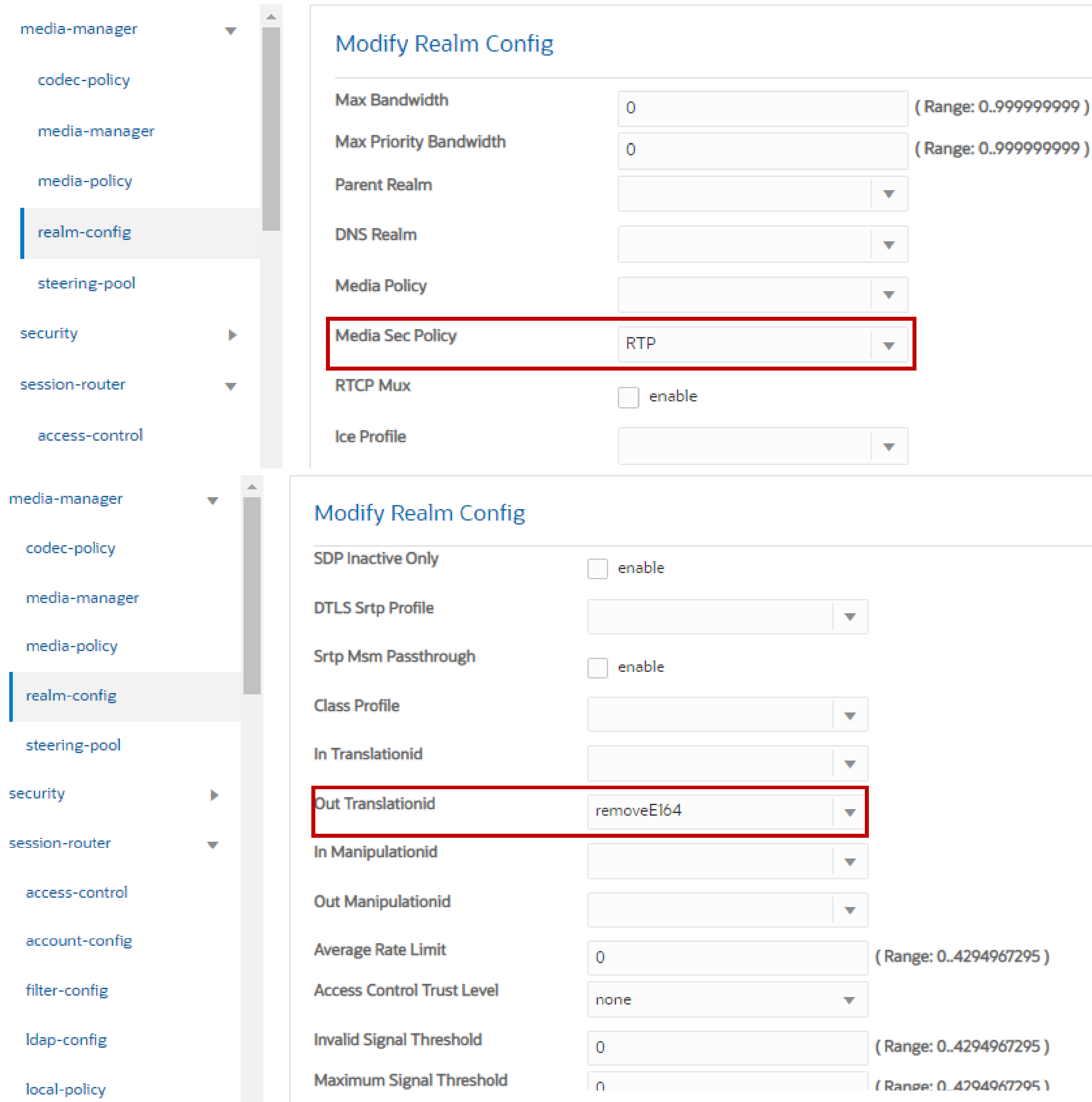


Figure 26: Realm Config towards PSTN Gateway Cont.

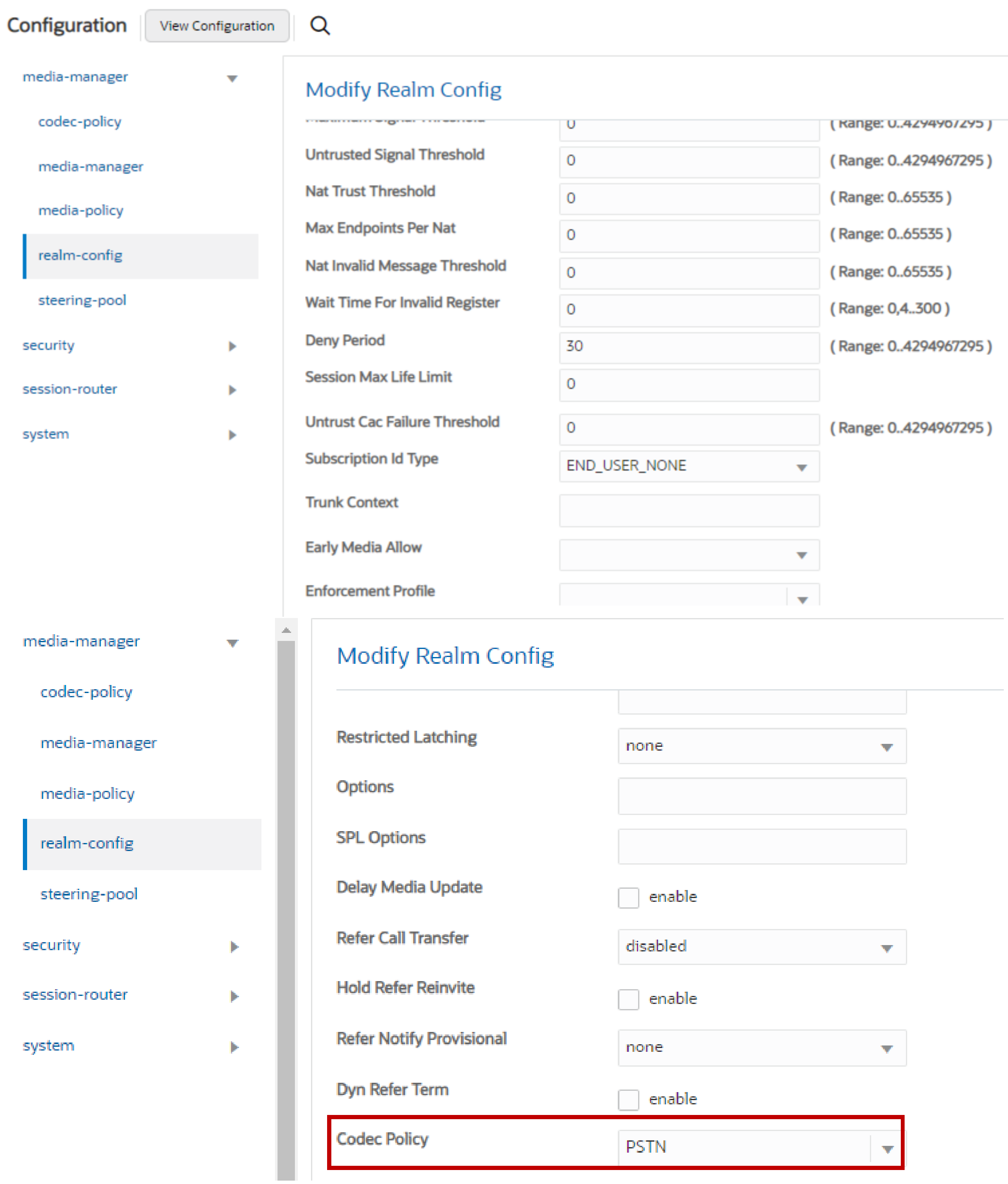


Figure 27: Realm Config towards PSTN Gateway Cont.

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

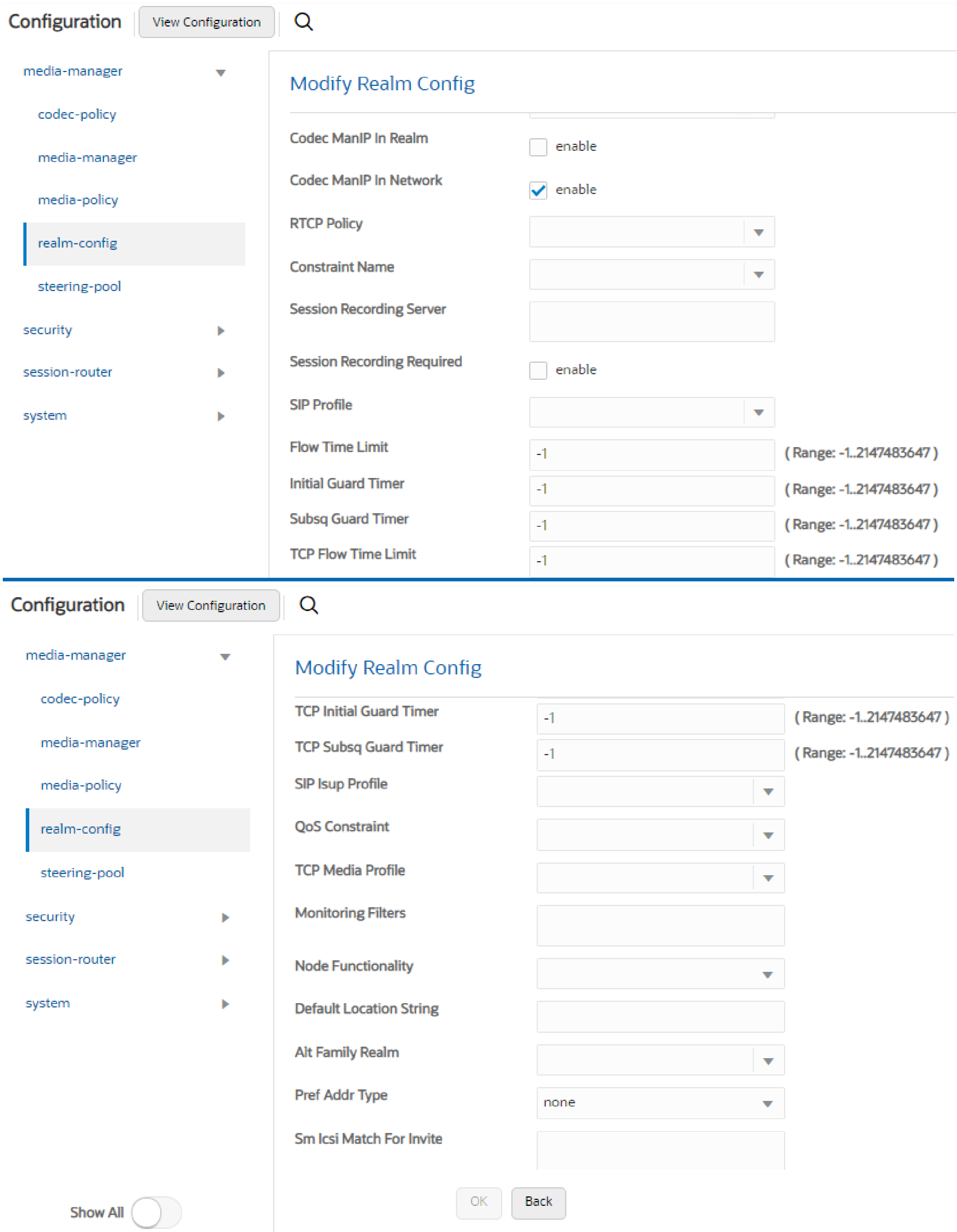


Figure 28: Realm Config towards PSTN Gateway Cont.

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.

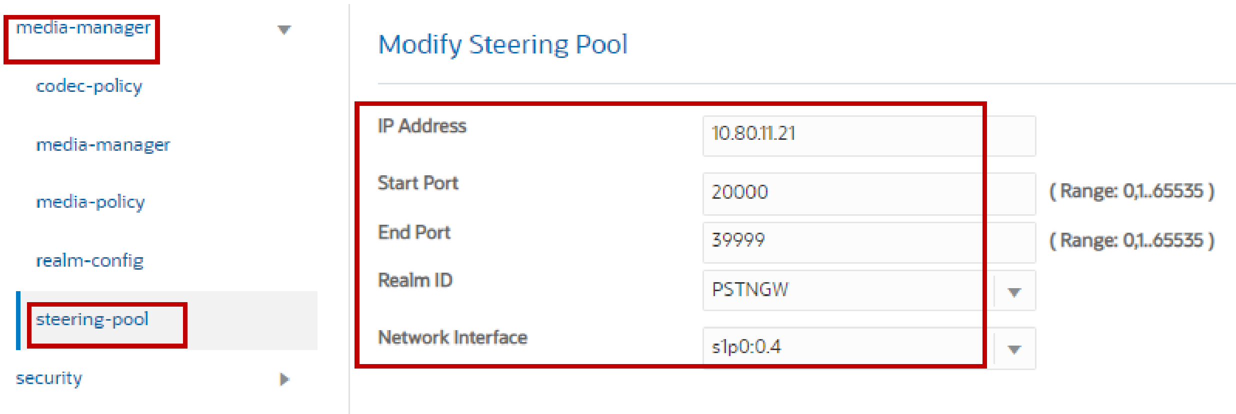


Figure 29: Steering Pool towards PSTN Gateway

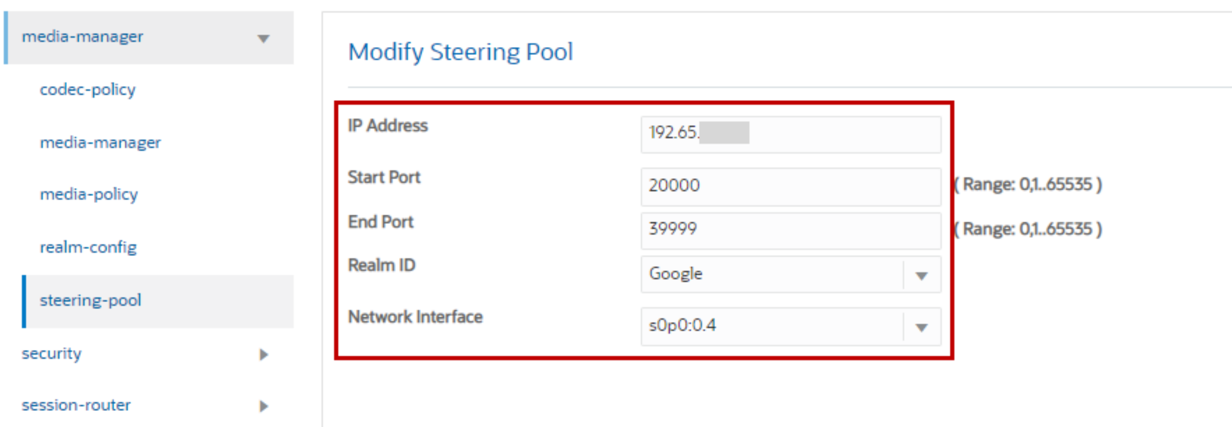


Figure 30: Steering Pool towards Google Voice

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.

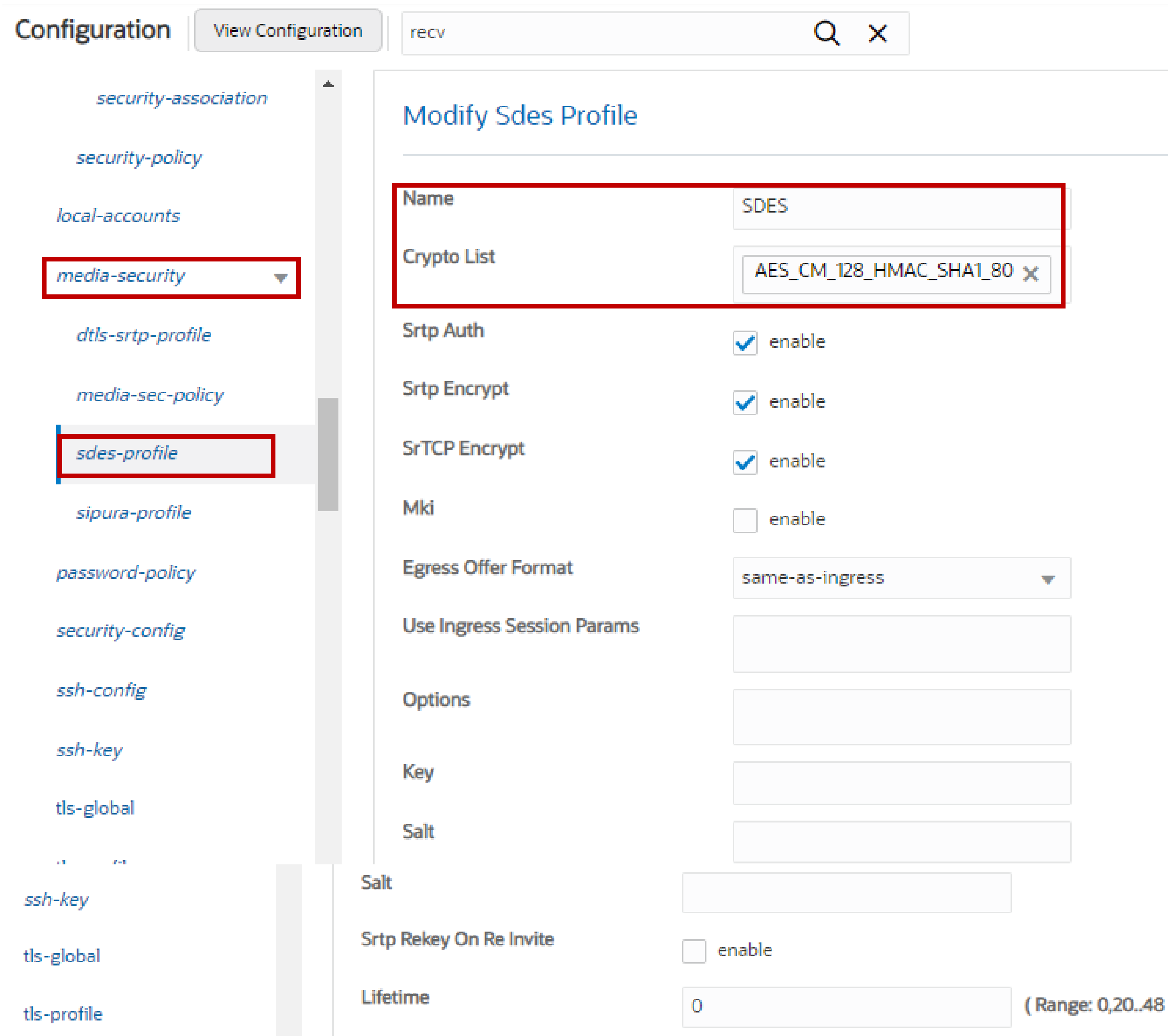


Figure 32: SDES Profile for TLS

6.4.10 Media Sec Policy

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

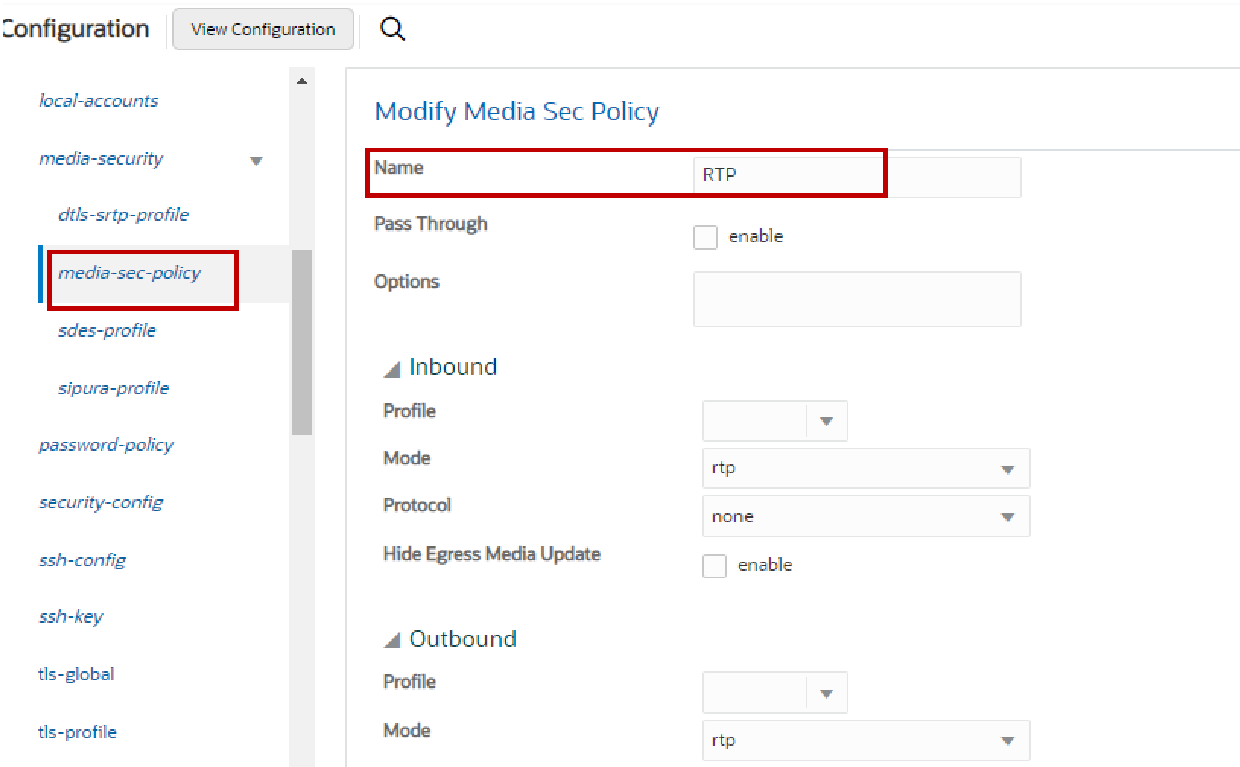


Figure 33: Media Security Policy for RTP

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

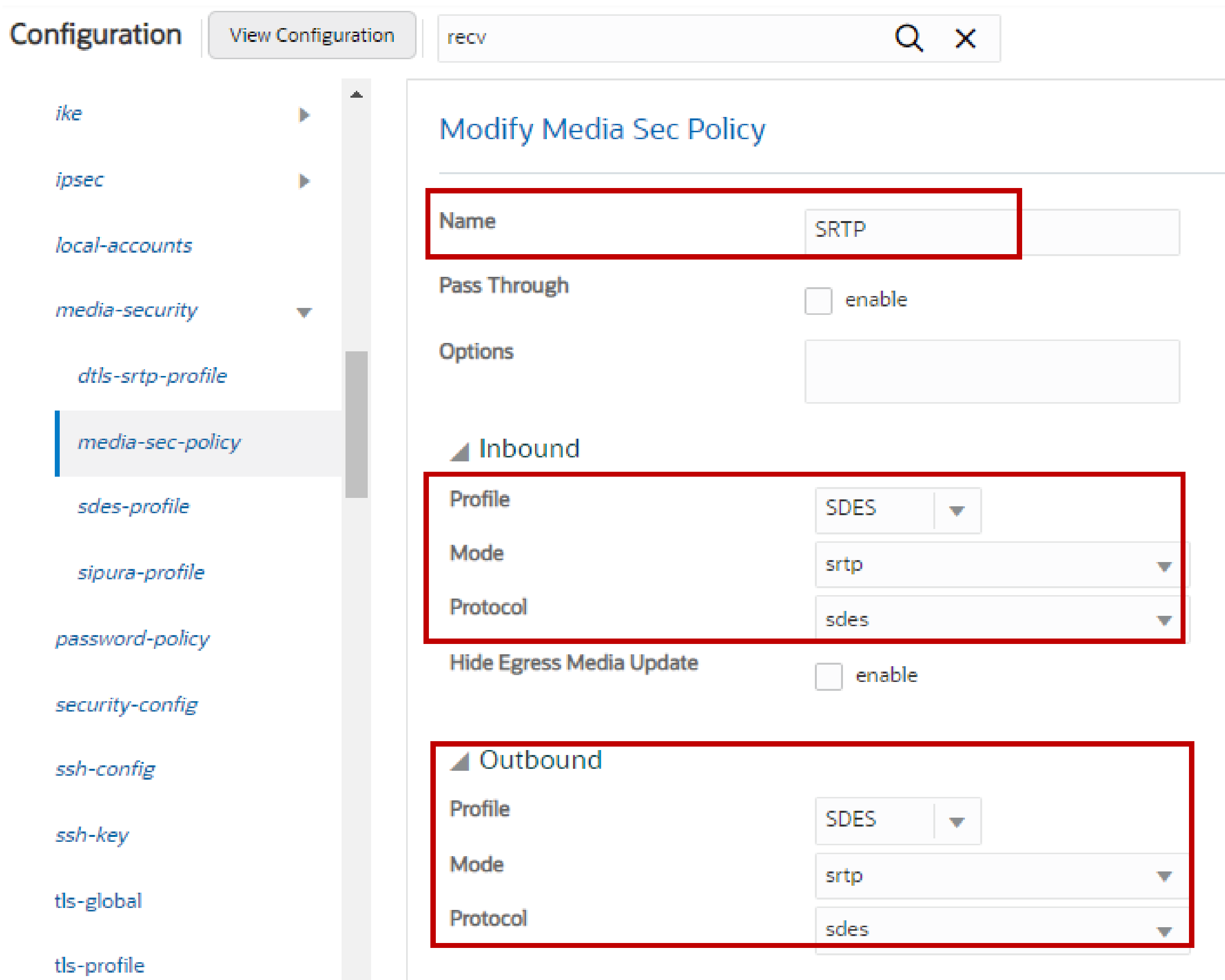


Figure 34: Media Security Policy for SRTP

6.4.11 TLS – Certificate Record

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

Create a certificate record for Google Voice as shown below.

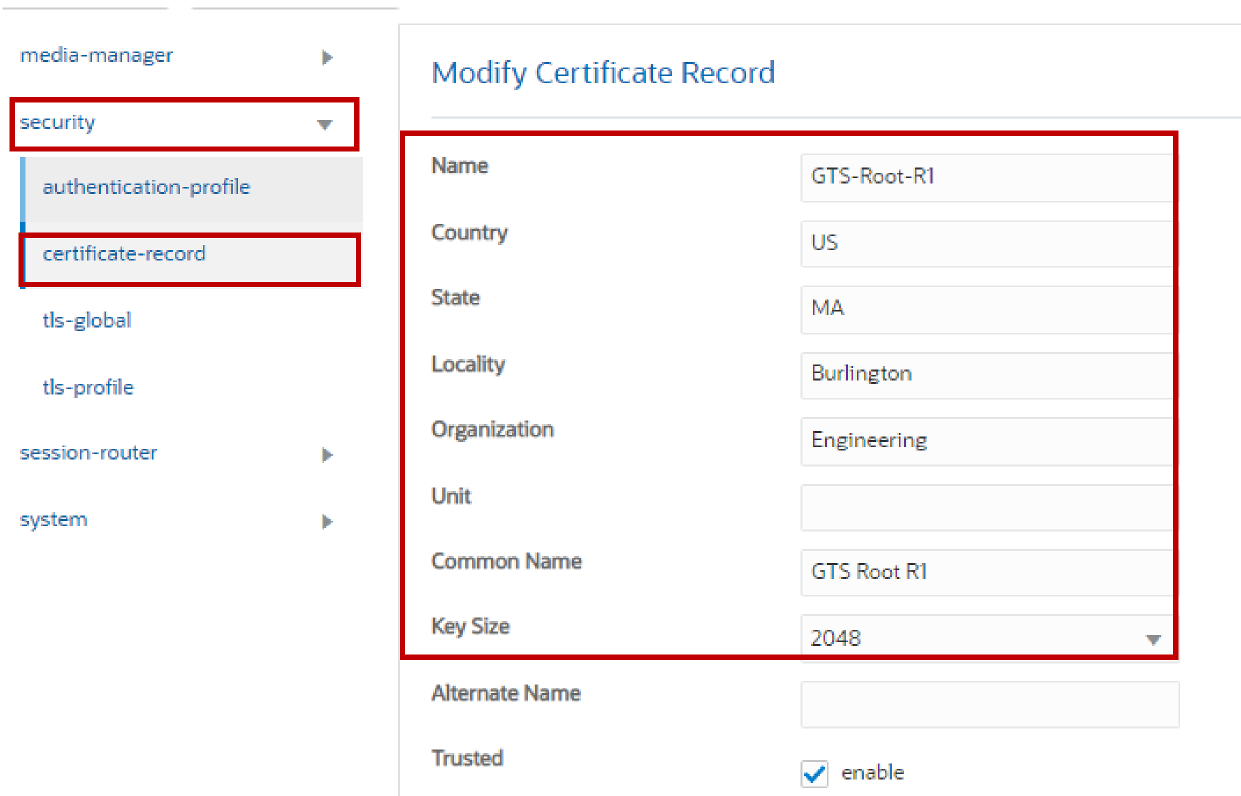


Figure 35: Create Certificate Record for Google Voice ROOT CA

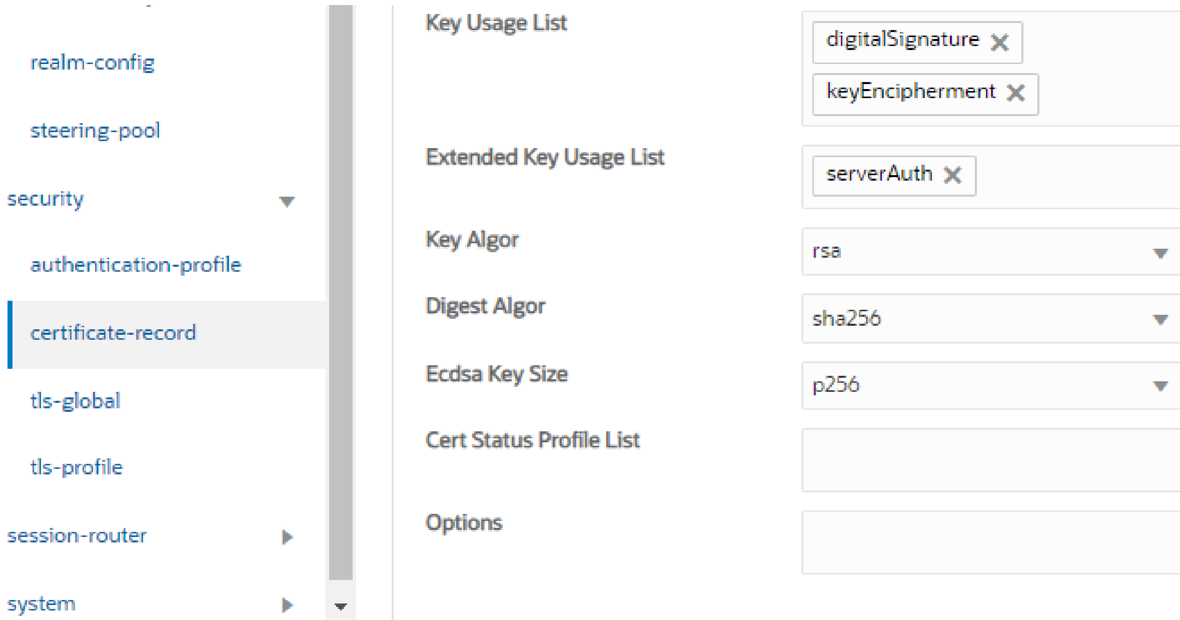


Figure 36: Create Certificate Record for Google Voice Root CA Cont.

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

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

Note

Refer to Google Voice SIP Link documentation for other compatible CAs

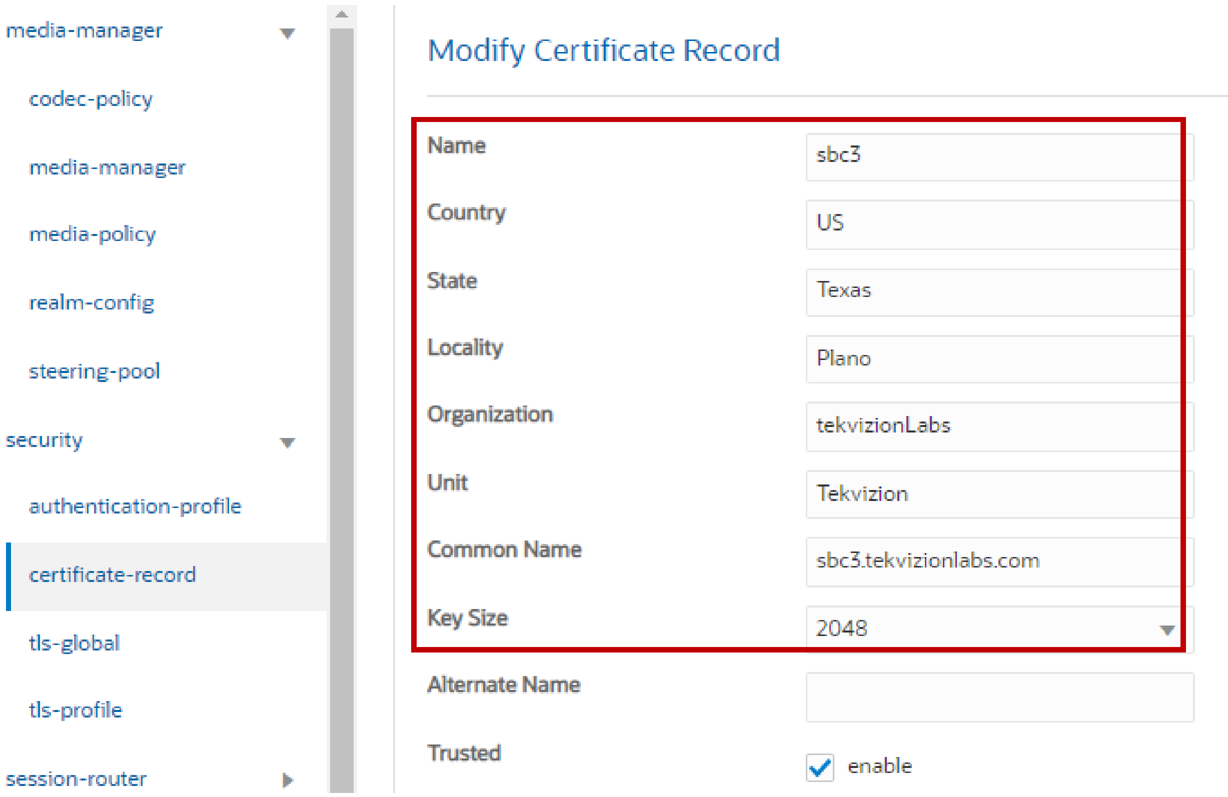


Figure 37: Create Certificate Record for Oracle E-SBC

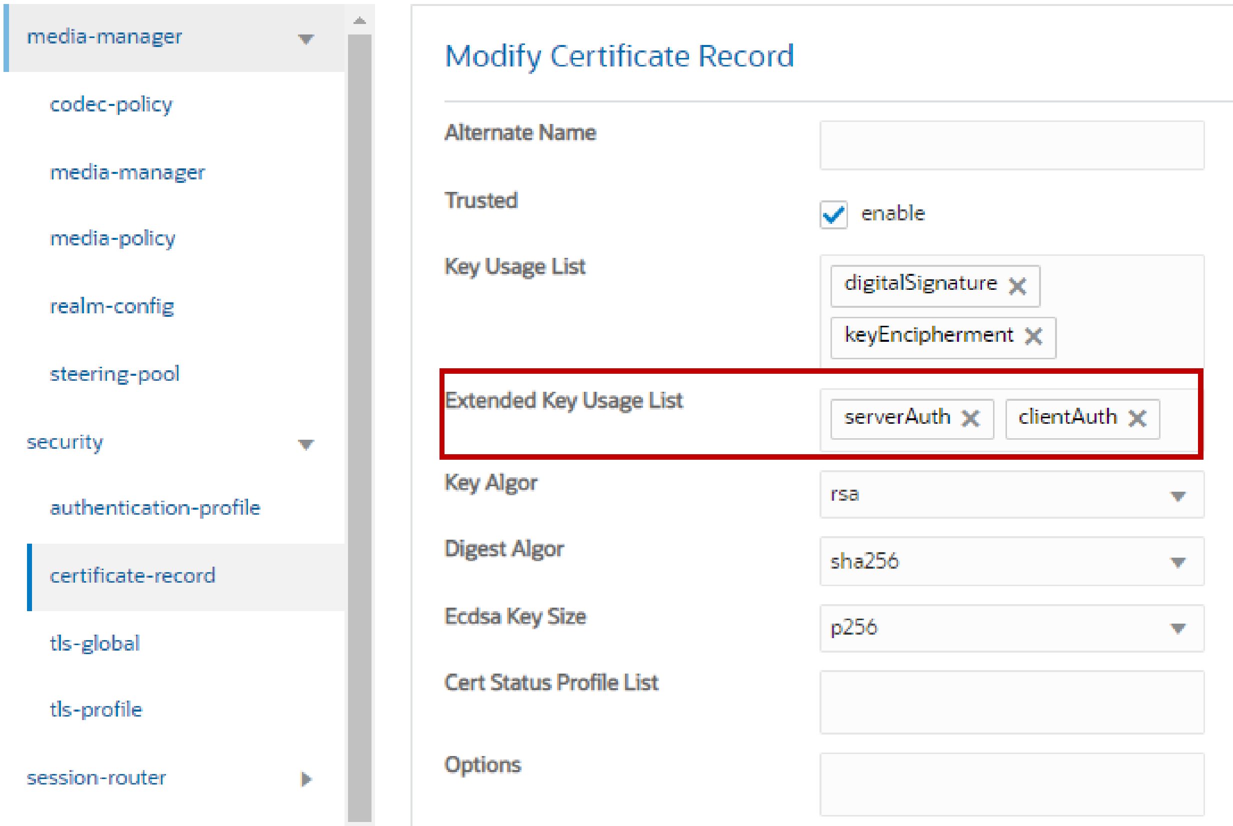
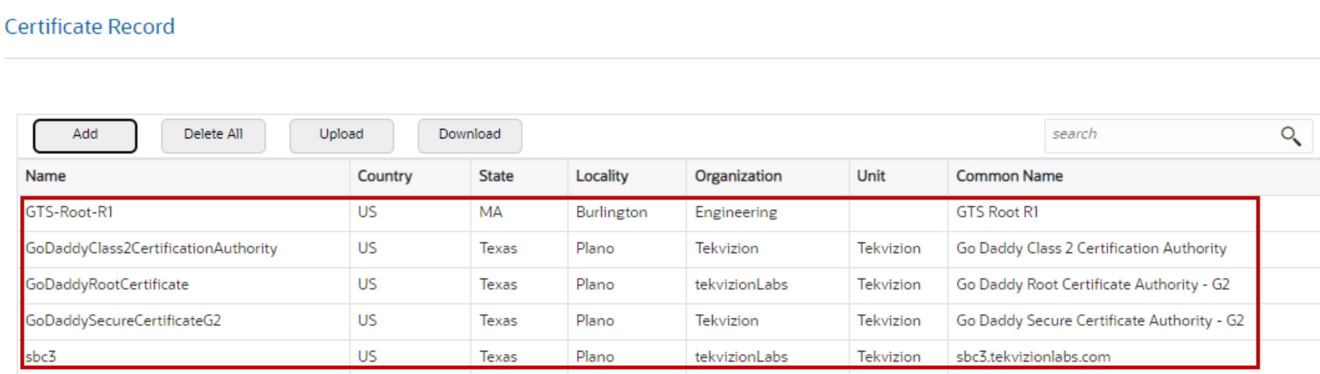


Figure 38: Create Certificate Record for Oracle E-SBC Cont.

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



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

Figure 39: Certificate Records

6.4.12 TLS – TLS Profile

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

Create a TLS profile for Google Voice as shown below.

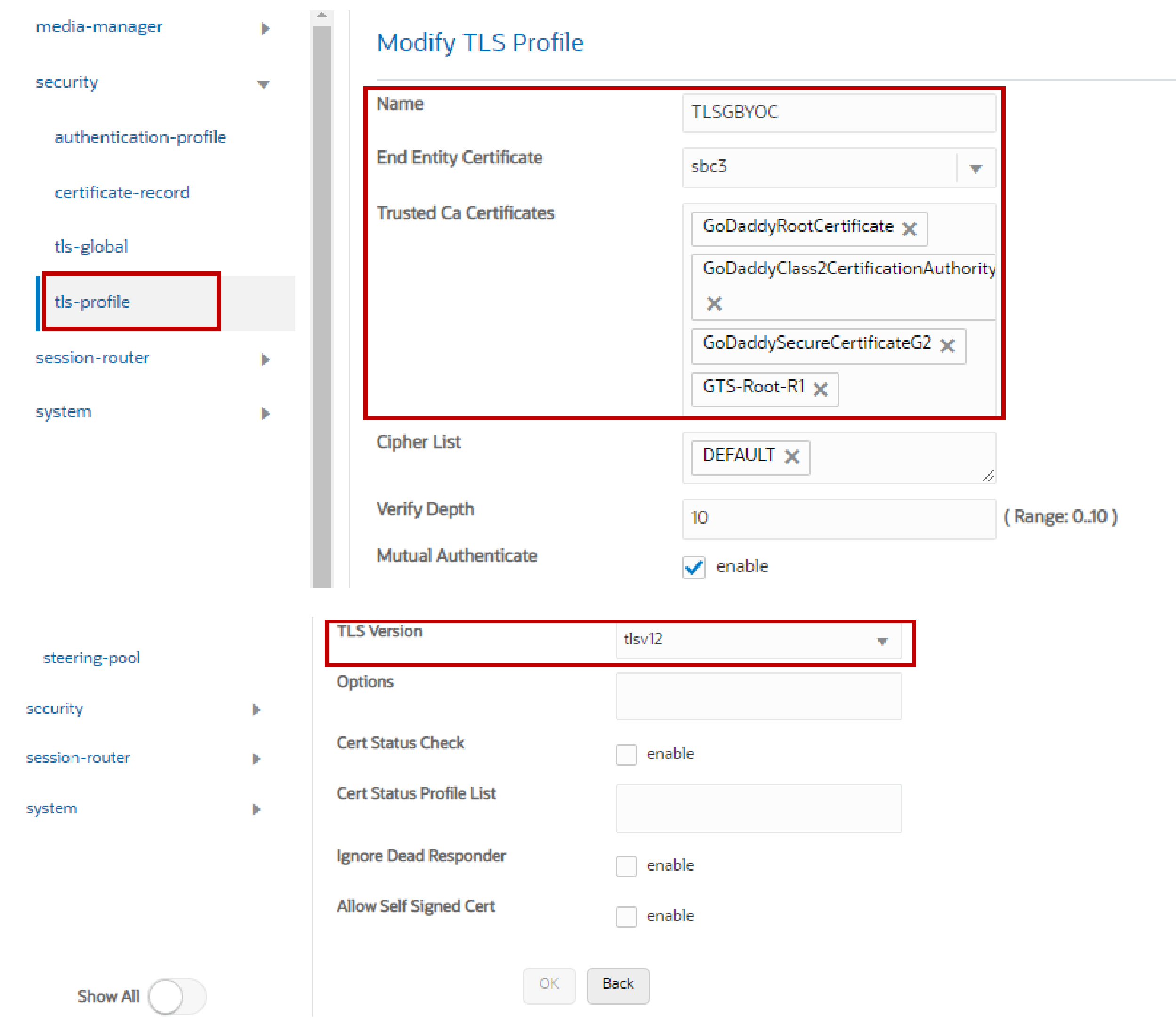


Figure 40: TLS Profile

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.

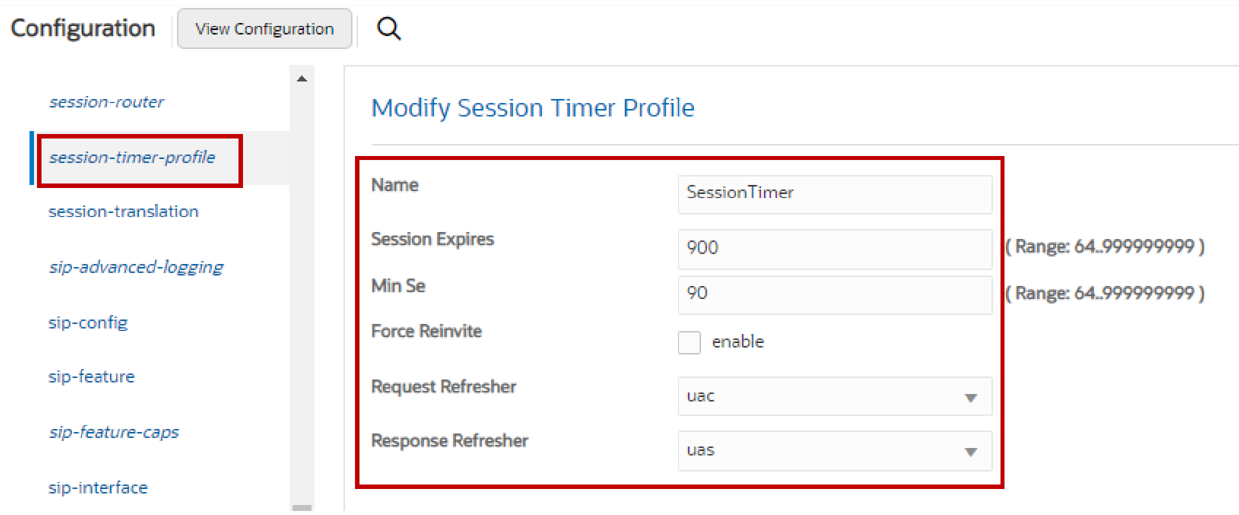


Figure 41: Session Timer

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.

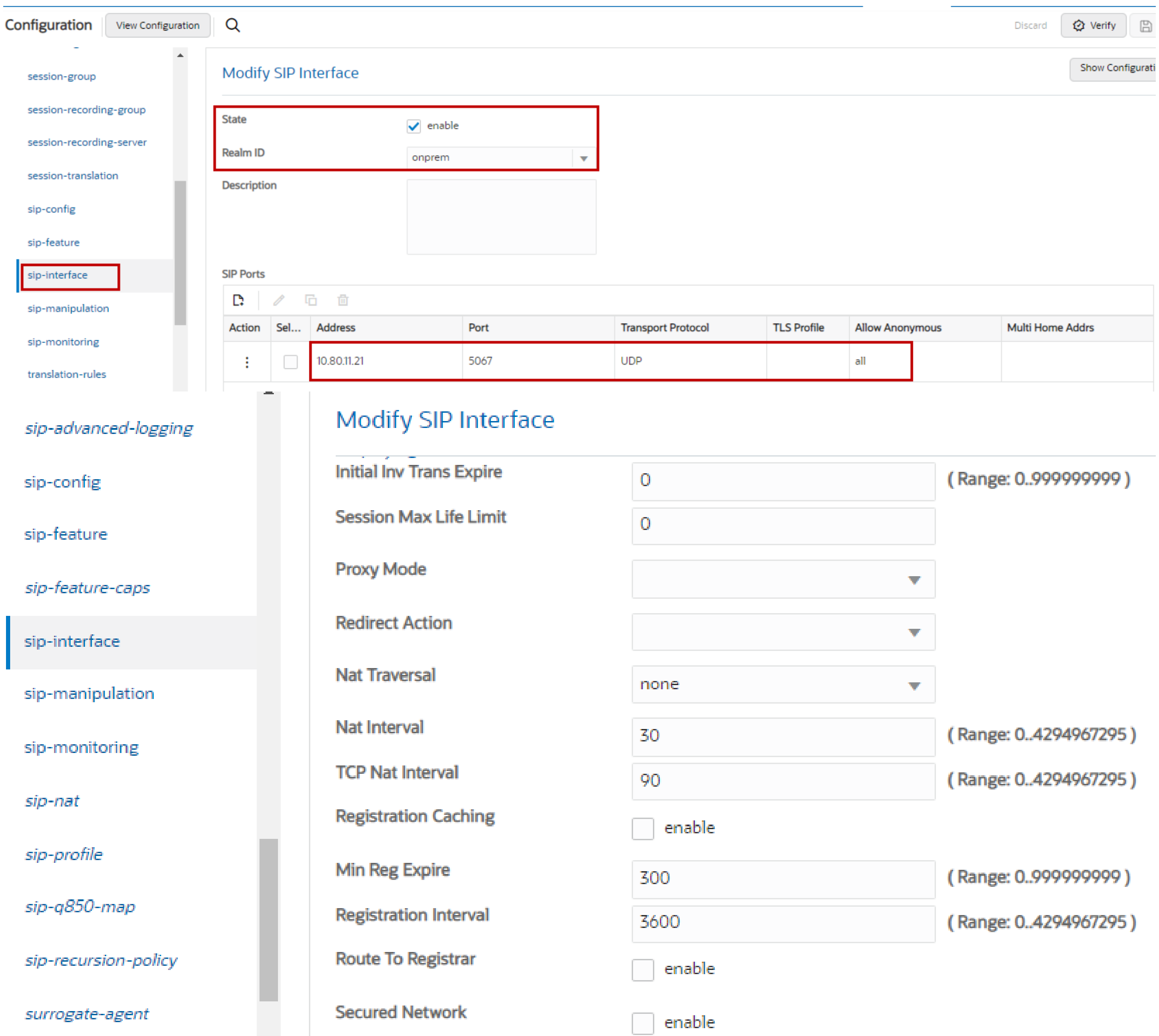


Figure 42: SIP Interface for OnPrem PBX

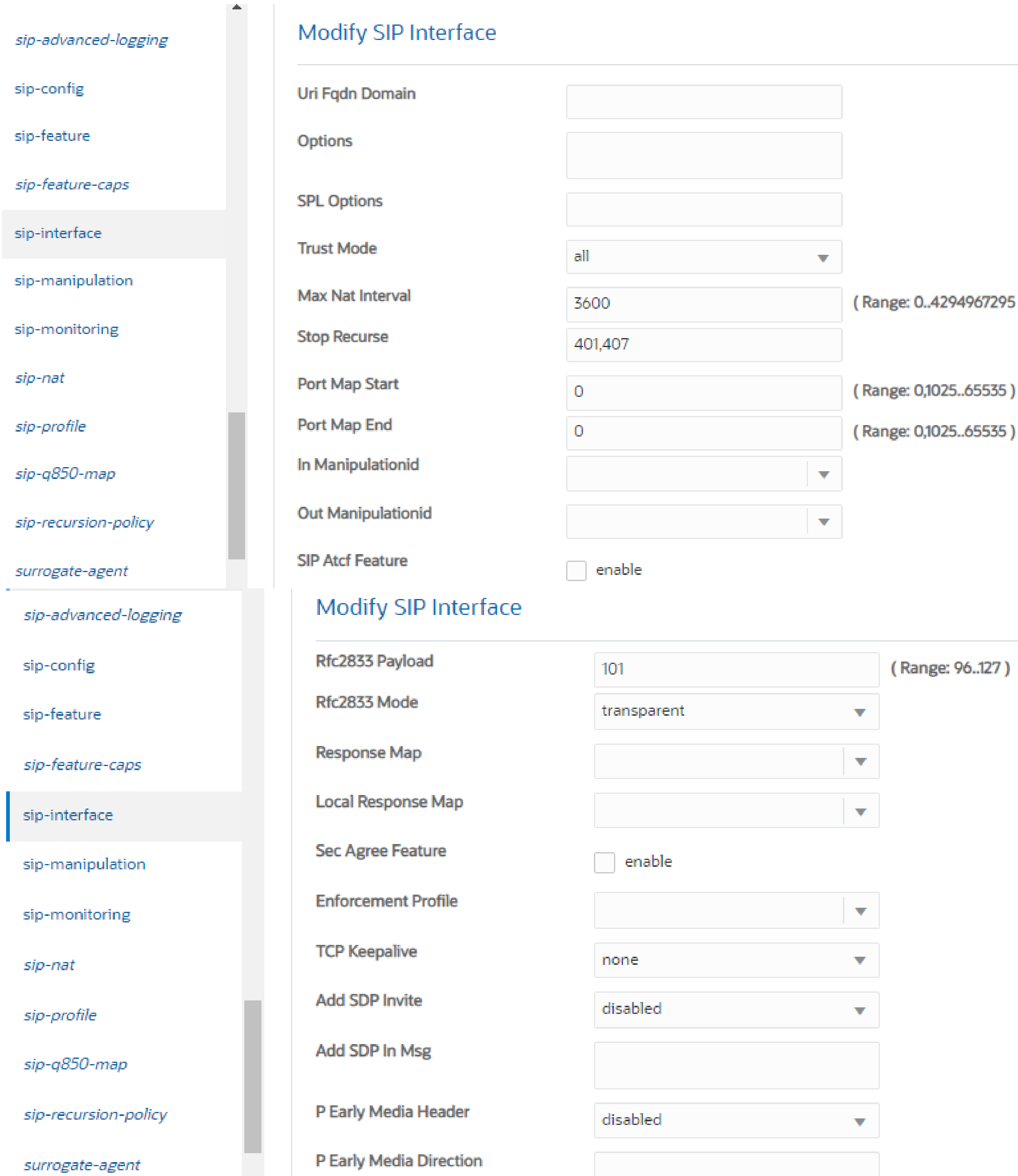


Figure 43: SIP Interface for OnPrem PBX Cont.

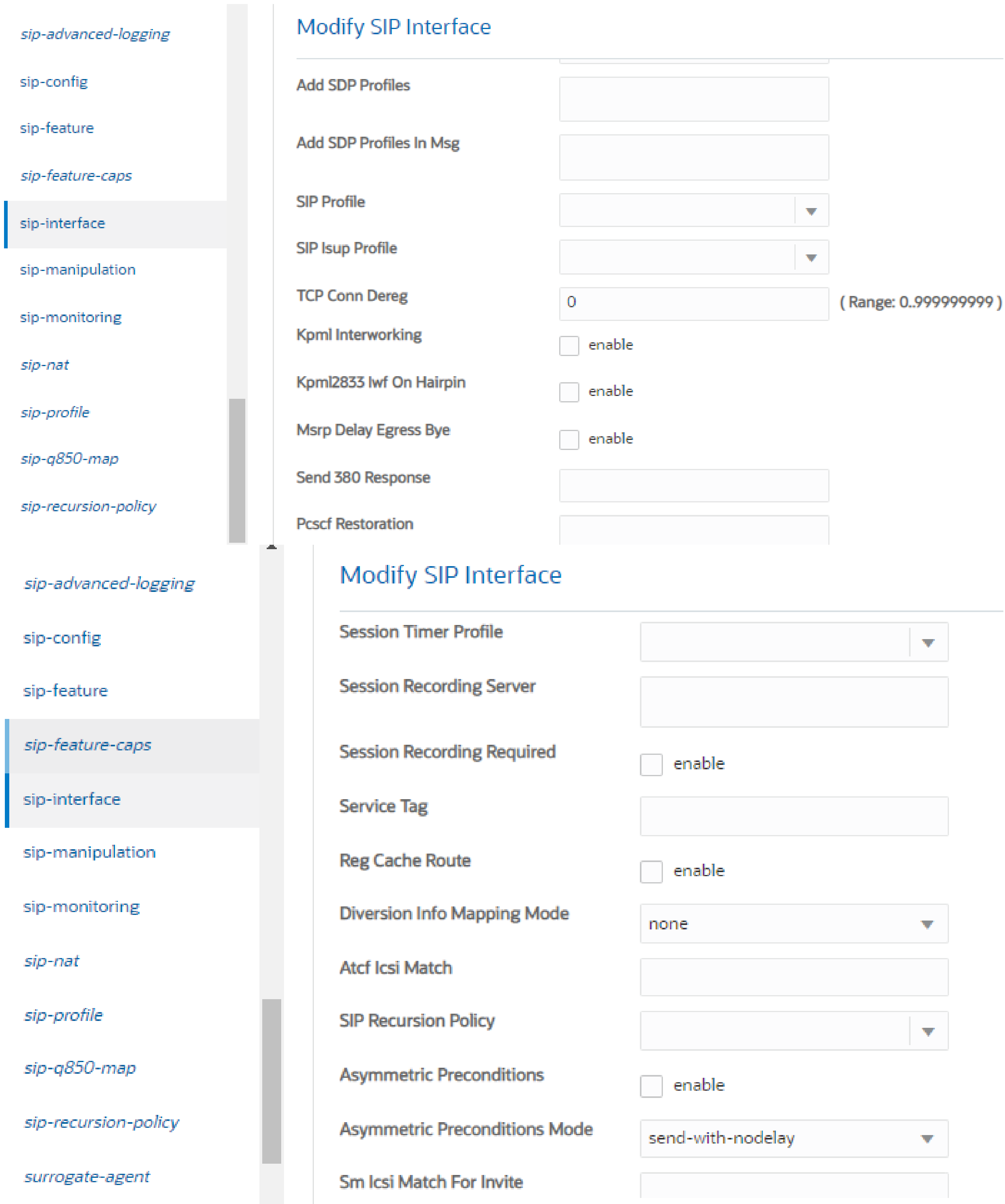


Figure 44: SIP Interface for OnPrem PBX Cont.

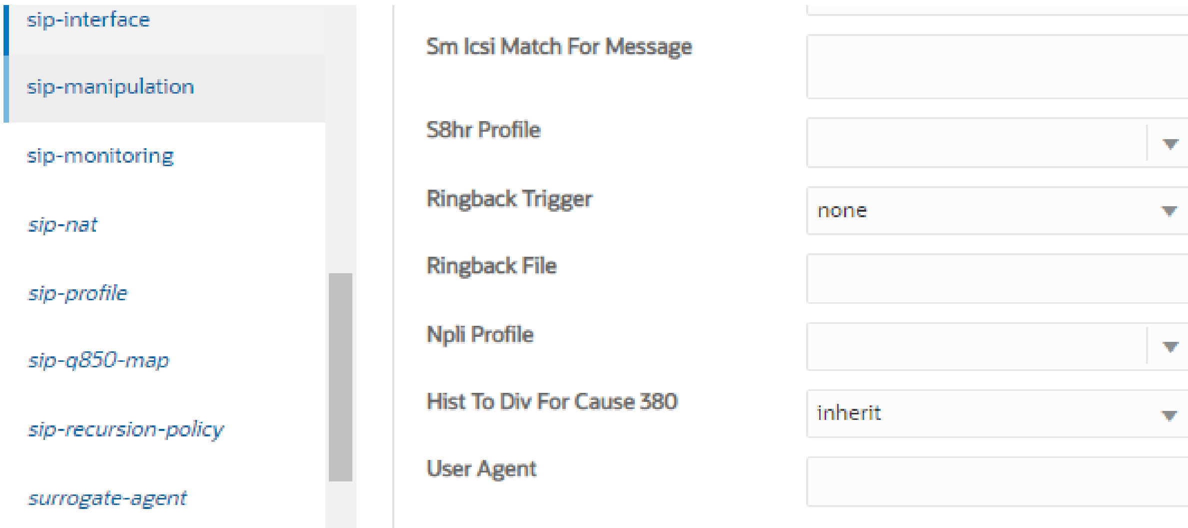


Figure 45: SIP Interface for OnPrem PBX Cont.

TLS Profile is configured with TLSGBYOC configured in Section 6.4.12

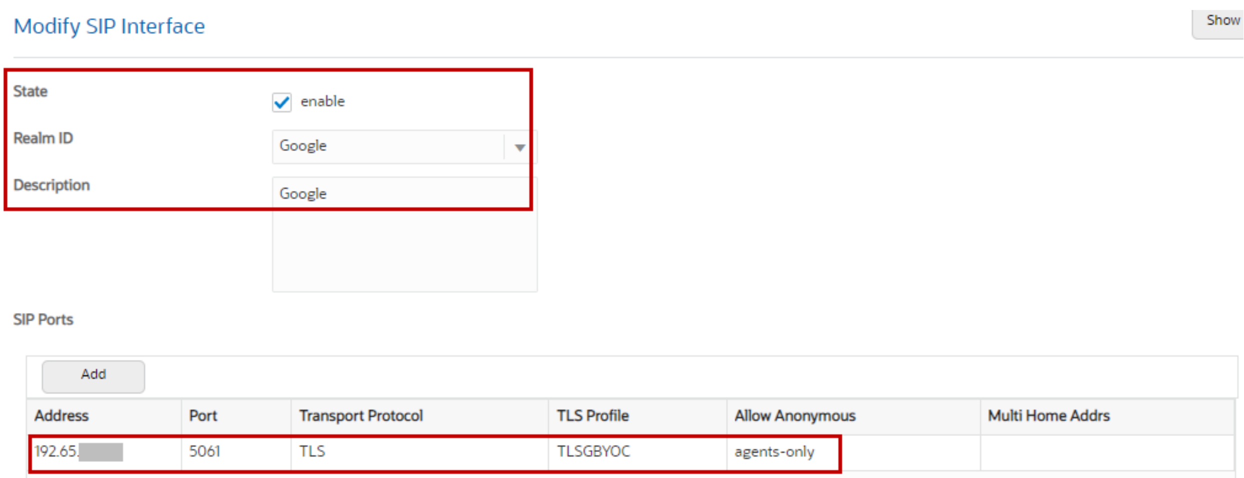
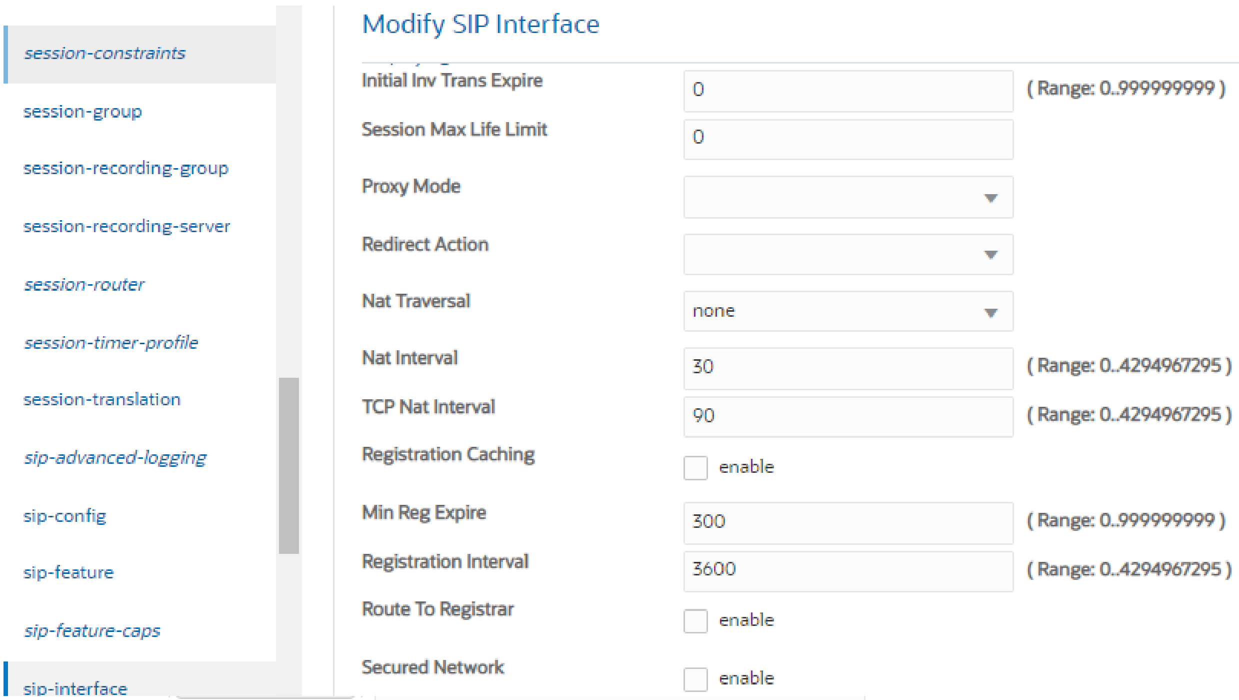


Figure 46: SIP Interface for Google Voice



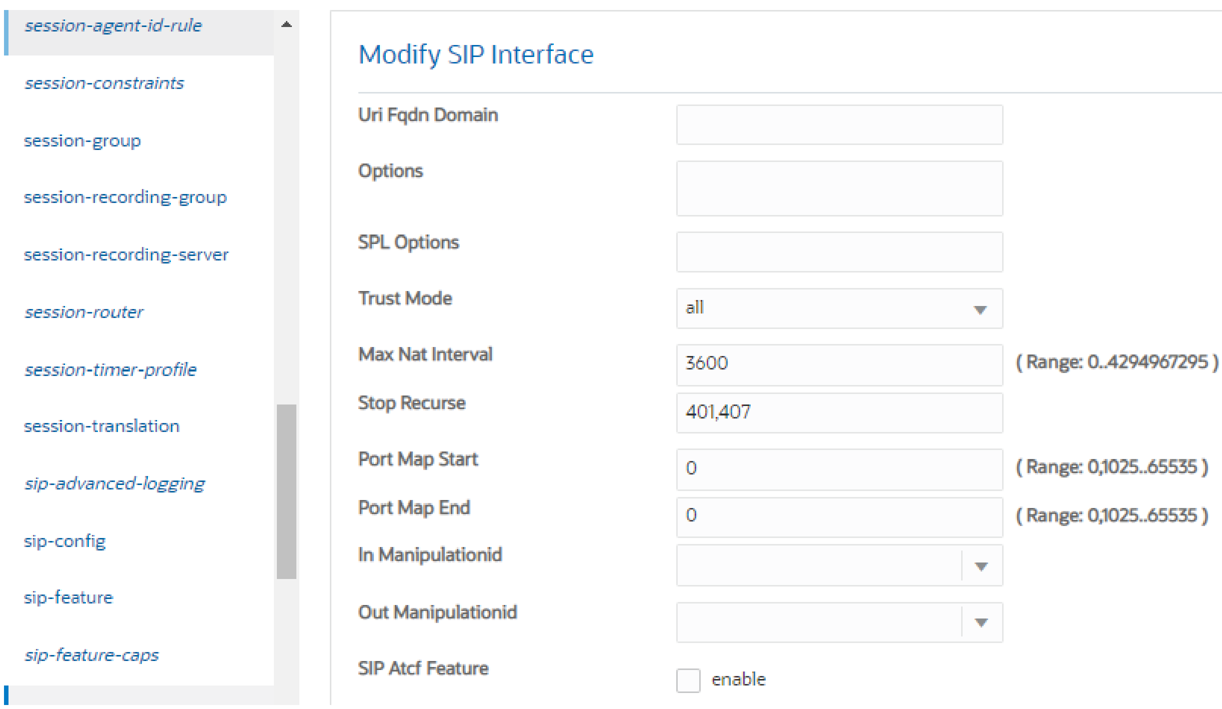


Figure 47: SIP Interface for Google Voice Cont.

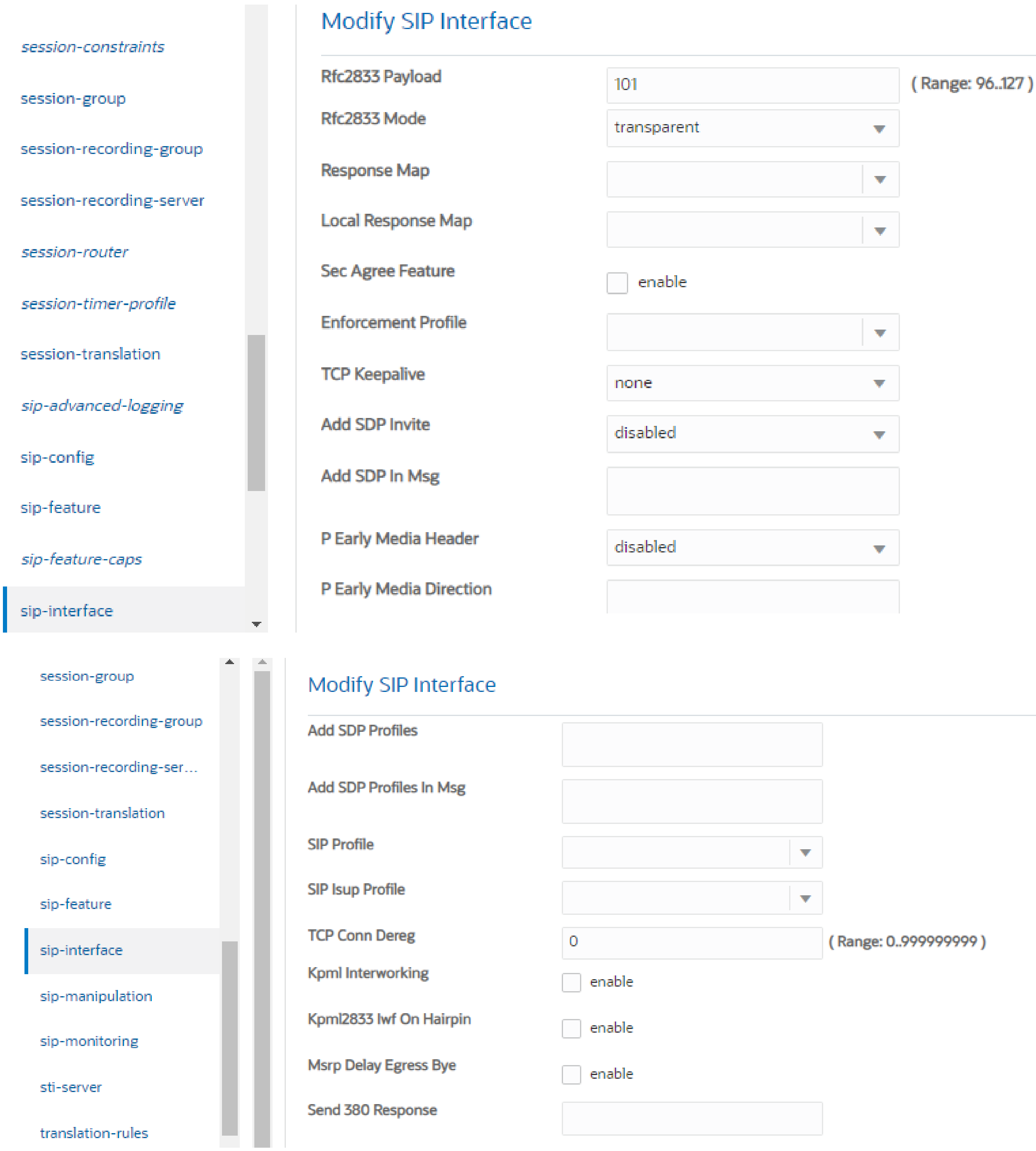


Figure 48: SIP Interface for Google Voice Cont.

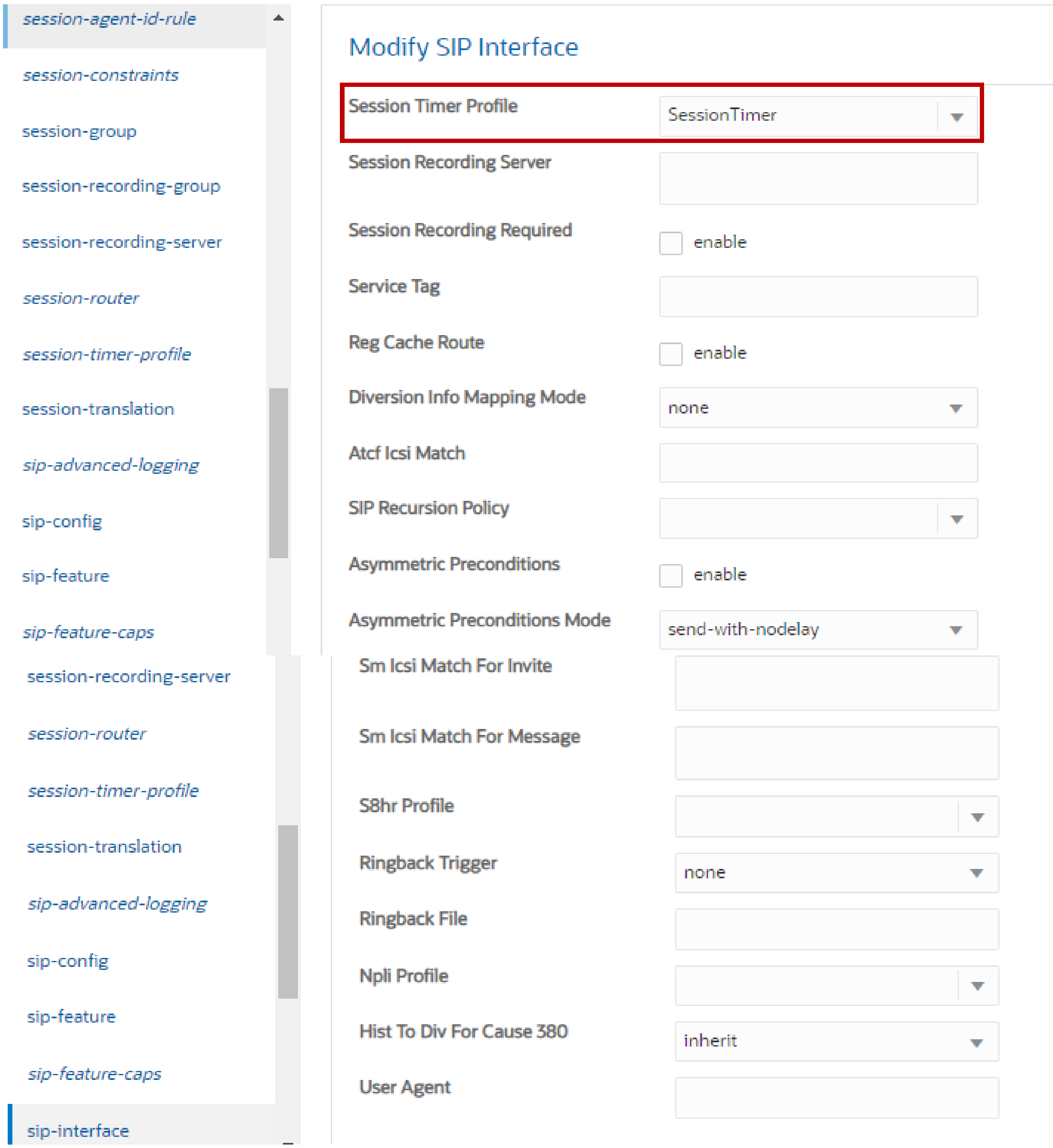


Figure 49: SIP Interface for Google Voice Cont.

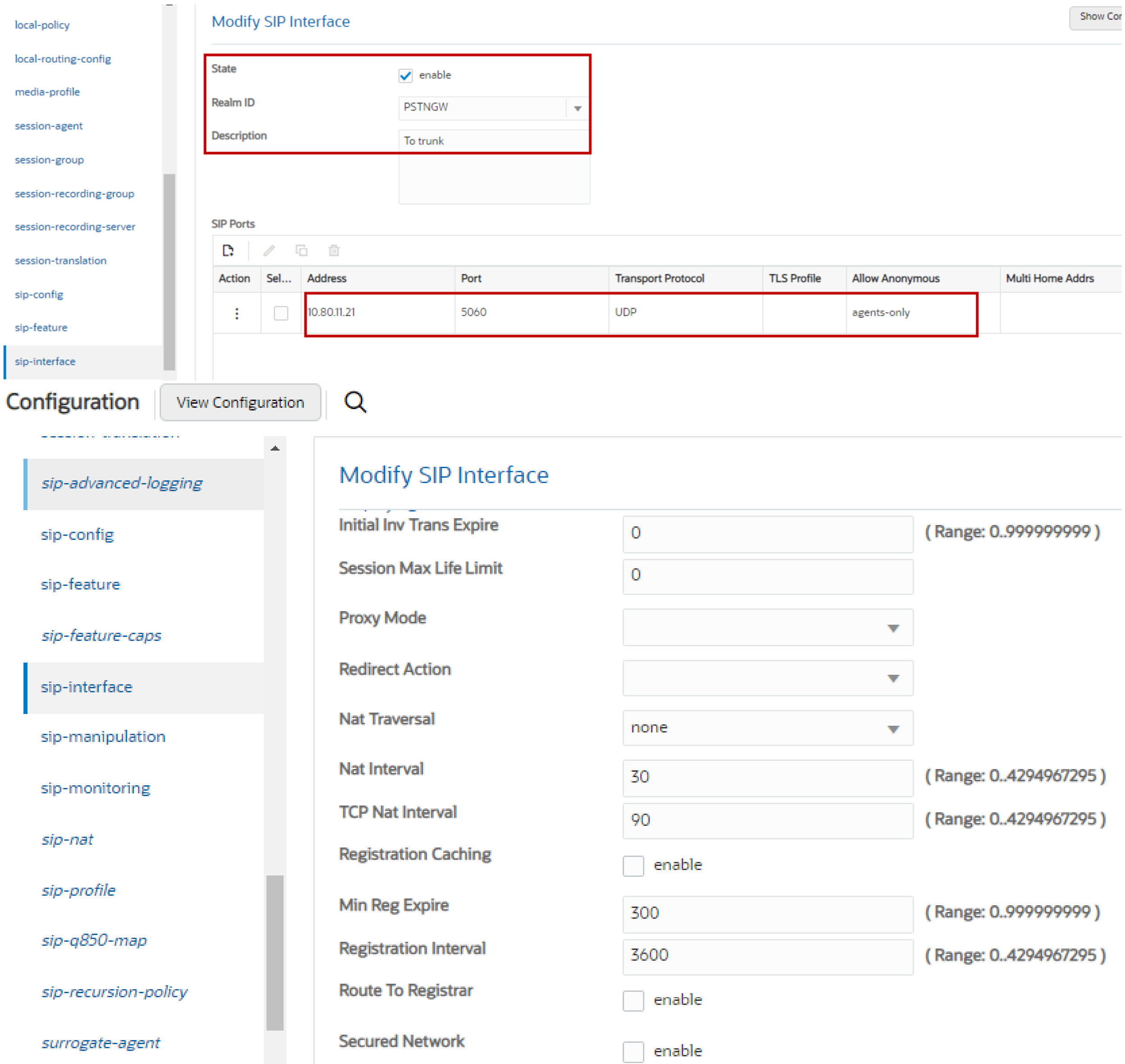


Figure 50: SIP Interface for PSTN Gateway

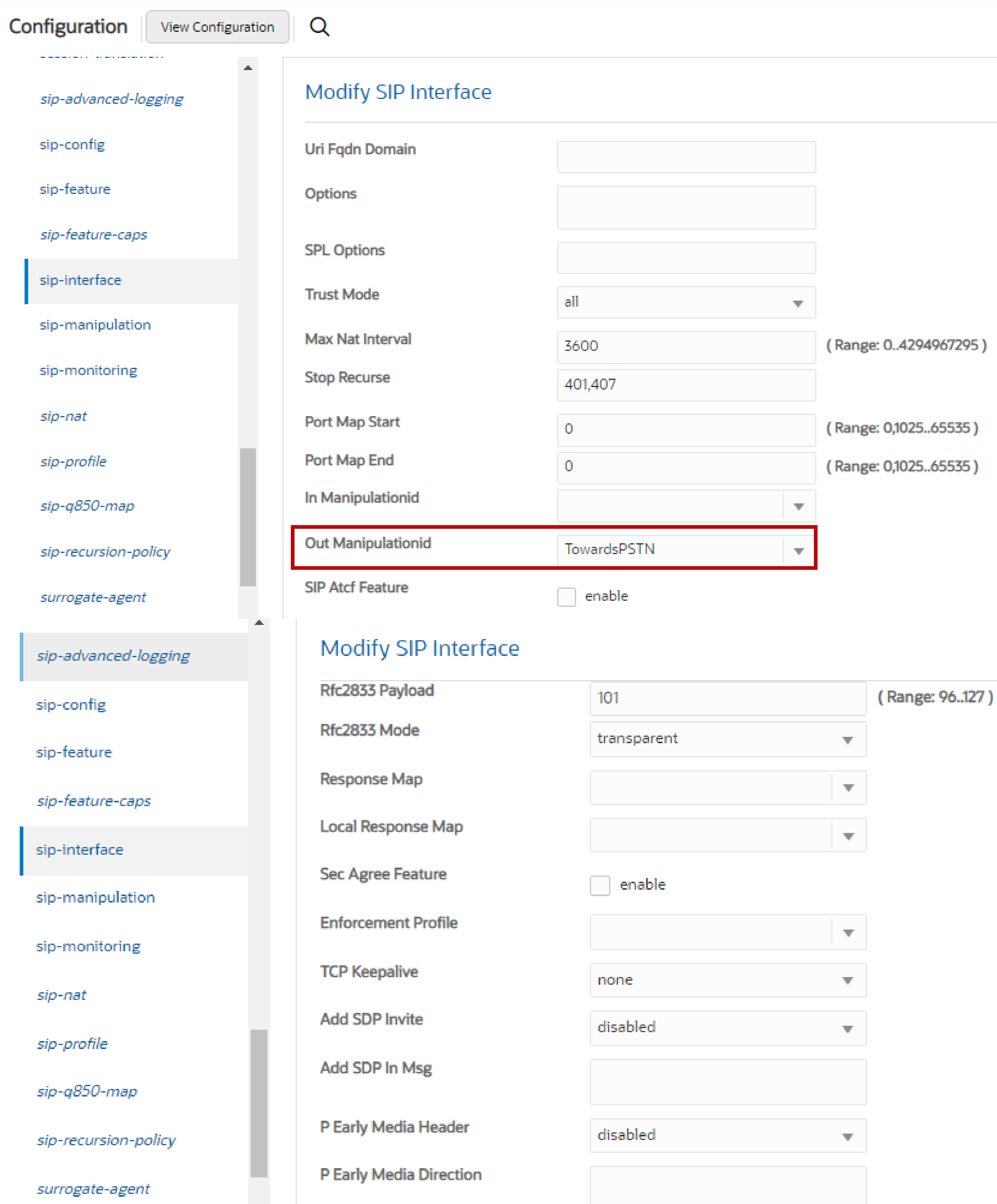
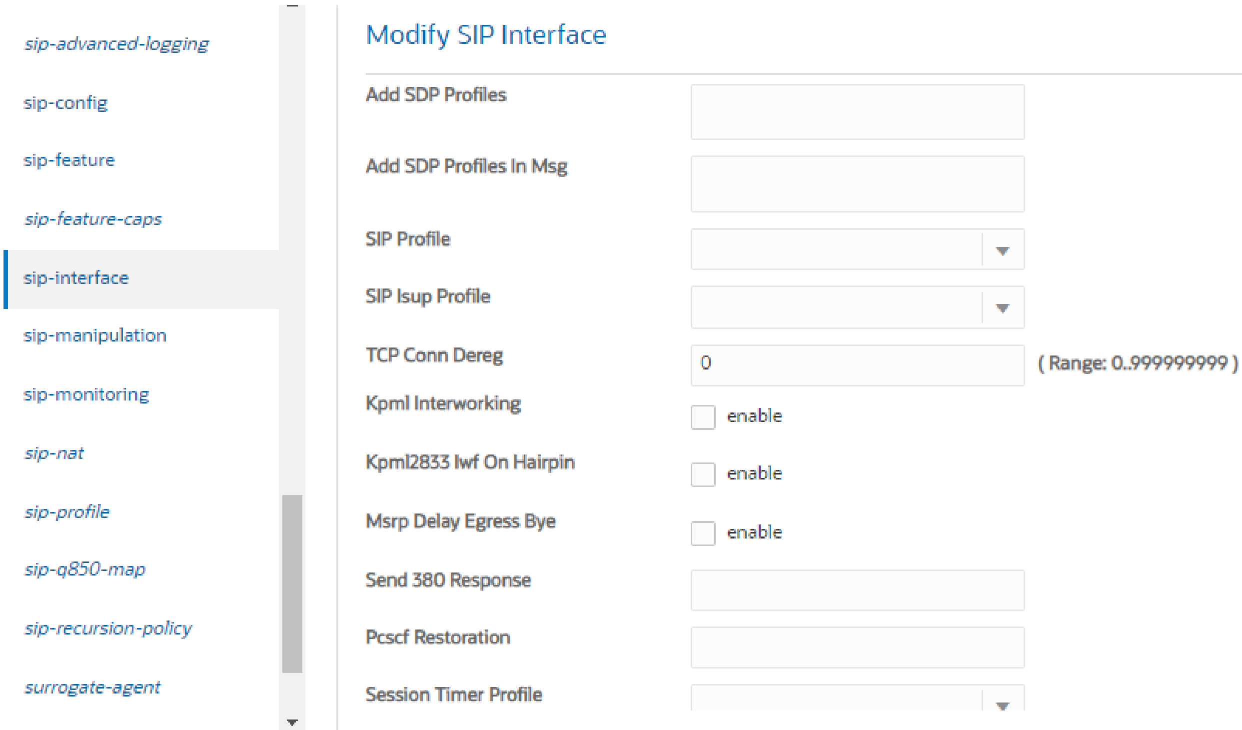


Figure 51: SIP Interface for PSTN Gateway Cont.



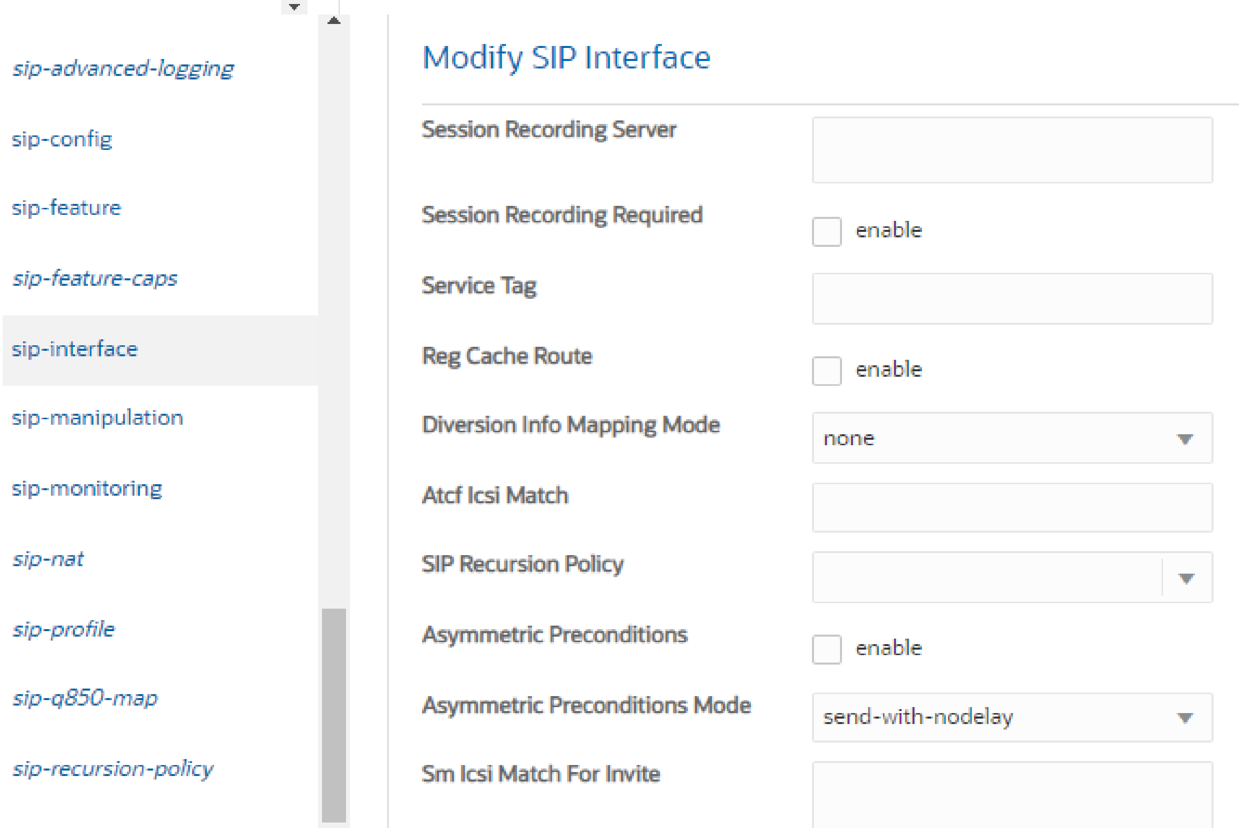


Figure 52: SIP Interface for PSTN Gateway Cont.

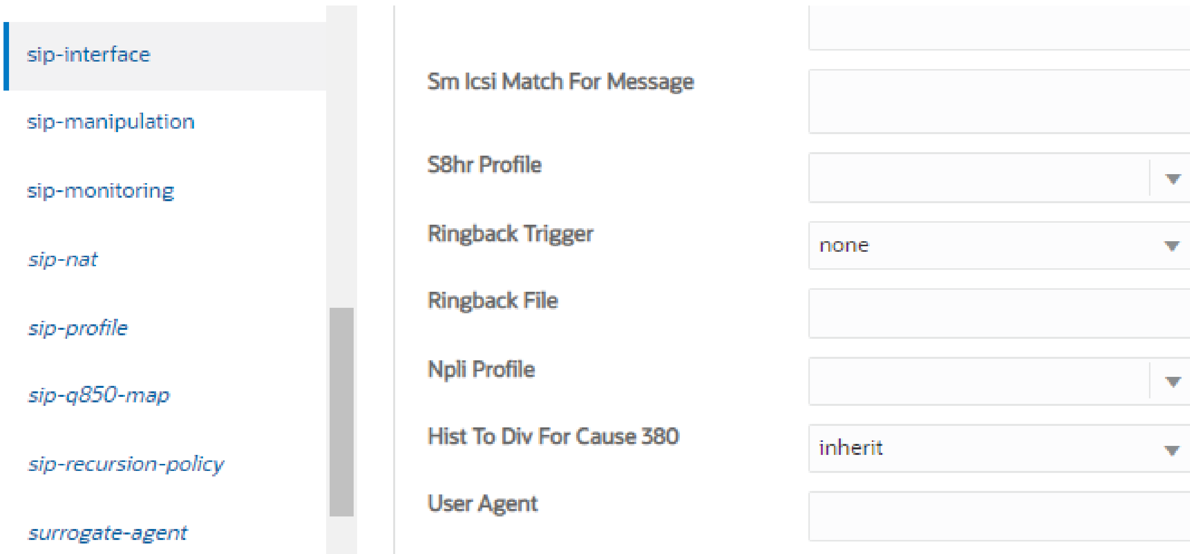


Figure 53: SIP Interface for PSTN Gateway Cont.

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.

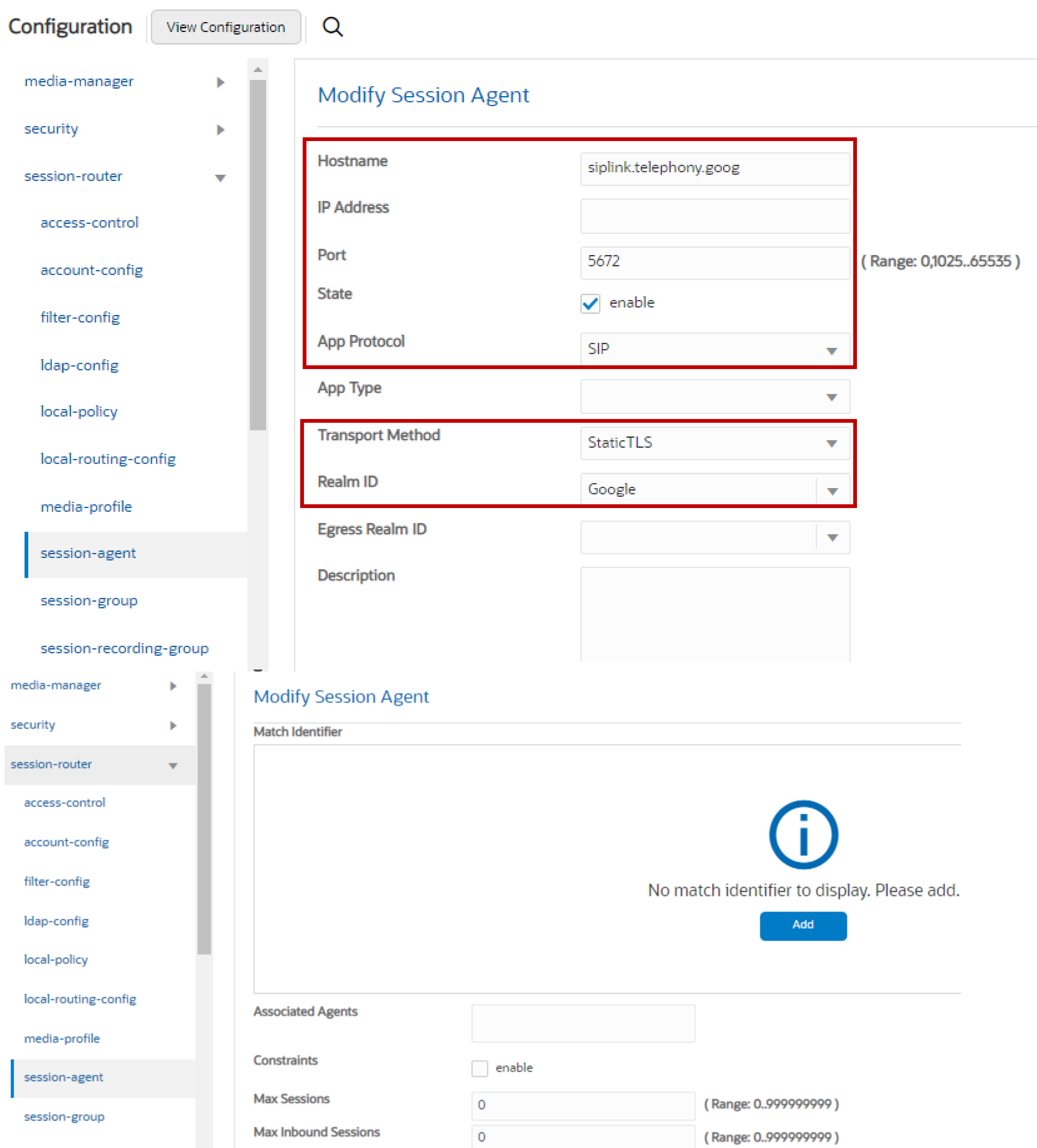
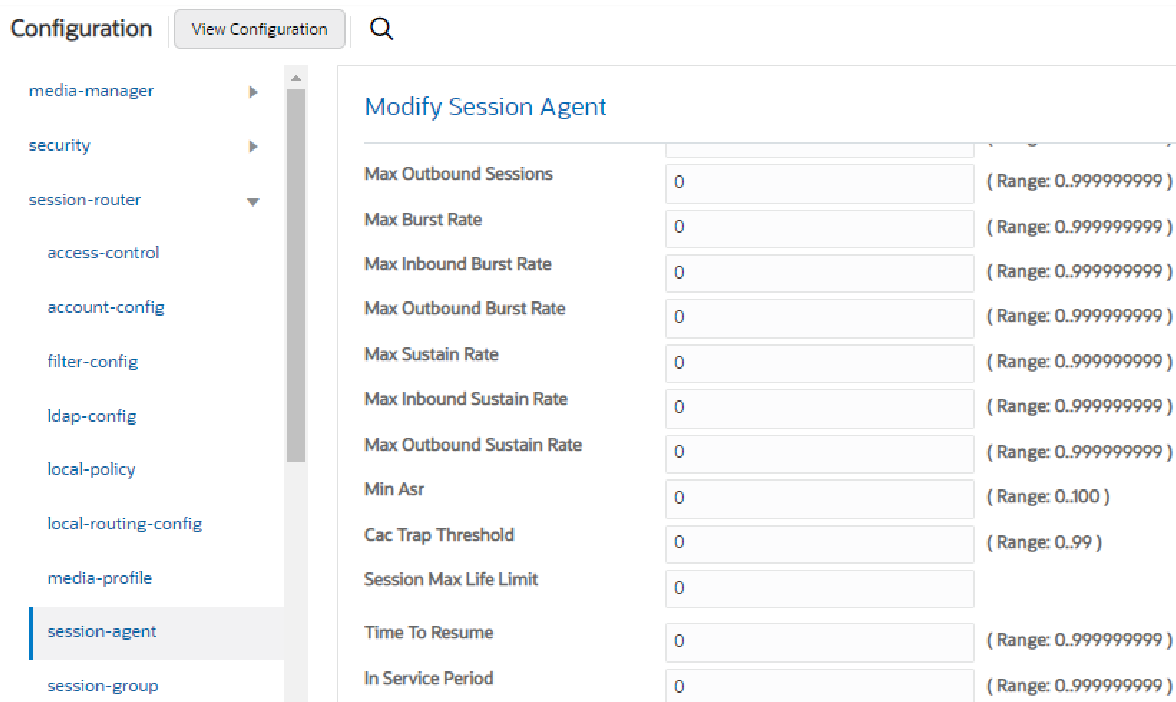


Figure 54: Session Agent for Google Voice



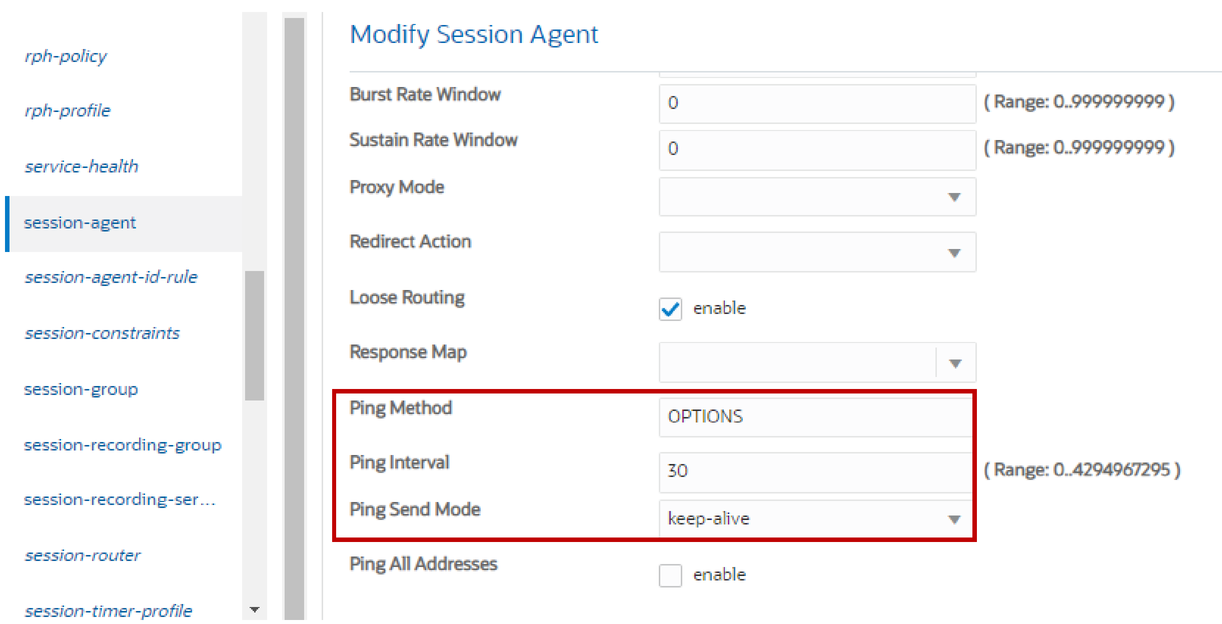


Figure 55: Session Agent for Google Voice Cont.

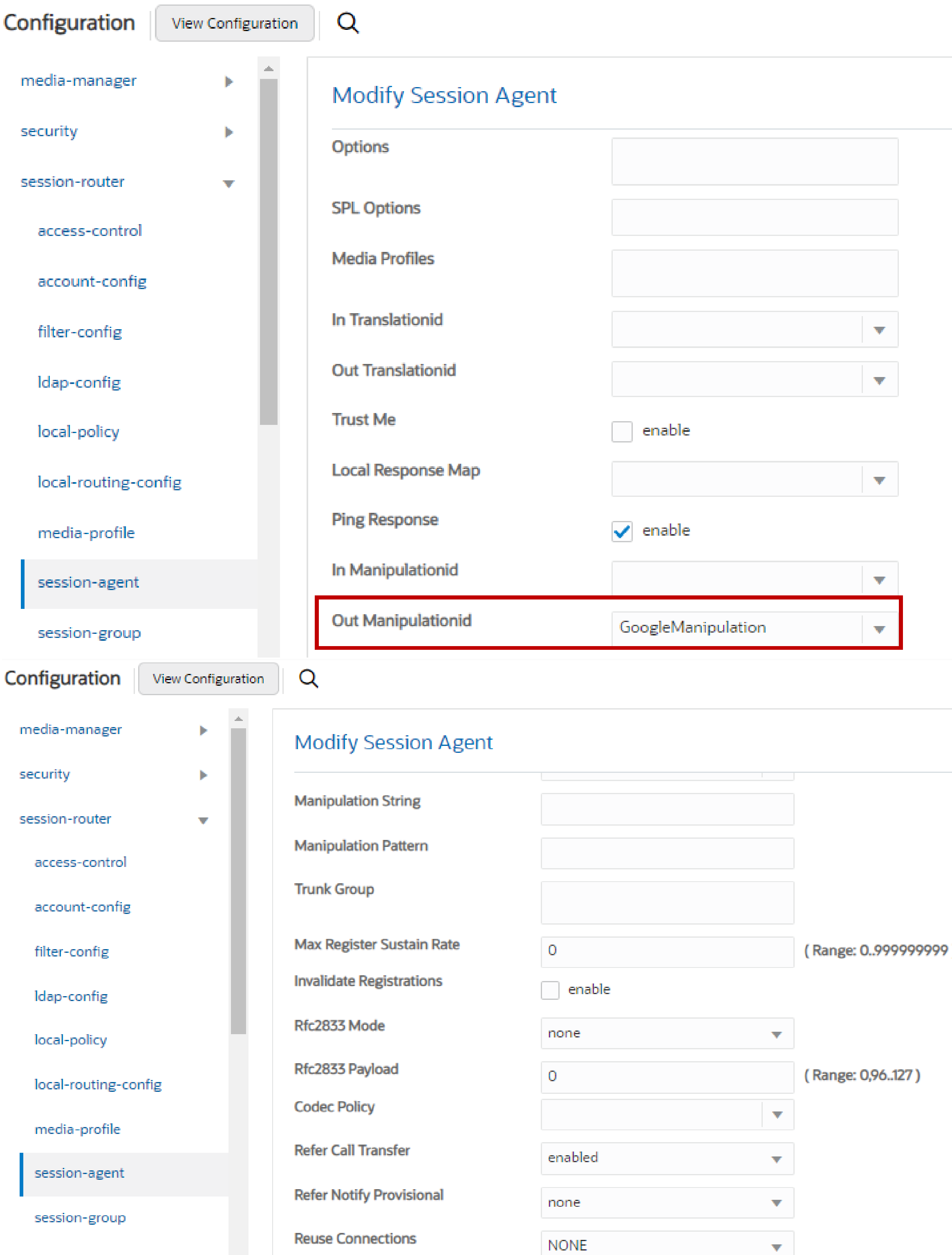


Figure 56: Session Agent for Google Voice Cont.

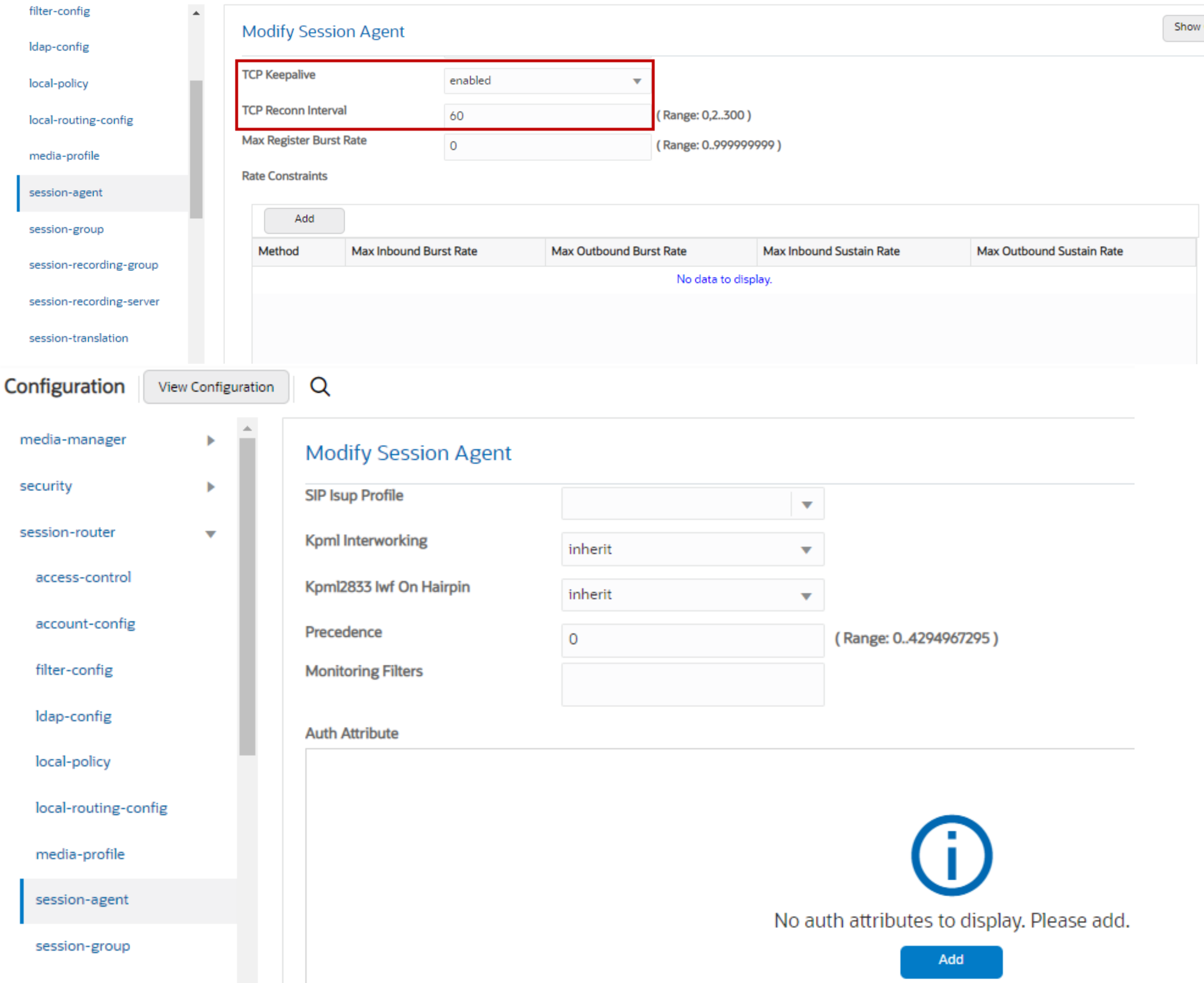


Figure 57: Session Agent for Google Voice Cont.

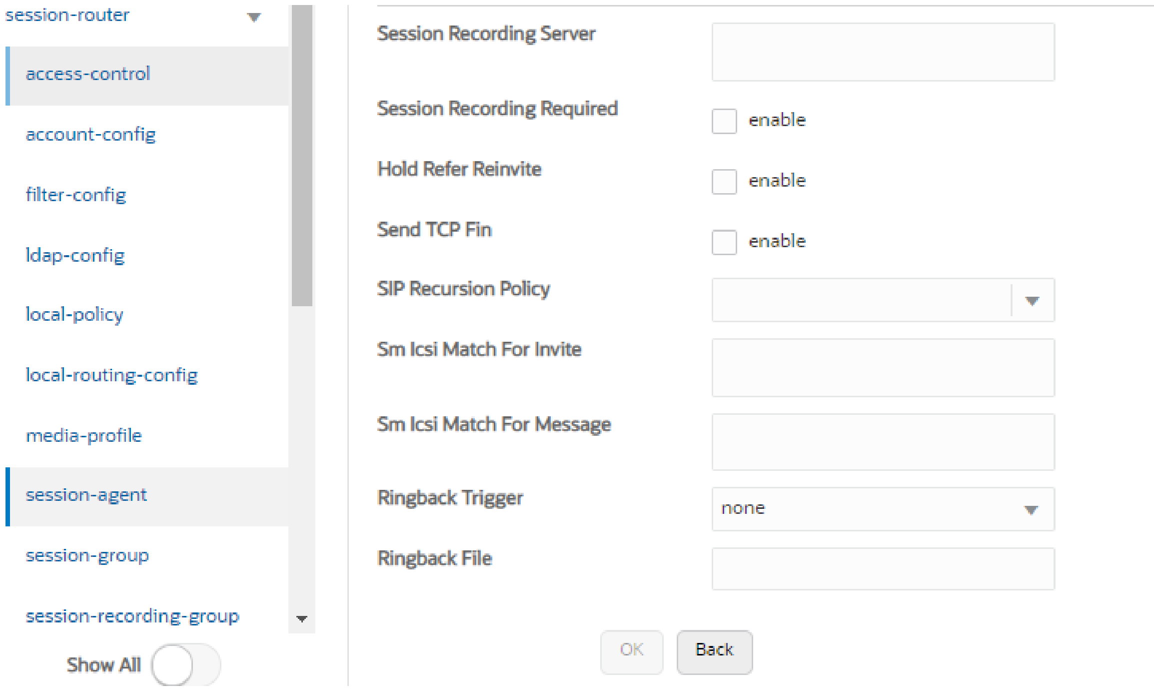


Figure 58: Session Agent for Google Voice Cont.

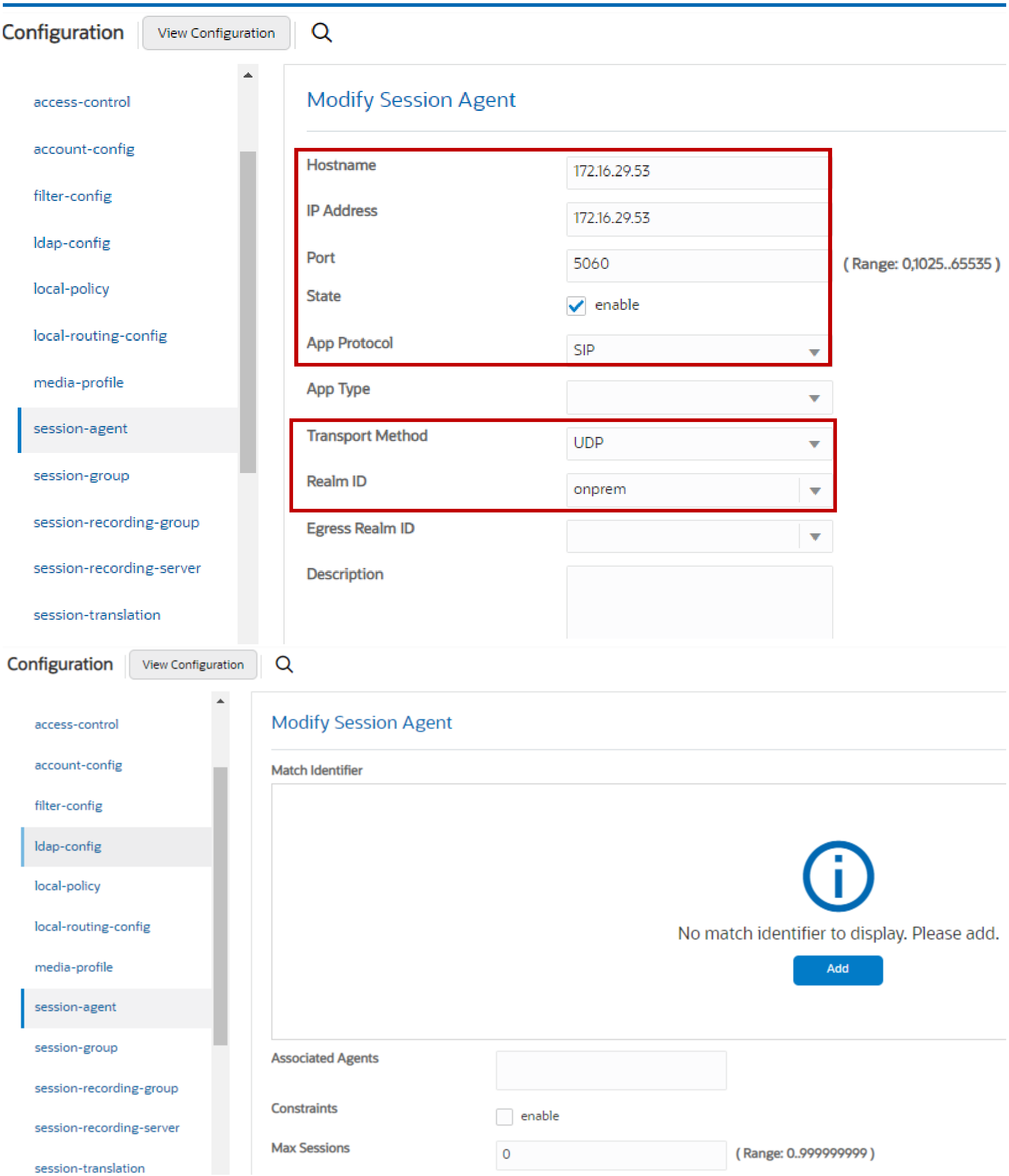


Figure 59: Session Agent for OnPrem PBX

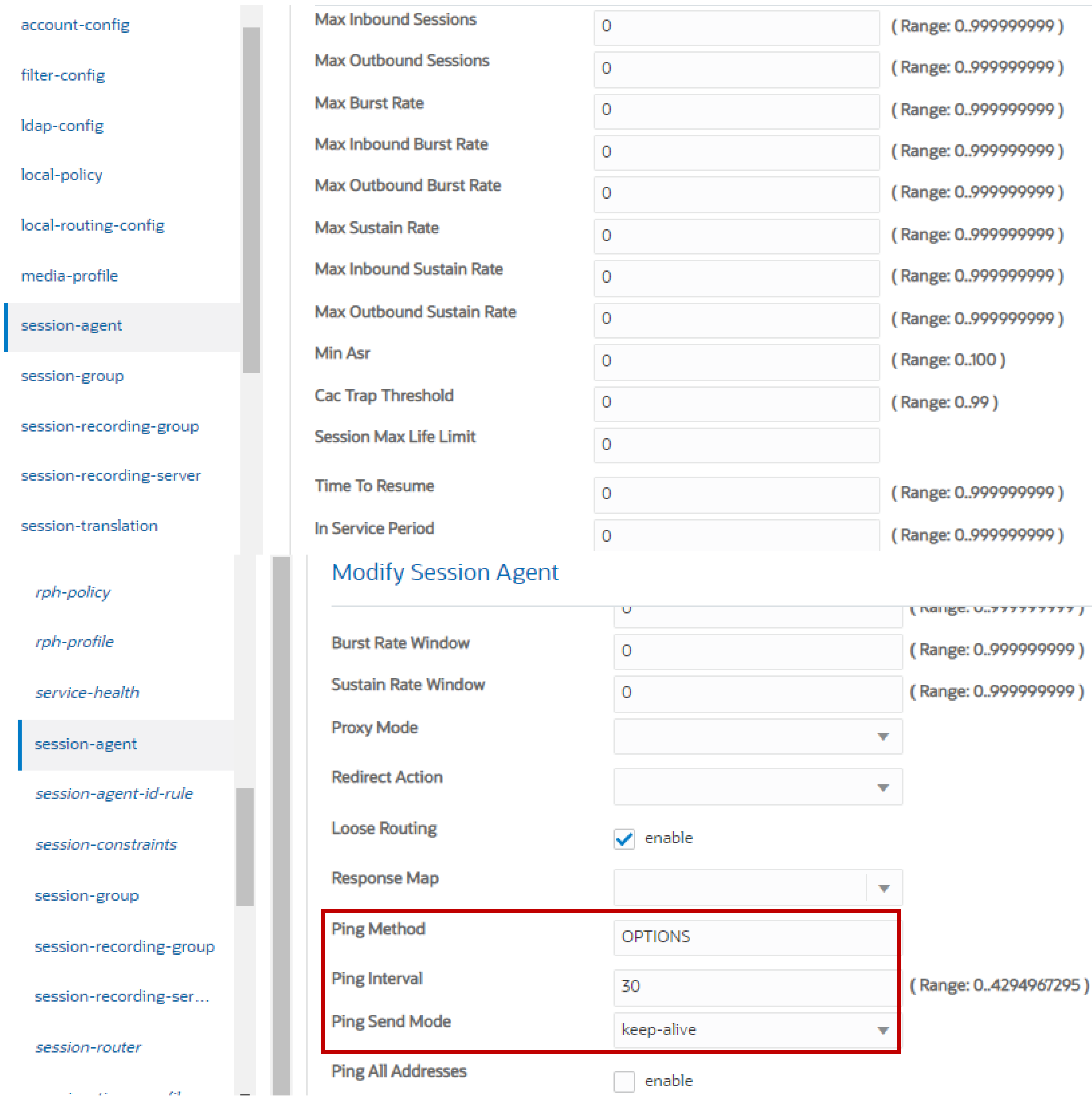


Figure 60: Session Agent for OnPrem PBX Cont.

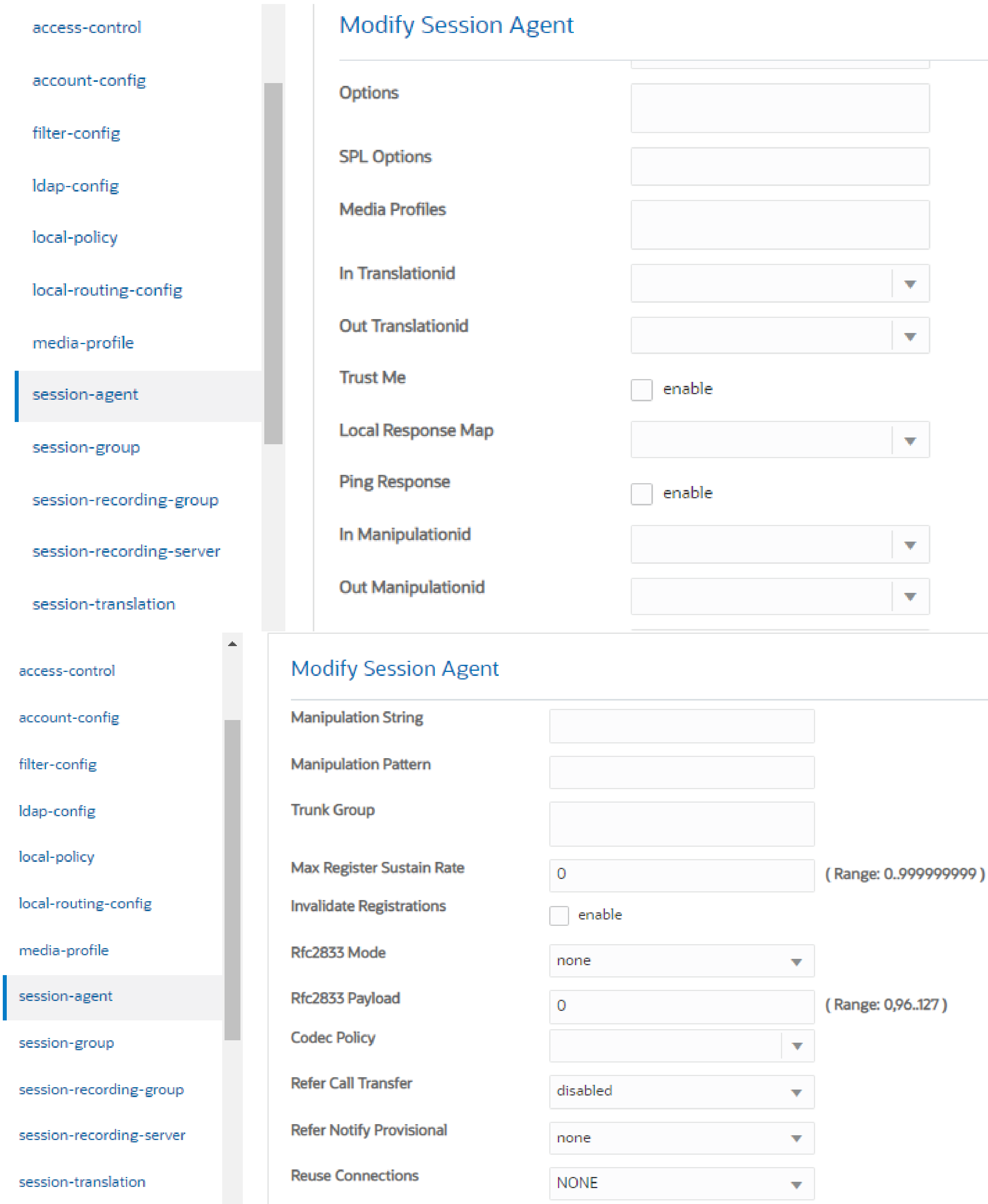


Figure 61: Session Agent for OnPrem PBX Cont.

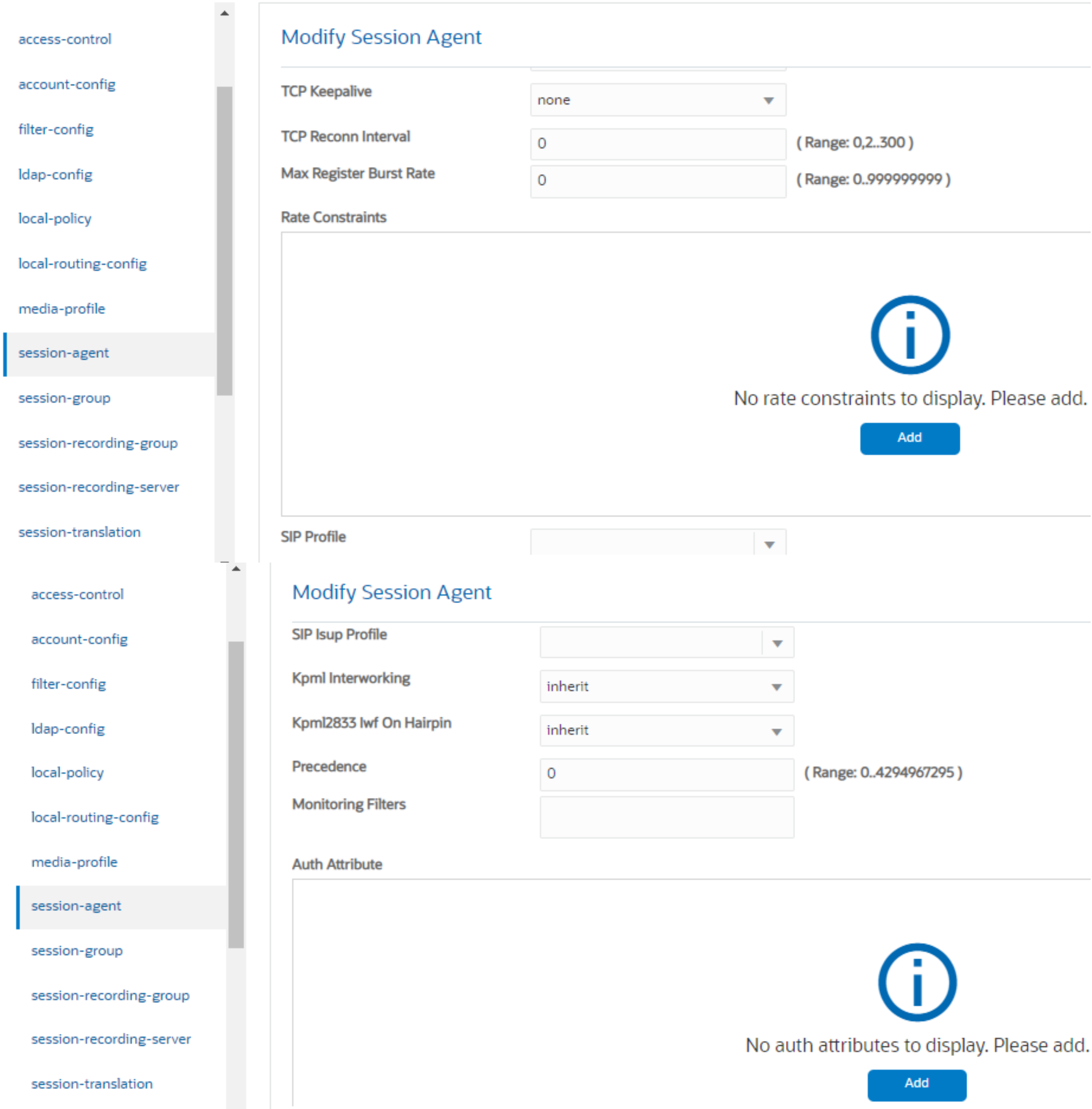


Figure 62: Session Agent for OnPrem PBX Cont.

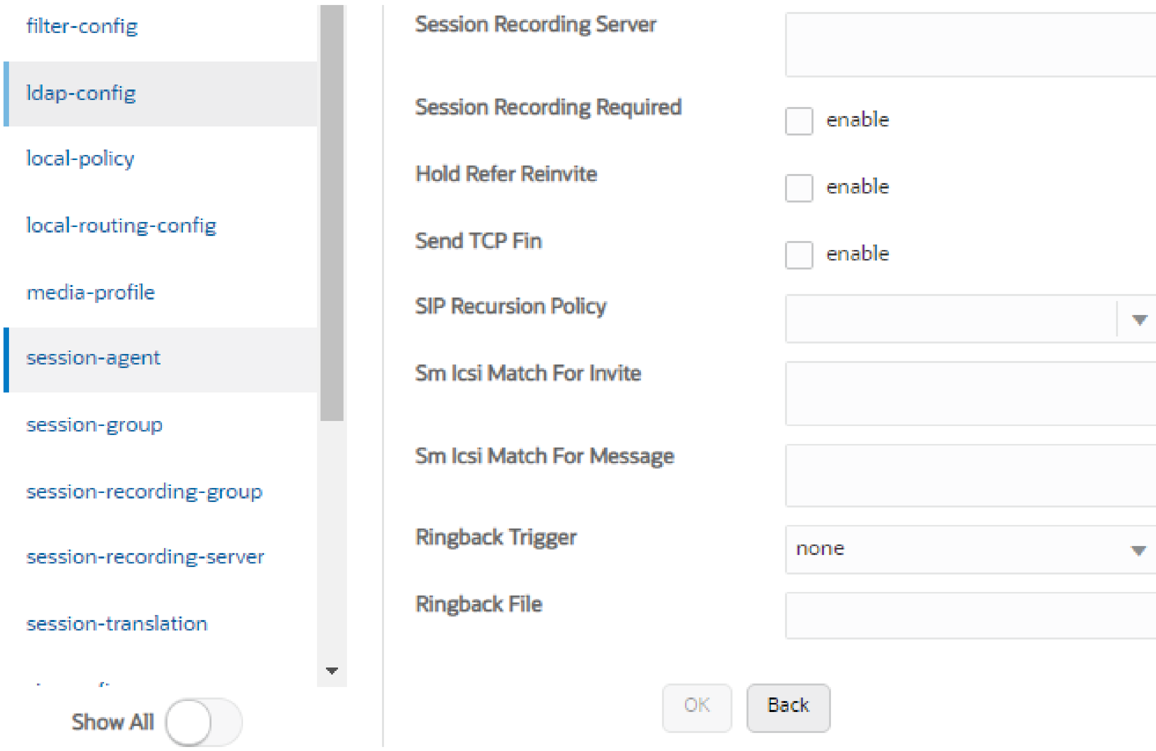


Figure 63: Session Agent for OnPrem PBX Cont.

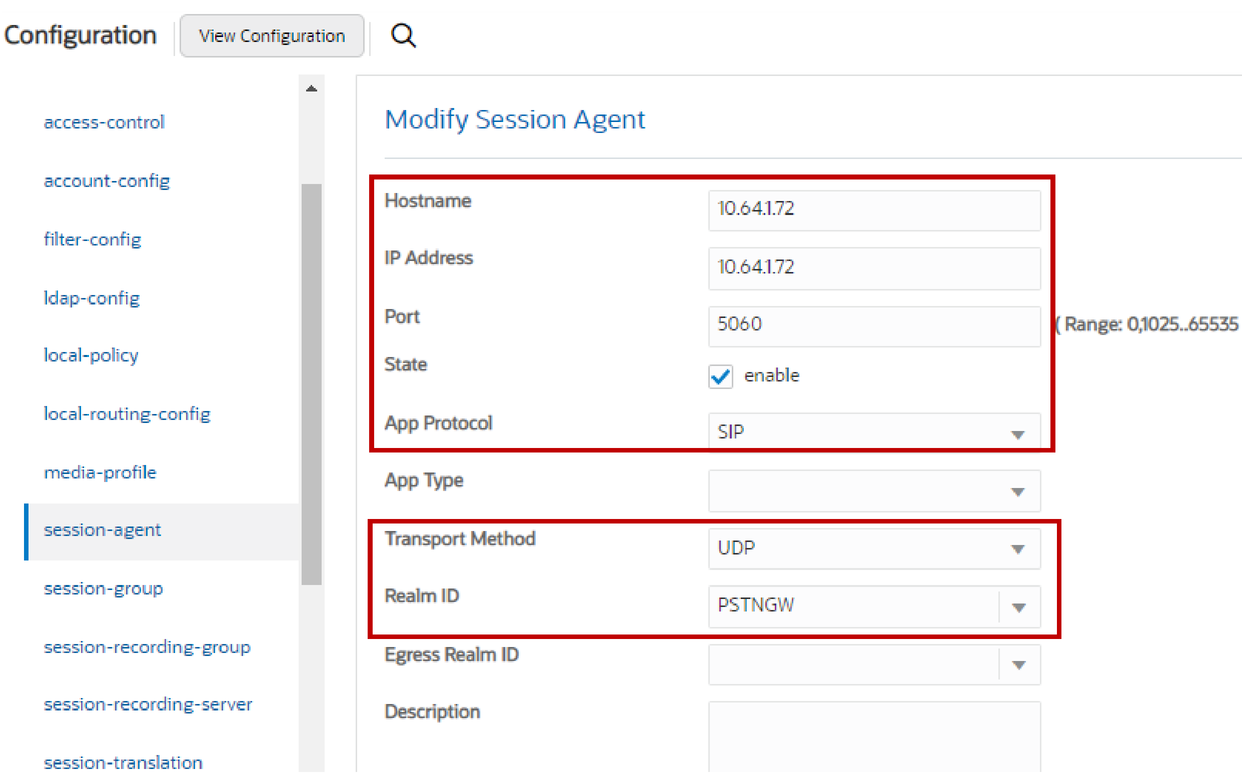


Figure 64: Session Agent for PSTN Gateway

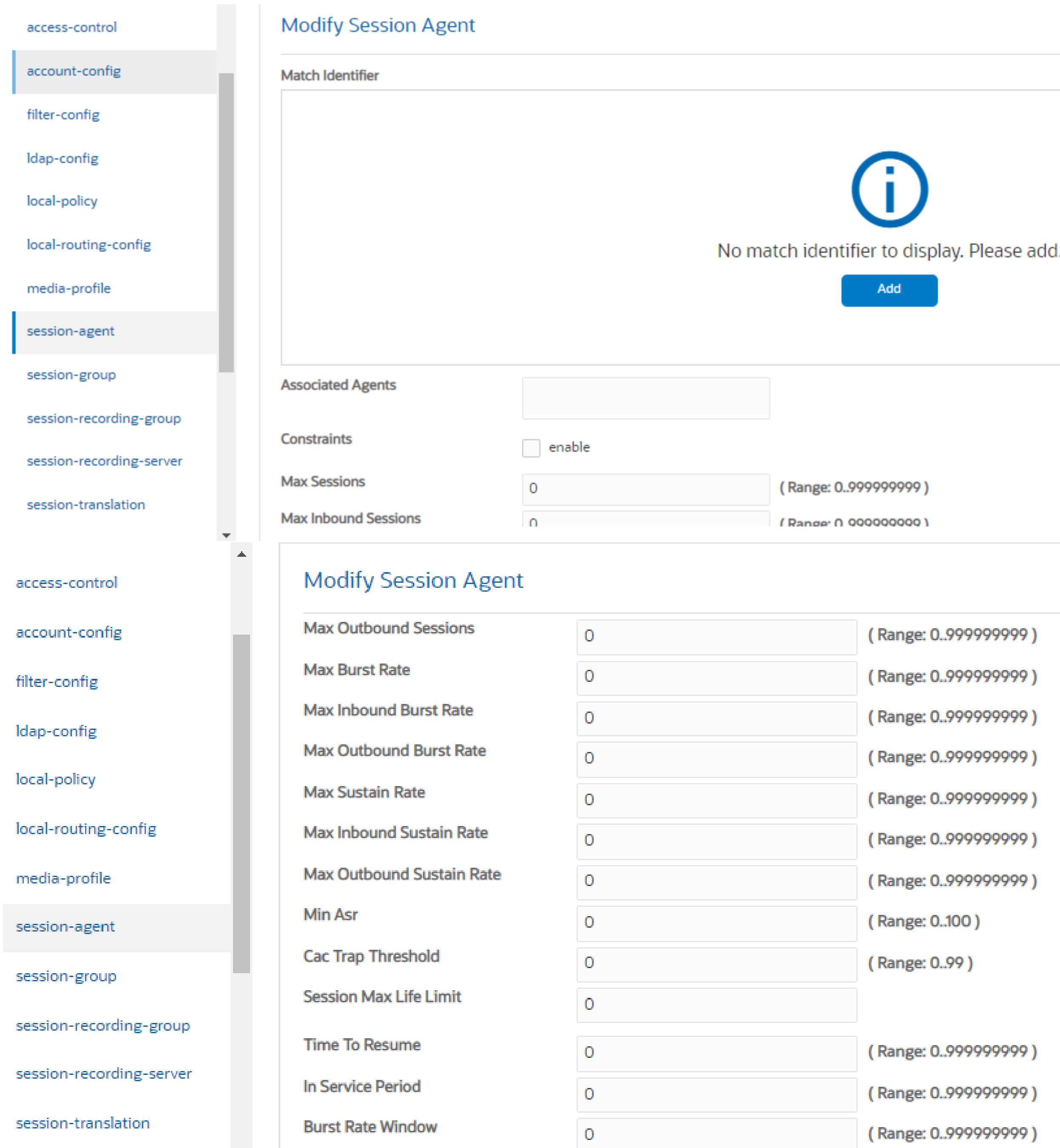


Figure 65: Session Agent for PSTN Gateway Cont.

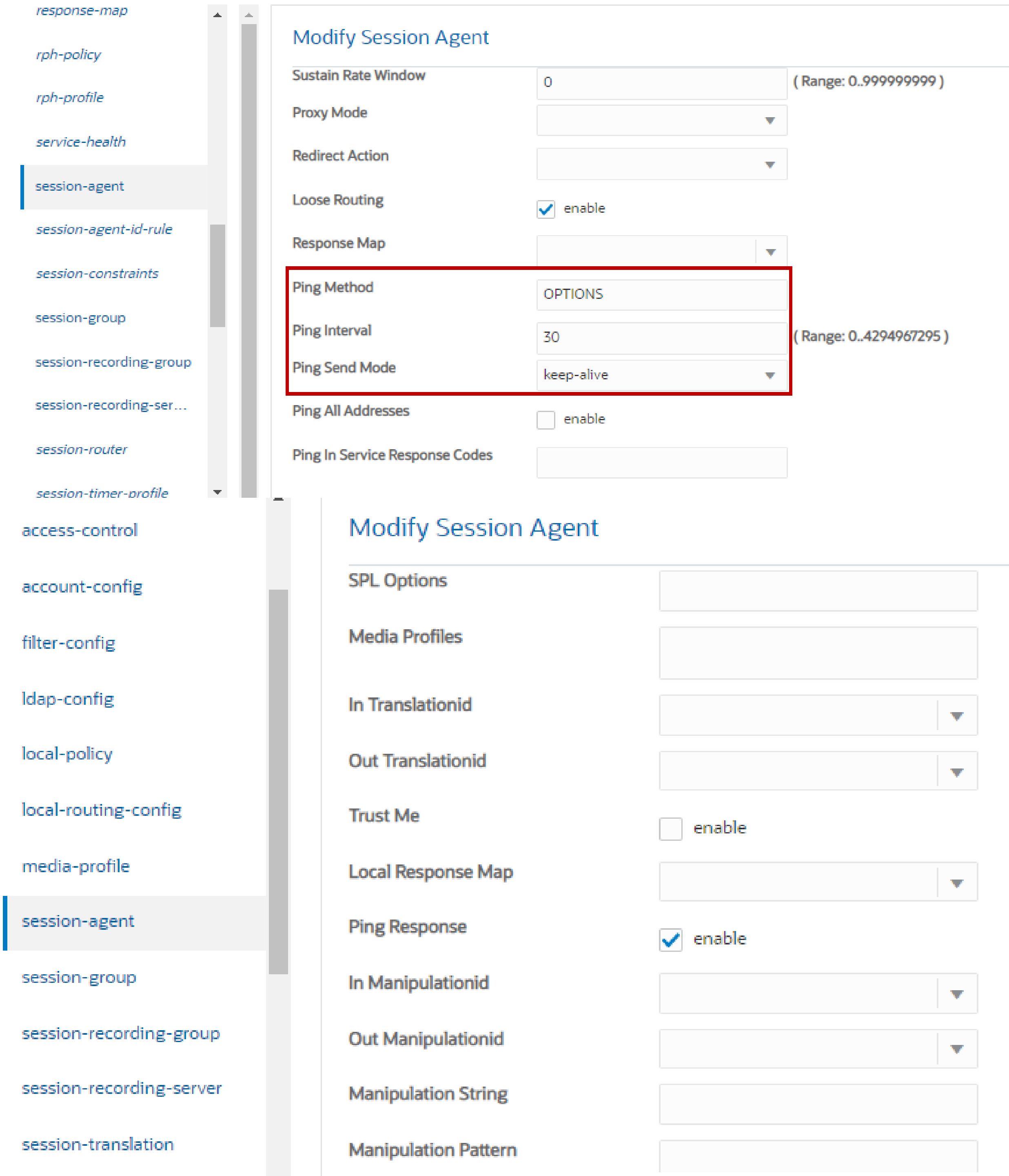


Figure 66: Session Agent for PSTN Gateway Cont.

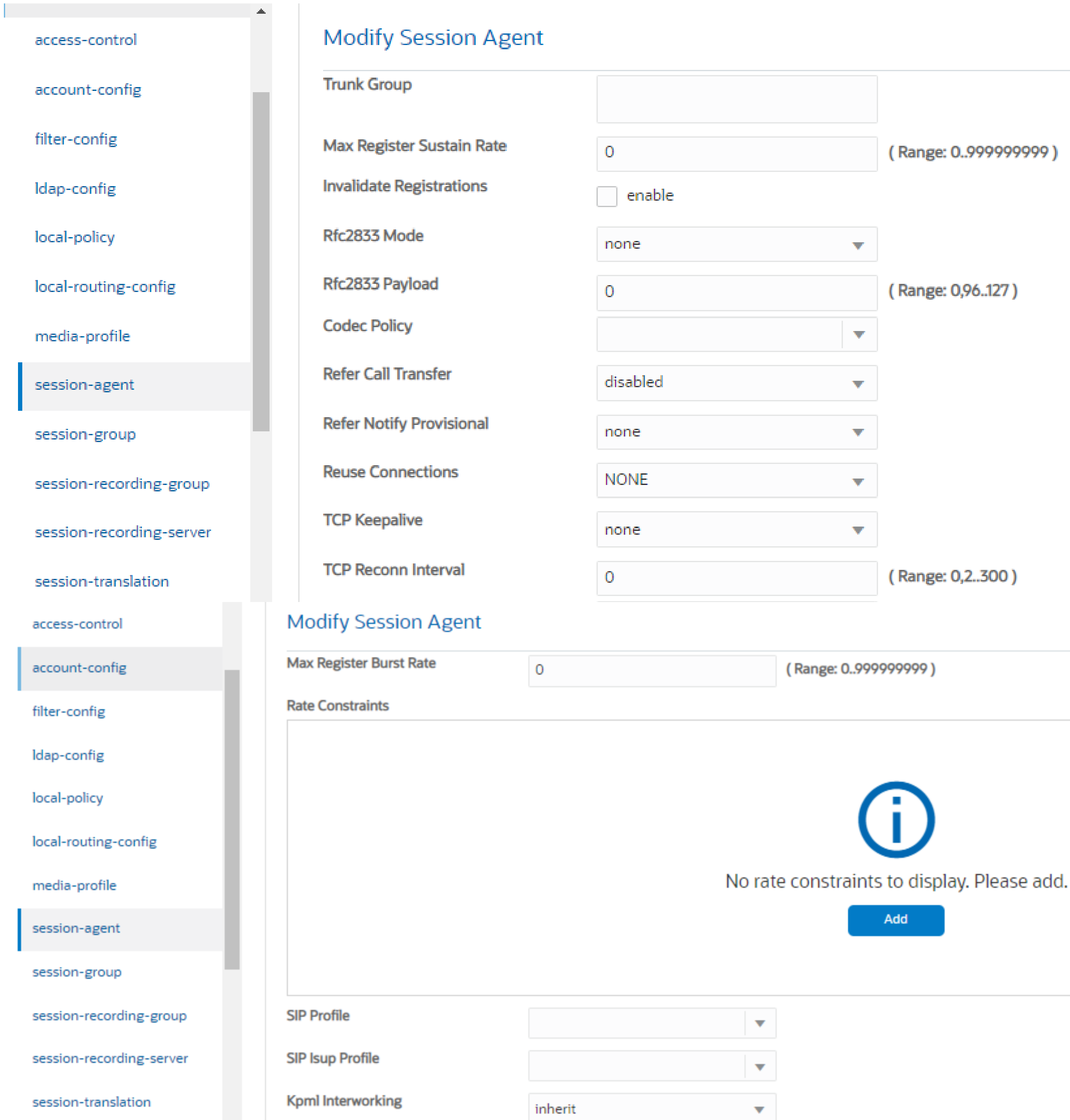


Figure 67: Session Agent for PSTN Gateway Cont.

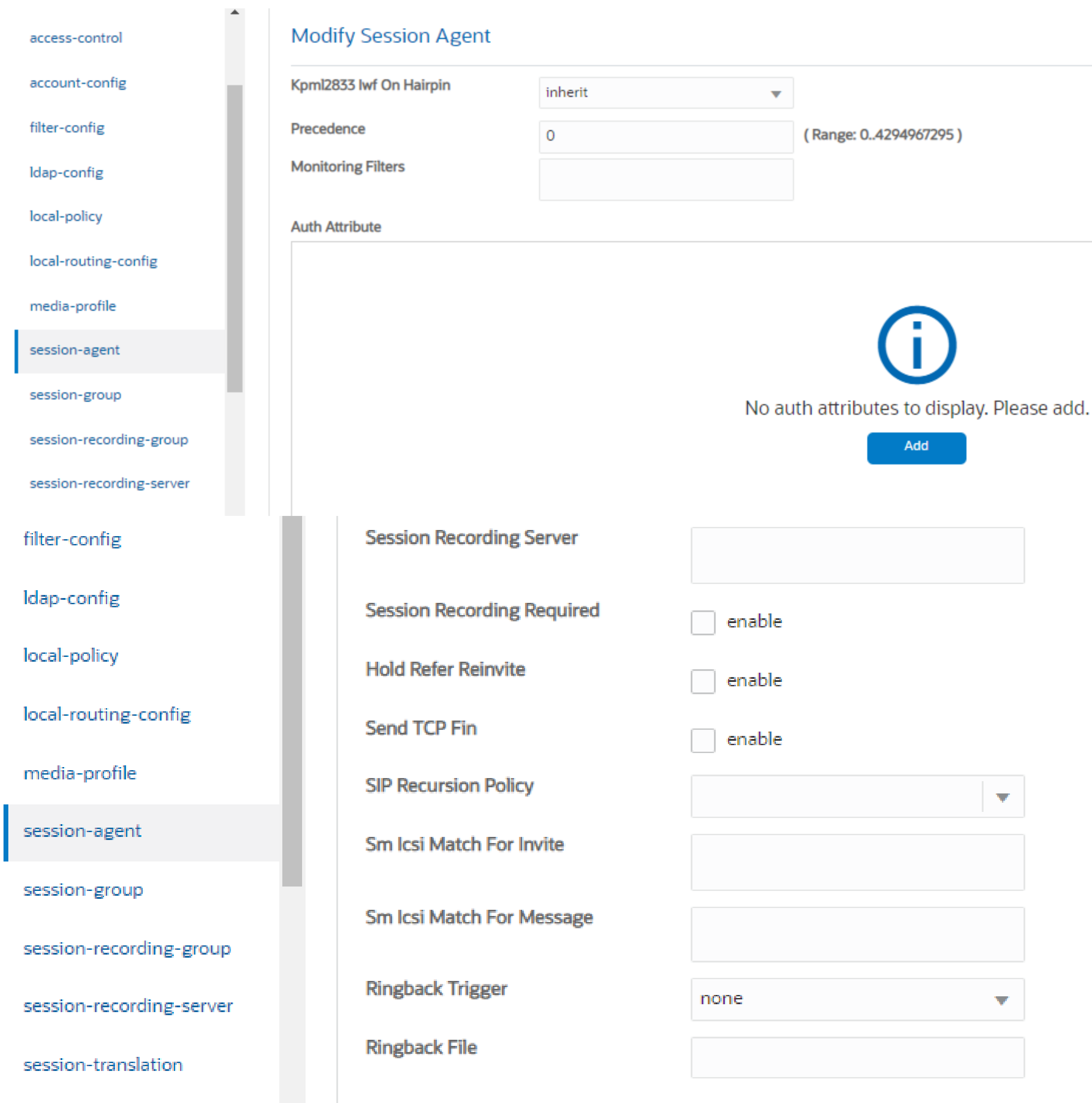


Figure 68: Session Agent for PSTN Gateway Cont.

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.

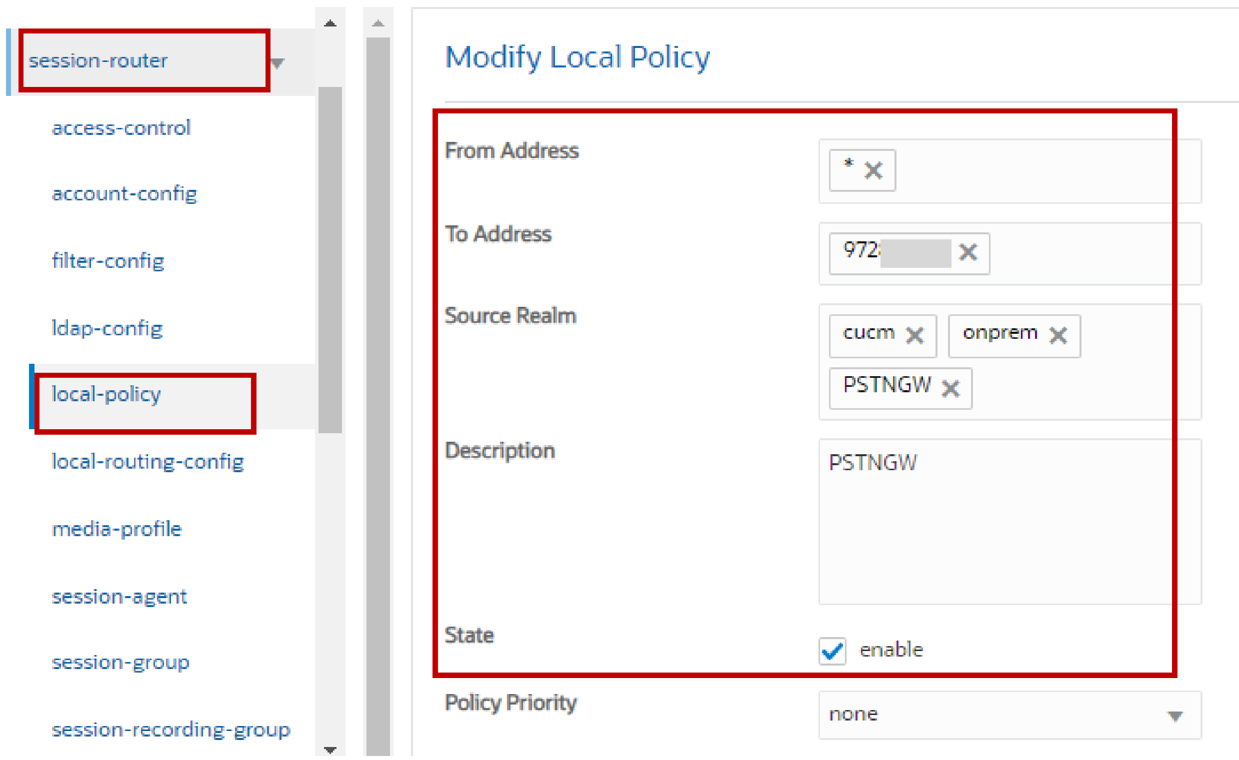


Figure 69: Local Policy towards Google Voice

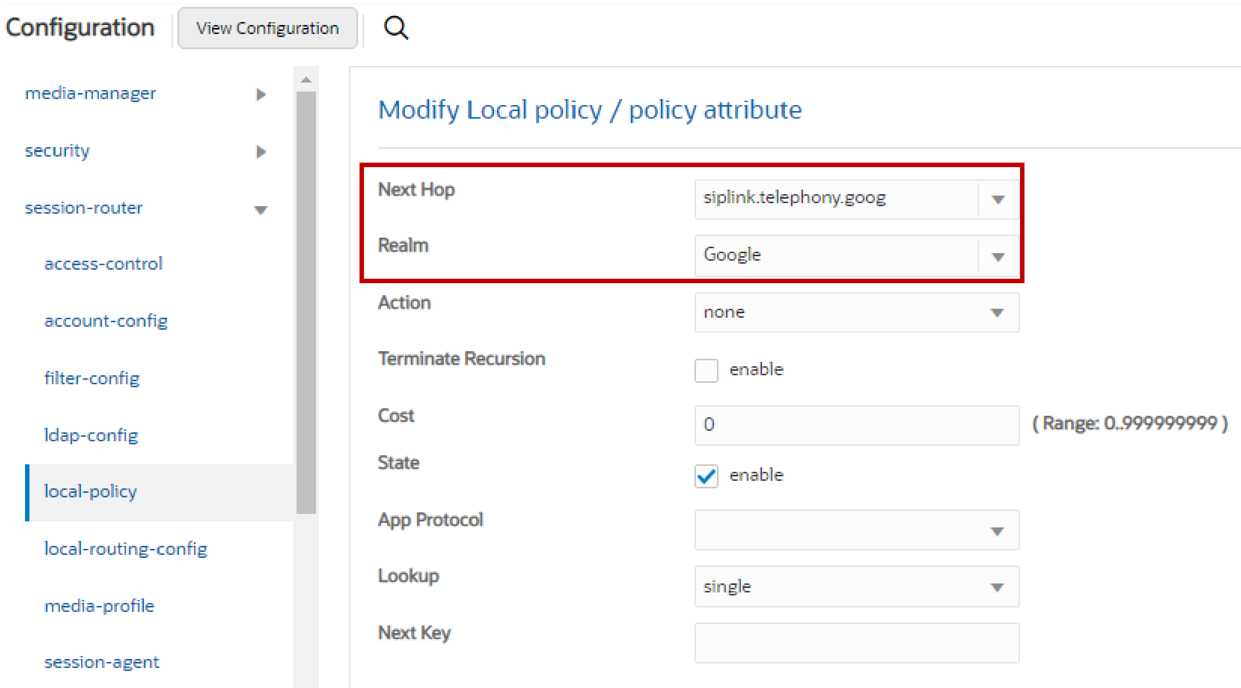


Figure 70: Local Policy towards Google Voice Cont.

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

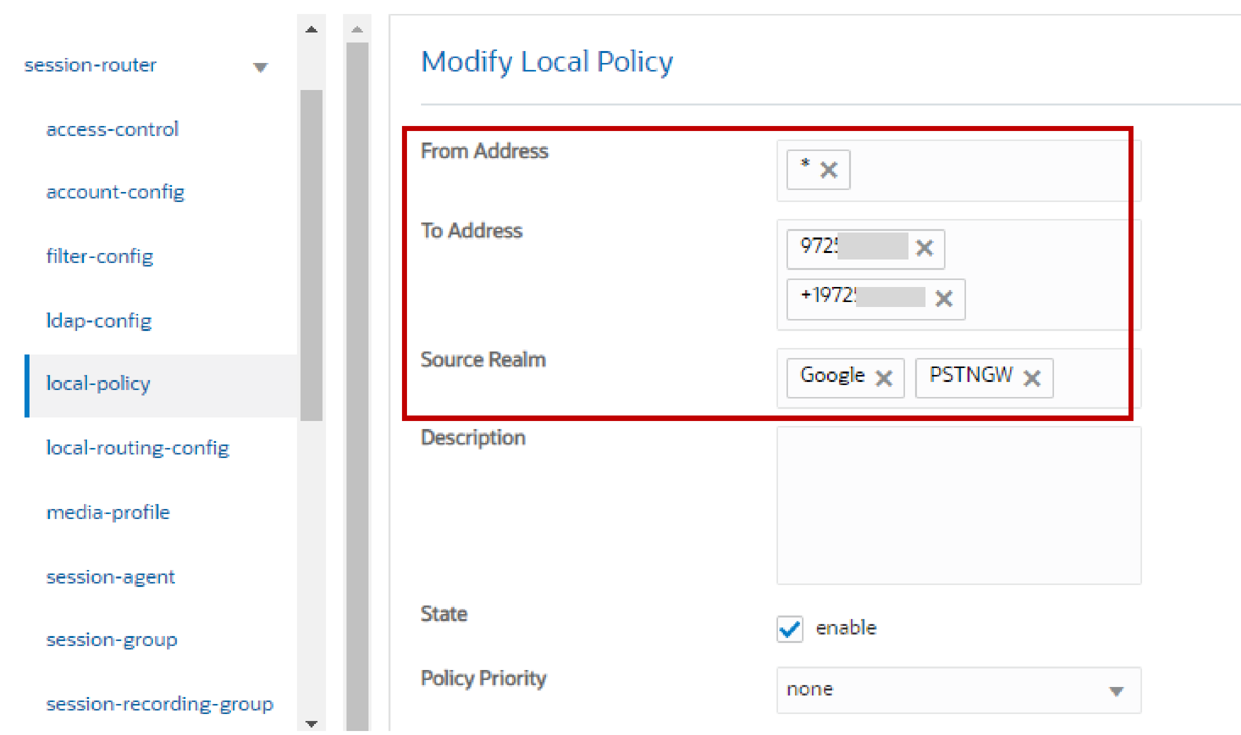


Figure 71: Local Policy towards OnPrem PBX

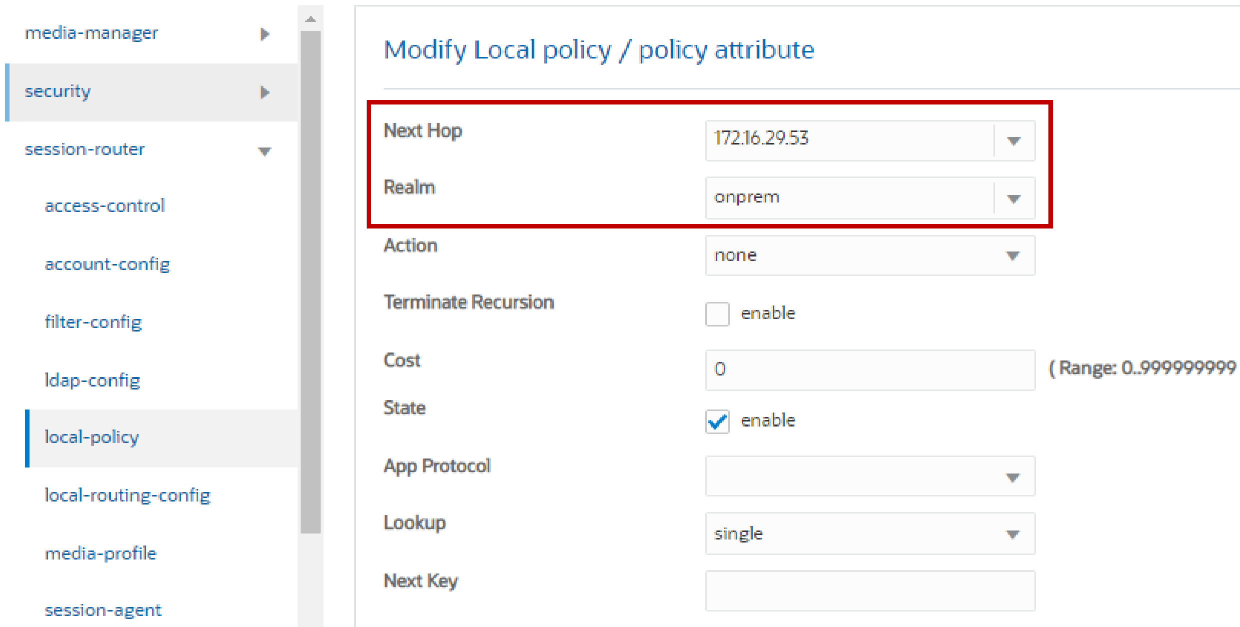


Figure 72: Local Policy towards OnPrem PBX Cont.

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

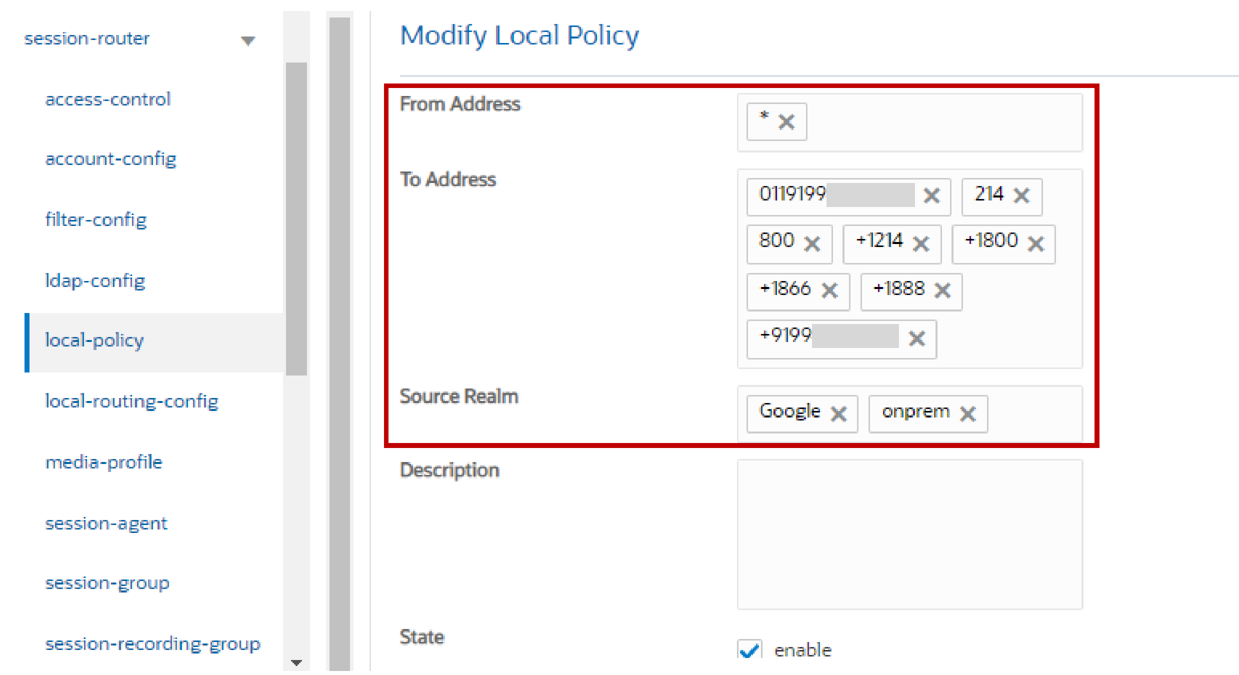


Figure 73: Local Policy towards PSTN Gateway

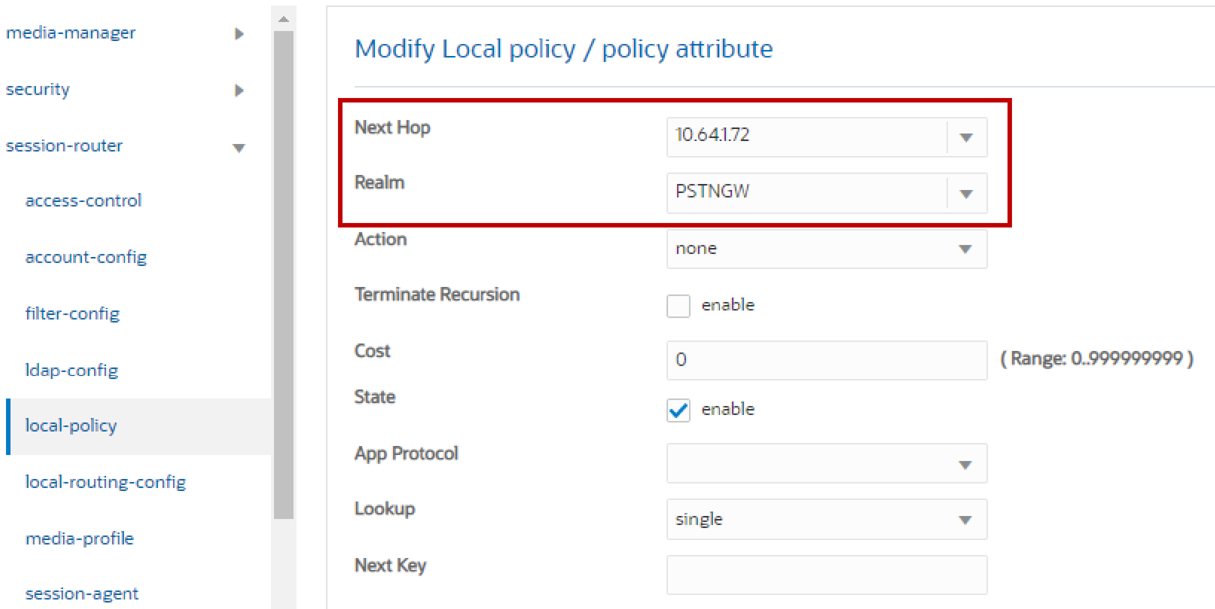


Figure 74: Local Policy towards PSTN Gateway Cont.

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

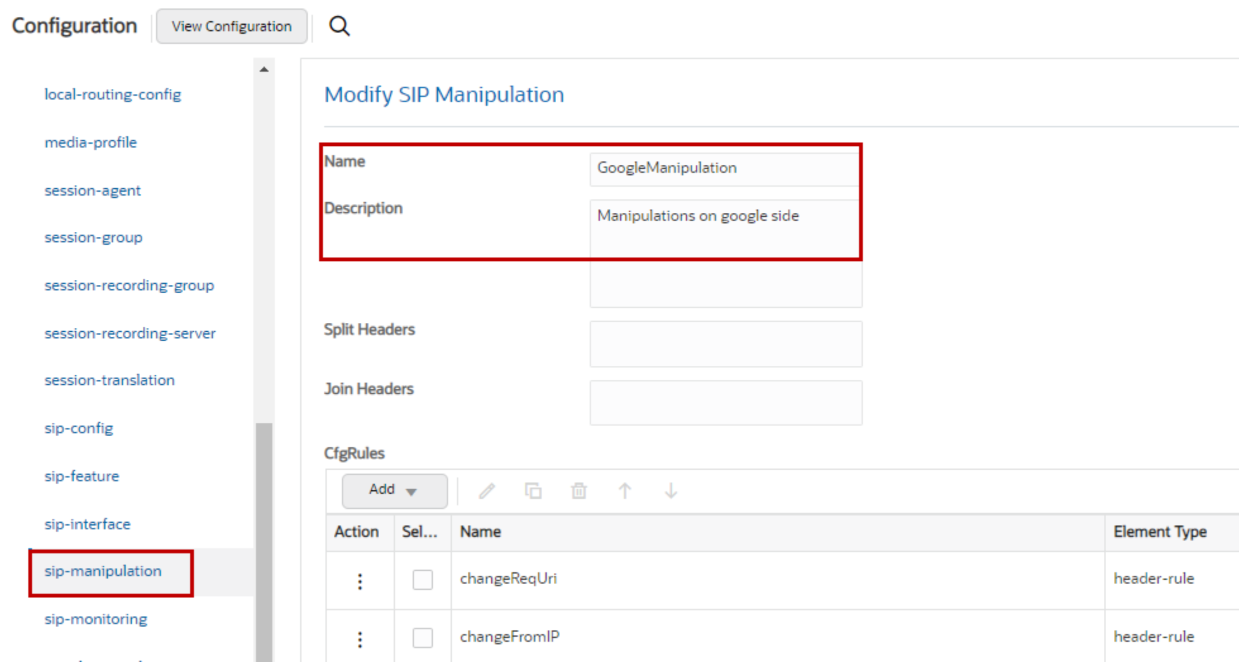


Figure 75: SIP Manipulation towards Google Voice

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

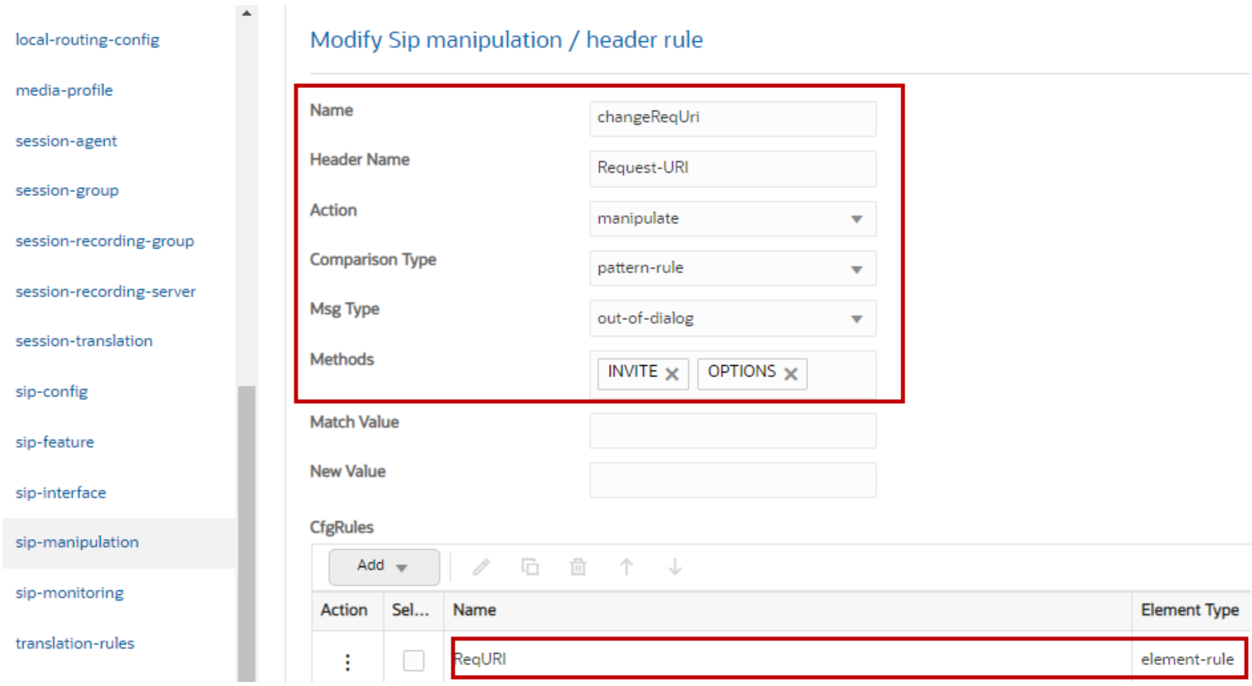


Figure 76: SIP Manipulation towards Google Voice to change Request-URI

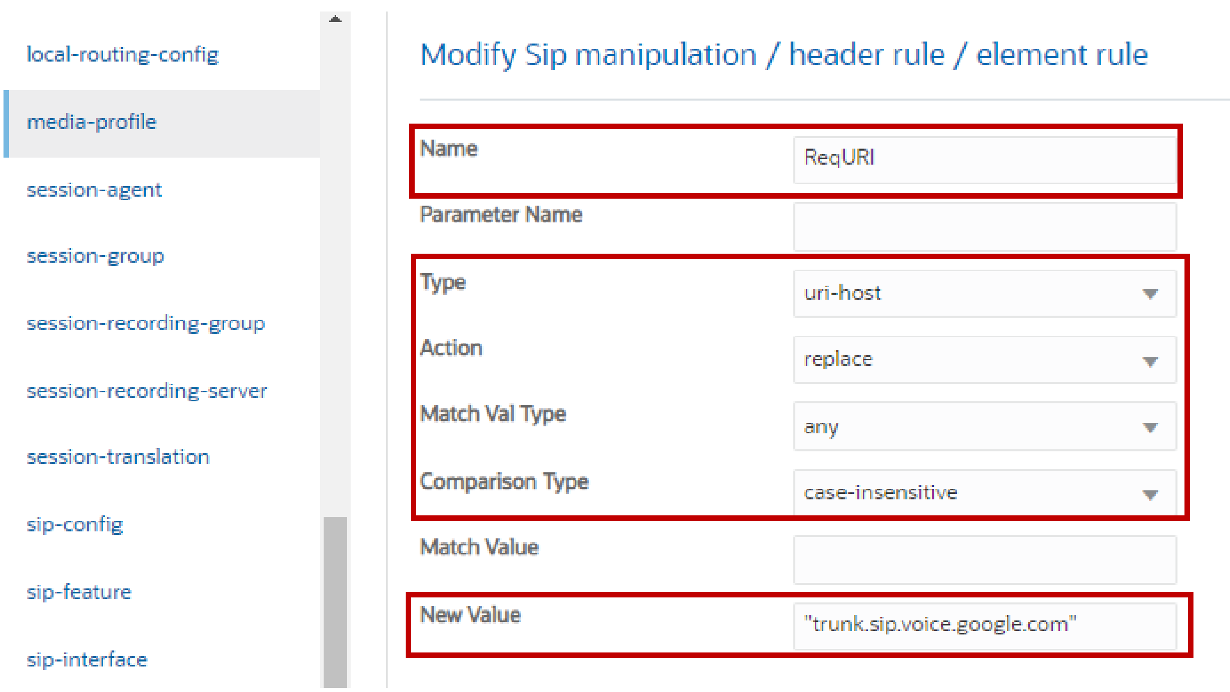


Figure 77: SIP Manipulation towards Google Voice to change Request-URI Cont.

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

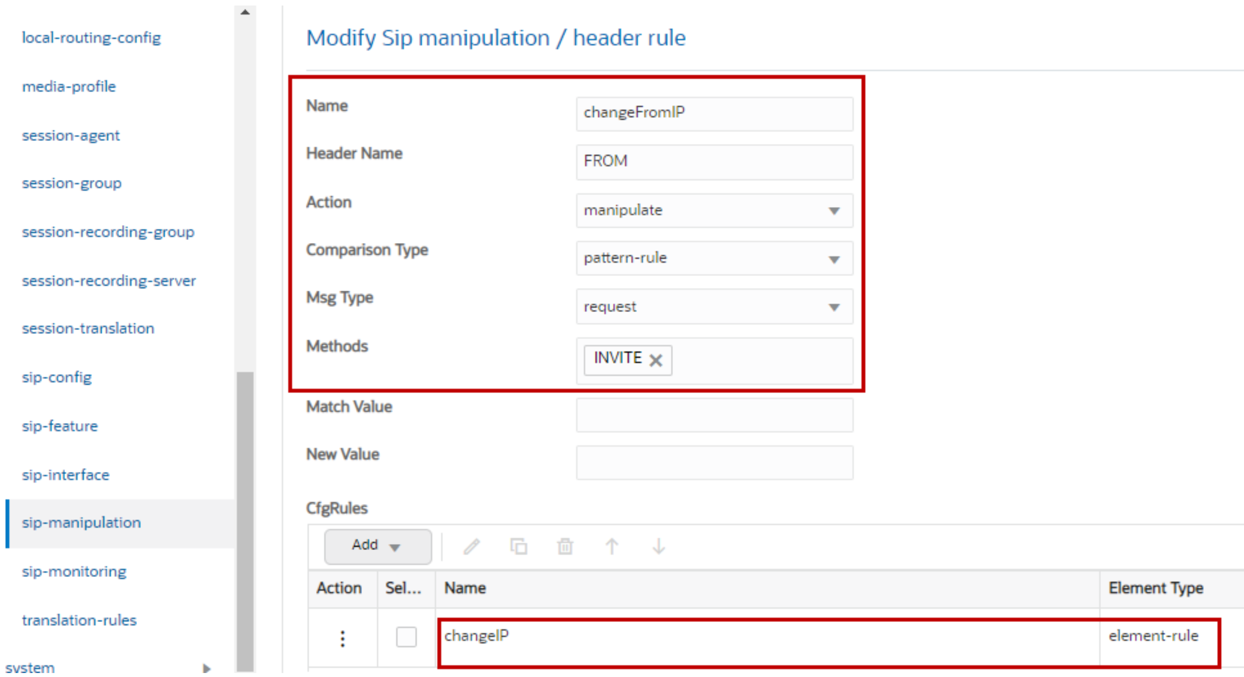


Figure 78: SIP Manipulation towards Google Voice to change host part of FROM IP

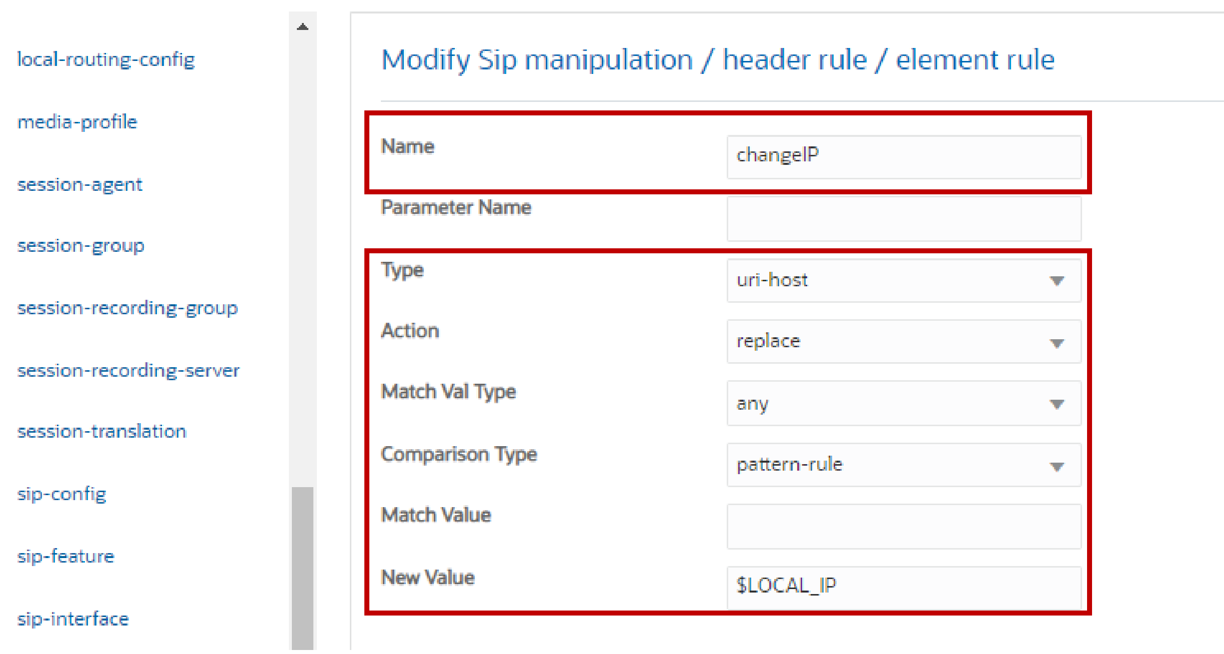


Figure 79: SIP Manipulation towards Google Voice to change host part of FROM IP Cont.

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

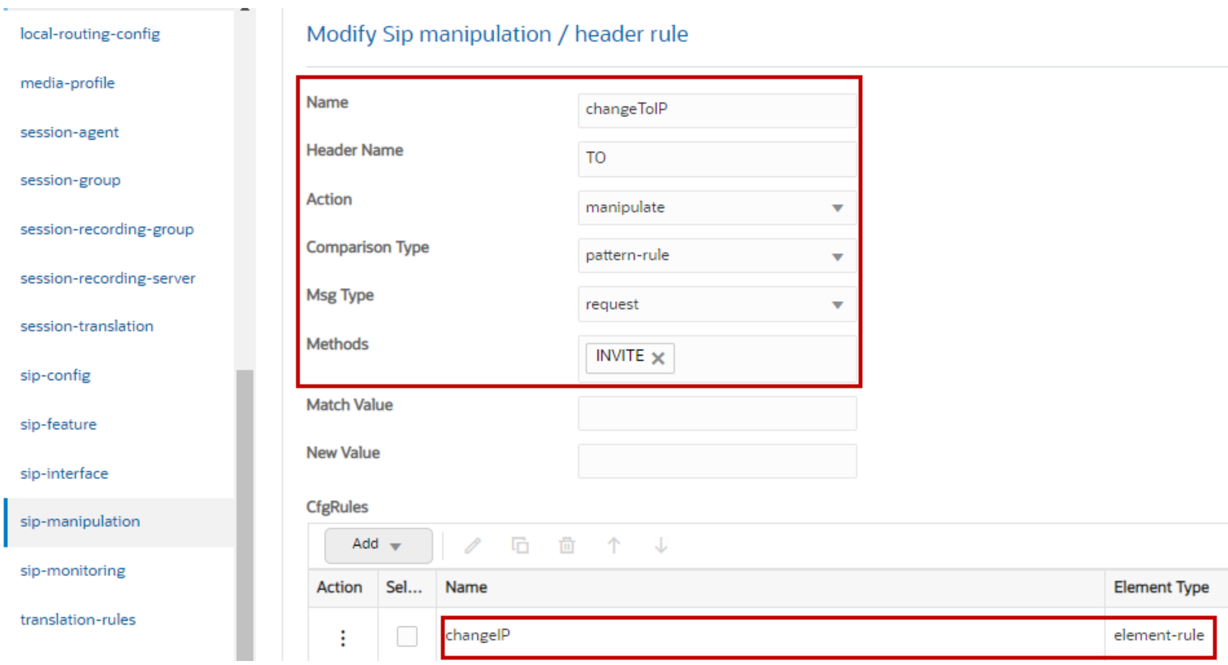


Figure 80: SIP Manipulation towards Google Voice to change host part of TO IP

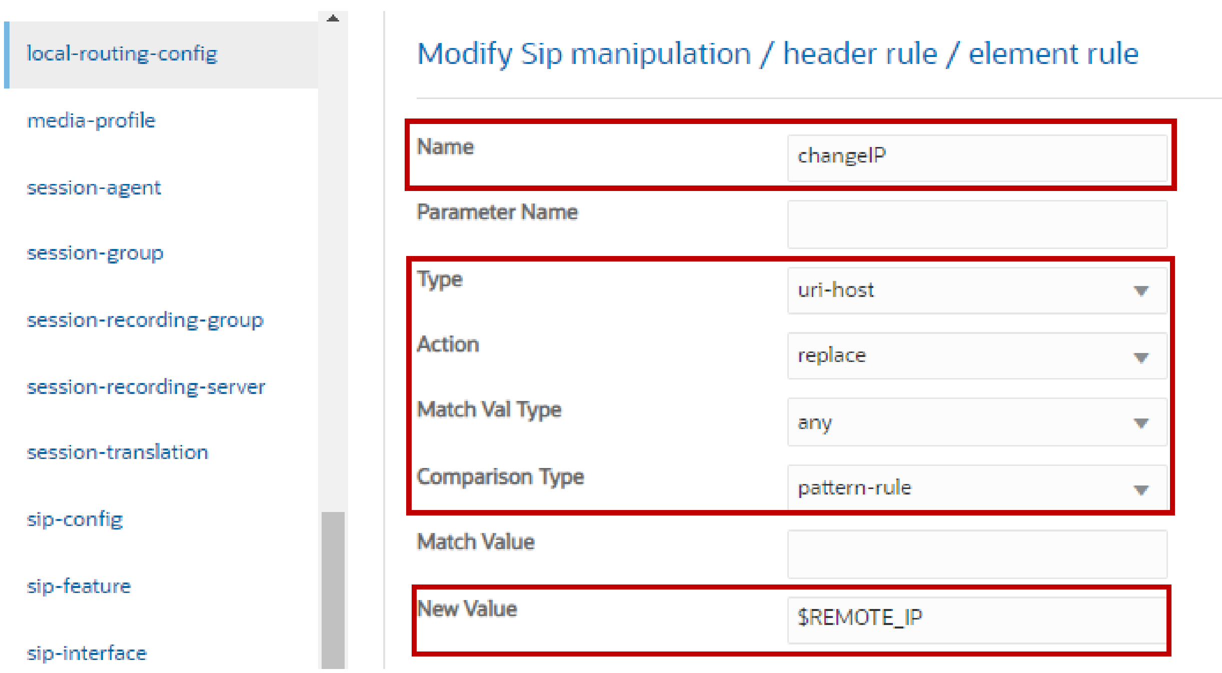


Figure 81: SIP Manipulation towards Google Voice to change host part of TO IP Cont.

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

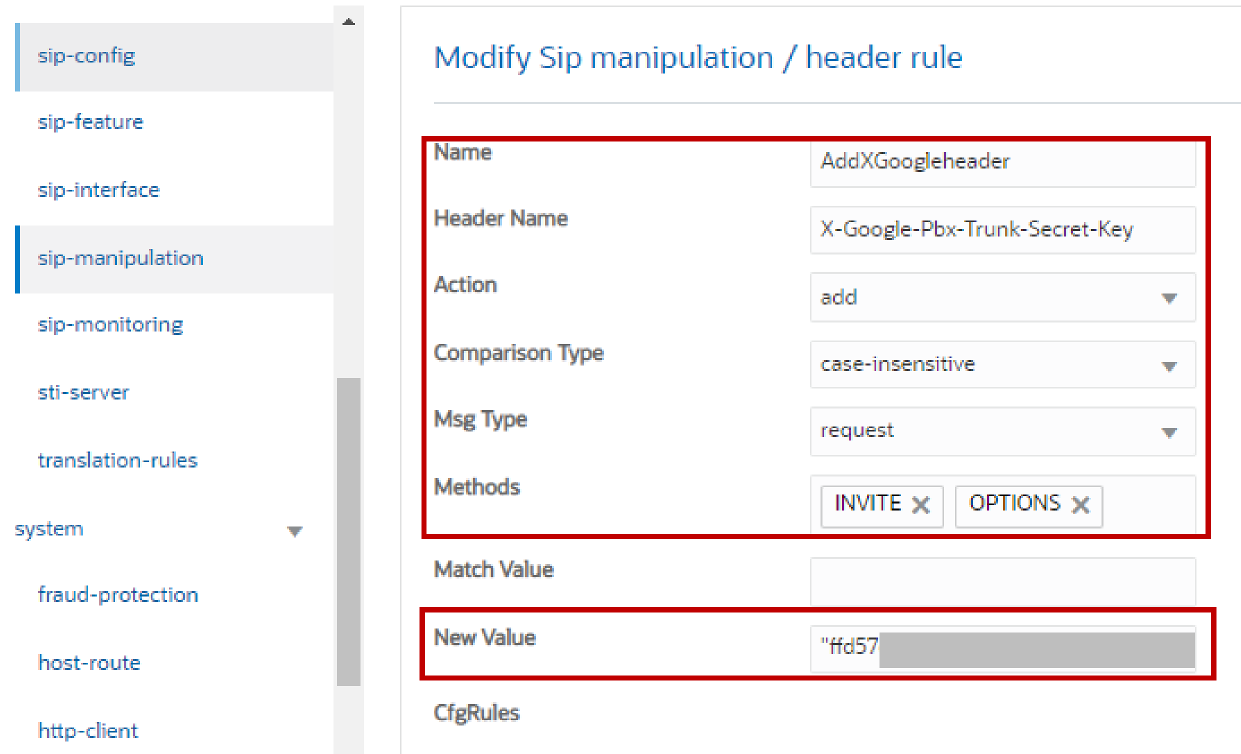
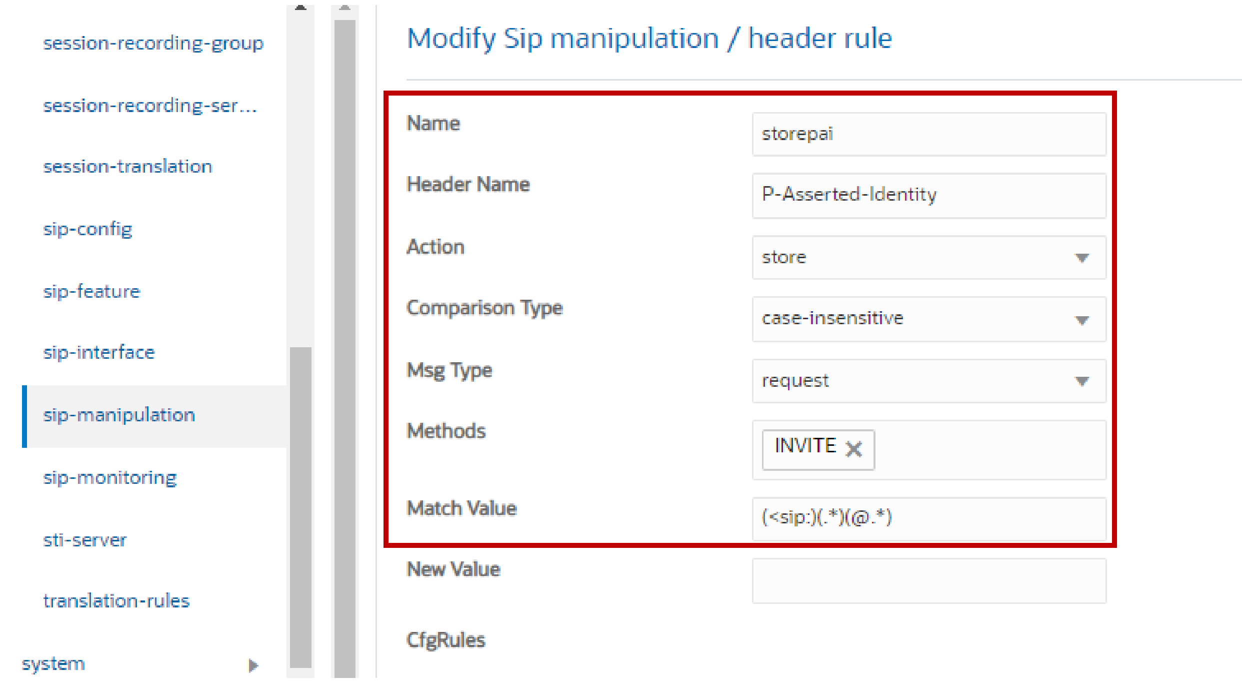


Figure 82: SIP Manipulation towards Google Voice to add X-Google-Pbx-Trunk-Secret-Key Header

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



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

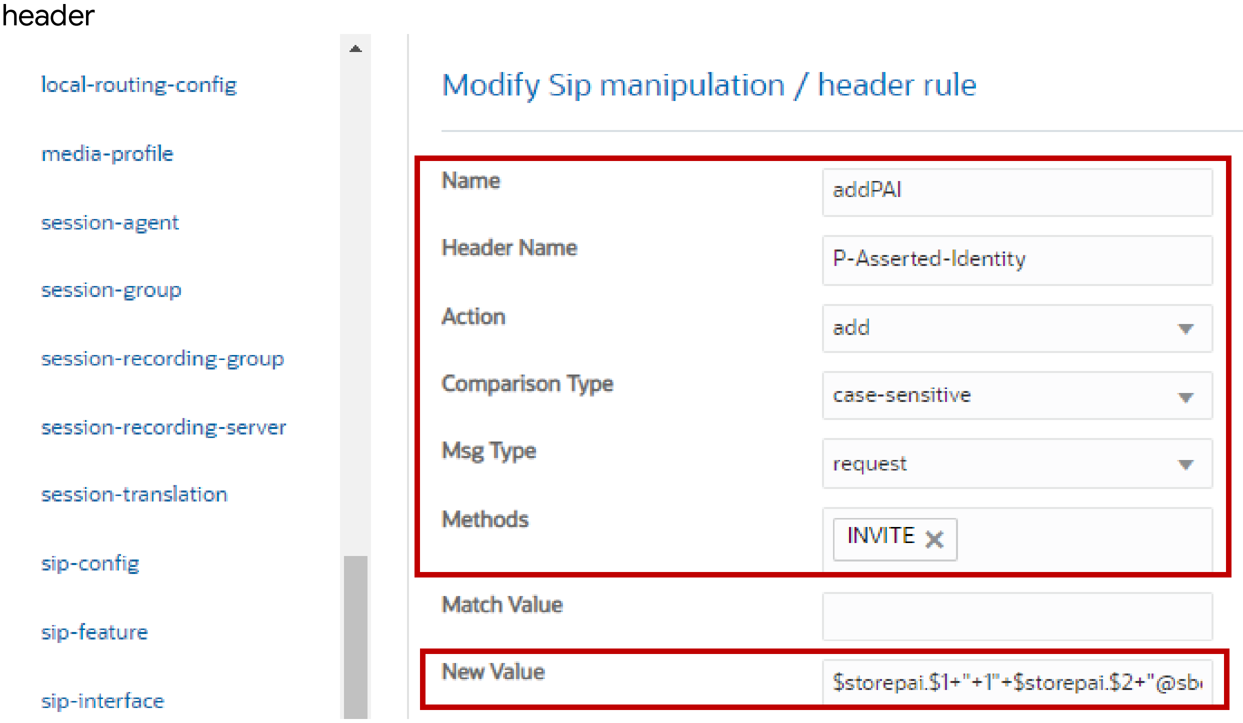


Figure 83: SIP Manipulation towards Google Voice to add P-Asserted-Identity with E.164 format

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

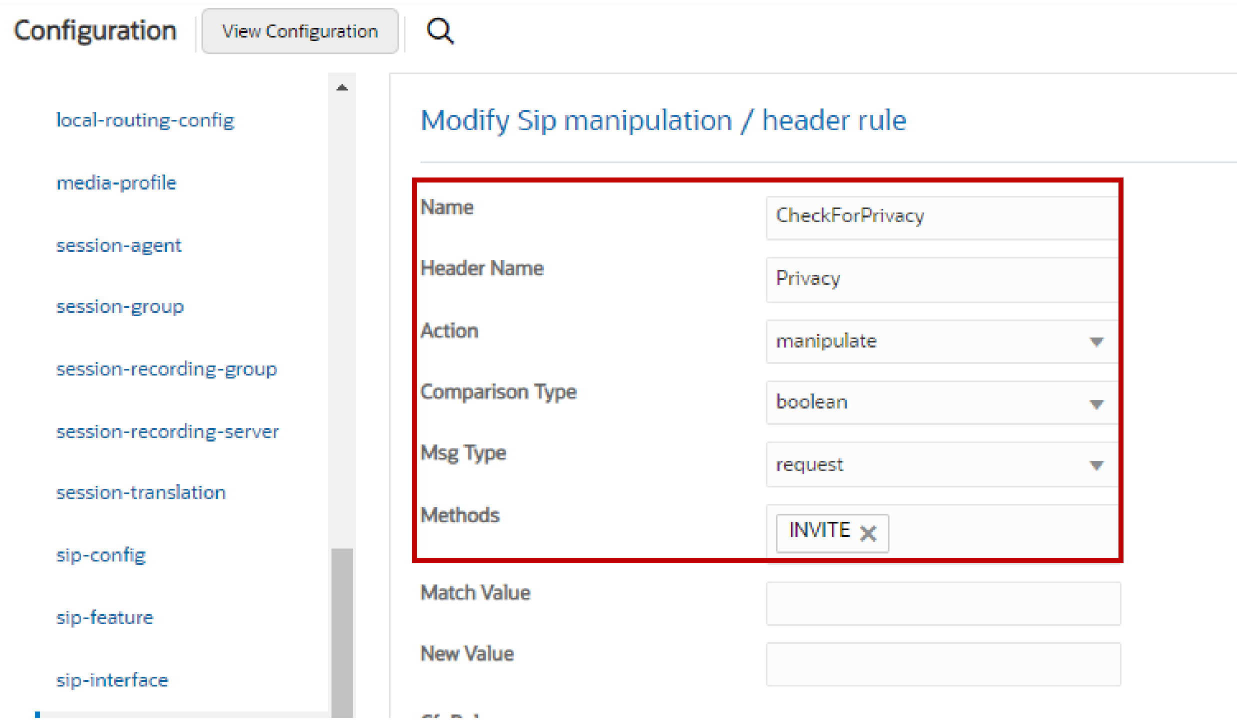


Figure 84: SIP Manipulation towards Google Voice to check for Privacy header

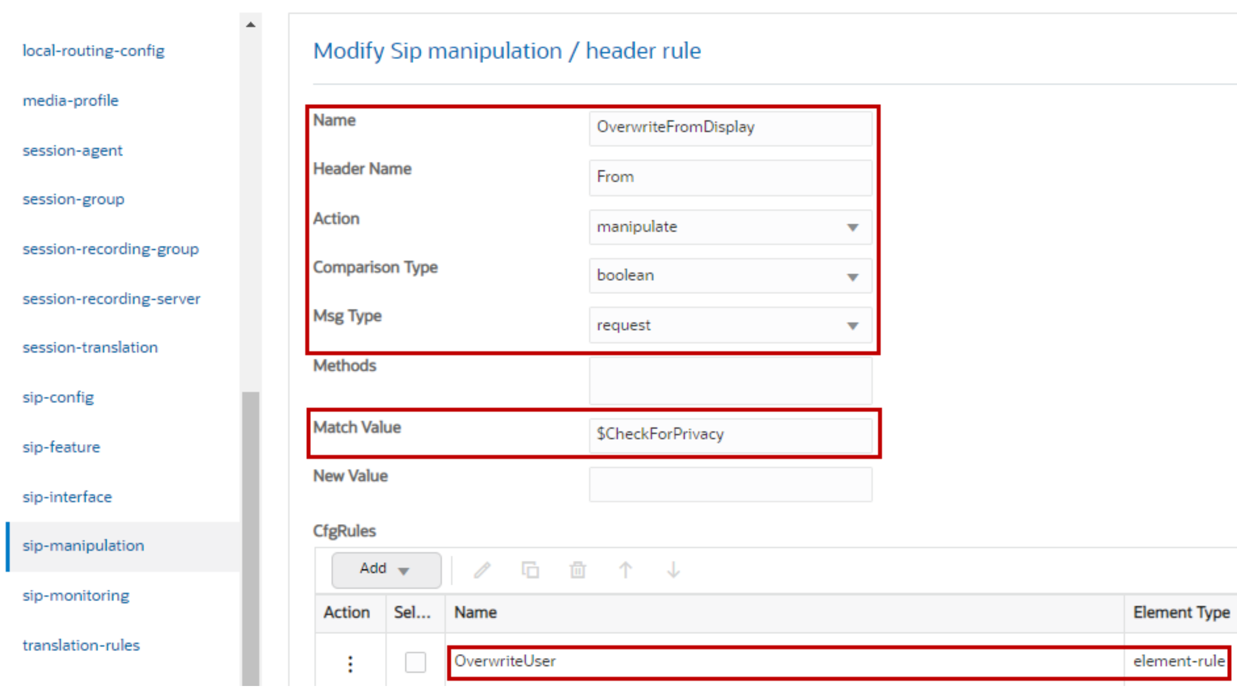


Figure 85: SIP Manipulation towards Google Voice to overwrite User part of FROM header to “anonymous”

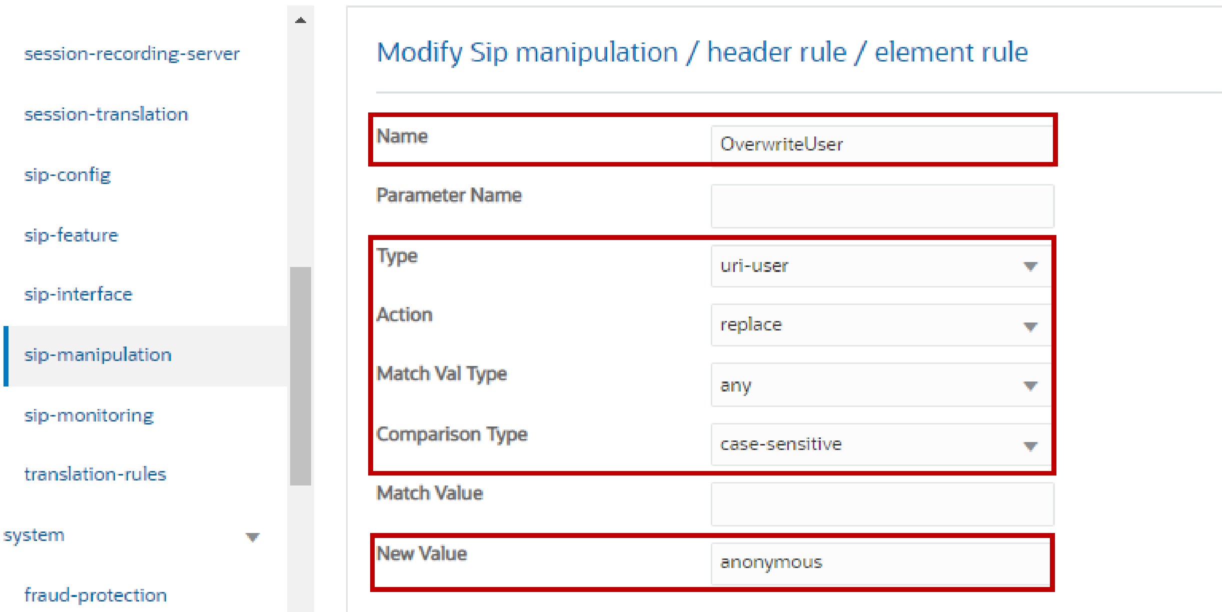


Figure 86: SIP Manipulation towards Google Voice to overwrite User part of FROM header to "anonymous" Cont.

SIP manipulation towards PSTN Gateway

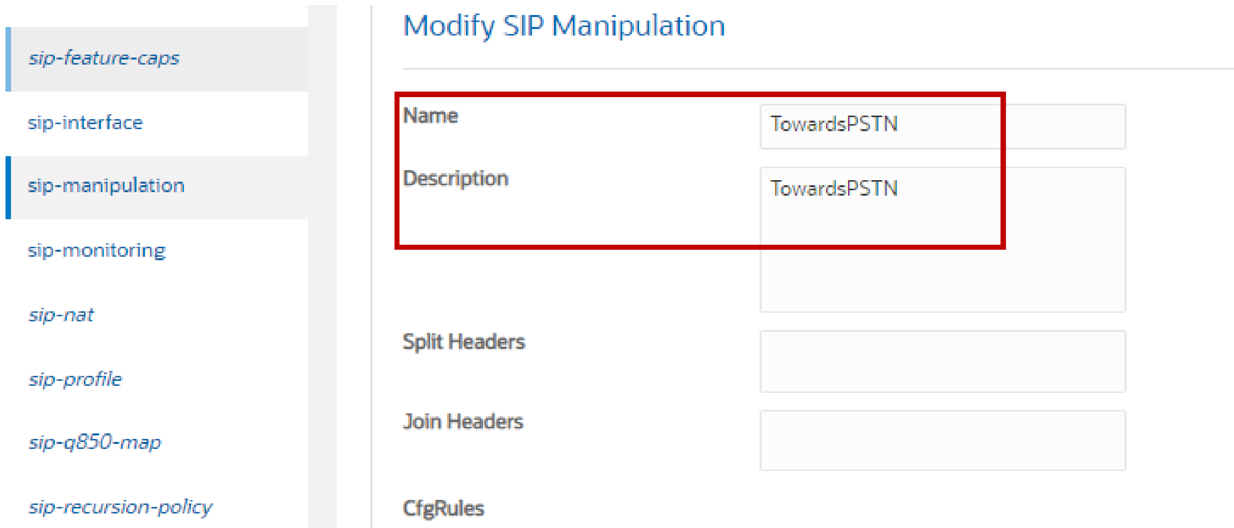


Figure 87: SIP Manipulation towards PSTN Gateway

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

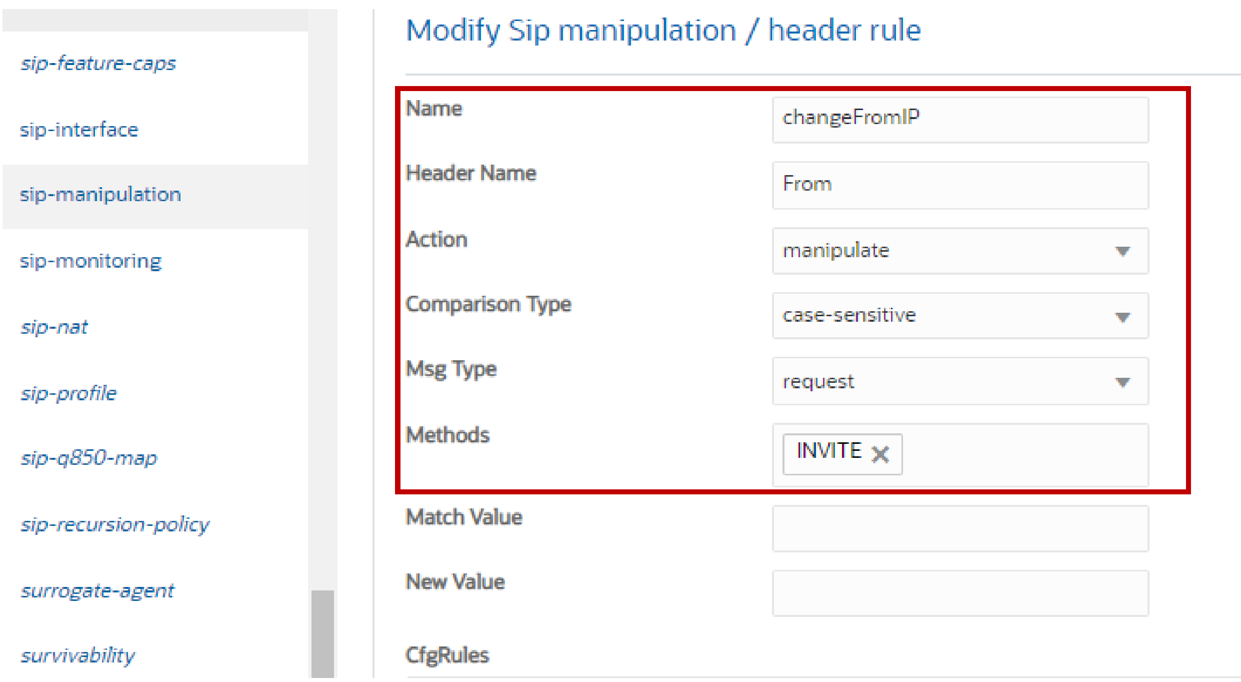


Figure 88: SIP Manipulation towards PSTN Gateway to change FROM header with SBC IP

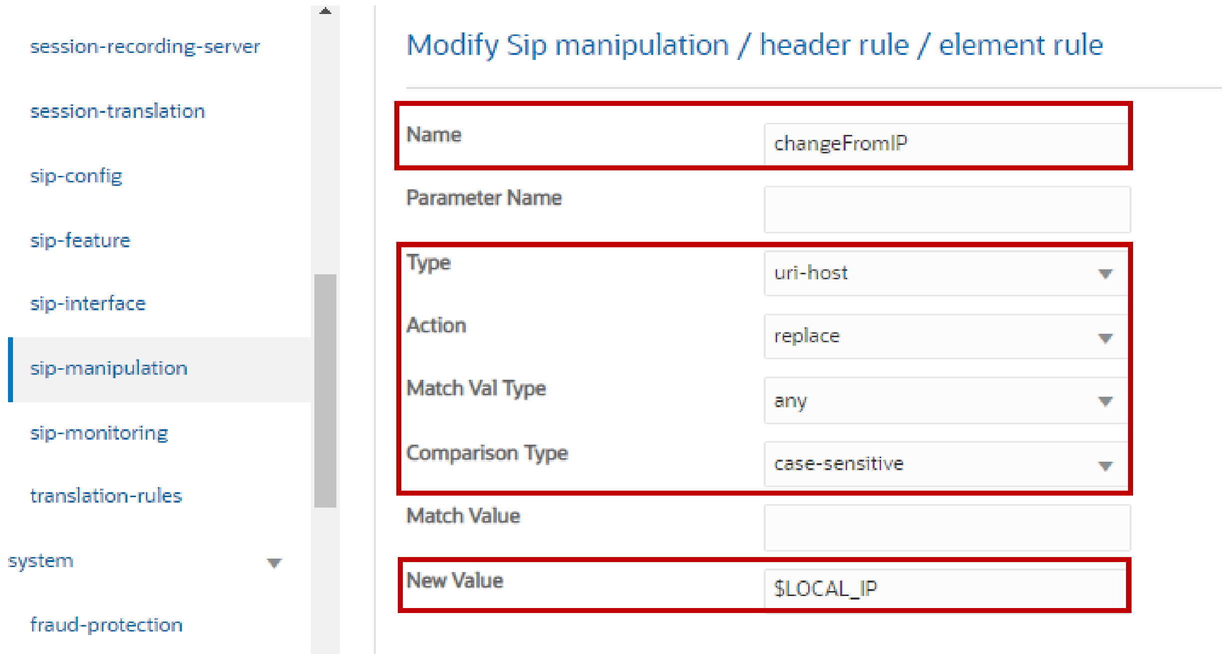


Figure 89: SIP Manipulation towards PSTN Gateway to change FROM header with SBC IP Cont.

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

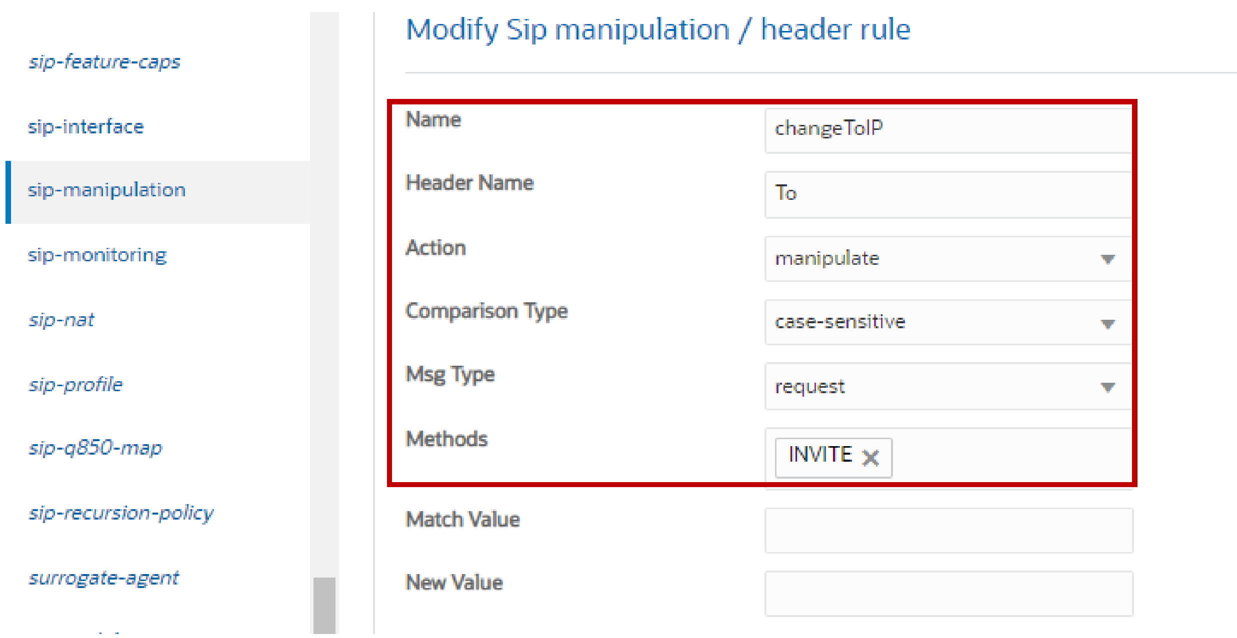


Figure 90: SIP Manipulation towards PSTN Gateway to change TO header

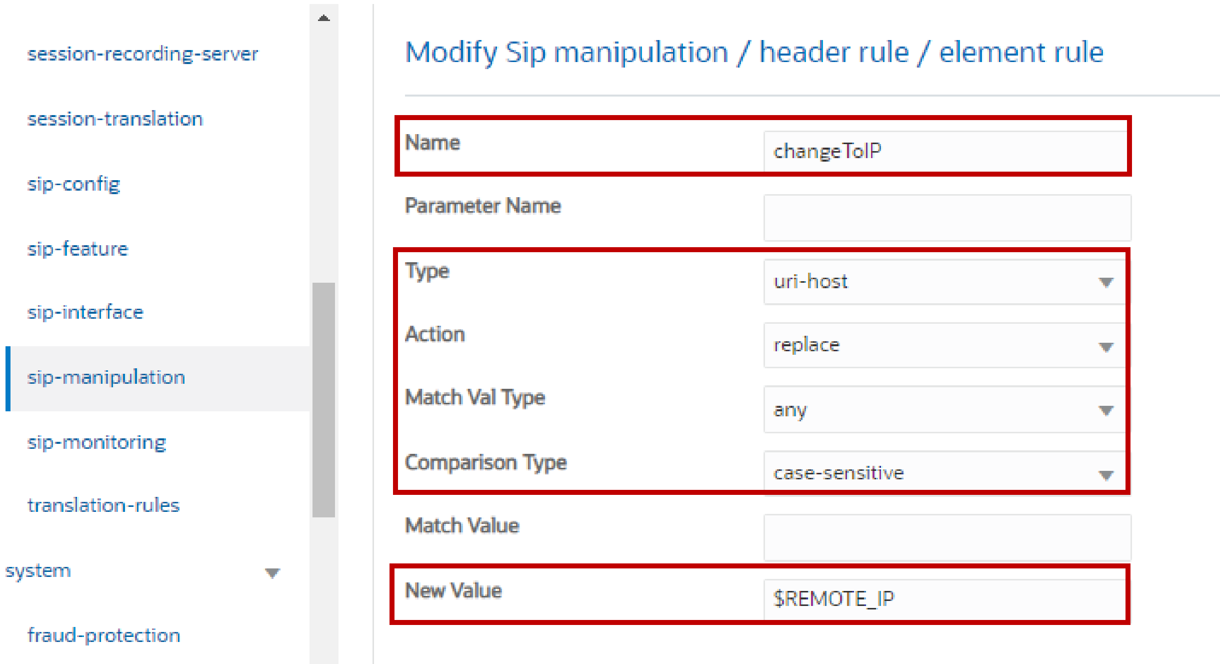


Figure 91: SIP Manipulation towards PSTN Gateway to change TO header Cont.

6.4.18 Redundancy Configuration

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

HA Topology Diagram

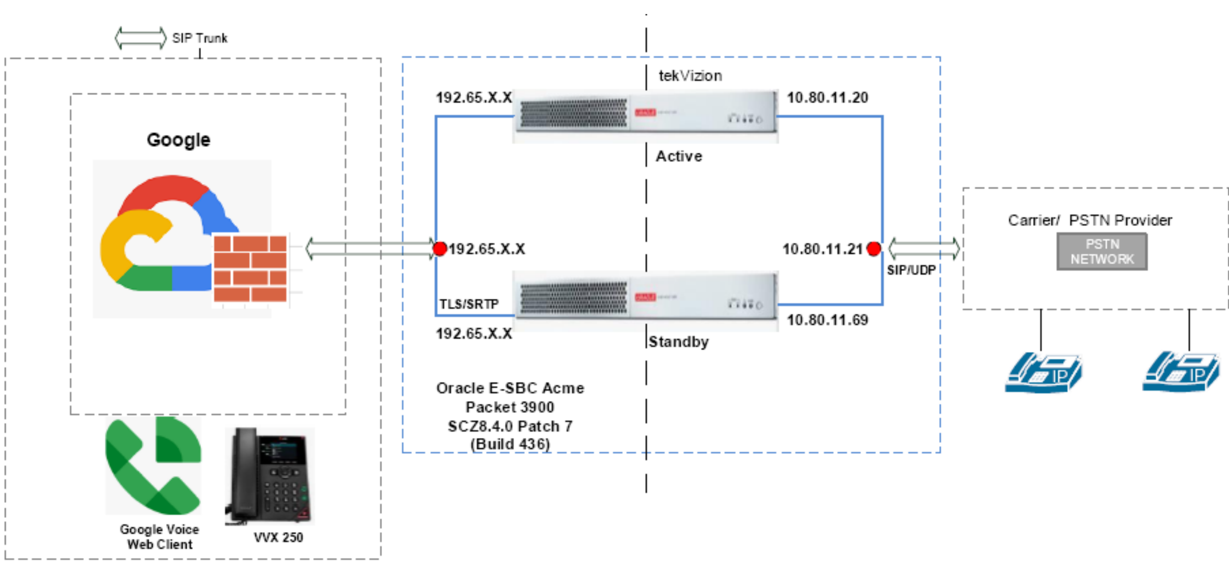


Figure 92: Topology Diagram – HA Pair

Create Physical Interface

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

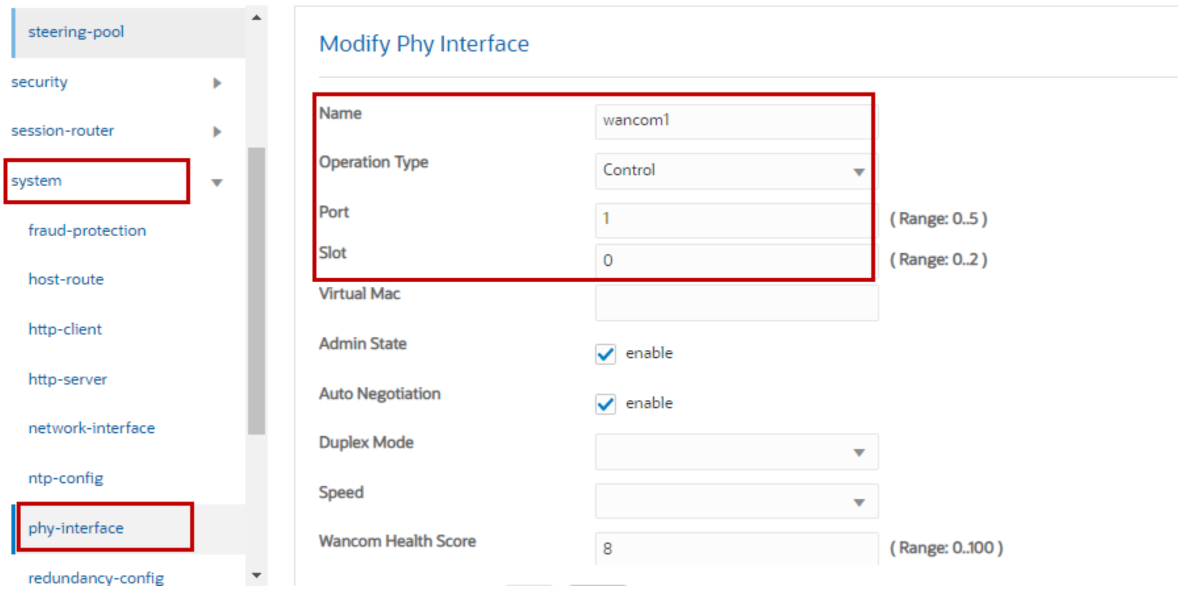


Figure 93: Create phy-interface – Wancom1

Modify Network Interface

Navigate to **System > network-interface**. Primary and Secondary utility IP addresses of Oracle SBC 1 and SBC 2 for Wancom1, S0P0 and S1P0 are configured as shown below.

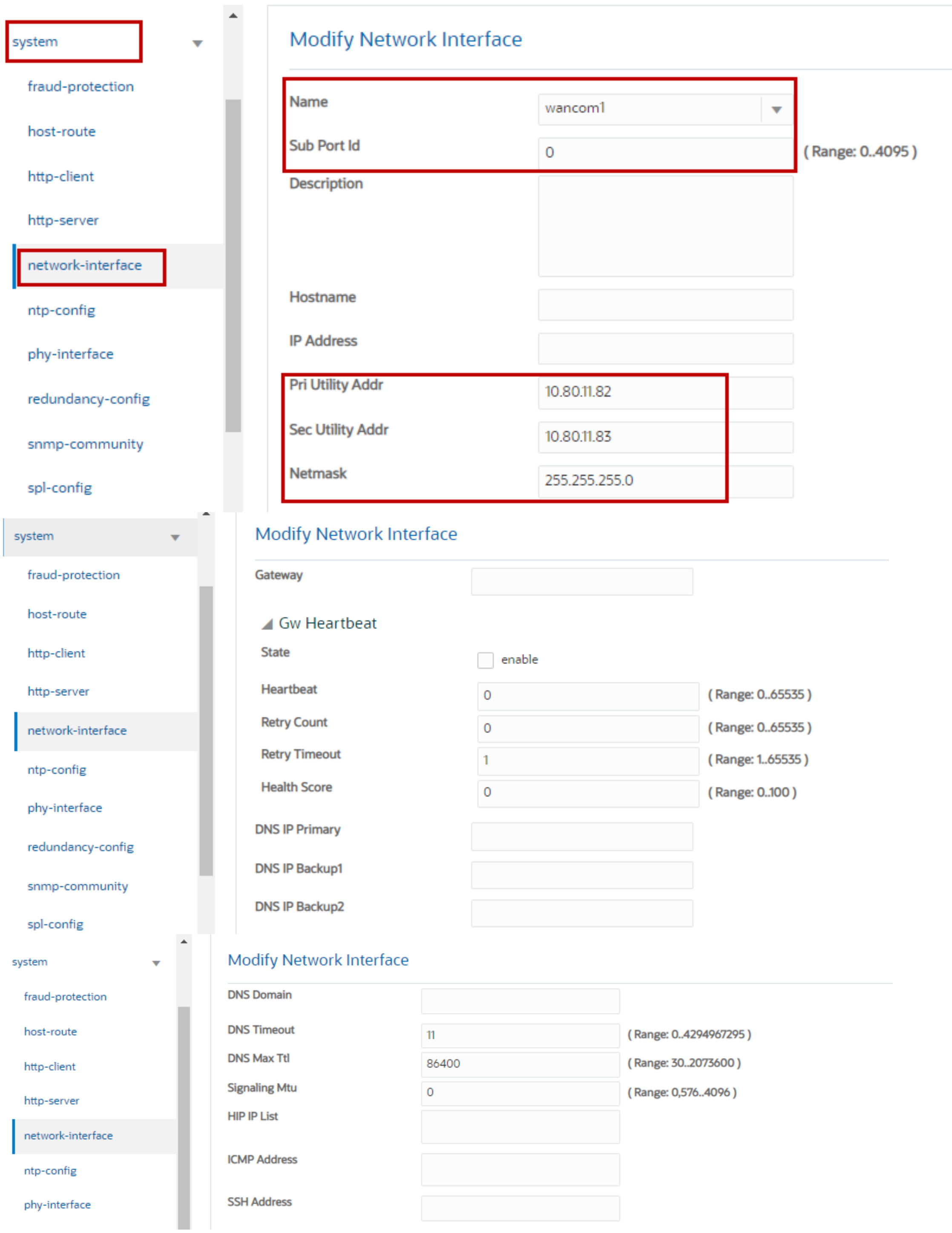


Figure 94: Modify Network-Interface for Wancom1

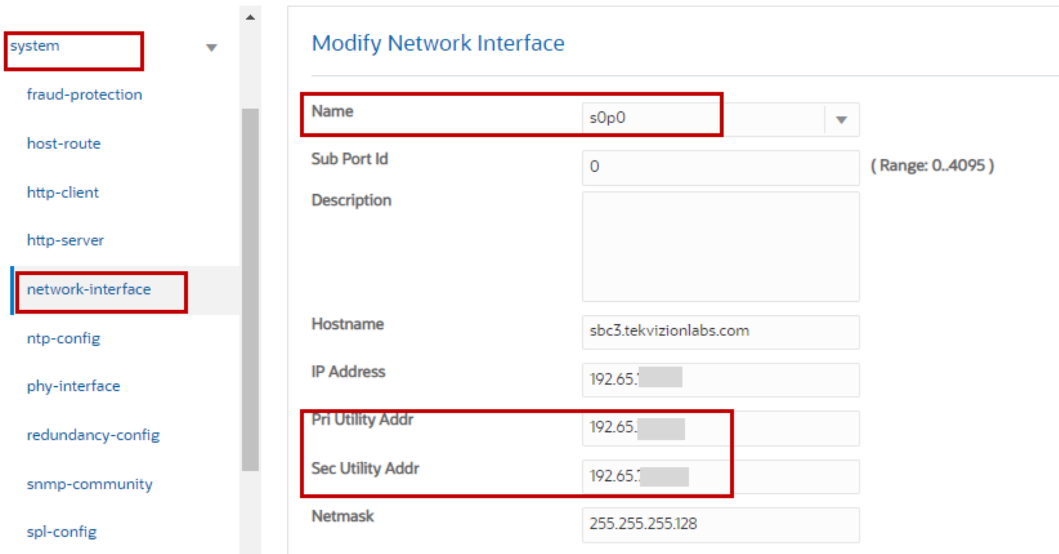


Figure 95: Modify Network-Interface of S0P0

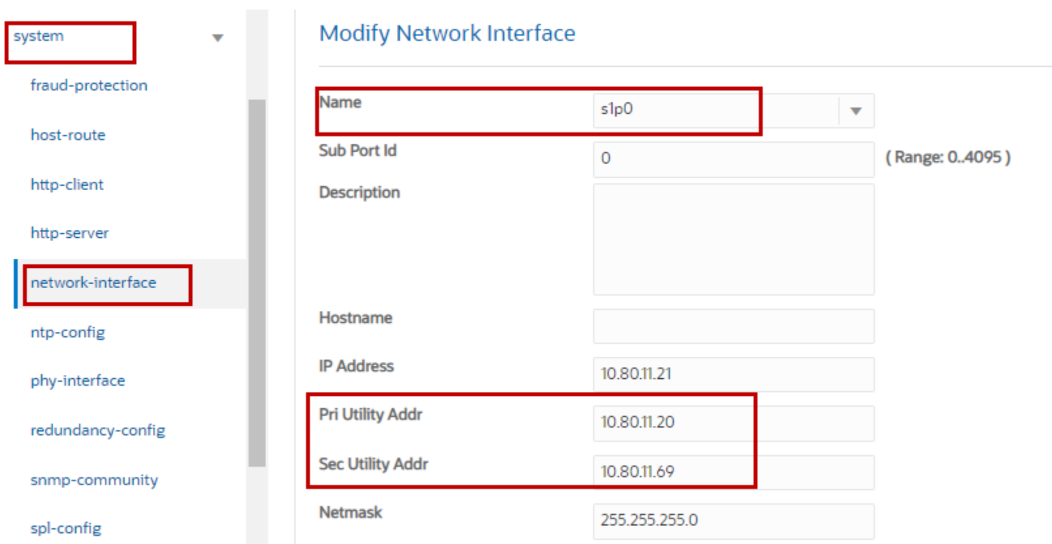


Figure 96: Modify Network-Interface of S1P0

Create Redundancy Config

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

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

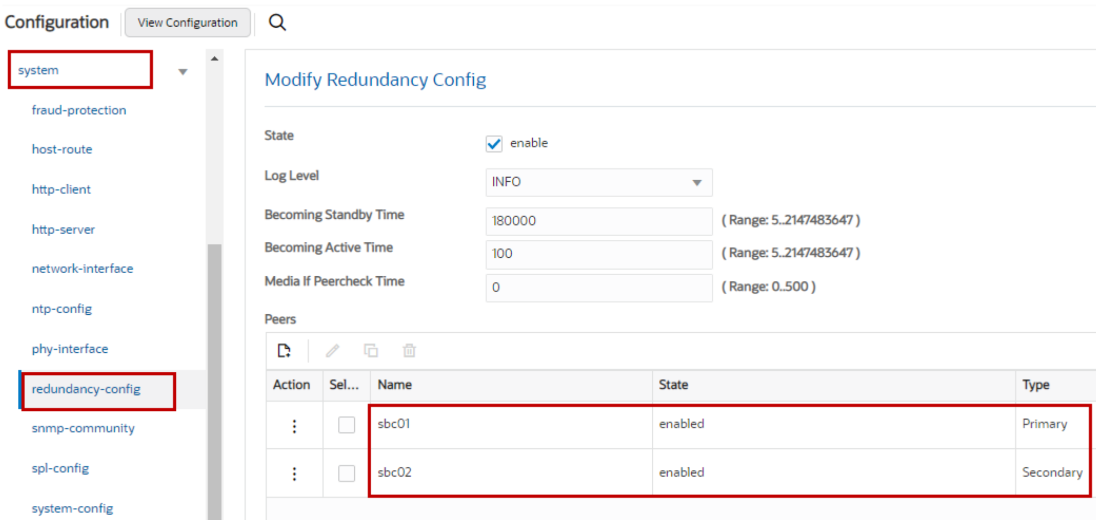


Figure 97: Create Redundancy configuration for SBC1 and SBC2

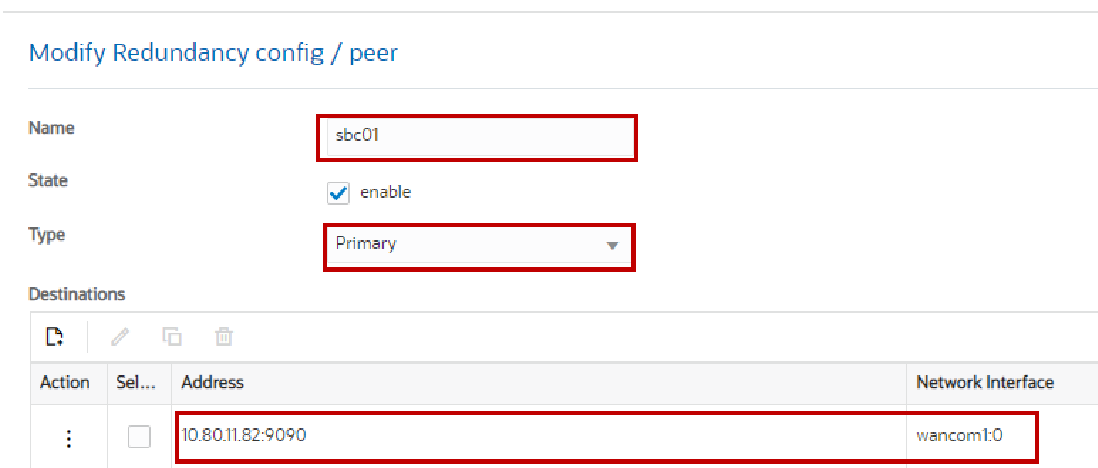


Figure 98: Configure SBC1 as Primary

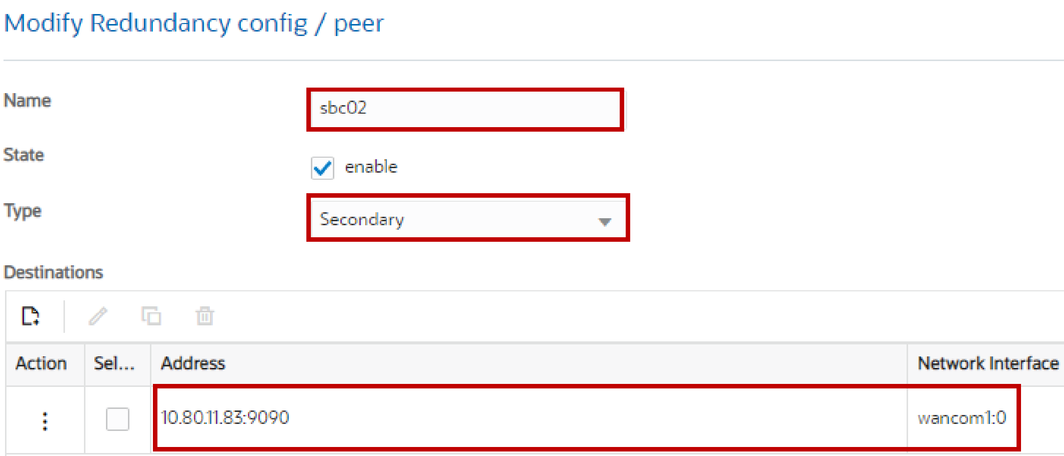


Figure 99: Configure SBC2 as Secondary

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

6.4.19 Oracle SBC deployed Behind NAT

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

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

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

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

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

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

Navigate to session-router > sip-interface.

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

HeaderNatPublicSipIfIp is the Public Interface IP.
HeaderNatPrivateSipIfIp is the Private IP.

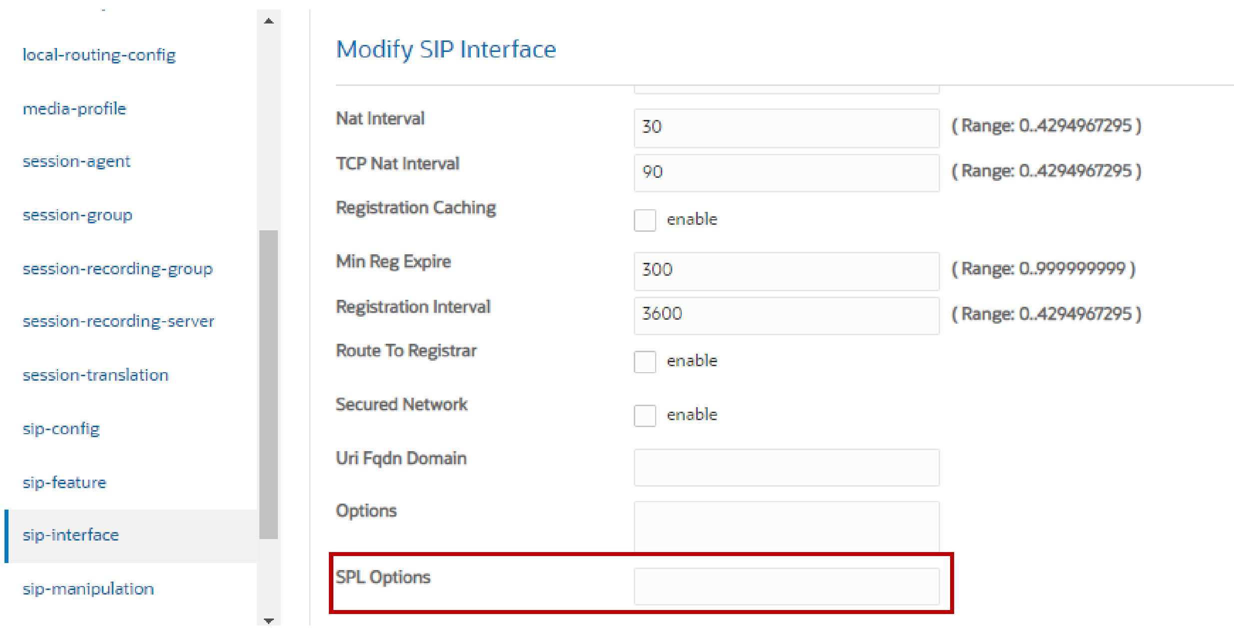


Figure 100: Oracle SBC Behind NAT – SPL Options configuration

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

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

7 Oracle E-SBC Running configuration

Attached the Oracle E-SBC running configuration.

Oracle_RunningCo
nfig.txt

8 Summary of Tests and Results

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

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
1.5	PBX Extension Number > PBX User > BYOT Phone Number > User Termination	Place a call from External network to a PBX phone number assigned to an PBX user. PBX user then transfers the call to BYOT phone number for User Termination. Ensure 2 way voice.	The GV client should ring and upon answering the call 2 way audio 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
			conversation 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 conversation should be possible. The PBX extension should be	PASSED	

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

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

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

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		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 Conference		
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 successfully leave a voicemail as well as able to exercise the various dtmf options in the voicemail menu	PASSED	
Outbound					
1.21	User Originated > BYOT Trunk Termination	GV endpoint user will place a call to External no. <> via BYOT trunk	The call should be successful with 2 way audio.	PASSED	
1.22	Short Code call > BYOT Trunk Termination	GV user to dial a short code to test outbound call via short codes via BYOT trunk	The call should be successful with 2 way audio.	NOT SUPPORTED	Short code dialing is not supported by Google Voice. Google updated that this feature is

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

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

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 or Called Party 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 established with bidirectional audio Verify call is disconnected properly	PASSED	
2.3	Inbound call: Called Party disconnects the call.	PSTN Calls GV user A GV user A answers the call. Allow the call to be connected more than 32 seconds Called party disconnects the call	Verify Call is established with bidirectional audio Verify call is disconnected properly	PASSED	
Inbound Call – Calling or Called Party 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 established with bidirectional audio Verify call is disconnected properly	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
2.5	Outbound call: Called Party disconnects the call.	GV user A Calls PSTN A PSTN A answers the call. Allow the call to be connected more than 32 seconds Called party disconnects the call	Verify Call is established with bidirectional audio Verify call is disconnected properly	PASSED	
Terminate the call before answer					
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 anonymous	PASSED	
Early 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 bidirectional audio for more than 60 mins Bidirectional audio is present after session audit or session refresh invite from the DUT. Call is properly disconnected when either party disconnects the call	PASSED	
2.11	Long duration call-Outbound Call- 1 hour max	GV user A Calls PSTN A PSTN A answers the call. Allow the call to be connected for 60 mins	Call is connected and bidirectional audio for more than 60 mins Bidirectional audio is	PASSED	

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

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

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
2.14	Terminate the call during hold	PSTN Calls GV user A. GV user A answers the call GV user A places the call on hold GV user A hangs up during hold	Call is properly disconnected when either party disconnects the call	PASSED	
Do Not 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	
Simultaneous Ring					
2.16	Simultaneous ring	PSTN A calls GV user A GV user sets simultaneous ring to PSTN B	PSTN B answers the call Bidirectional Audio is present	NOT SUPPORTED	Google Voice does not support Simultaneous ring.
Call Forward					
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 Free					
2.19	Calls to 800 numbers (Toll Free Numbers)	GV user A Calls TOLL FREE number starts with 800	Verify Call is established with bidirectional audio Verify call is disconnected properly	PASSED	
2.20	Calls to 877 numbers (Toll Free Numbers)	GV user A Calls TOLL FREE number starts with 877	Verify Call is established with bidirectional audio Verify call is disconnected properly	PASSED	
2.21	Calls to 866 numbers (Toll Free Numbers)	GV user A Calls TOLL FREE number starts with 866	Verify Call is established with bidirectional audio Verify call is disconnected properly	PASSED	
2.22	Calls to 888 numbers (Toll Free Numbers)	GV user A Calls TOLL FREE number starts with 888	Verify Call is established with bidirectional audio Verify call is disconnect	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
			ed properly		
2.23	Call to 5551212	GV user A Calls 5551212	Verify Call is established with bidirectional audio Verify call is disconnected properly	NOT SUPPORTED	Google Voice does not support short-code dialing. Google updated that this feature is under work in progress.
2.24	Calls to 0 operator assistance.	GV user A Calls 0 Operator answers the call	Verify Call is established with bidirectional audio Verify call is disconnected properly	NOT SUPPORTED	Google Voice does not support Operator code dialing. Google updated that this feature is under work in progress.
Handling E164 and Non 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 established with bidirectional audio Verify call is disconnected properly	NOT SUPPORTED	Google Voice does not support Non E164. Google updated that this feature is under work in progress.
International 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 established with bidirectional audio Verify call is disconnected 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 Waiting					
2.28	Call waiting	GV user A Calls PSTN A PSTN A answer the call. Allow the call to be connected PSTN B makes an incoming call to GV user A, GV user A receives call waiting indication GV user A answers PSTN B PSTN A Placed on hold and hears MOH PSTN B hangs up the call GV user A resumes PSTN A. GV user A hangs up the call.	Verify Call is established with bidirectional audio Verify Call waiting indication Verify MOH on PSTNA during hold Verify call is disconnected properly	NOT SUPPORTED	When a second call is made from PSTN B to GV User A, GV user does not see call waiting notification. PSTN B hears ringback. Google updated that this is a known behavior as per design. The behavior is same for iOS and Android phones.

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
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.
Handling 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	
Handling 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 established with bidirectional audio and G711 U law is negotiated Verify call is disconnect	PASSED	

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

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

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		to PBX Auto Attendant > PBX Auto Attendant > BYOT Ring Group > BYOT User Termination 2	with BYOT and any customer PBX are not impacted		
2.42	Codec	BYOT Phone Number> Auto Attendant (mulaw) > BYOT User Termination Web Client (opus) > Transfer to BYOT Deskphone (G.722) > Outbound call to PBX (mulaw)	Media should connect and call must progress with 2 way audio	PASSED	
2.43	Third party IVR	3rd Party IVR calling BYOT Phone Number >User Termination > Press DTMF	Should be able to hear the external IVR and DTMF navigation should be successful. Ensure Telephony -events being passed for DTMF interaction . Test DTMF from a called party 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	HA	Unplug the LAN side cable of Primary SBC	Unplug the LAN side cable of Primary SBC, verify both incoming and outgoing calls work through secondary SBC	PASSED	
2.46	HA	Plug the LAN cable back for Primary SBC, verify the incoming/outgoing call going through the Secondary SBC. (Note: Can also be executed by shutting/unshutting the interfaces)	Plug the LAN cable back for Primary SBC, verify the incoming/outgoing call going through the Secondary SBC	PASSED	
2.47	HA	Unplug the WAN side cable of Primary SBC, verify both incoming and outgoing calls work through secondary SBC(Note: Can also be executed by shutting/unshutting the interfaces)	Unplug the WAN side cable of Primary SBC, verify both incoming and outgoing calls work through secondary SBC	PASSED	
2.48	HA	Plug the WAN cable back for Primary SBC, verify the incoming/outgoing	Plug the WAN cable back for Primary	PASSED	

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

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