Configuration Guide for Google CES Call Recording Using Oracle E-SBC Acme Packet 4600 SCZ 9.3.0 GA (Build 46)

Table of Contents

1 Audience 4
1.1 Introduction 4
1.1.1 TekVizion Labs 4
2 SIP Trunking Network Components 5
3 Hardware Components 6
4 Software Requirements 6
5 Google CES Certified Oracle E-SBC Versions 6
6 Features 6
6.1 Features tested for Google CES Call Recording 6
6.2 Features Not tested for Google CES Call Recording 6
6.3 Caveats and Limitations 6
6.4 Failed Testcase 6
7 Configuration 7
7.1 Configuration Checklist 7
7.2 IP Address Worksheet 8
7.3 Google CES API Configuration 9
7.4 Oracle E-SBC Configuration 10
7.4.1 Media Manager 10
7.4.2 Physical Interface 12
7.4.3 Network Interface 13
7.4.4 SIP Config 17
7.4.5 System-Config 19
7.4.6 SIP Monitoring 21
7.4.7 HTTP Server 22
7.4.8 Codec Policy 22
7.4.9 Translation Rules 23
7.4.10 Session Translation 24
7.4.11 Session Recording Server 25
7.4.12 Realm Config 25
7.4.13 Steering Pool 31
7.4.14 SDES Profile 32
7.4.15 Media Sec Policy 33
7.4.16 TLS – Certificate Record 34
7.4.17 TLS – TLS Profile 37

7.4.18 Session Timer 38
7.4.19 SIP Interface 38
7.4.20 Session Agent 43
7.4.21 Local Policy 50
7.4.22 SIP Manipulation 52
8 SIP INVITE To Google CES 62
8.1 SIP INVITE for SIPREC call 62
8.2 SIP INVITE for GTP call 62
9 Oracle E-SBC Running configuration 63
10 Summary of Tests and Results 64

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 CES Call Recording** using **Oracle Enterprise Session Border Controller Acme Packet 4600 SCZ9.3.0 GA (Build 46)**

1.1.1 TekVizion Labs

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

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

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

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

For more information on TekVizion and its practice areas, please visit TekVizion Labs website.

2 SIP Trunking Network Components

The network for the SIP Trunk reference configuration is illustrated below and is representative of Google CES Call Recording with Oracle Enterprise Session Border Controller (E-SBC) Acme Packet 4600 SCZ9.3.0 GA(Build46) configuration.

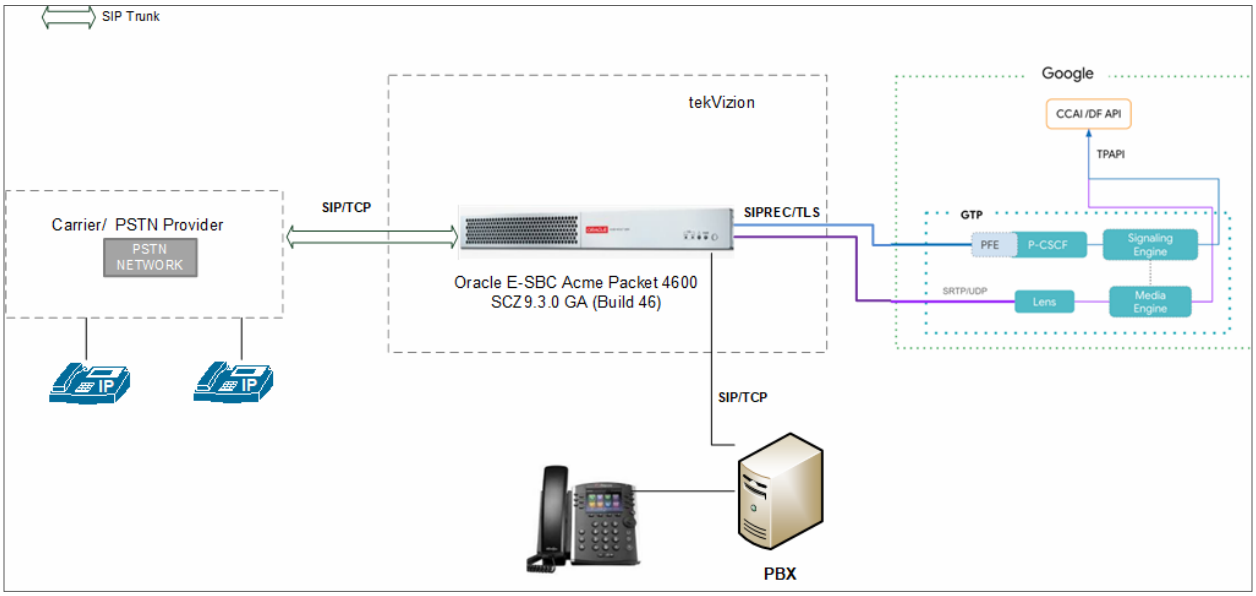

Figure 1: SIP Trunk Lab Reference Network.

The lab network consists of the following components.

- Google CES cloud Environment
- Oracle E-SBC Acme Packet 4600
- OnPrem PBX

3 Hardware Components

- Oracle E-SBC Acme Packet 4600

4 Software Requirements

- Oracle E-SBC Acme Packet 4600 SCZ 9.3.0 GA (Build46)

5 Google CES Certified Oracle E-SBC Versions

Table 1 – Google CES Certified Oracle E-SBC Version

Google CES Certified Oracle E-SBC Version	
Oracle E-SBC 4600	SCZ 9.3.0 GA (Build46)
Oracle E-SBC 3900	SCZ 9.3.0 GA (Build46)
Oracle E-SBC 3900	SCZ 8.4.0 Patch 2 (Build 151)

6 Features

6.1 Features tested for Google CES Call Recording

- Basic Inbound calls
- Call Hold and Resume
- Call Transfer
- Conference

6.2 Features Not tested for Google CES Call Recording

- None

6.3 Caveats and Limitations

DTLS	DTLS towards Google CES is not tested

6.4 Failed Testcase

- None

7

Configuration

7.1 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 7.4.1
Step 2	Physical Interface	Section 7.4.2
Step 3	Network Interface	Section 7.4.3
Step 4	SIP Config	Section 7.4.4
Step 5	System-Config	Section 7.4.5
Step 6	SIP Monitoring	Section 7.4.6
Step 7	HTTP Server	Section 7.4.7
Step 8	Codec Policy	Section 7.4.8
Step 9	Translation Rules	Section 7.4.9
Step 10	Session Translation	Section 7.4.10
Step 11	Session Recording Server	Section 7.4.11
Step 12	Realm Config	Section 7.4.12
Step 13	Steering Pool	Section 7.4.13
Step 14	SDES Profile	Section 7.4.14
Step 15	Media Sec Policy	Section 7.4.15
Step 16	TLS – Certificate Record	Section 7.4.16
Step 17	TLS – TLS Profile	Section 7.4.17
Step 18	Session Timer	Section 7.4.18
Step 19	SIP Interface	Section 7.4.19
Step 20	Session Agent	Section 7.4.20
Step 21	Local Policy	Section 7.4.21
Step 22	SIP Manipulation	Section 7.4.22

7.2 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 are for **illustrative purposes only**.

Table 3 - IP Address Worksheet

Component	IP Address
Google CES	
Signaling	us.telephony.goog:5672
Media	74.125.X.X
OnPrem PBX	
LAN IP Address	172.16.X.X
Oracle E-SBC	
LAN IP Address	10.80.X.X
WAN IP Address	192.65.X.X

7.3

Google CES API Configuration

Below link can be referred to configure Google CES API configuration for Call recording.

---------Link provided by Google team----------

https://cloud.google.com/contact-center/insights/docs/troubleshooting

7.4 Oracle E-SBC Configuration

The following is the example configuration of Oracle E-SBC for Google CES Call Recording.

7.4.1 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**

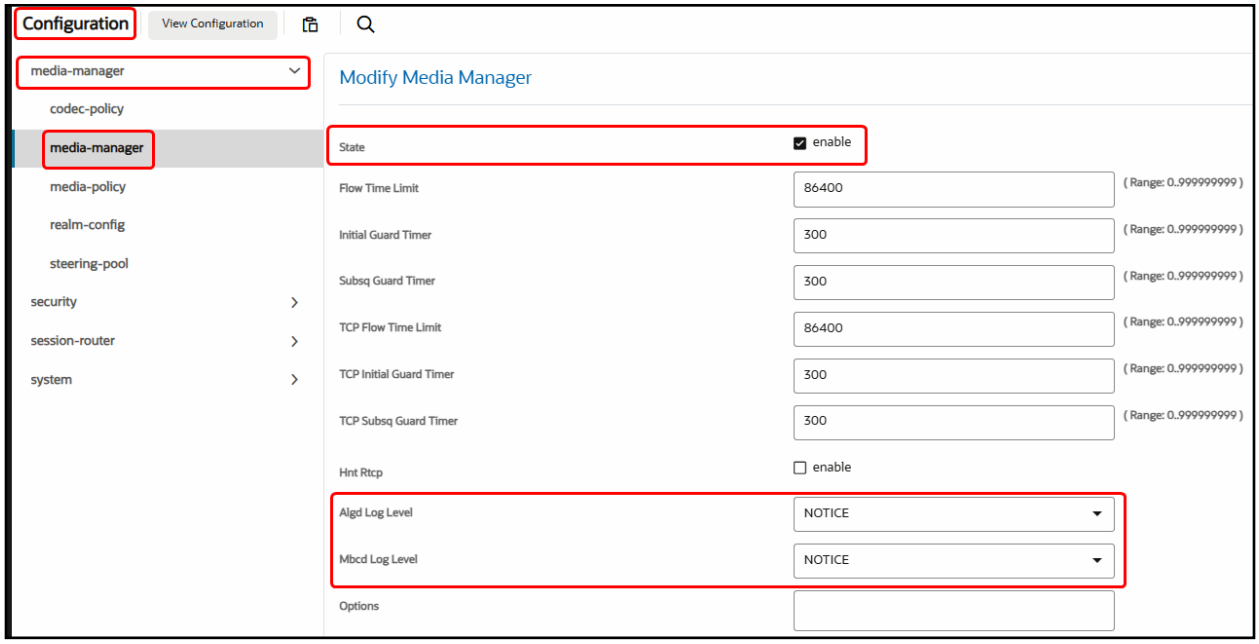

Figure 2: Media Manager Configuration.

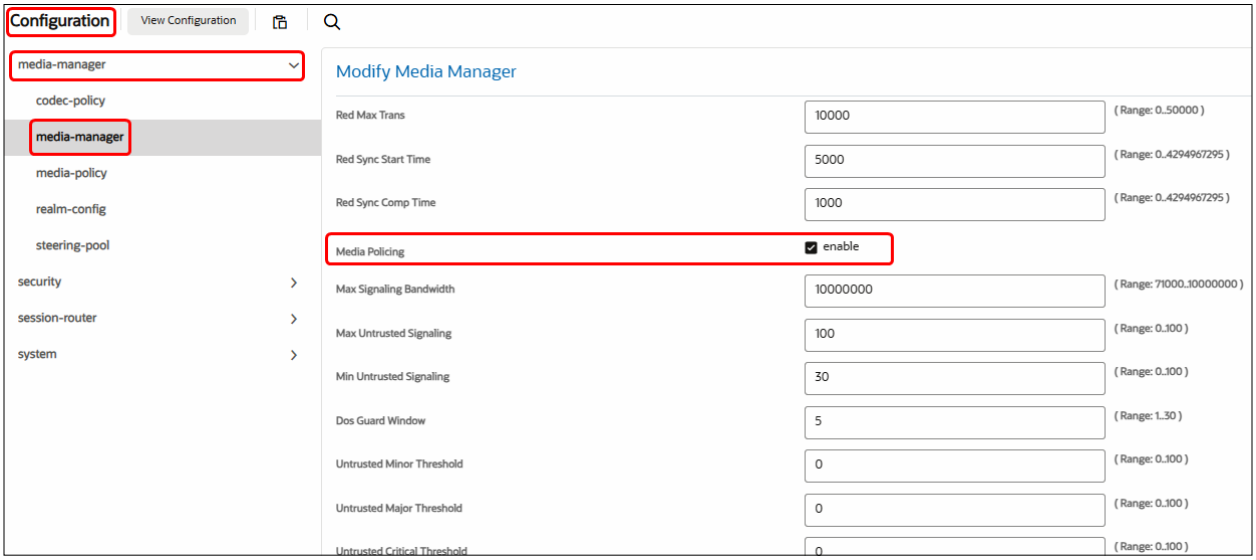

Figure 2.1: Media Manager Configuration.

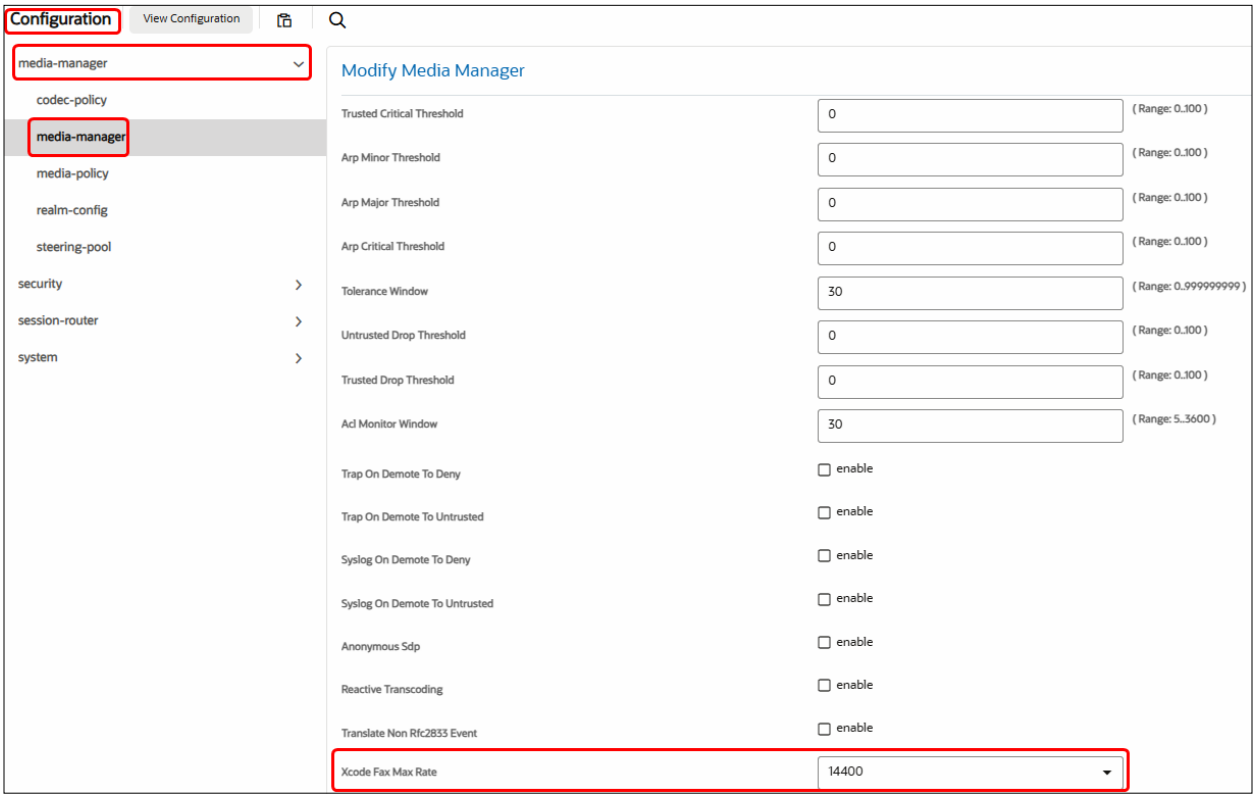

Figure 2.2: Media Manager Configuration (Cont.)

7.4.2 Physical Interface

- Navigate to **Configuration → system → phy-interface**.
- Configure Physical interface towards Google CES, OnPrem PBX and PSTN Gateway as shown below.

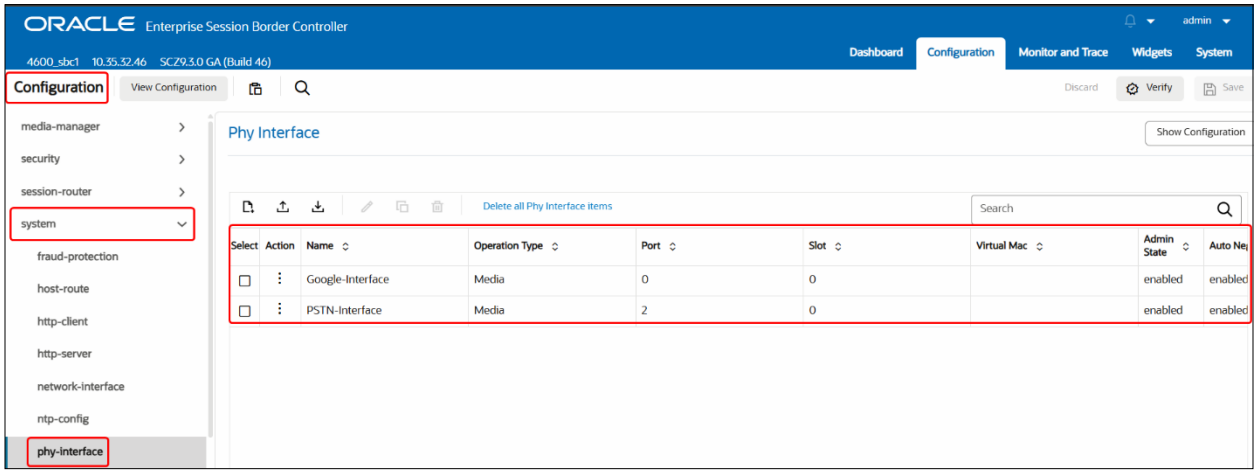

Figure 3: Physical Interfaces.

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

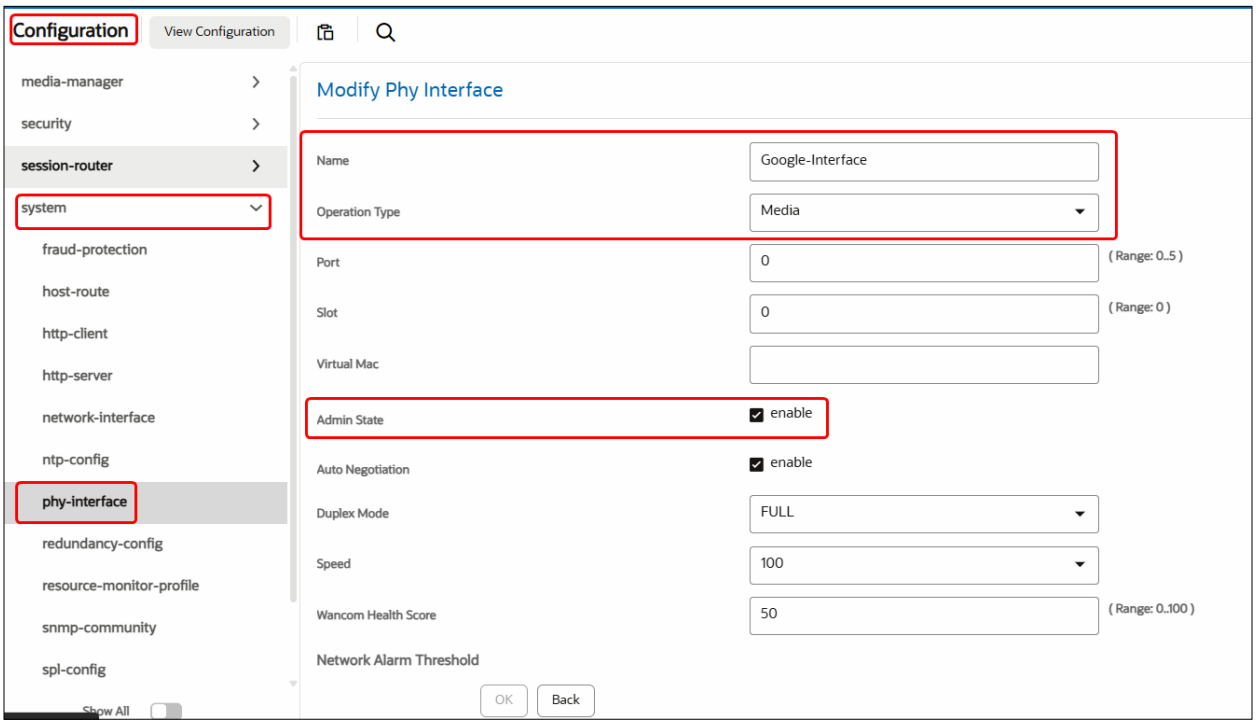

Figure 3.1: Physical Interface towards Google CES (Cont.)

- The interface designated towards PSTN Gateway and OnPrem PBX are named as s0p2 (Slot 0, port 2).

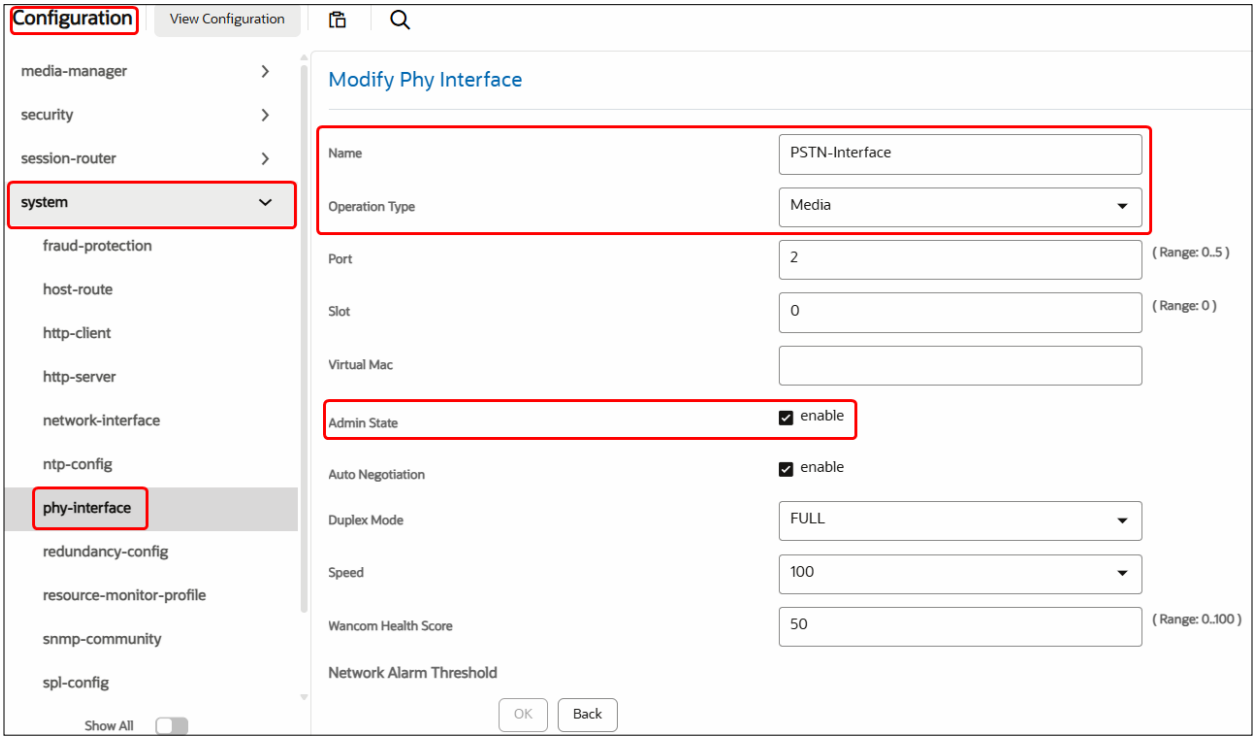

Figure 3.2: Physical Interface towards PSTN Gateway and OnPrem PBX.

7.4.3 Network Interface

- Navigate to **Configuration → system → network-interface**.
- Configure network interface towards Google CES, OnPrem PBX and PSTN Gateway as shown below

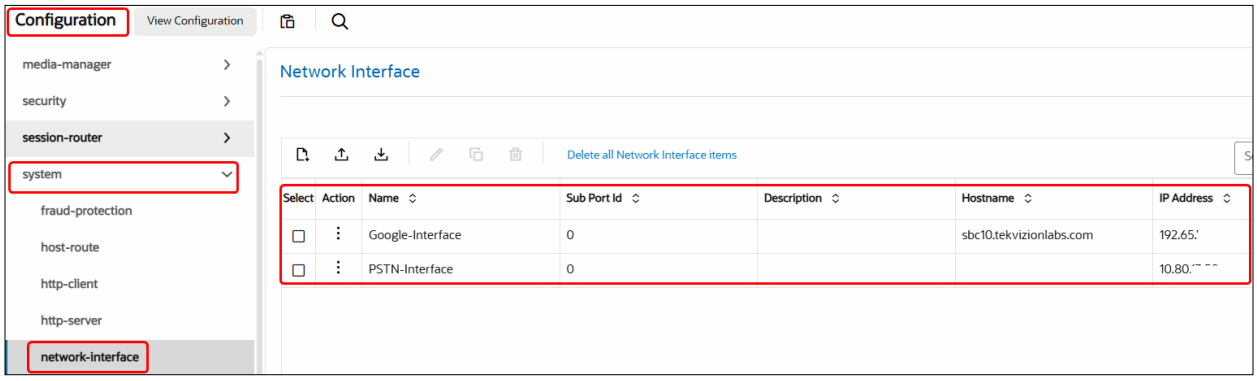

Figure 4: Network Interfaces.

- Configure Network interface towards Google CES as shown below.

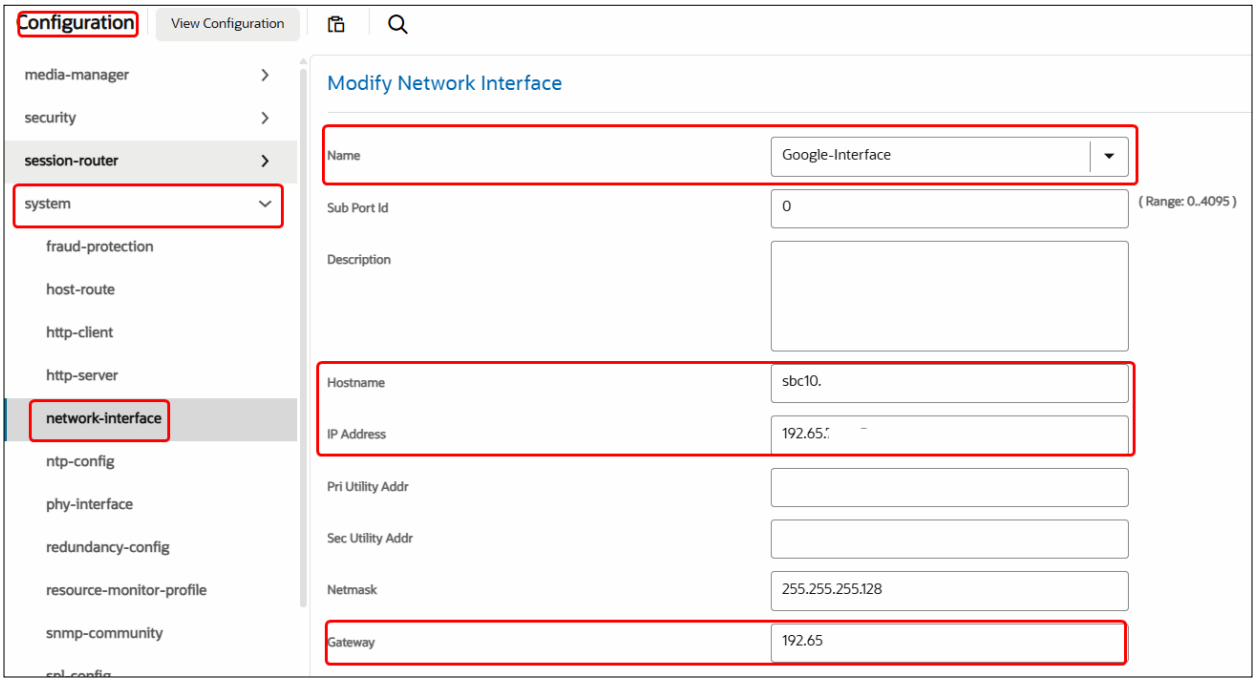

Figure 4.1: Network Interface towards Google CES.

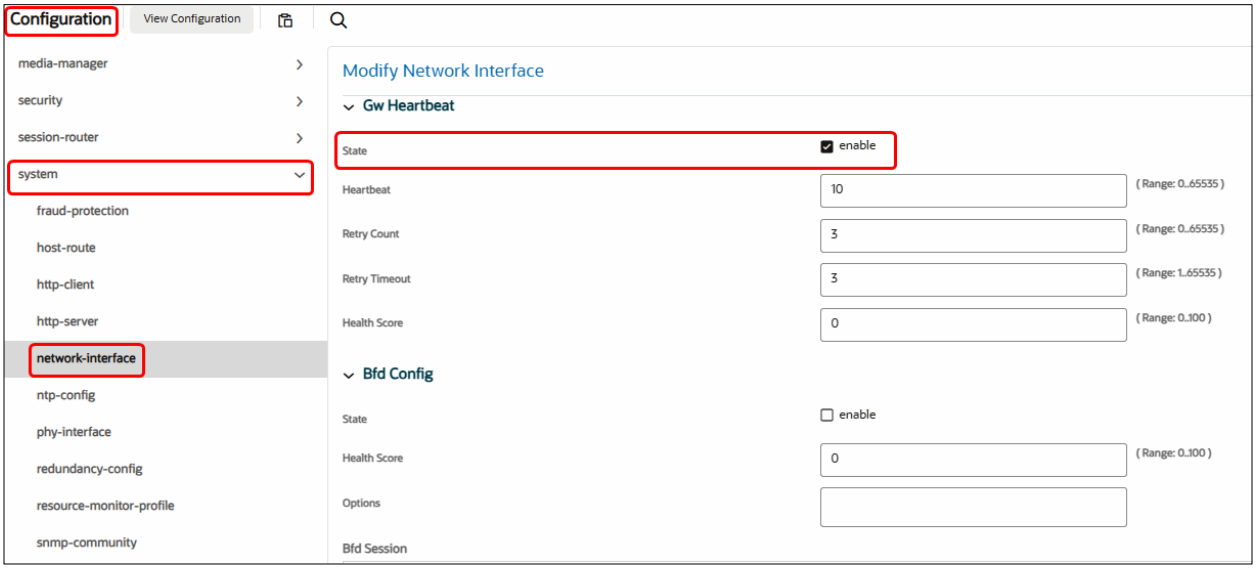

Figure 4.2: Network Interface towards Google CES (Cont.)

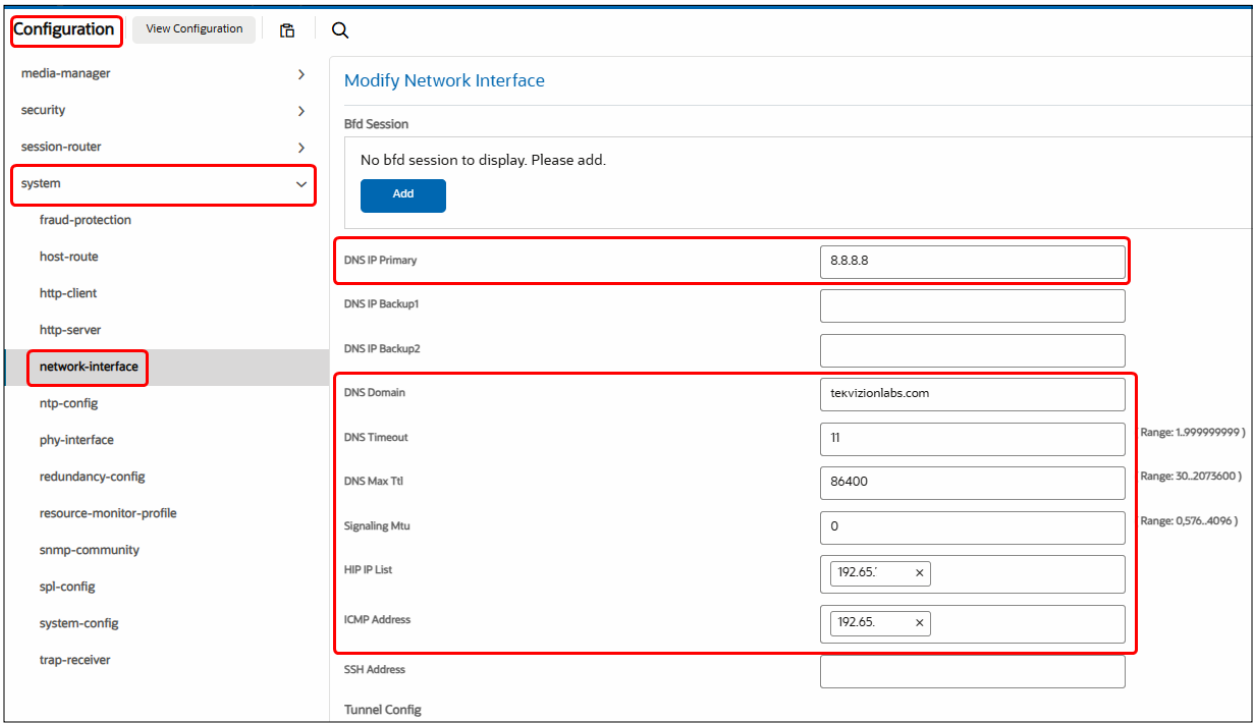

Figure 4.3: Network Interface towards Google CES (Cont.)

- Configure the Network interface towards OnPrem PBX and PSTN Gateway as shown below.

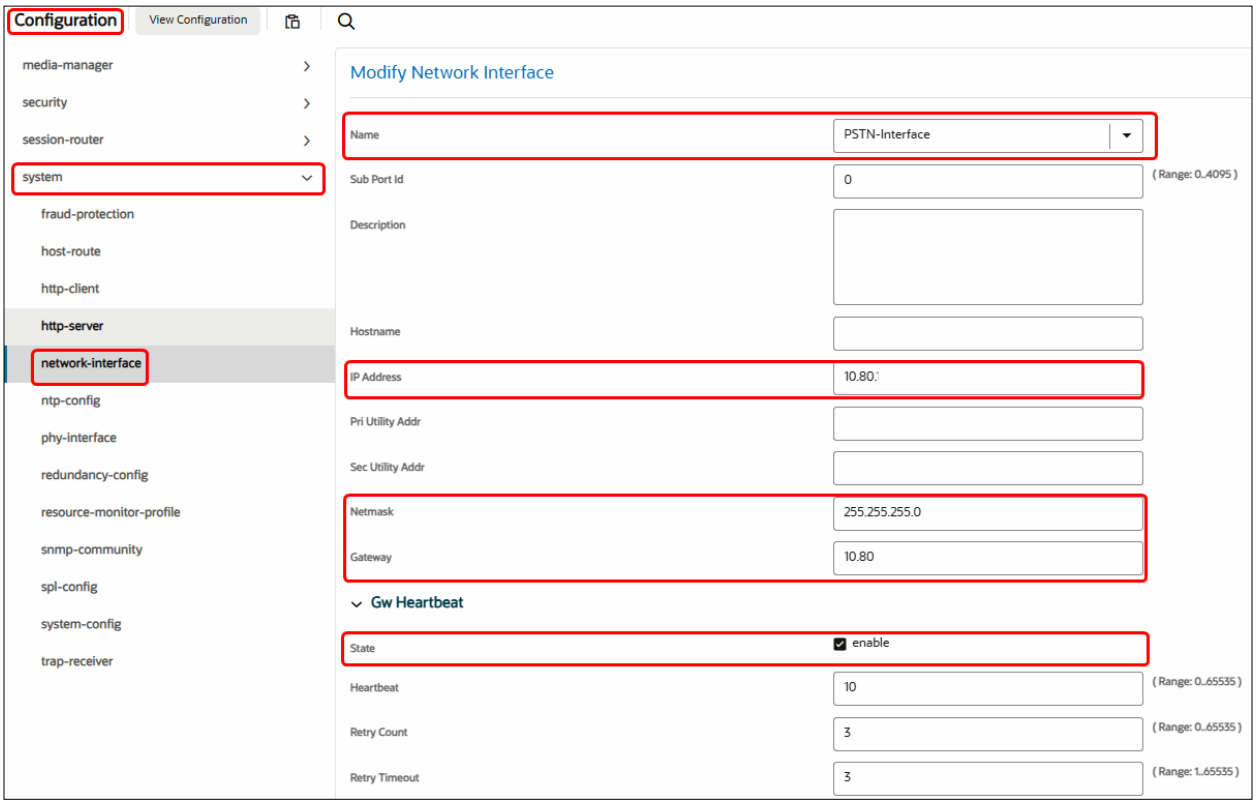

Figure 4.4: Network Interface towards OnPrem PBX and PSTN Gateway.

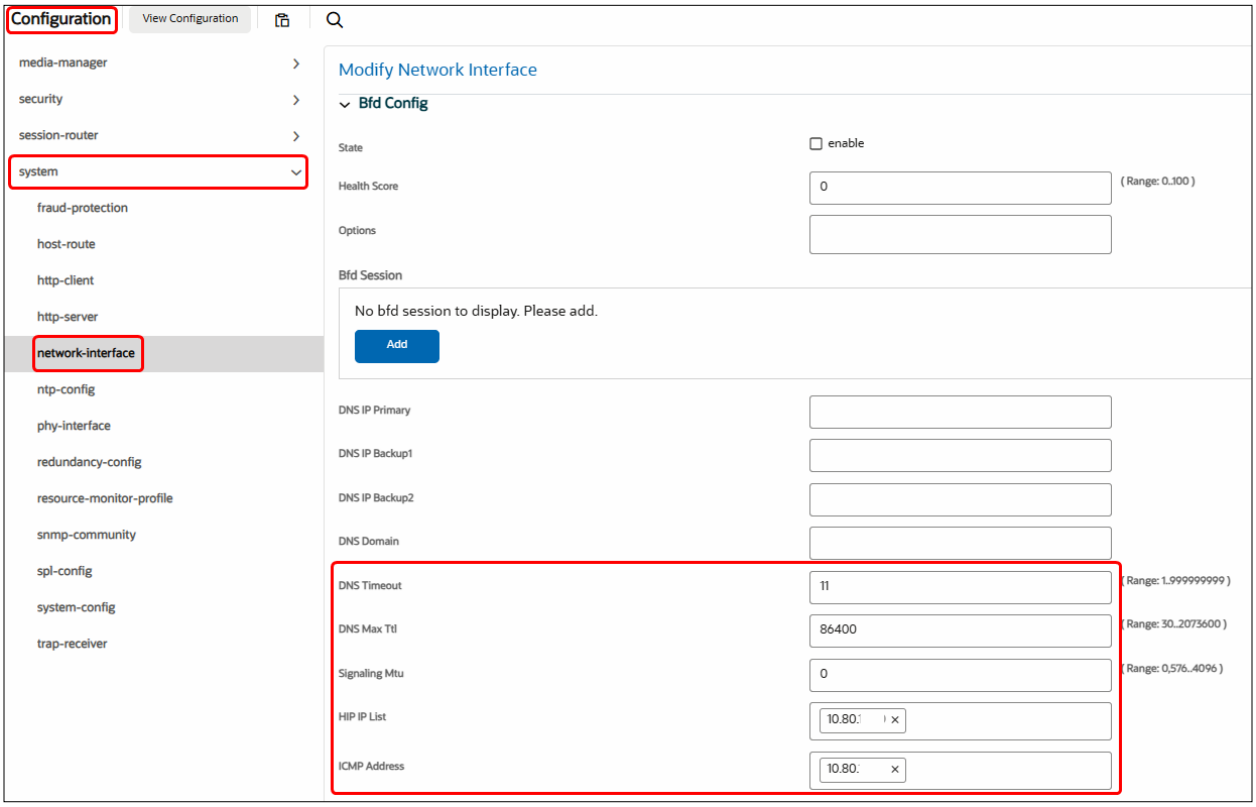

Figure 4.5: Network Interface towards OnPrem PBX and PSTN Gateway (Cont.)

7.4.4 SIP Config

Navigate to **Configuration → session-router → sip-config** for SIP configuration as shown below.

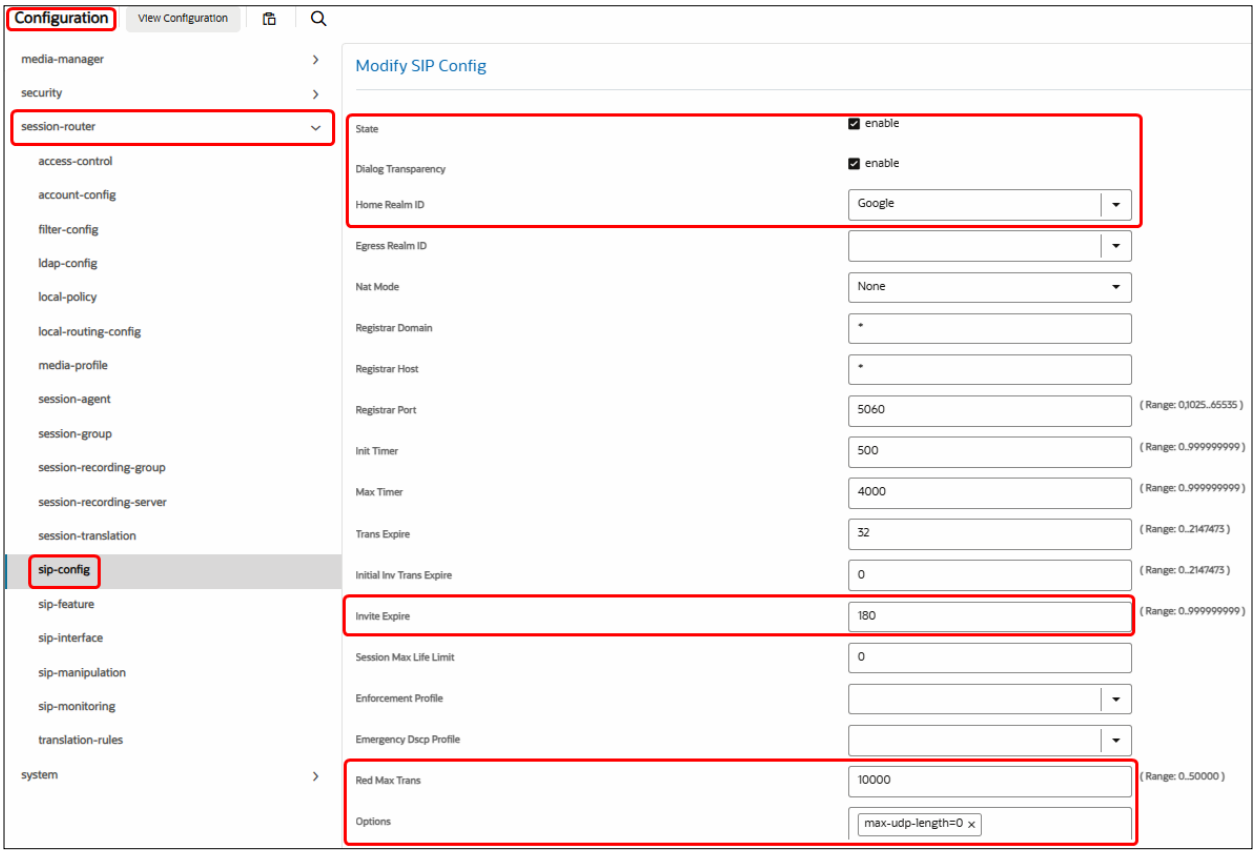

Figure 5: SIP-Config.

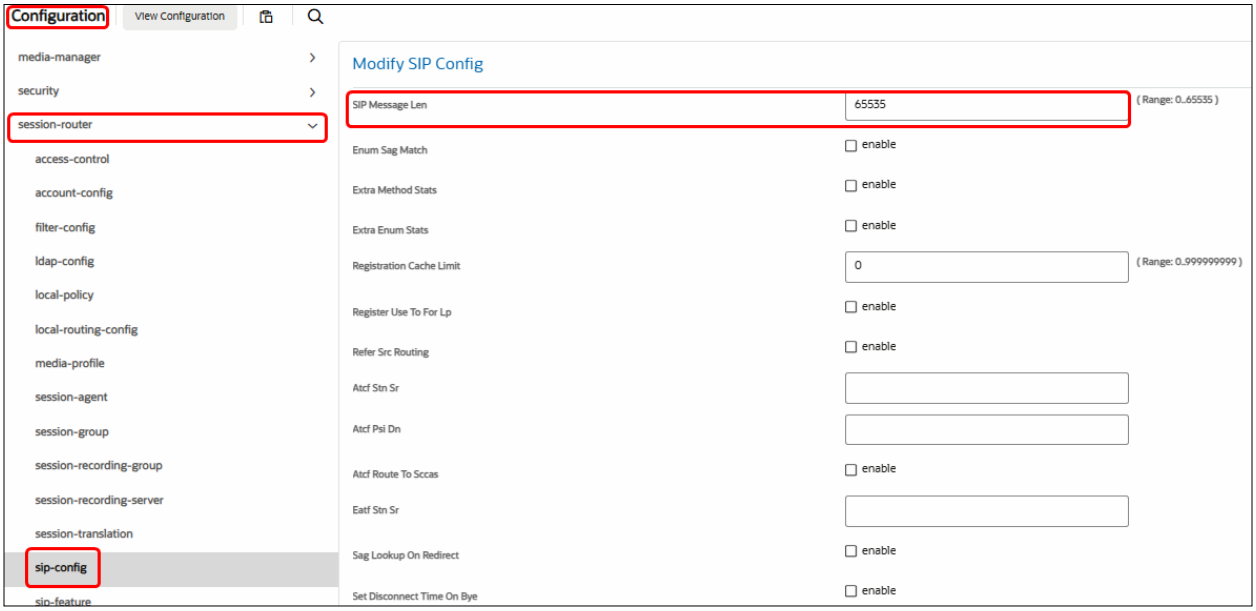

Figure 5.1: SIP-Config (cont.)

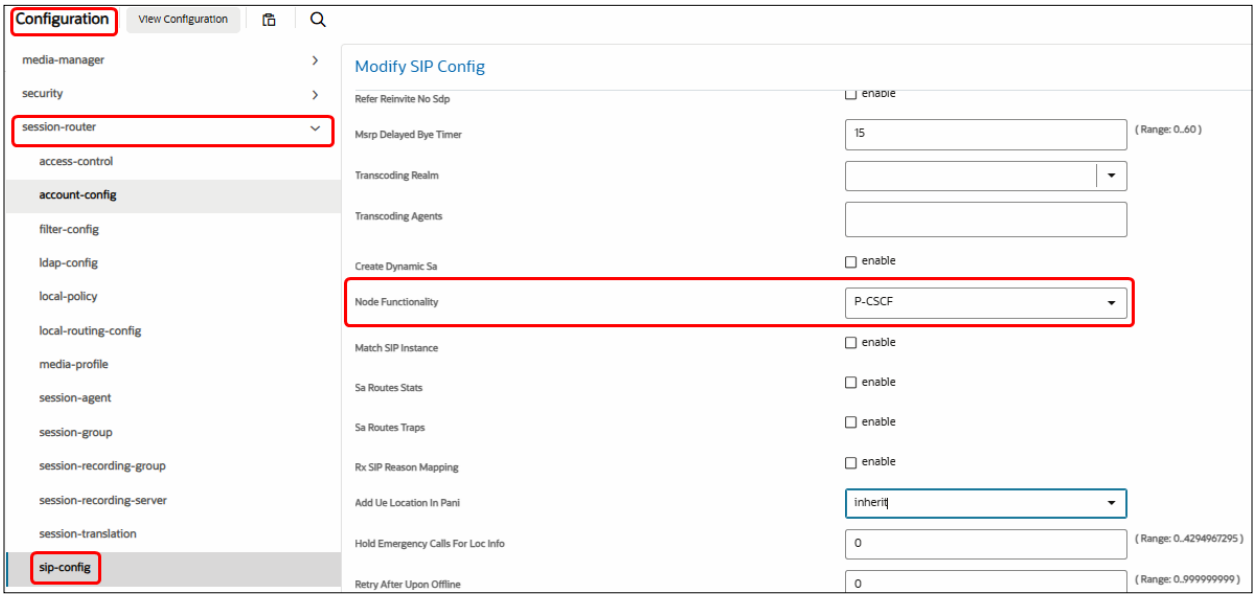

Figure 5.2: SIP-Config (cont.)

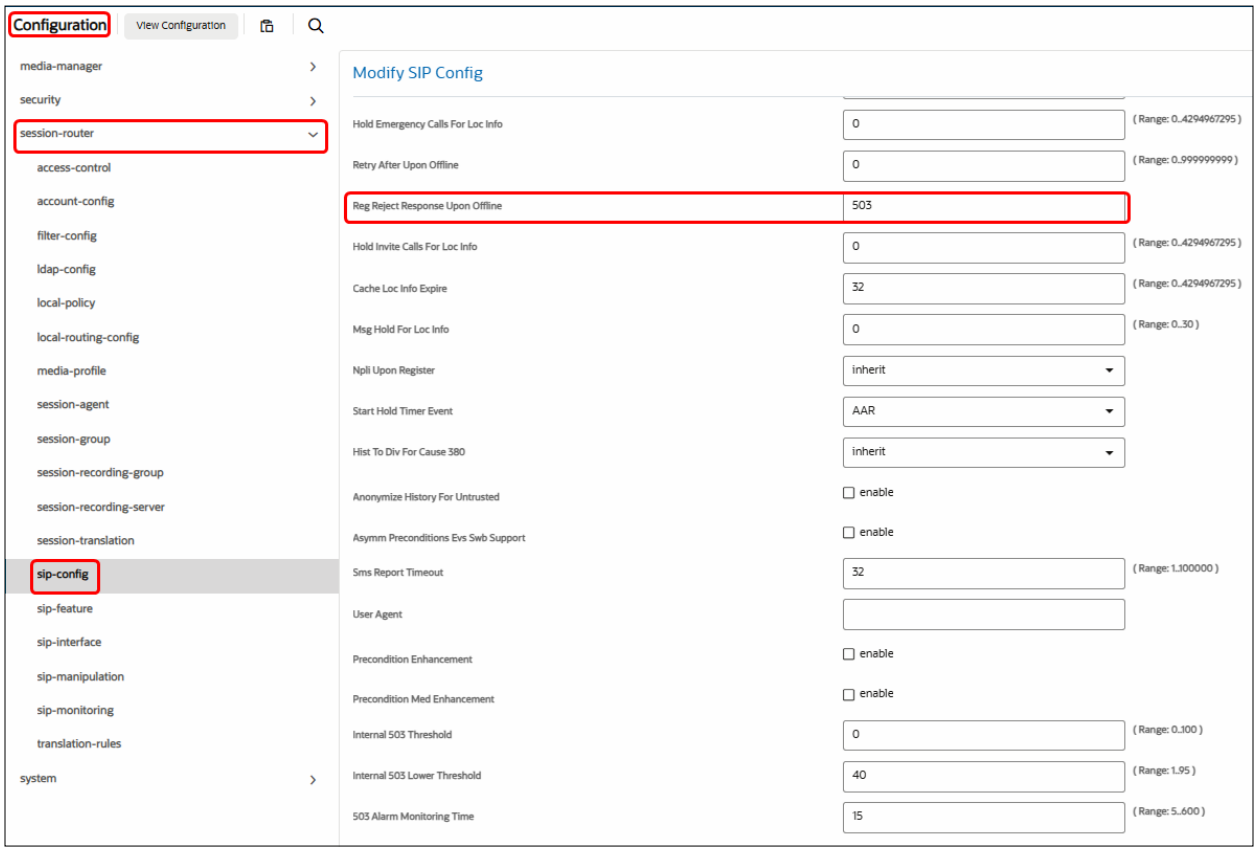

Figure 5.3: SIP-Config (cont.)

7.4.5 System-Config

- Navigate to **Configuration → system→ system-config** for system configuration as shown below.

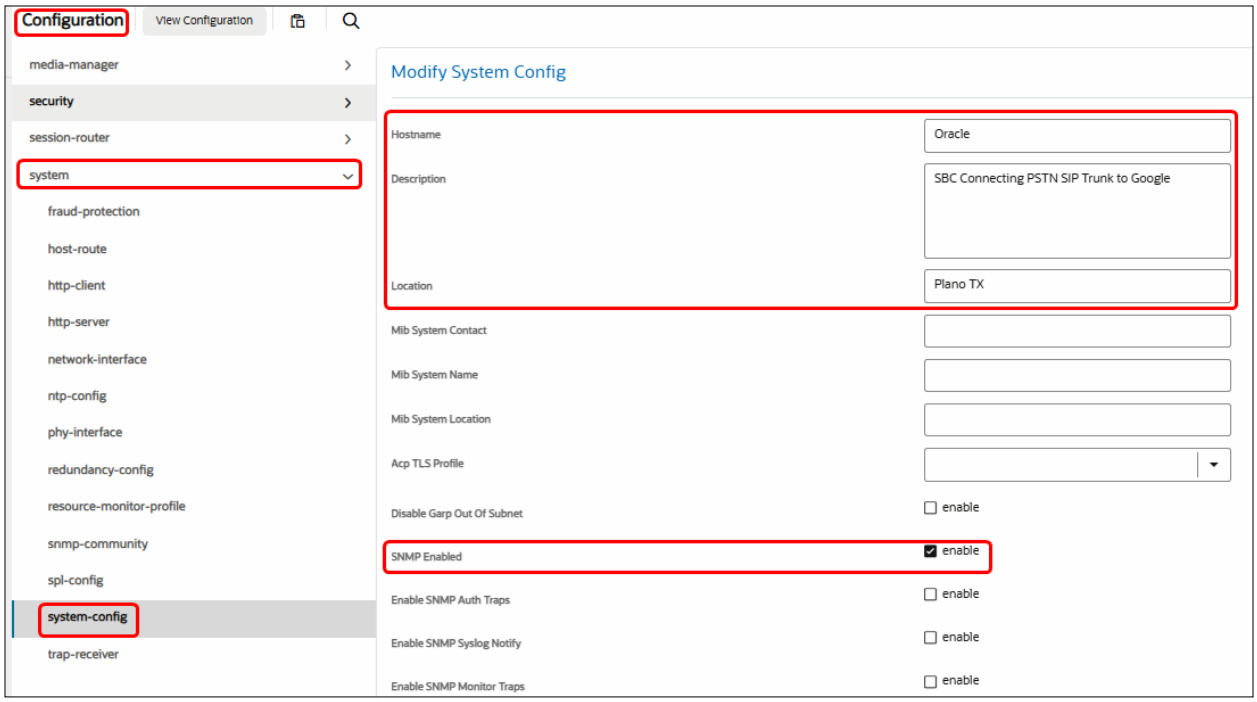

Figure 6: System-Config

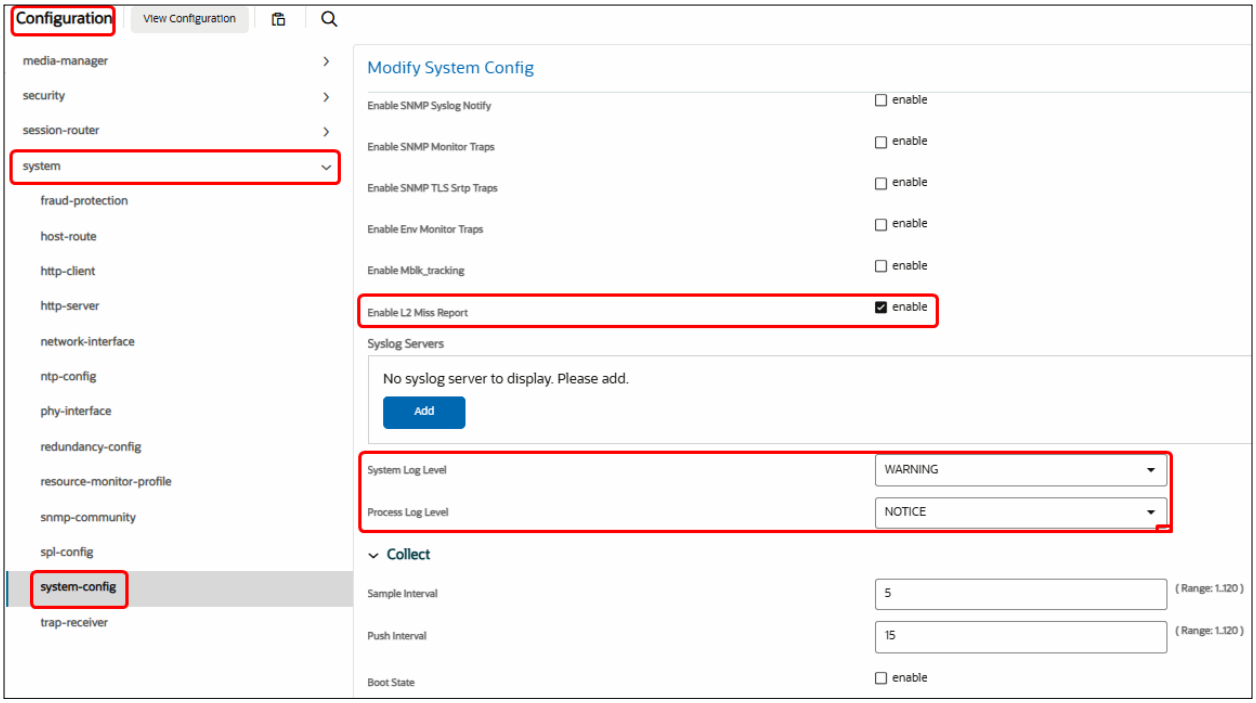

Figure 6.1: System-Config (Cont.)

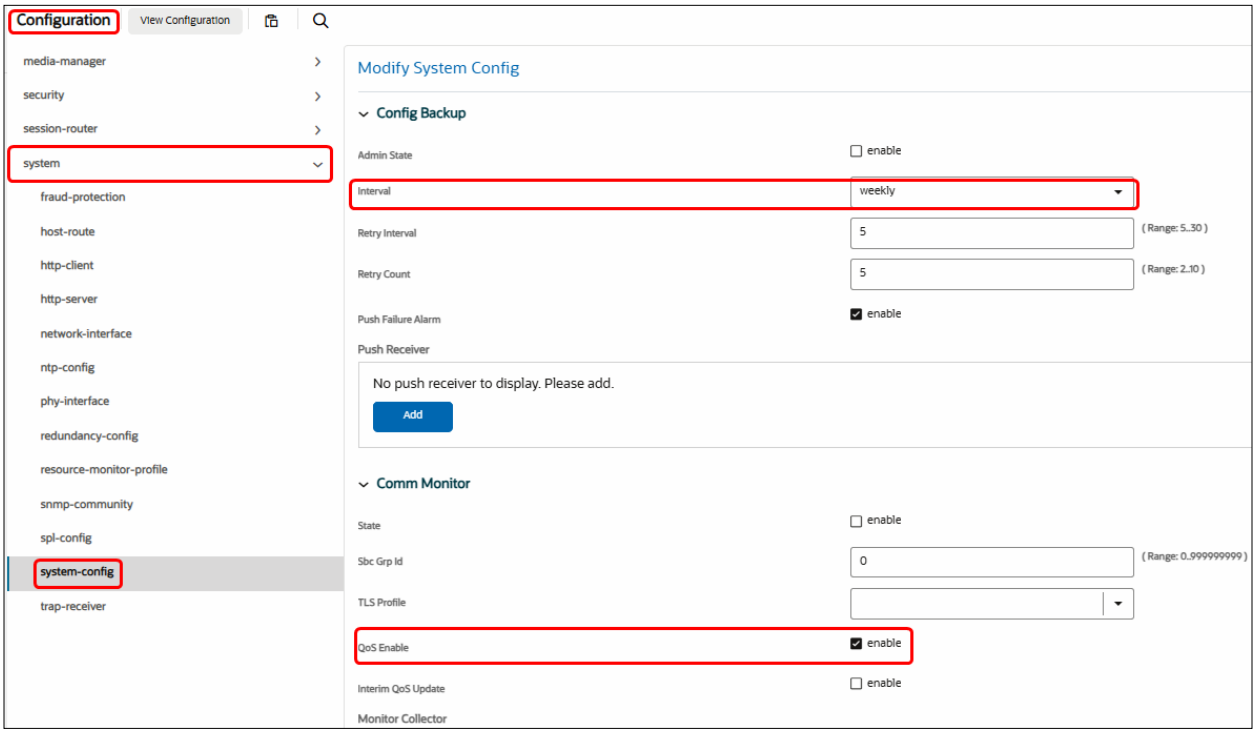

Figure 6.2: System-Config (Cont.)

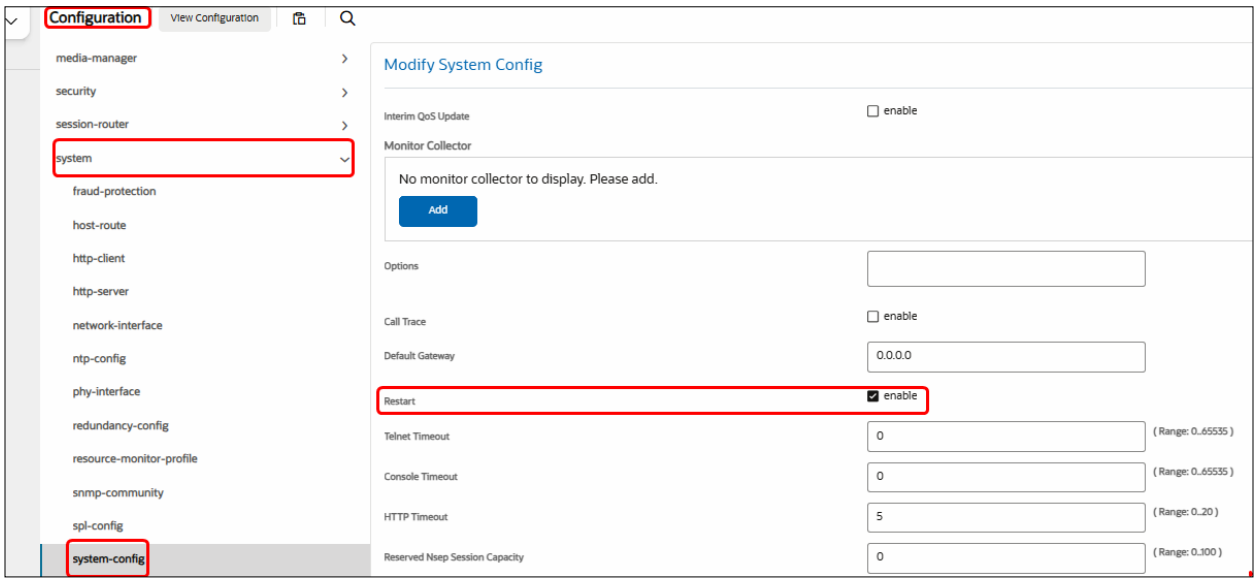

Figure 6.3: System-Config (Cont.)

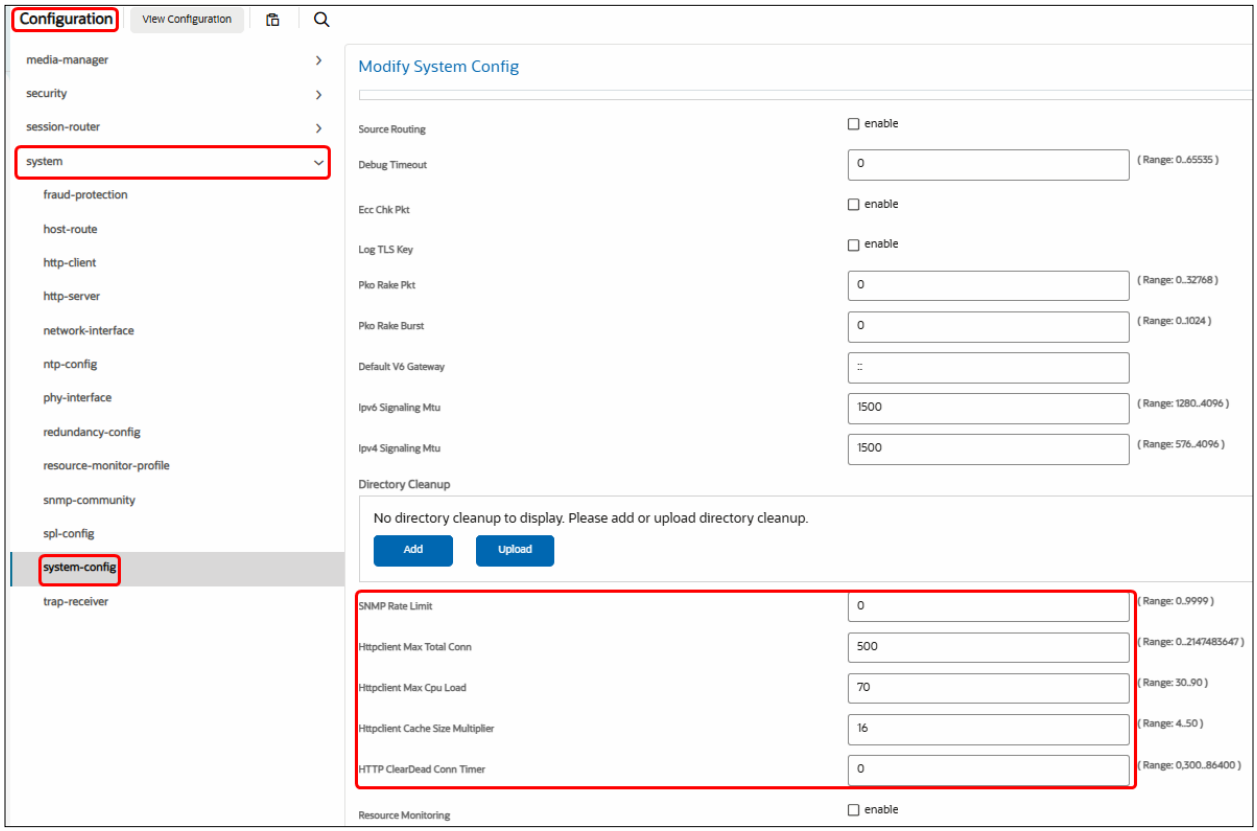

Figure 6.4: System-Config (Cont.)

7.4.6 SIP Monitoring

- Navigate to **Configuration** → **session-router** → **sip-monitoring** and configure SIP monitoring for capturing trace as shown below.

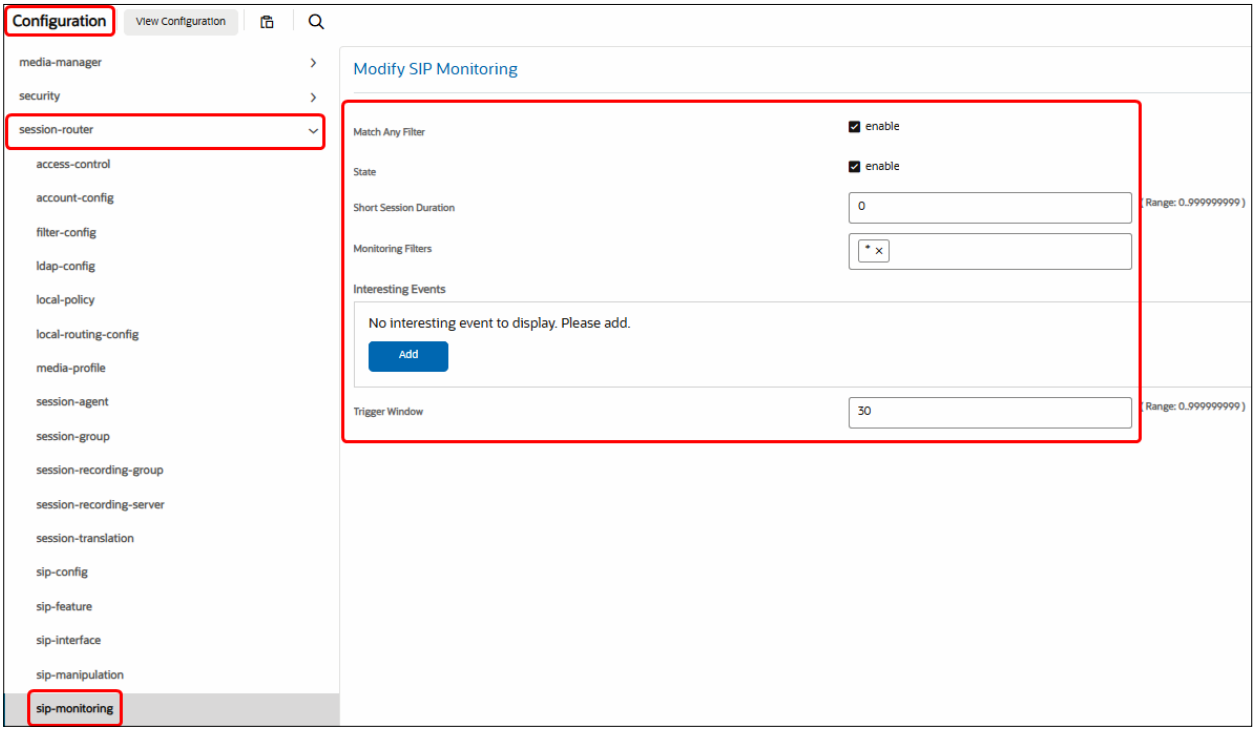

Figure 7: SIP Monitoring.

7.4.7 HTTP Server

- Navigate to **Configuration → system → http-server** and configure HTTP Server for GUI access to Oracle SBC as shown below.

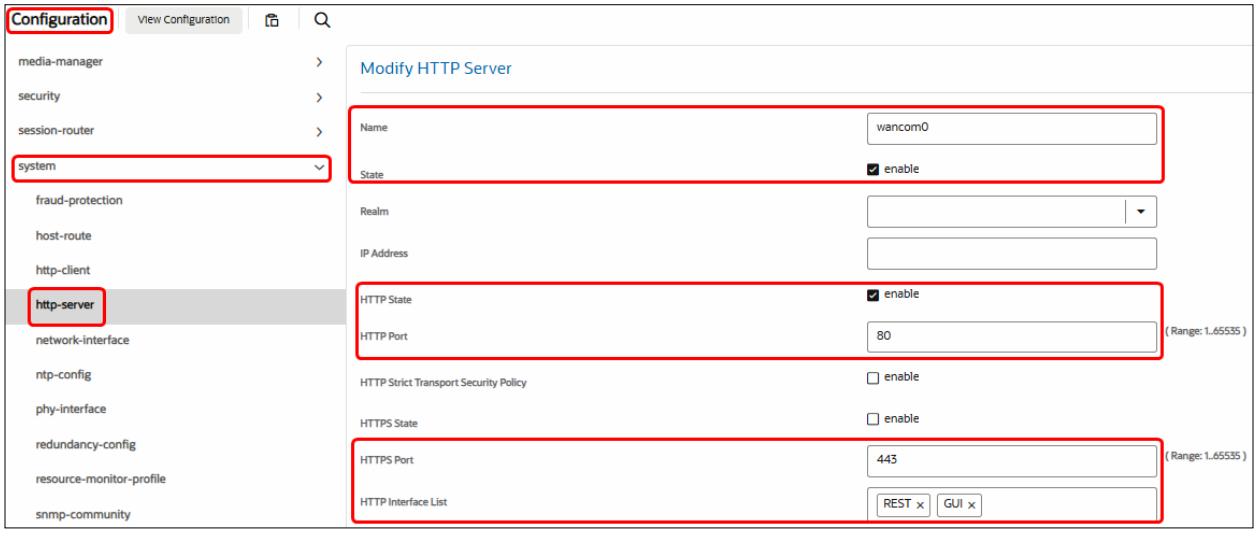

Figure 8: HTTP Server.

7.4.8 Codec Policy

- Navigate to **Configuration → media-manager → codec-policy** and configure codec policy for Google CES as shown below.

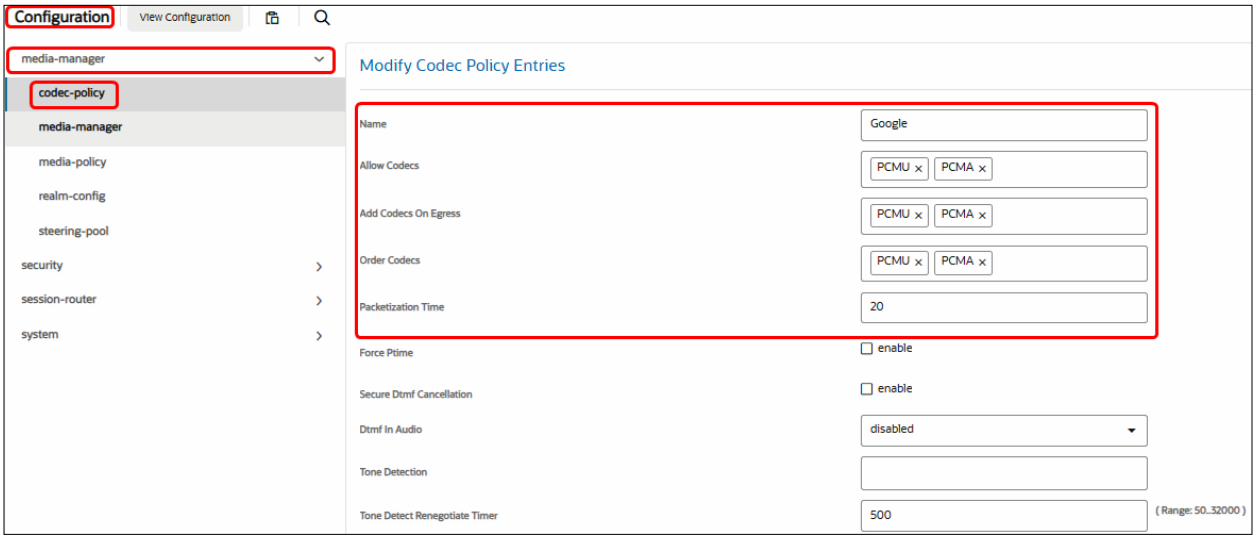

Figure 9: Codec Policy for Google CES.

7.4.9 Translation Rules

- Navigate to **Configuration → session-router → translation-rules** and **configure translation rules** for Google CES as shown below.
- Translation rule is created to send E.164 number format towards Google CES.

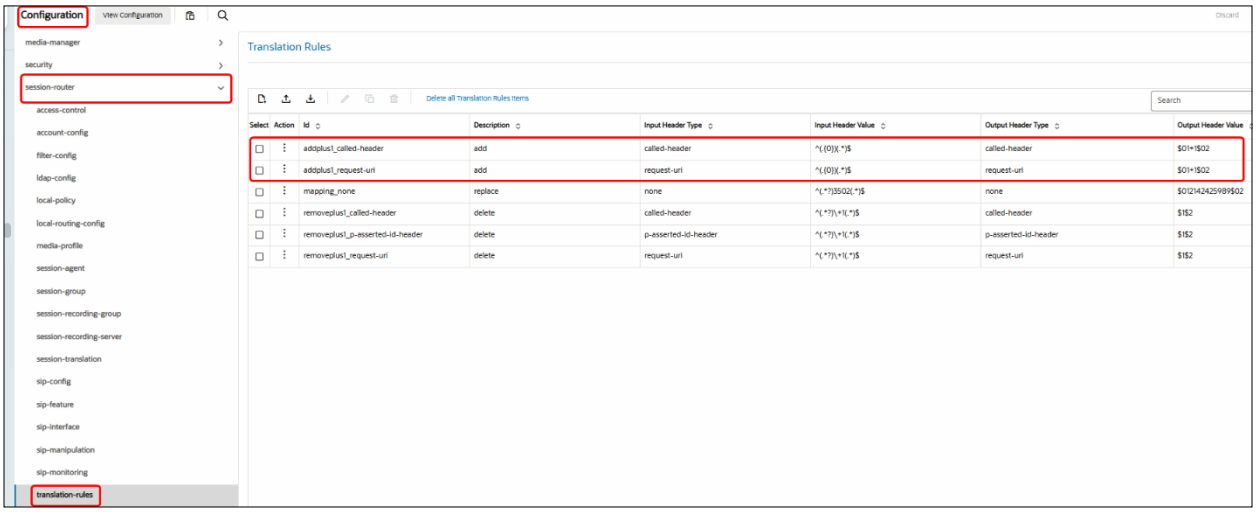

Figure 10: Translation Rule to add send E.164 towards Google CES.

7.4.10 Session Translation

- Navigate to **Configuration → session-router → session-translation.** The translation rules configured in Section 7.4.9 is mapped to Google CES is shown below.

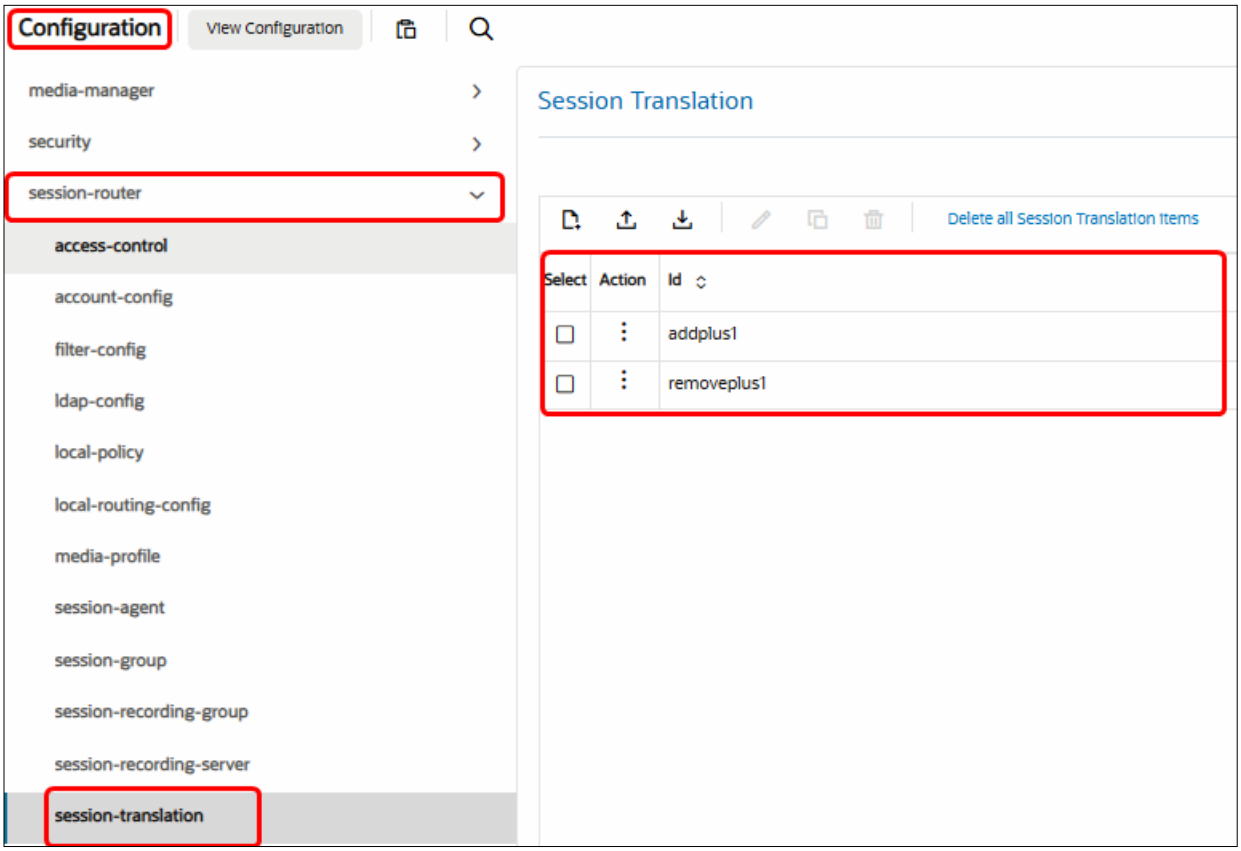

Figure 11: Session Translation towards Google CES.

7.4.11

Session Recording Server

- Navigate to **Configuration → session-router → session-recording-server** and select the destination as Google FQDN
- SIPREC profile for Google CES is created using the Session Recording Server

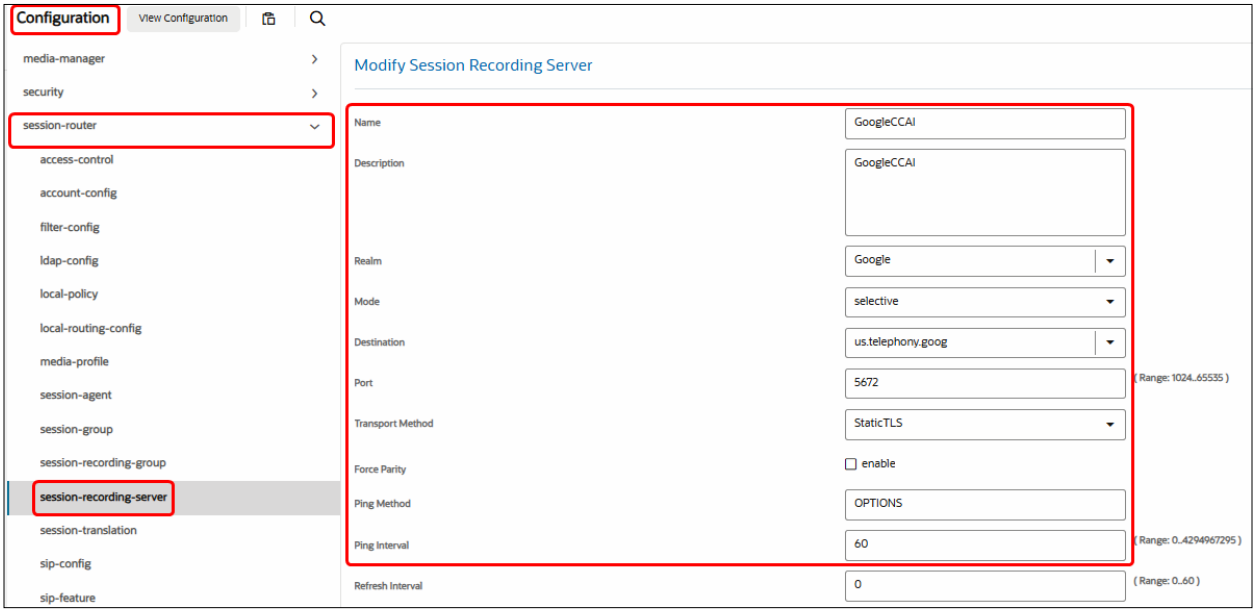

Figure 12: Session Recording Server towards Google CES.

7.4.12 Realm Config

- Navigate to **Configuration → media-manager → realm-config**.
- Realm Config towards Google CES is shown below.

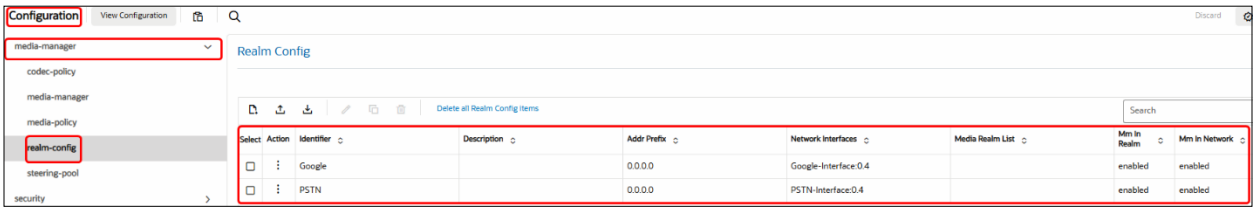

Figure 13: Realm Config towards Google CES.

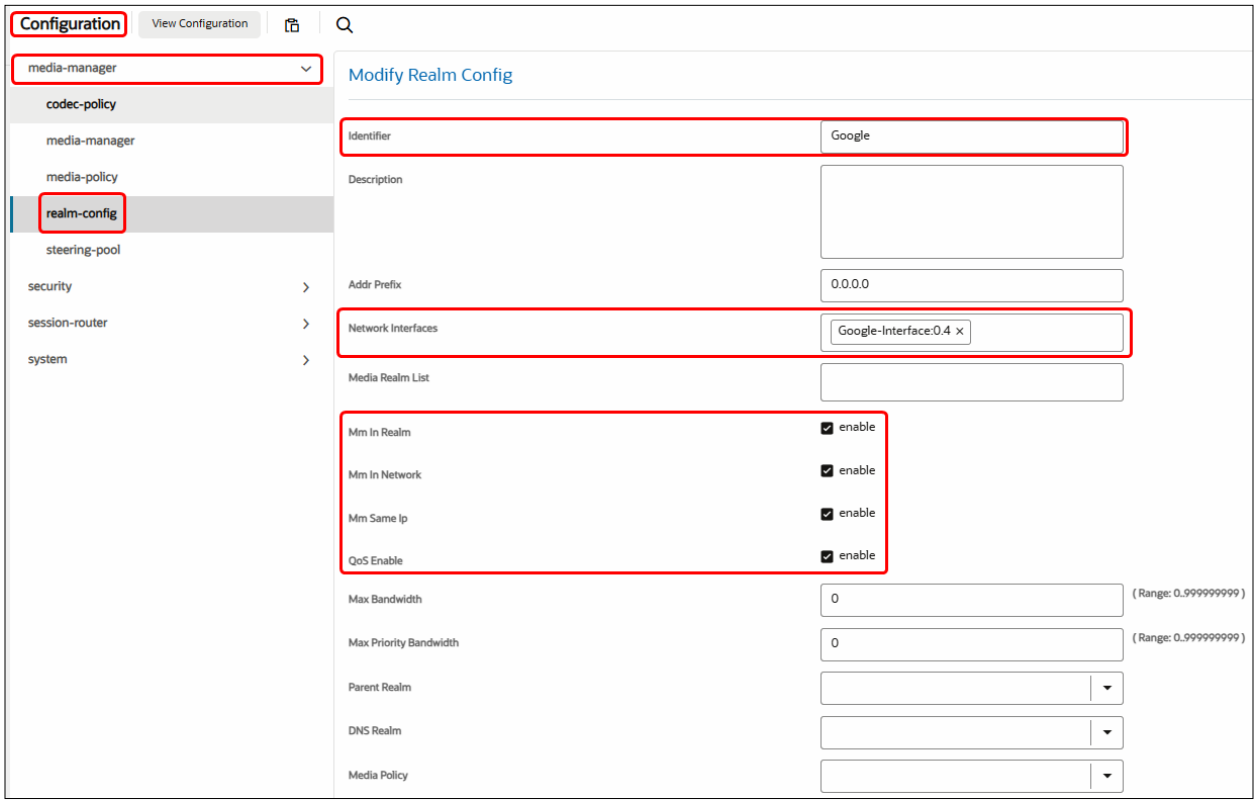

Figure 13.1: Realm Config towards Google CES (Cont.)

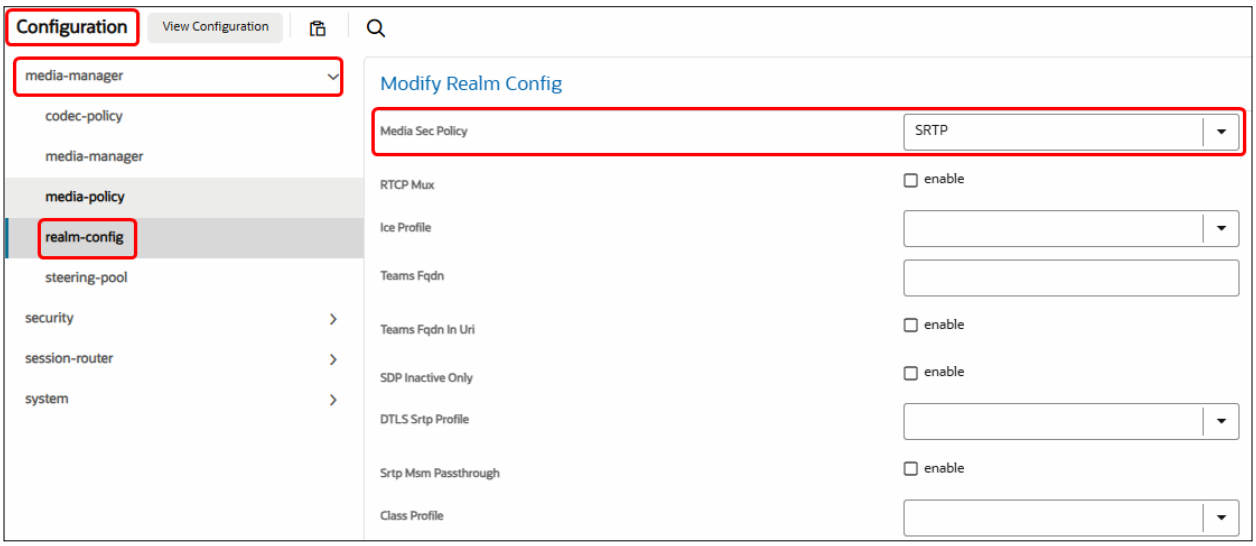

Figure 13.2: Realm Config towards Google CES (Cont.)

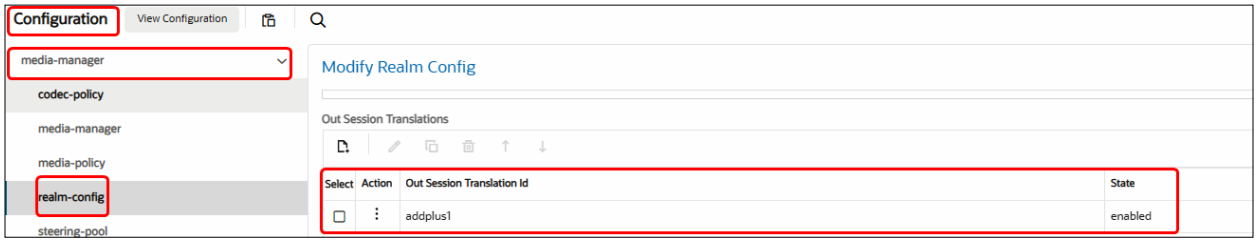

Figure 13.3: Realm Config towards Google CES (Cont.)

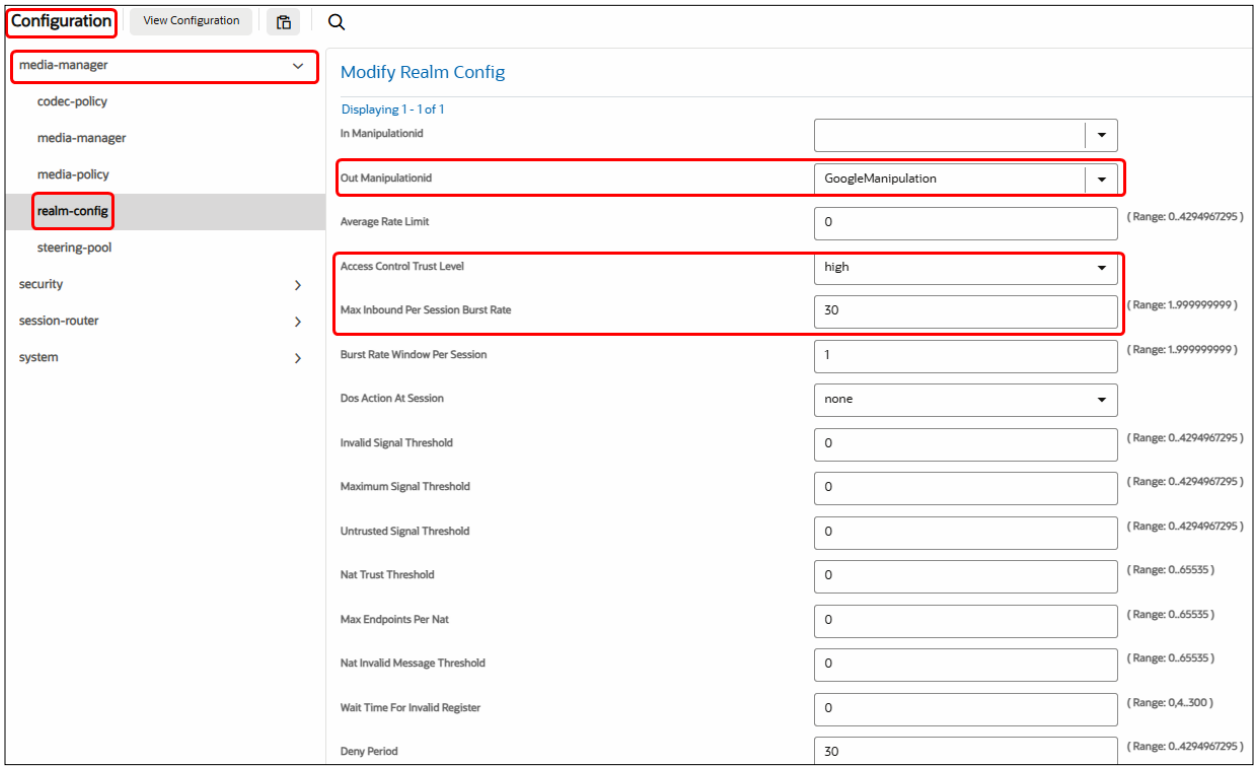

Figure 13.4: Realm Config towards Google CES (Cont.)

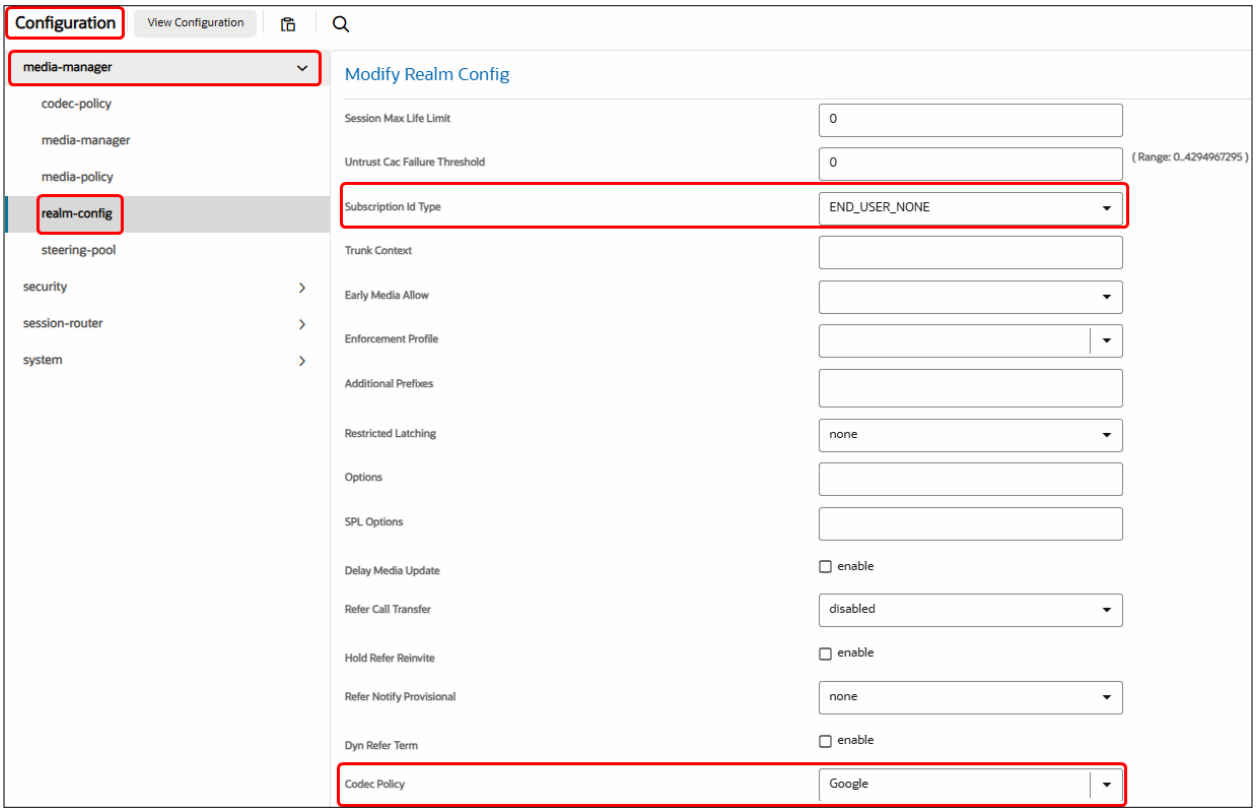

Figure 13.5: Realm Config towards Google CES (Cont.)

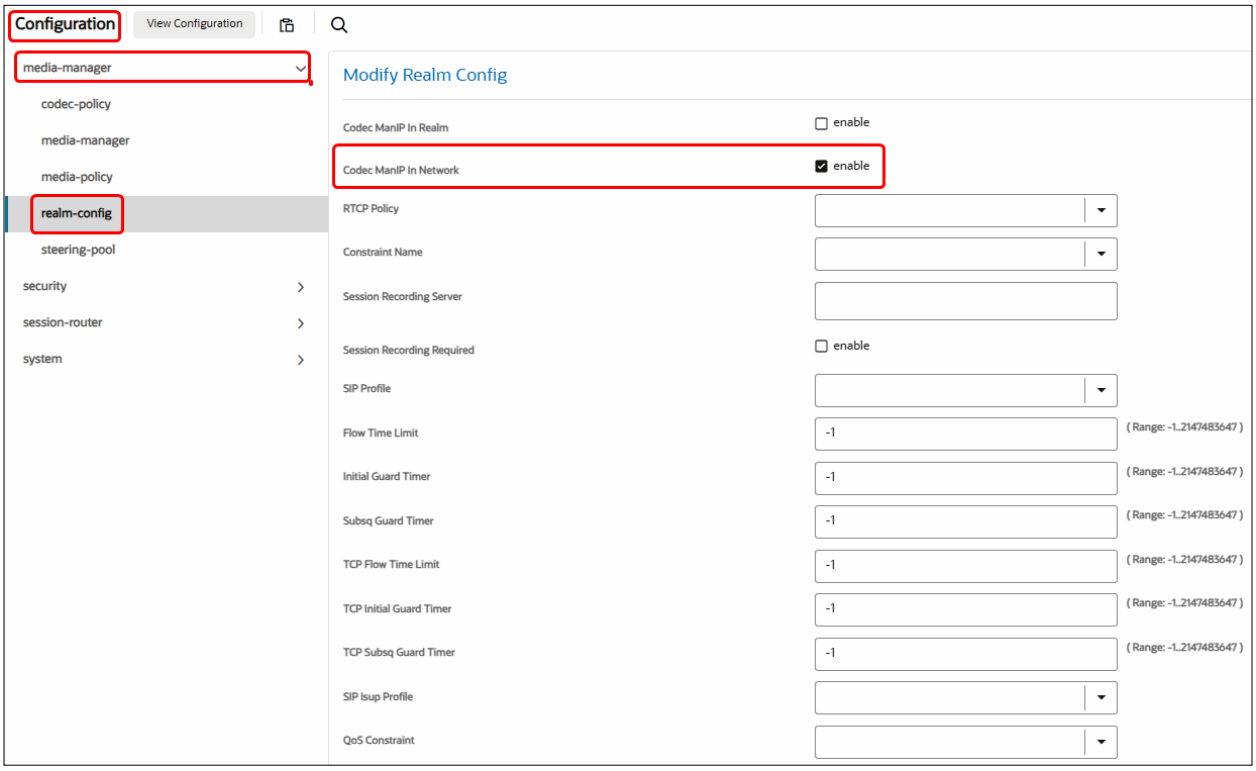

Figure 13.6: Realm Config towards Google CES (Cont.)

- Realm Config towards OnPrem PBX and PSTN Gateway is shown below.

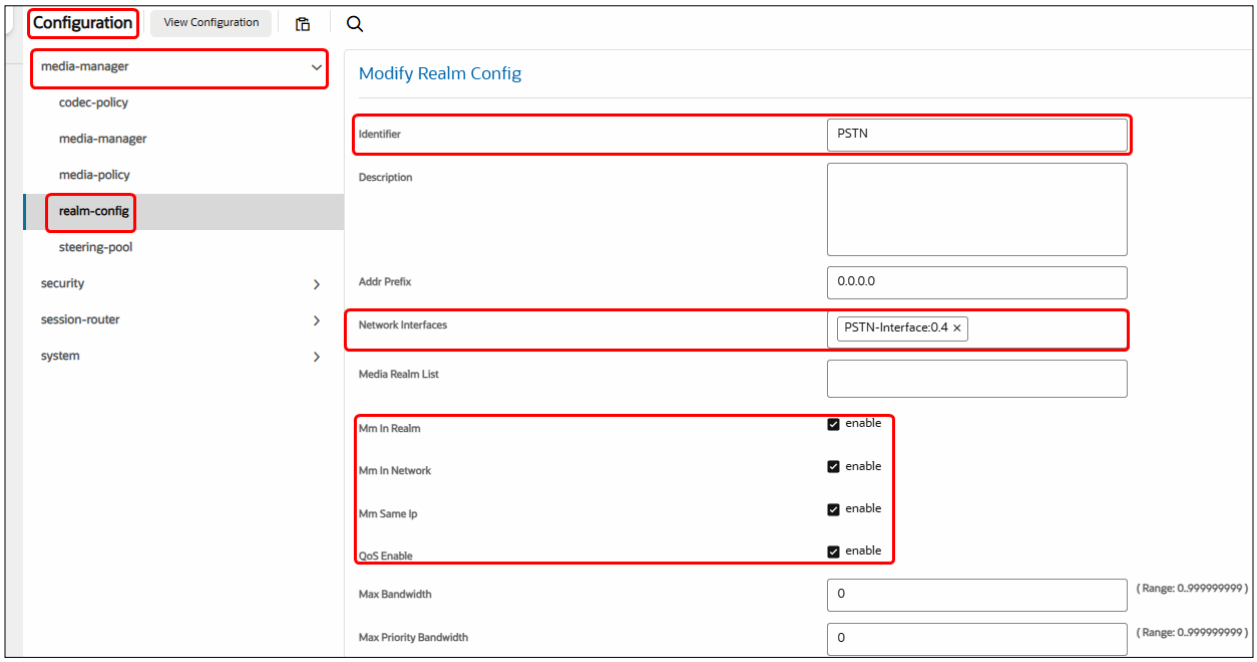

Figure 14: Realm Config towards OnPrem PBX and PSTN Gateway.

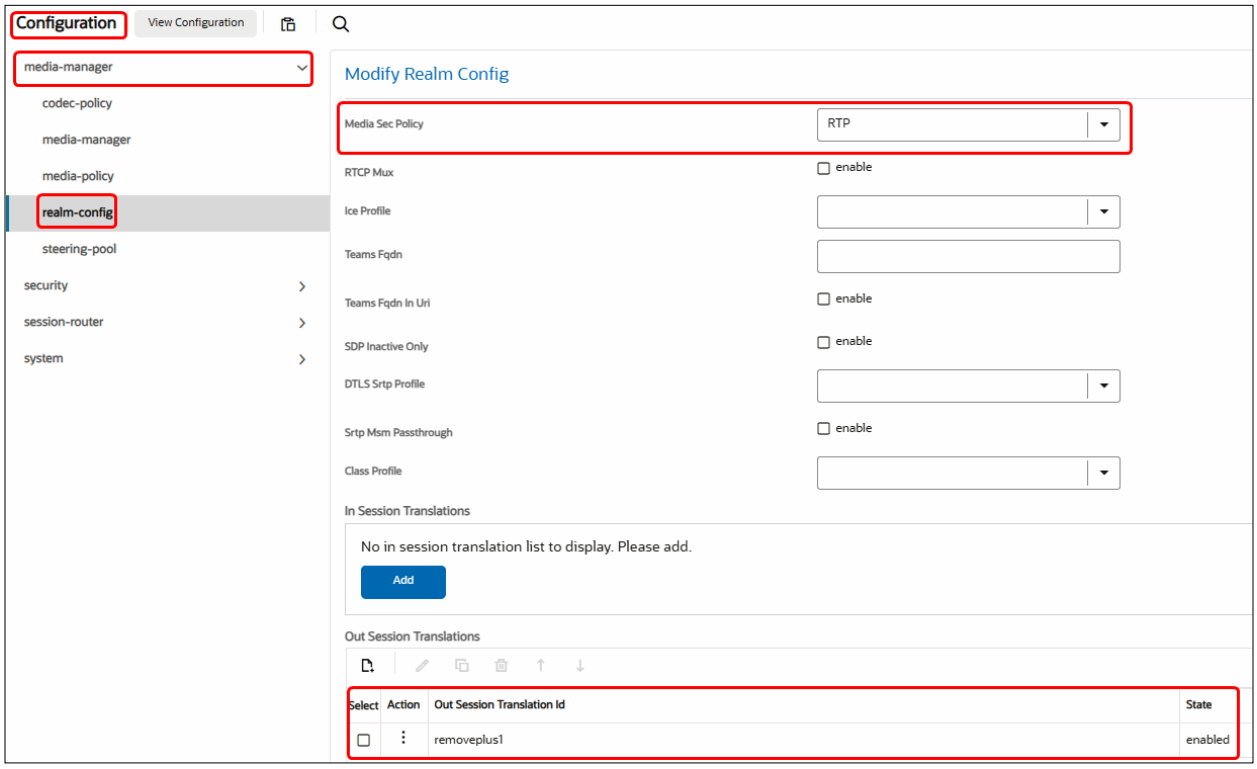

Figure 14.1: Realm Config towards OnPrem PBX and PSTN Gateway.

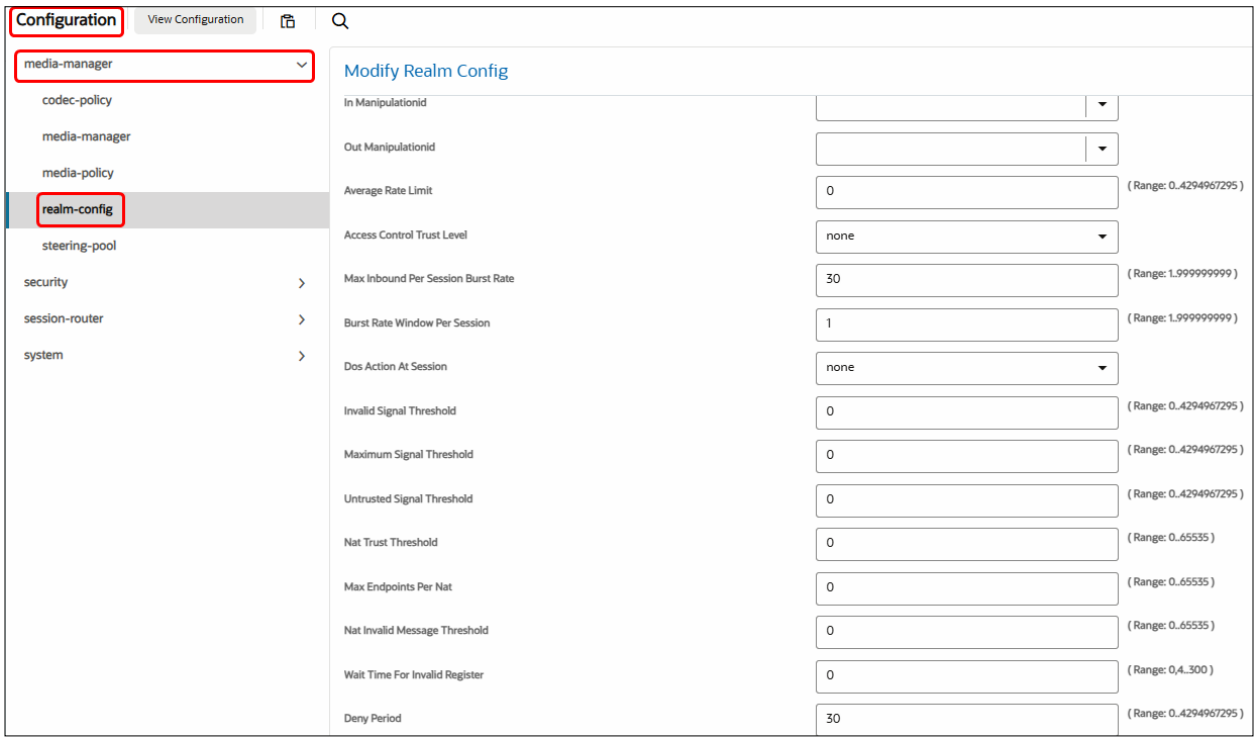

Figure 14.2: Realm Config towards OnPrem PBX and PSTN Gateway.

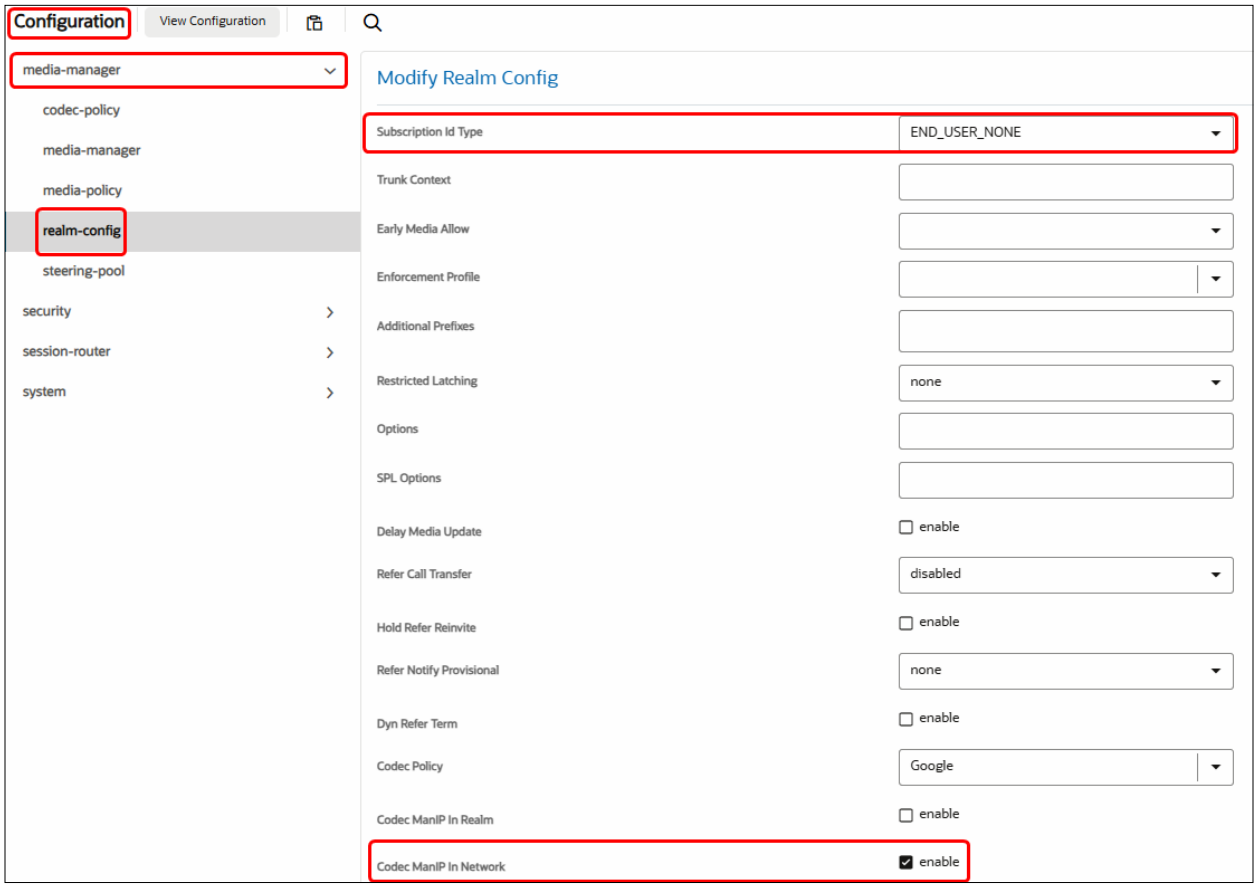

Figure 14.3: Realm Config towards OnPrem PBX and PSTN Gateway Cont.

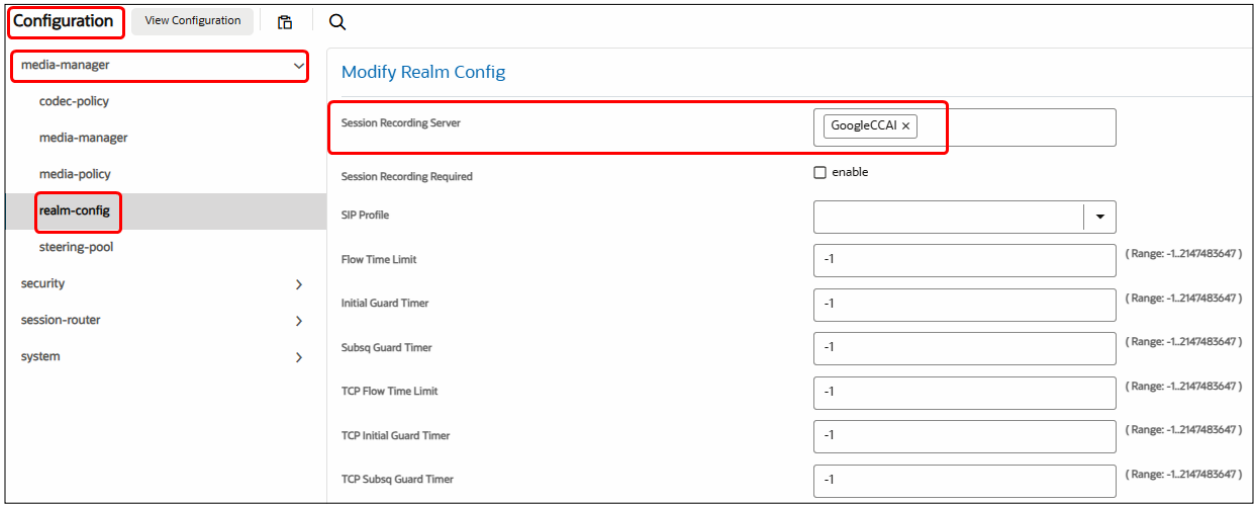

Figure 14.4: Realm Config towards OnPrem PBX and PSTN Gateway Cont.

7.4.13 Steering Pool

- Navigate to **Configuration → media-manager → steering-pool**.
- Steering pool allows configuration to assign IP address, ports, and a realm.
- Steering Pool Configuration towards OnPrem PBX and PSTN Gateway are shown below.

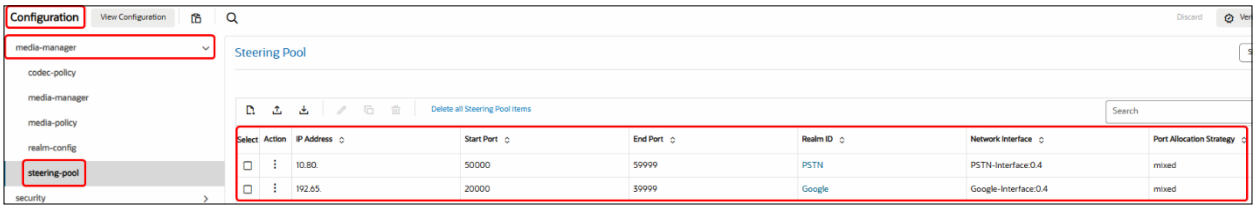

Figure 15: Steering Pool.

- Steering Pool Configuration towards Google CES is shown below.

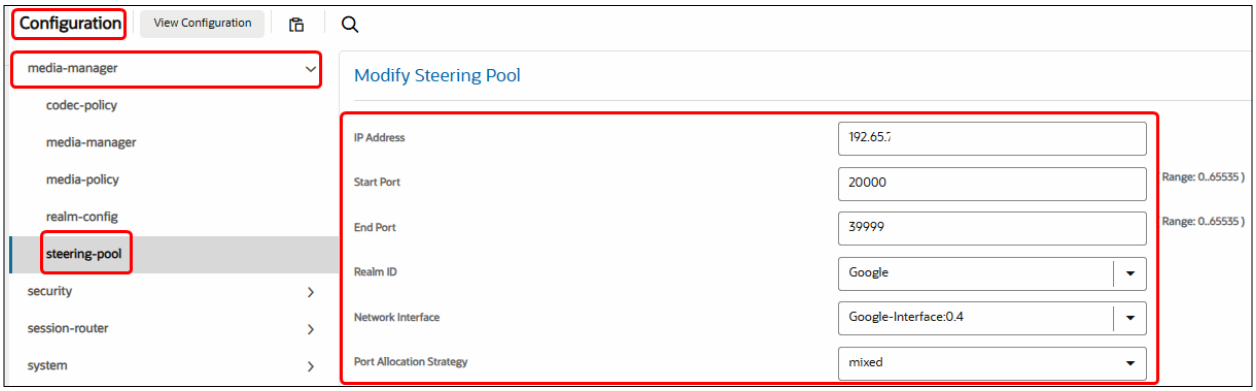

Figure 15.1: Steering Pool towards Google CES.

- Steering Pool Configuration towards Onprem PBX & PSTN is shown below

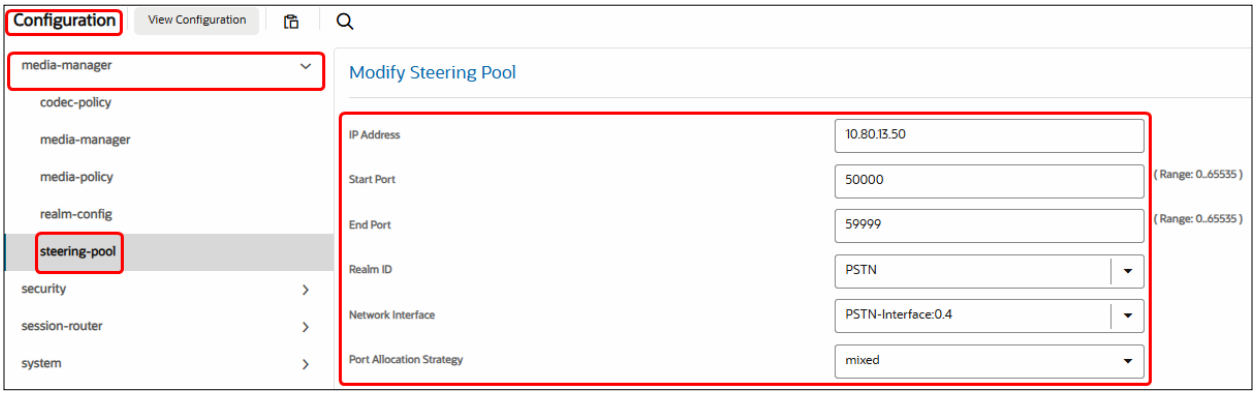

Figure 15.2: Steering Pool towards Google CES.

7.4.14 SDES Profile

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

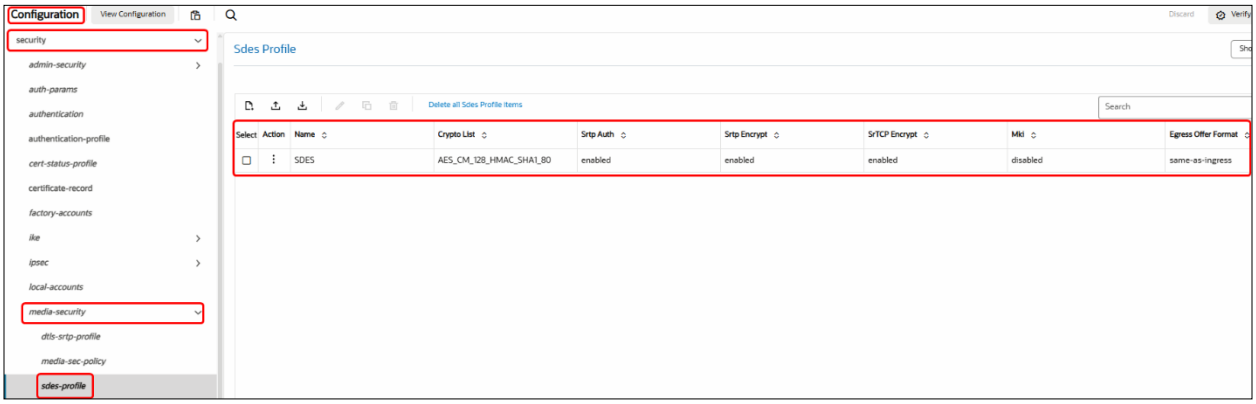

Figure 16: SDES Profile for TLS.

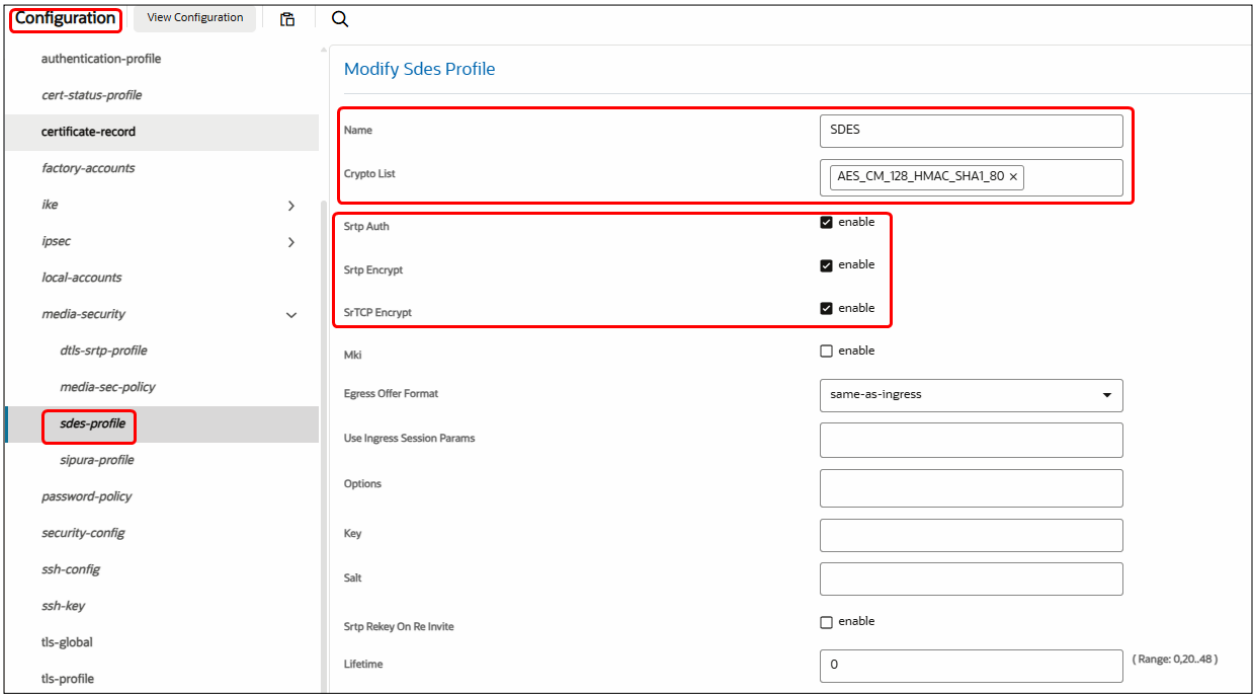

Figure 16.1: SDES Profile for TLS.

7.4.15 Media Sec Policy

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

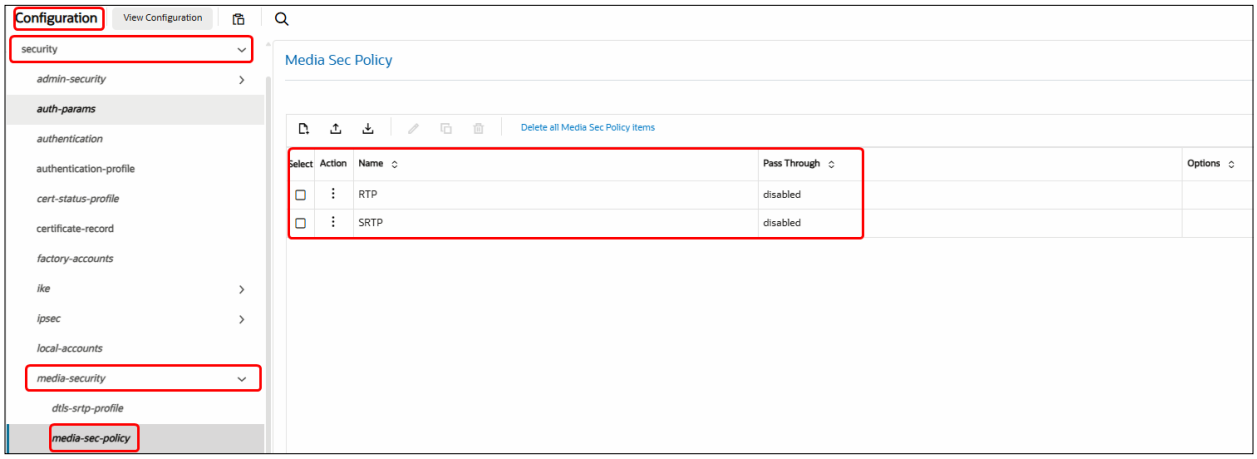

Figure 17: Media Security Policy

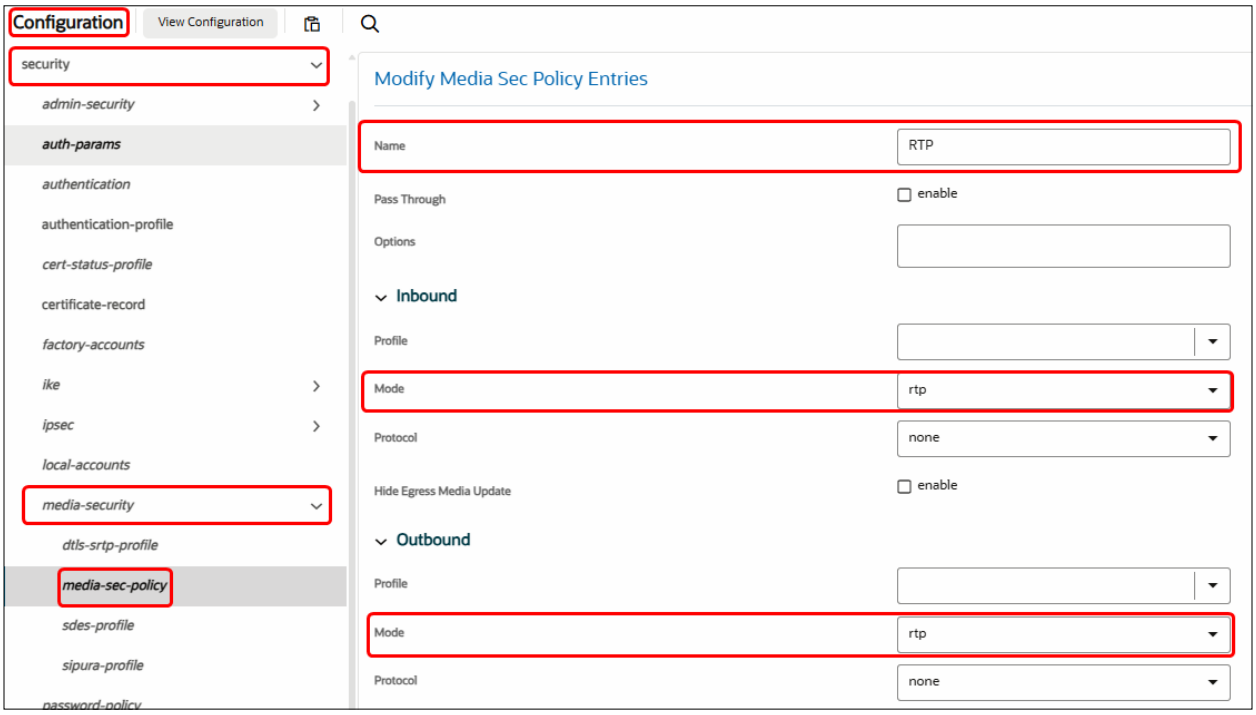

Figure 17.1: Media Security Policy for RTP.

- SDES profile created in Section 7.4.14 is associated with Media Security Policy for SRTP below.

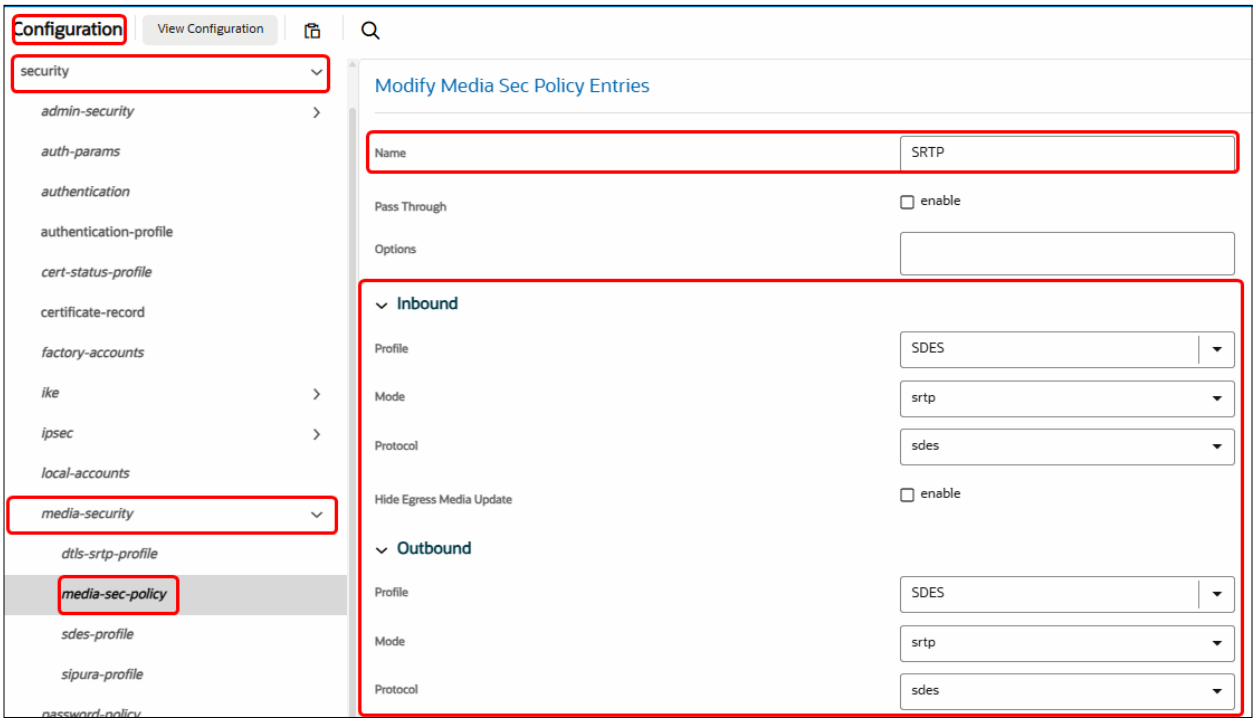

Figure 17.2: Media Security Policy for SRTP.

7.4.16 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 Oracle E-SBC as shown below.
    - Select the Certificate record and Click **Generate icon** to generate CSR.
    - Get the CSR signed and click **Import** to import the signed certificate.

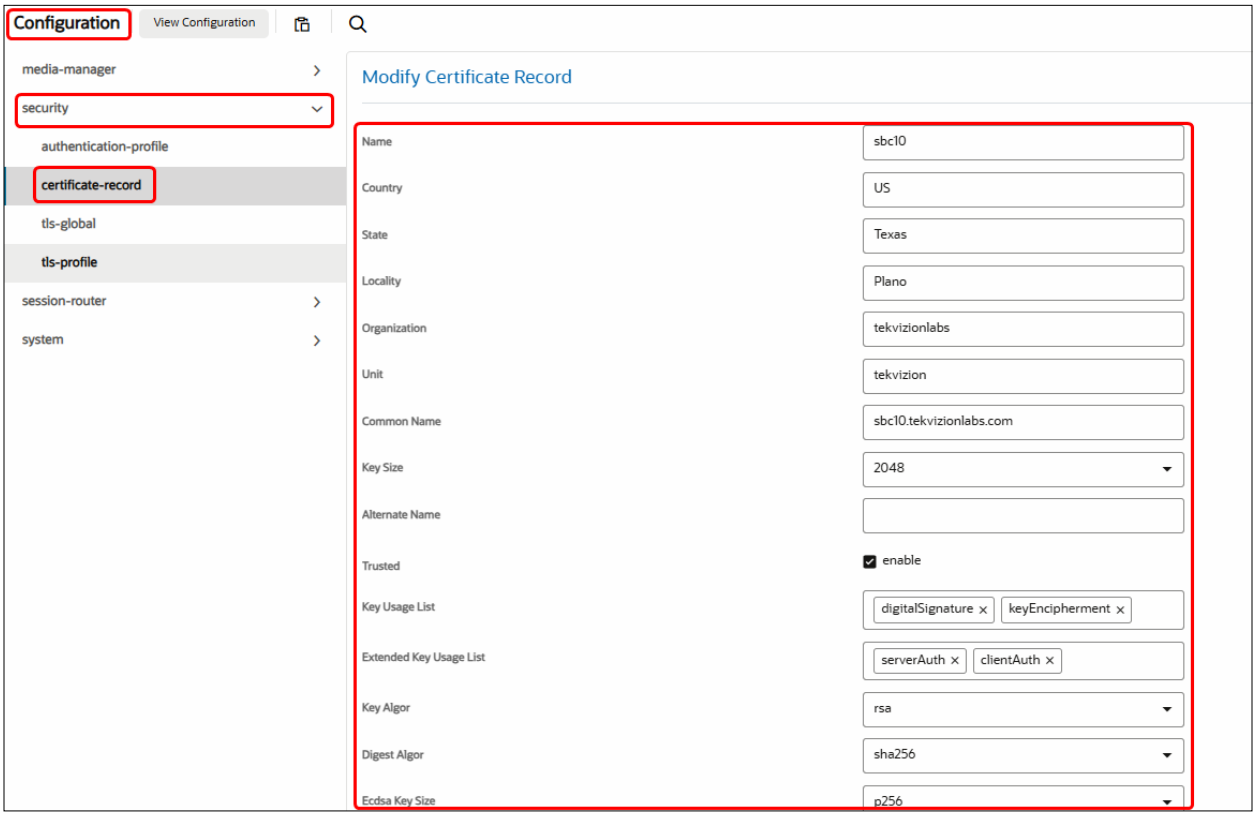

Figure 18: Create Certificate Record for Oracle E-SBC

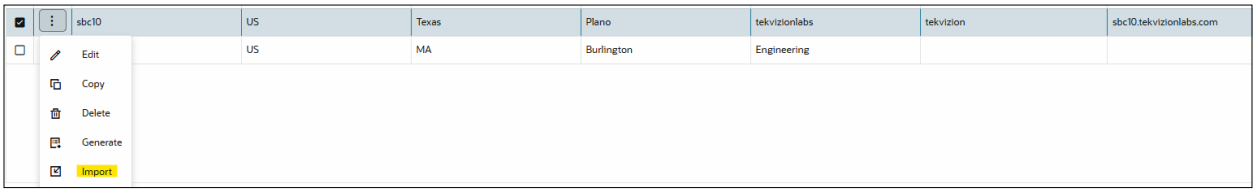

Figure 18.1: Import Certificate Record for Oracle E-SBC

- Download the Google certificate via link https://pki.goog/roots.pem
- Create a certificate record GTS Root R1 for Google CES.

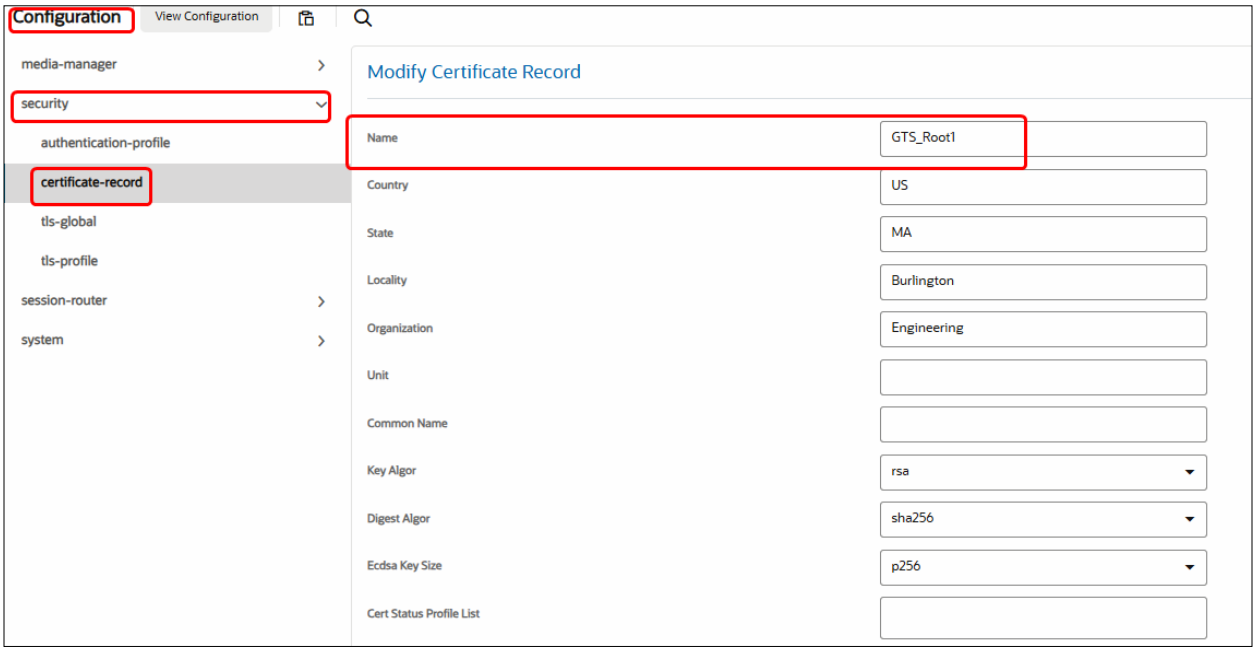

Figure 18.2: Create Certificate Record for Google GTS Root R1 certificate

- Right click on the Certificate Record and Click Import. Import the root certificate stored in the local machine and click Import as shown below.

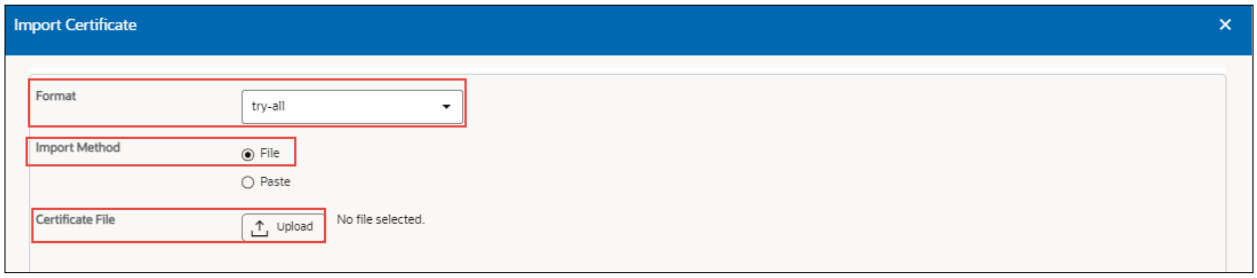

Figure 18.3: Import Google GTS Root R1 certificate.

- Similarly create other certificate records for the SBC leaf certificate and the Certificate Authority Root certificate, ensuring the entire certificate chain from leaf to root is present as shown below. The following certificate-records are required on the Oracle SBC to connect with Google CES.

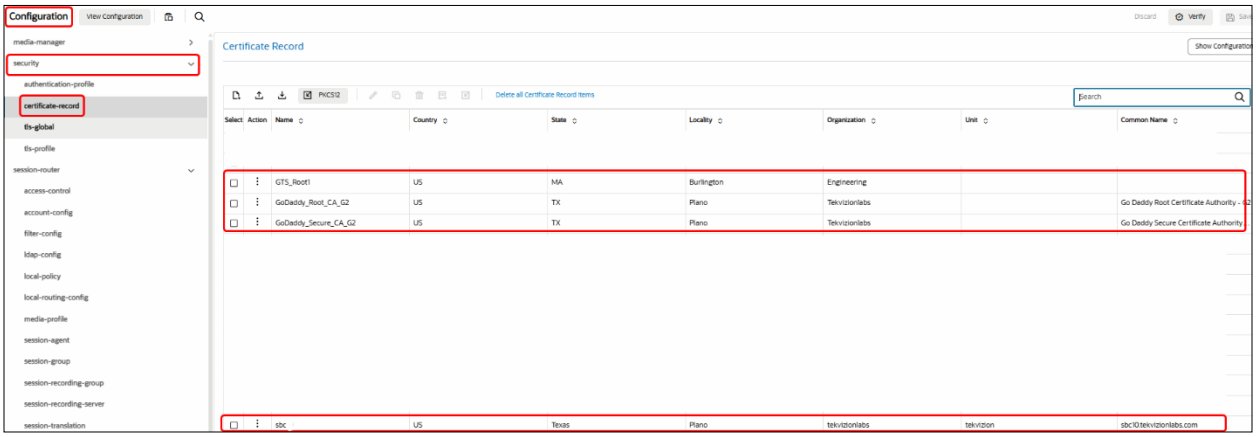

Figure 18.4: Certificate Records.

7.4.17 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 CES as shown below.

Figure 19: TLS Profile.

- Intermediate Certificates and Google GTS Root R1 certificate is added for TLS exchange.

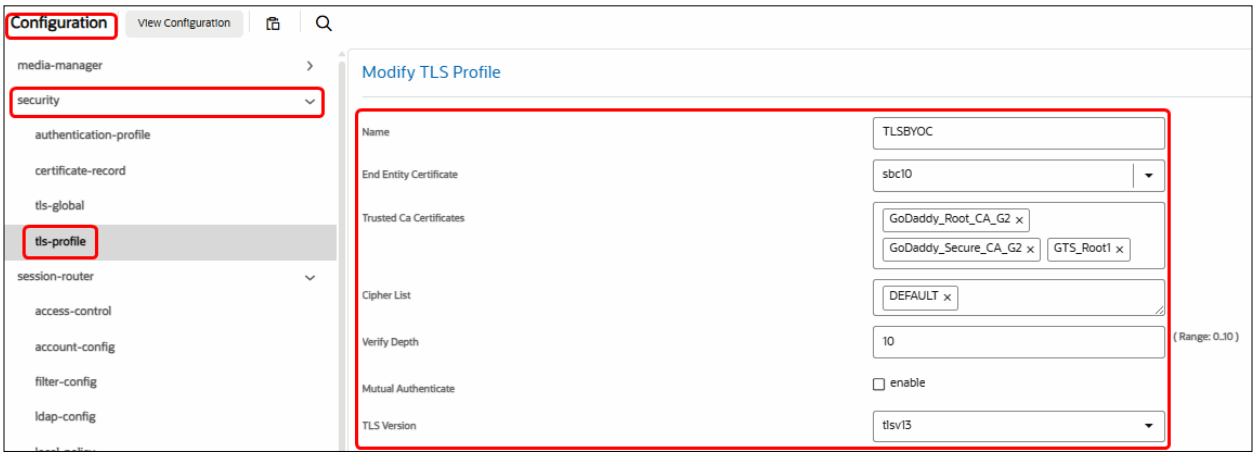

Figure 19.1: TLS Profile

7.4.18 Session Timer

- Navigate to **Configuration → session-router → session-timer-profile**.
- Configure session timer for Google CES as shown below.

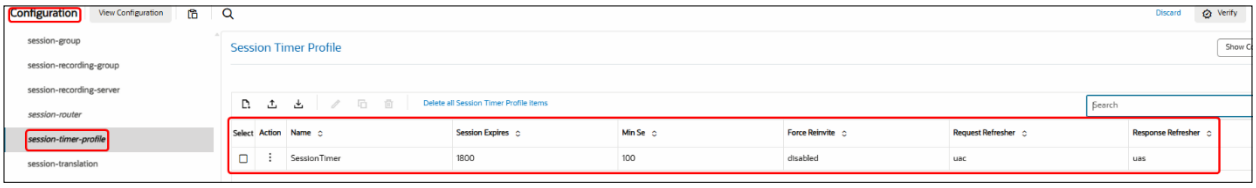

Figure 20: Session Timer

7.4.19 SIP Interface

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

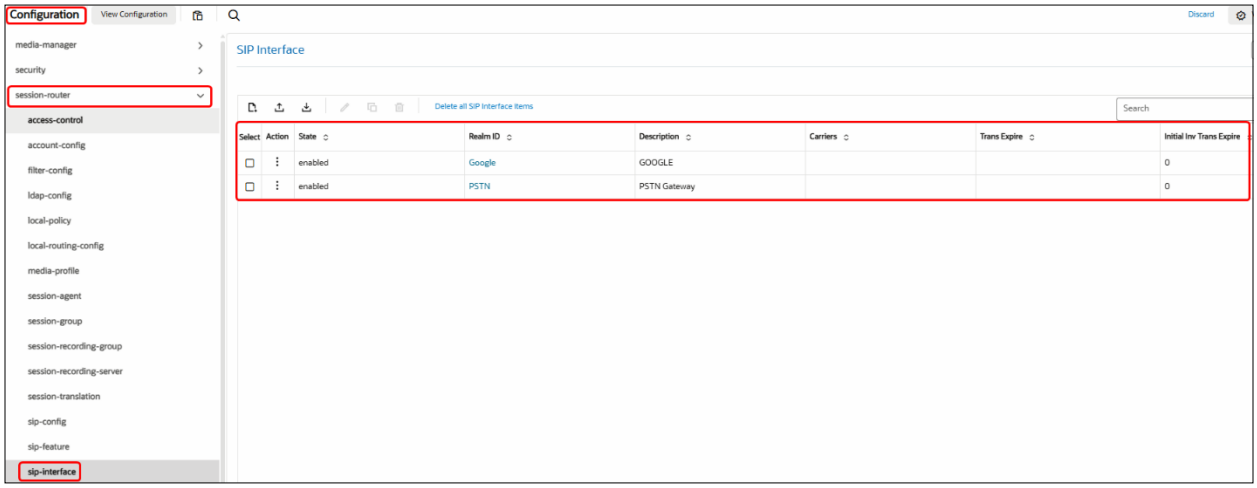

Figure 21: SIP Interface.

- Create SIP interface towards PSTN Gateway and OnPrem PBX by adding SIP Ports as shown below

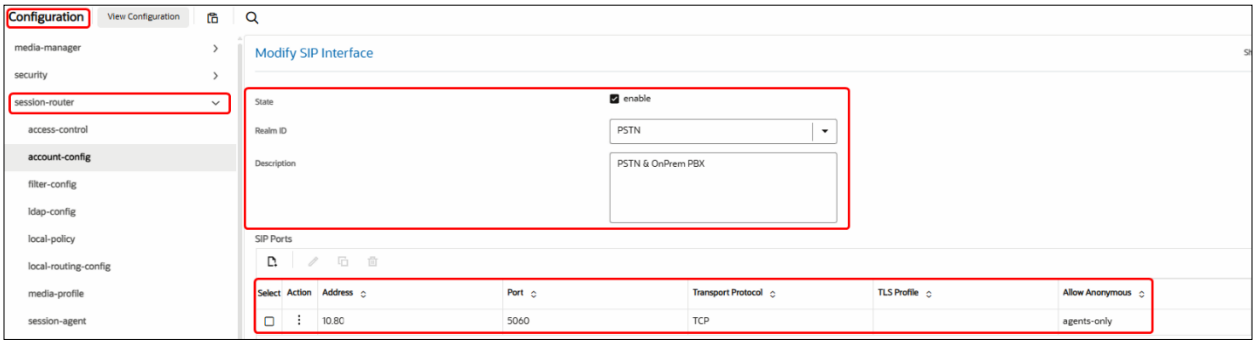

Figure 21.1: SIP Interface for PSTN &OnPrem PBX (Cont.)

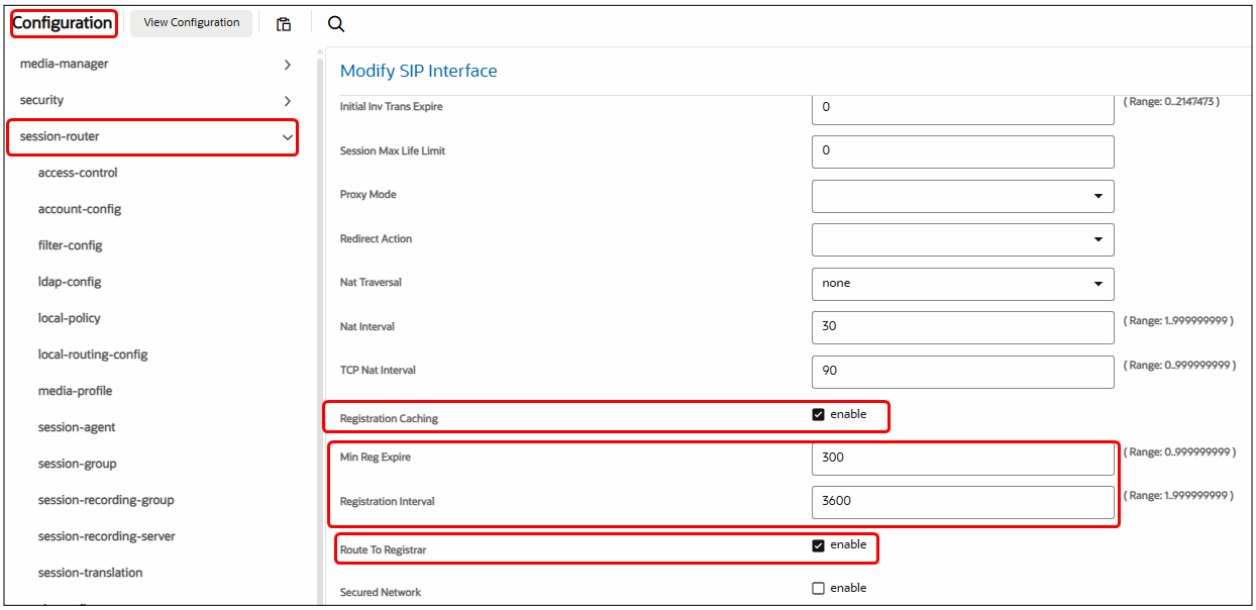

Figure 21.2: SIP Interface for PSTN &OnPrem PBX (Cont.)

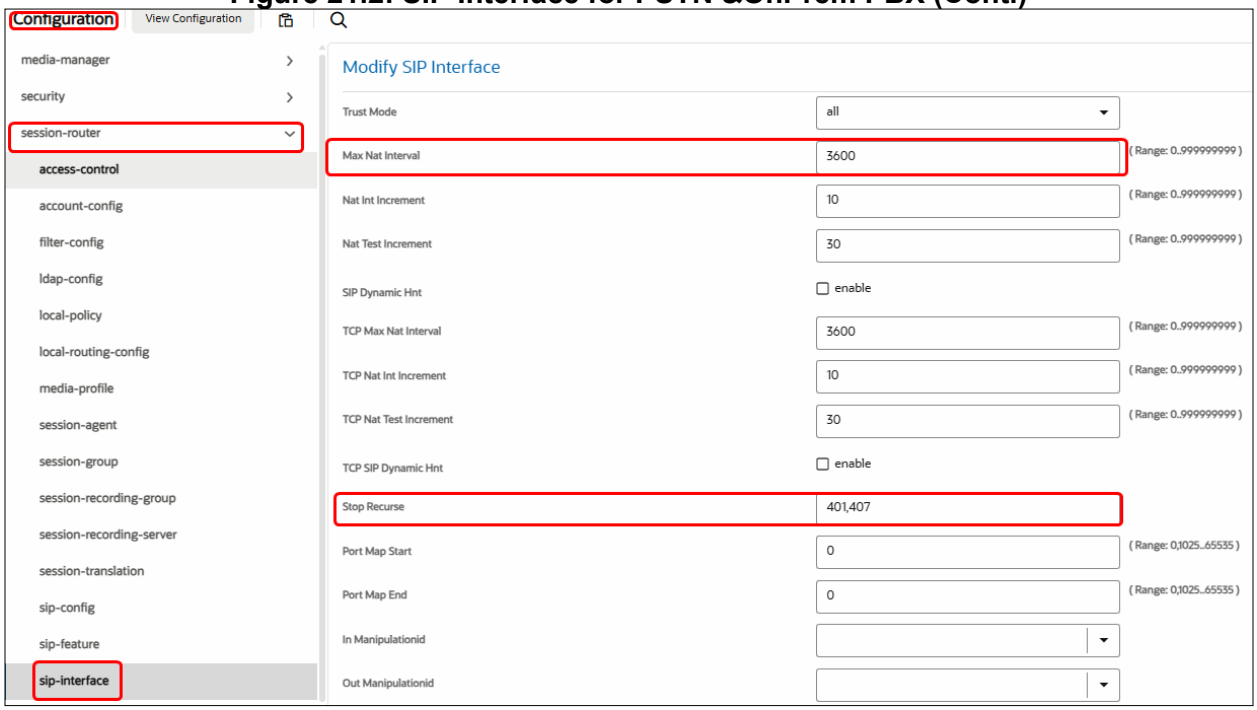

Figure 21.3: SIP Interface for PSTN &OnPrem PBX (Cont.)

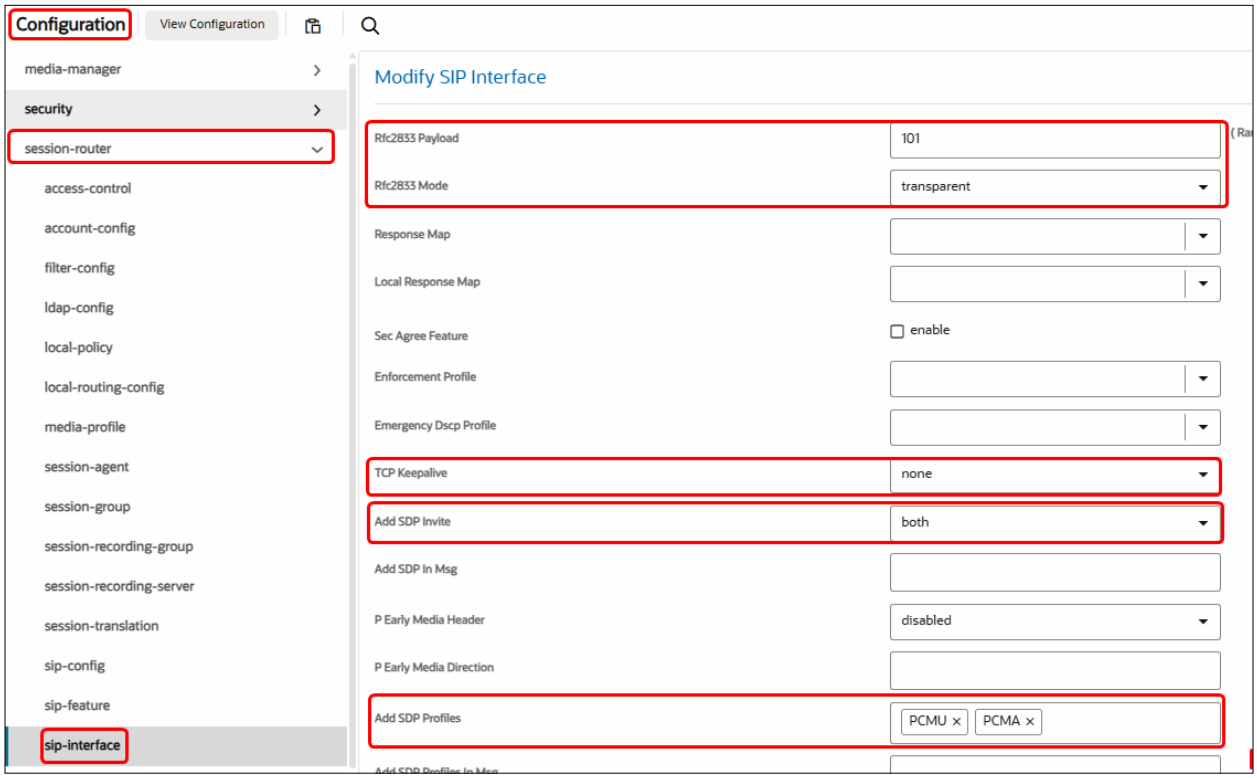

Figure 21.4: SIP Interface for PSTN &OnPrem PBX (Cont.)

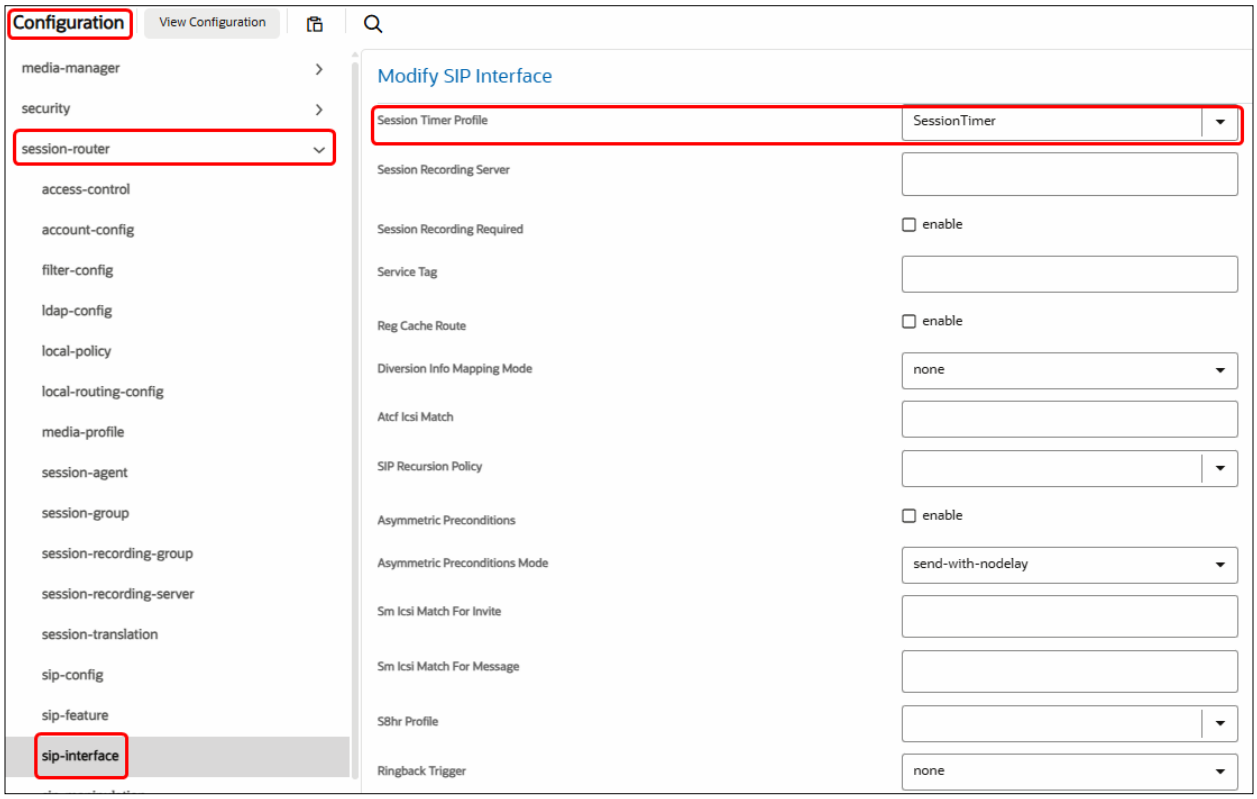

Figure 21.5: SIP Interface for PSTN &OnPrem PBX (Cont.)

- Create SIP interface towards Google CES by adding SIP Ports as shown below

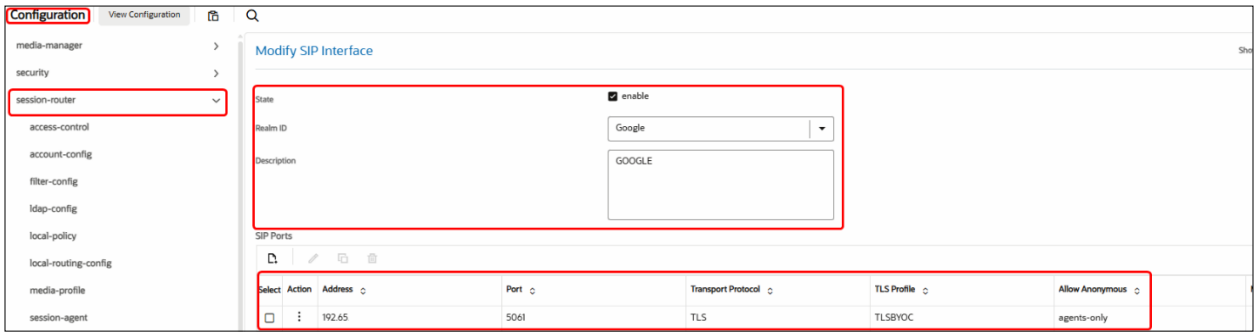

Figure 22: SIP Interface for Google CES

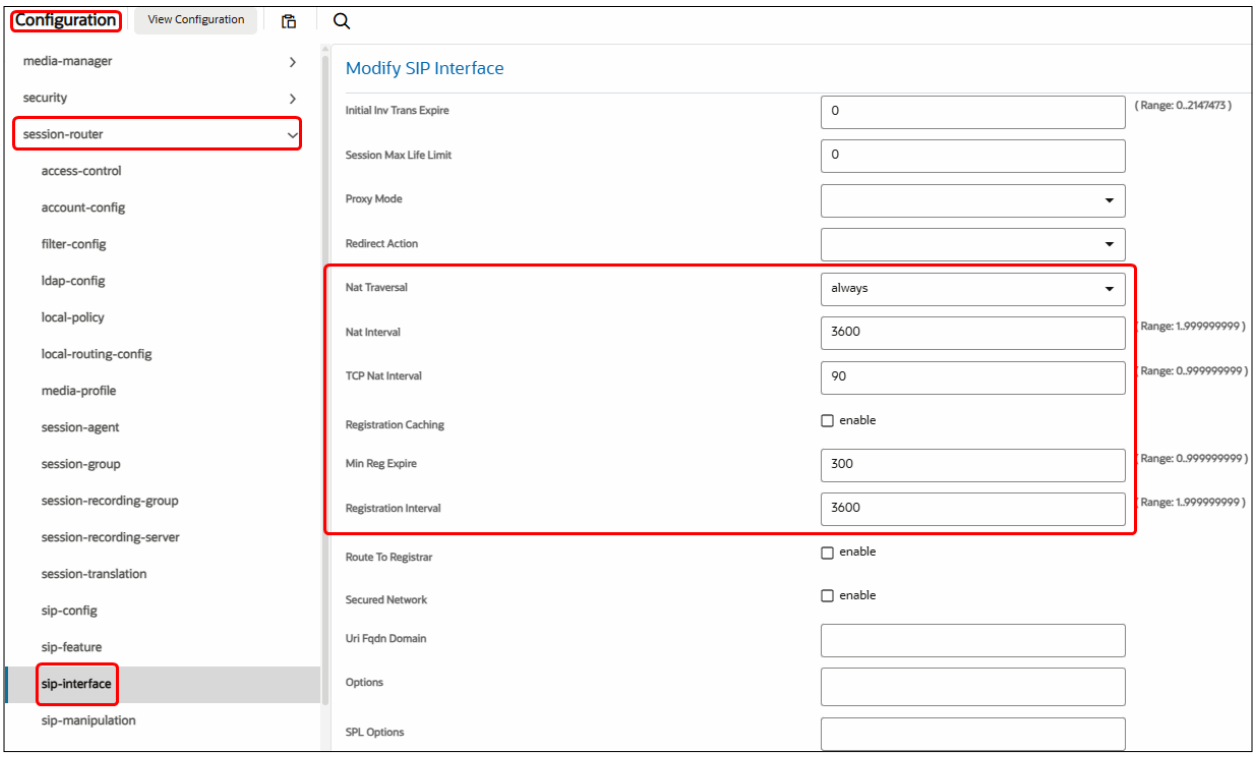

Figure 22.1: SIP Interface for Google CES (Cont.)

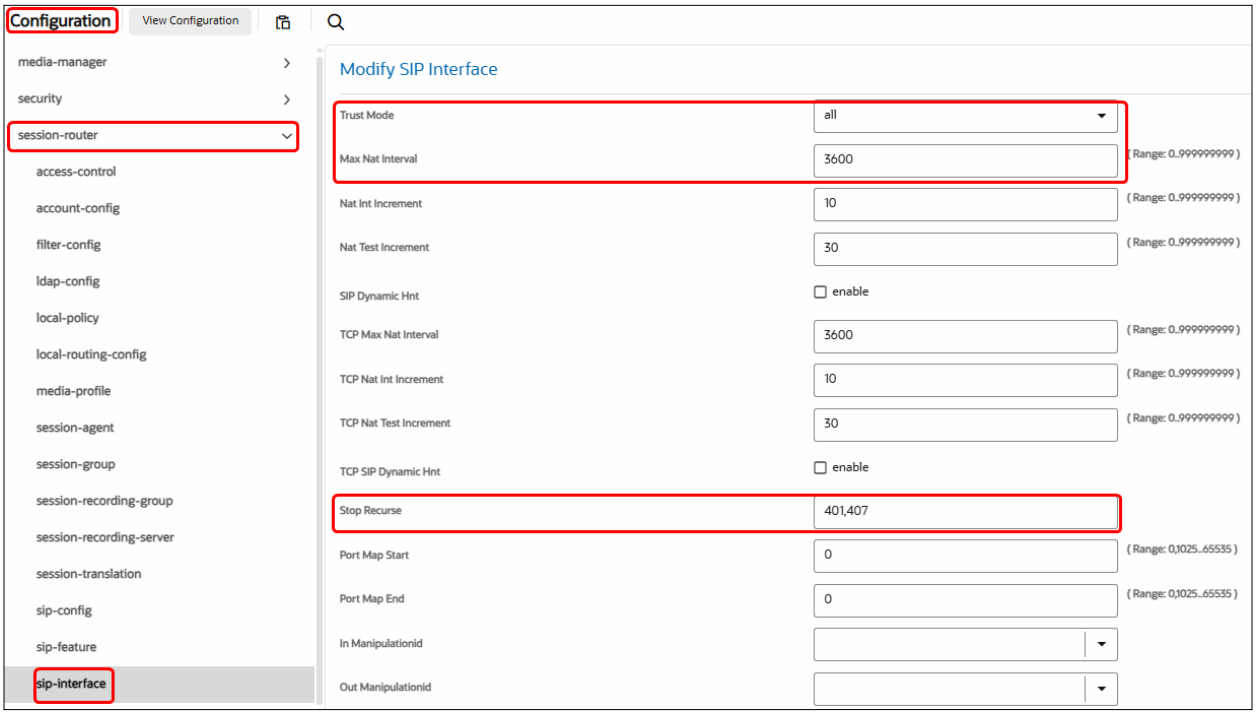

Figure 22.2: SIP Interface for Google CES (Cont.)

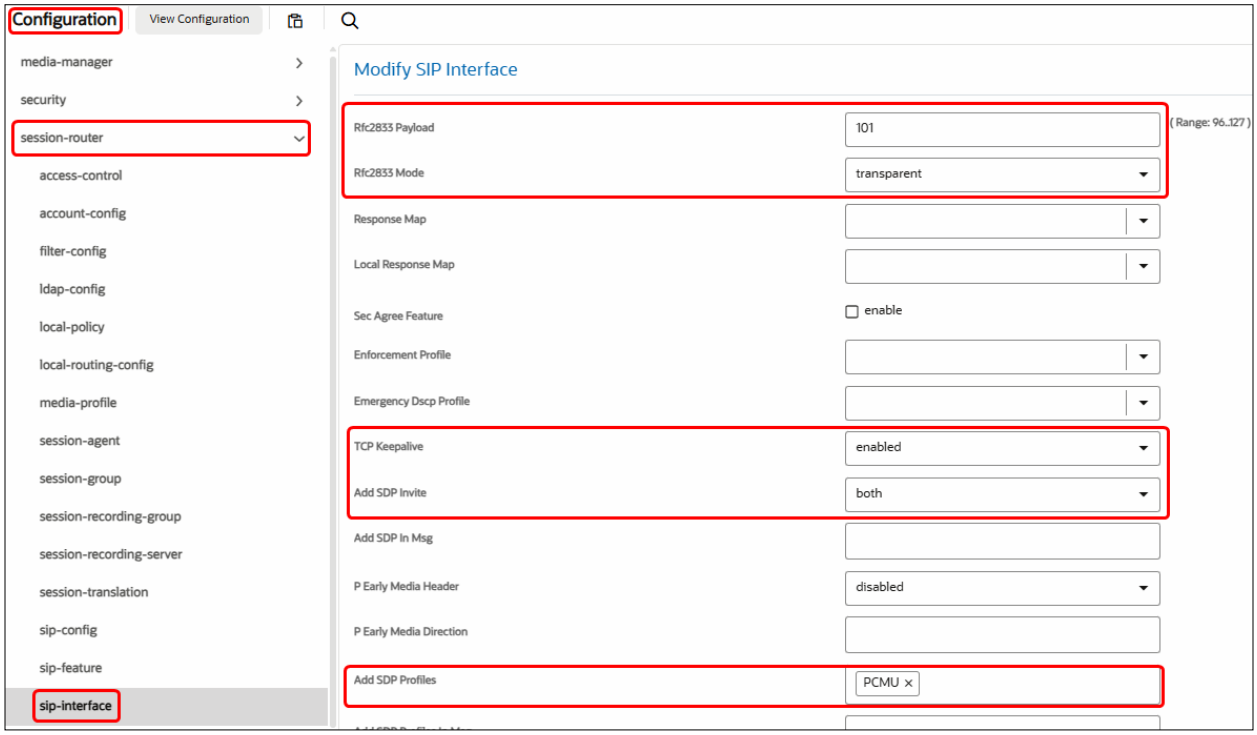

Figure 22.3: SIP Interface for Google CES (Cont.)

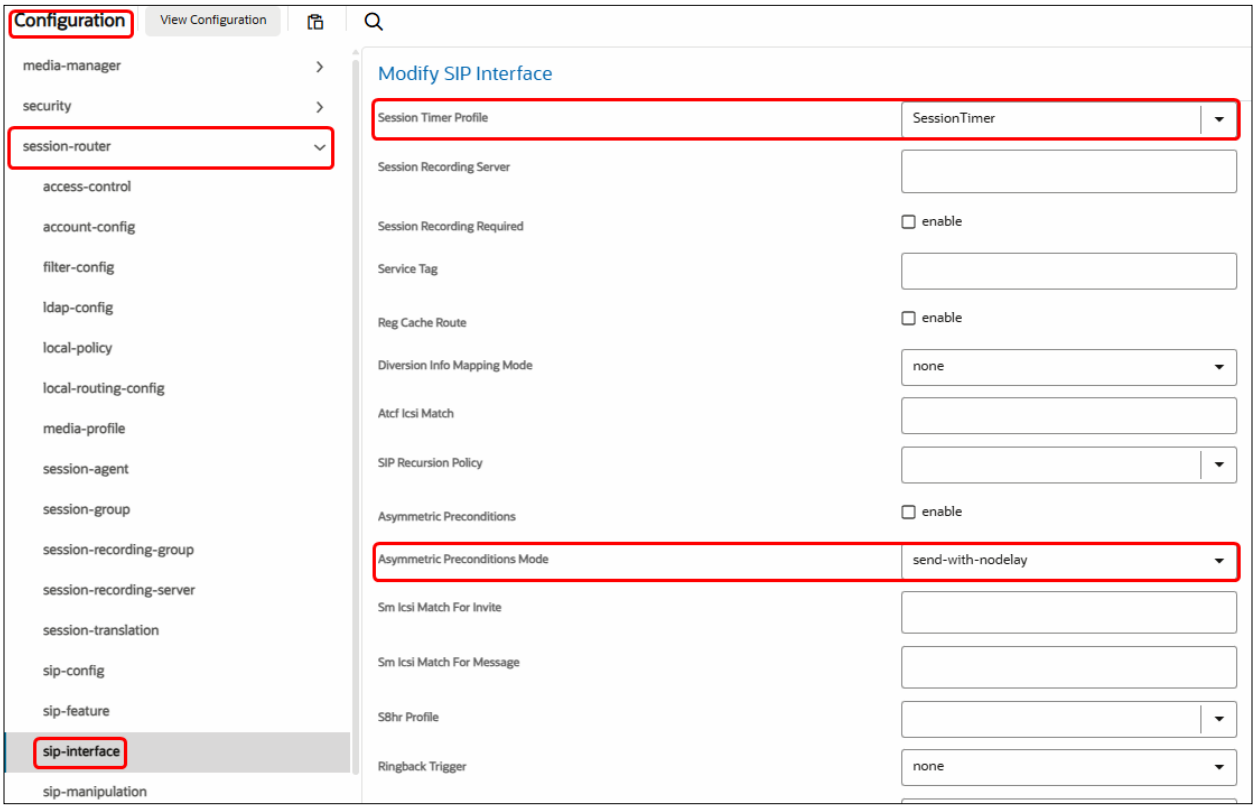

Figure 22.4: SIP Interface for Google CES (Cont.)

7.4.20 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 CES as shown below.

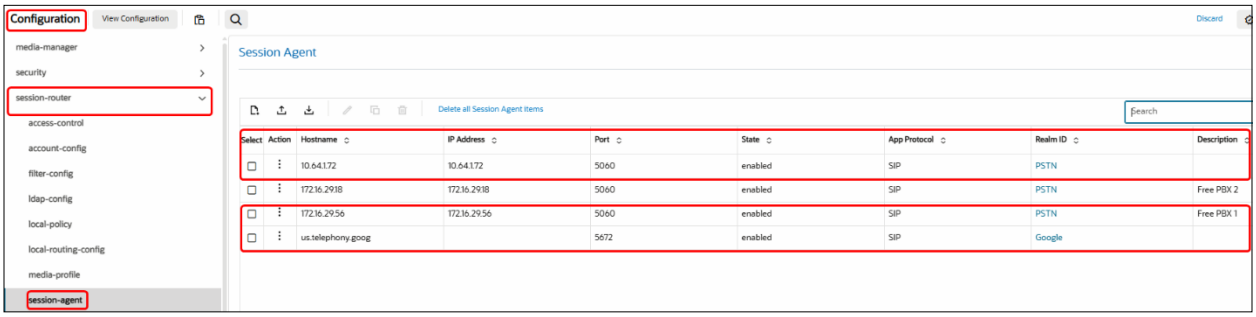

Figure 23: Session Agent.

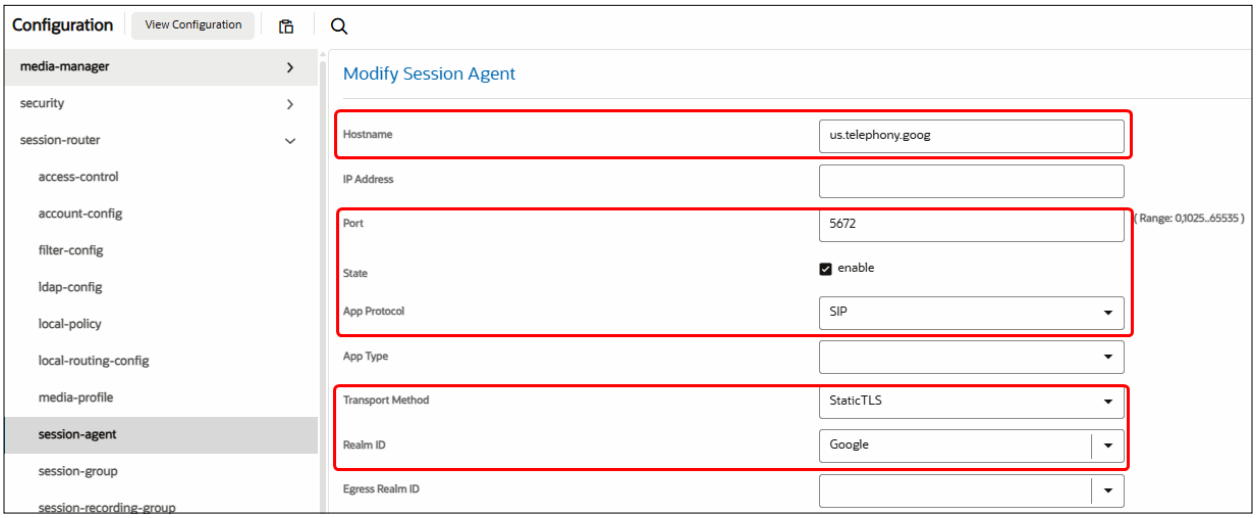

Figure 23.1: Session Agent for Google CES (Cont.)

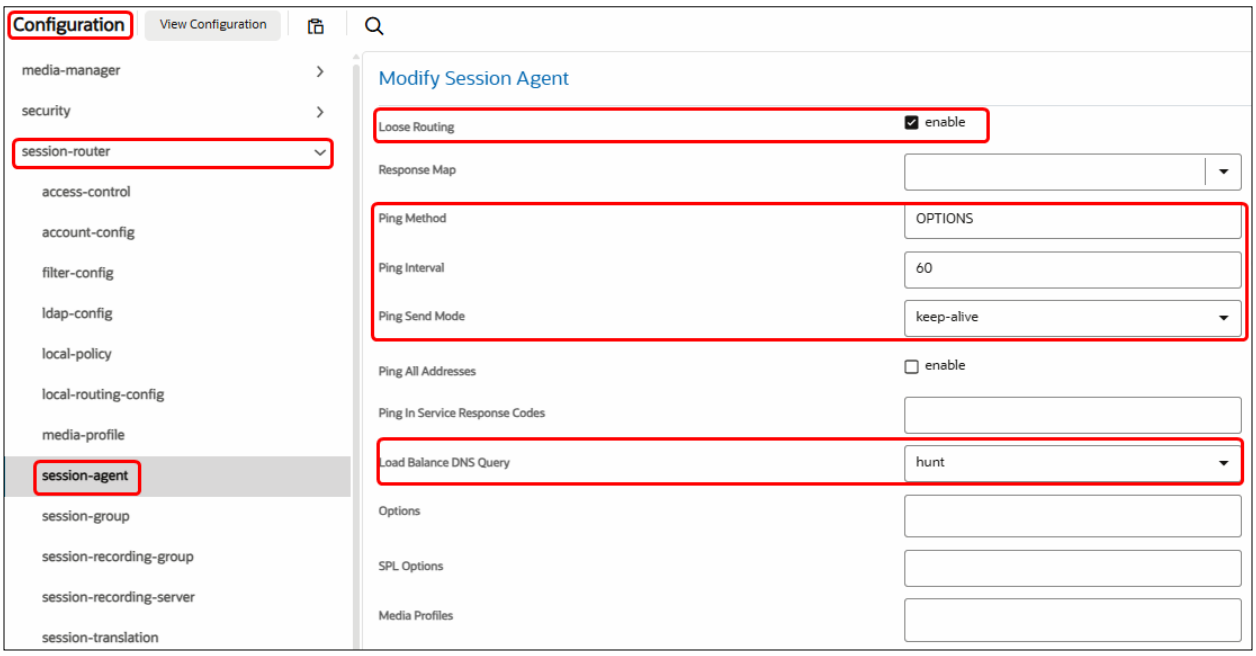

Figure 23.2: Session Agent for Google CES (Cont.)

Figure 23.3: Session Agent for Google CES (Cont.)

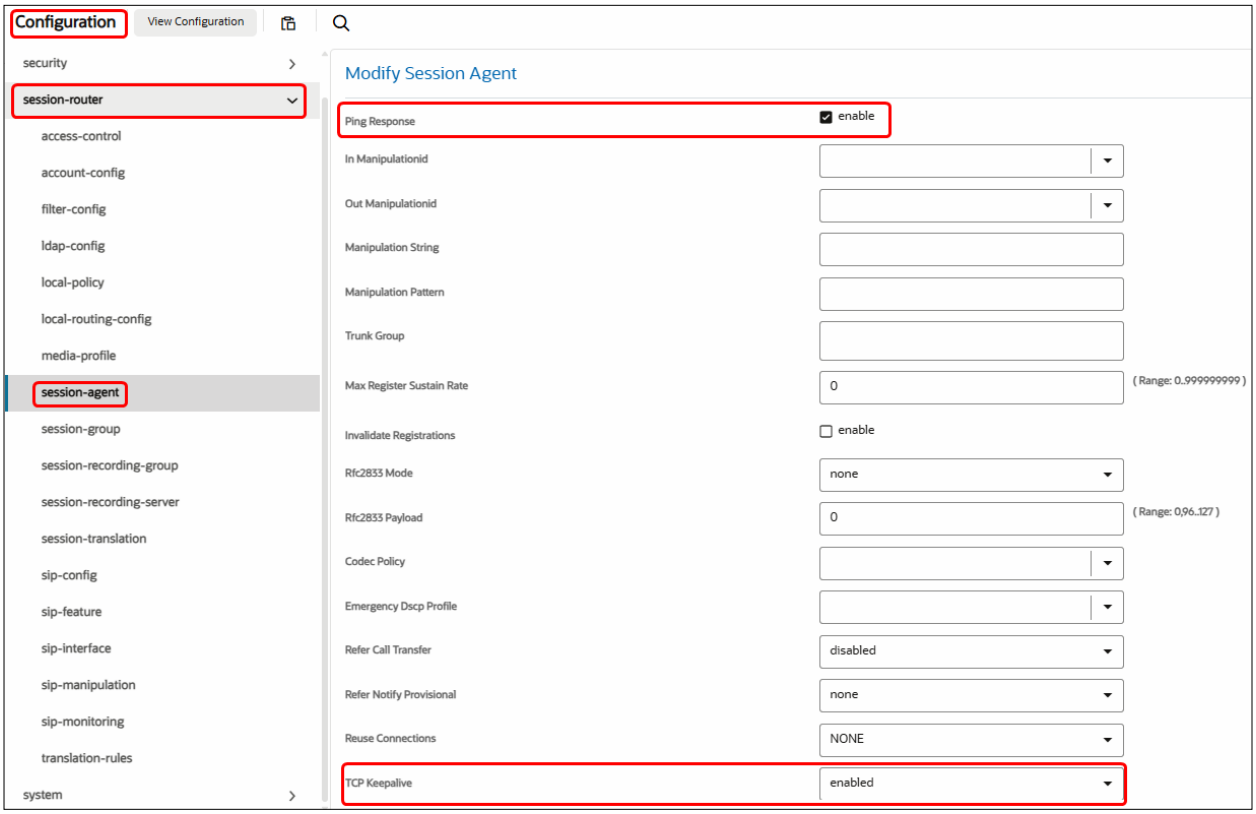

Figure 23.4: Session Agent for Google CES (Cont.)

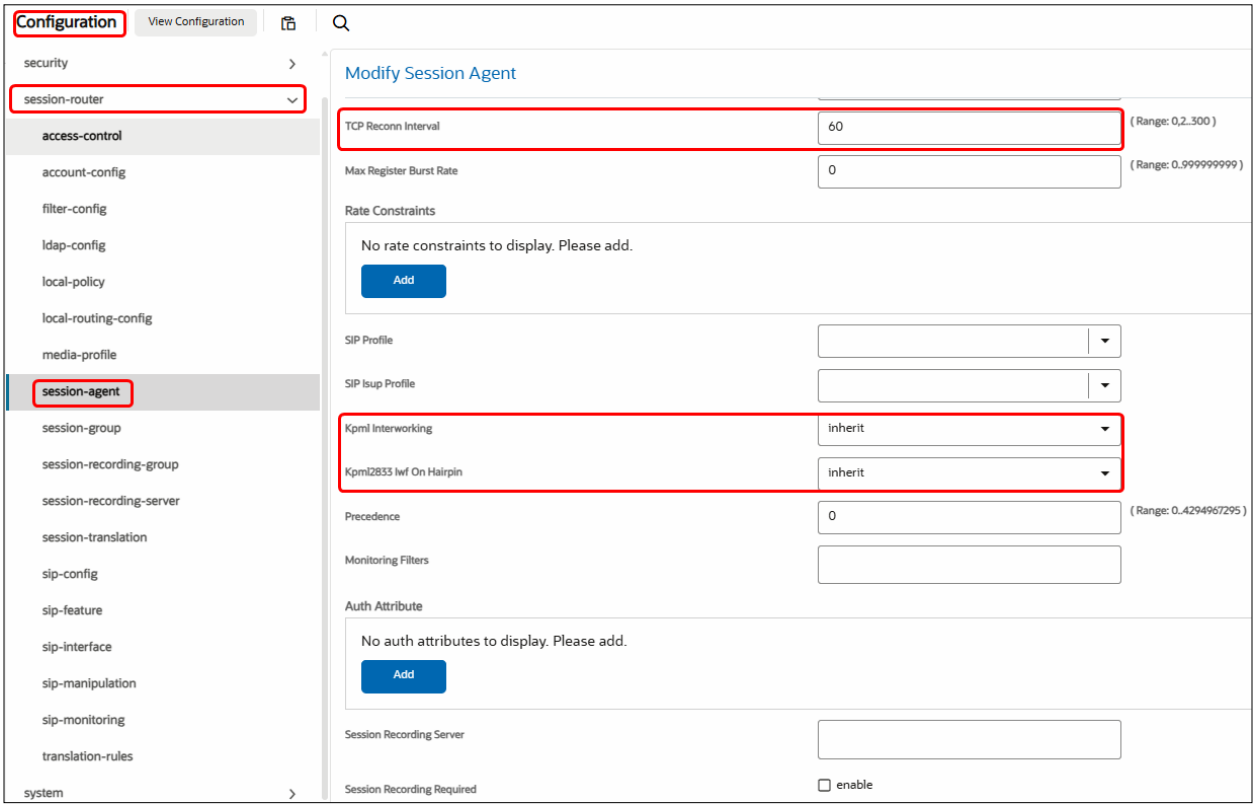

Figure 23.5: Session Agent for Google CES (Cont.)

- Configure the Session Agent for OnPrem PBX as shown below

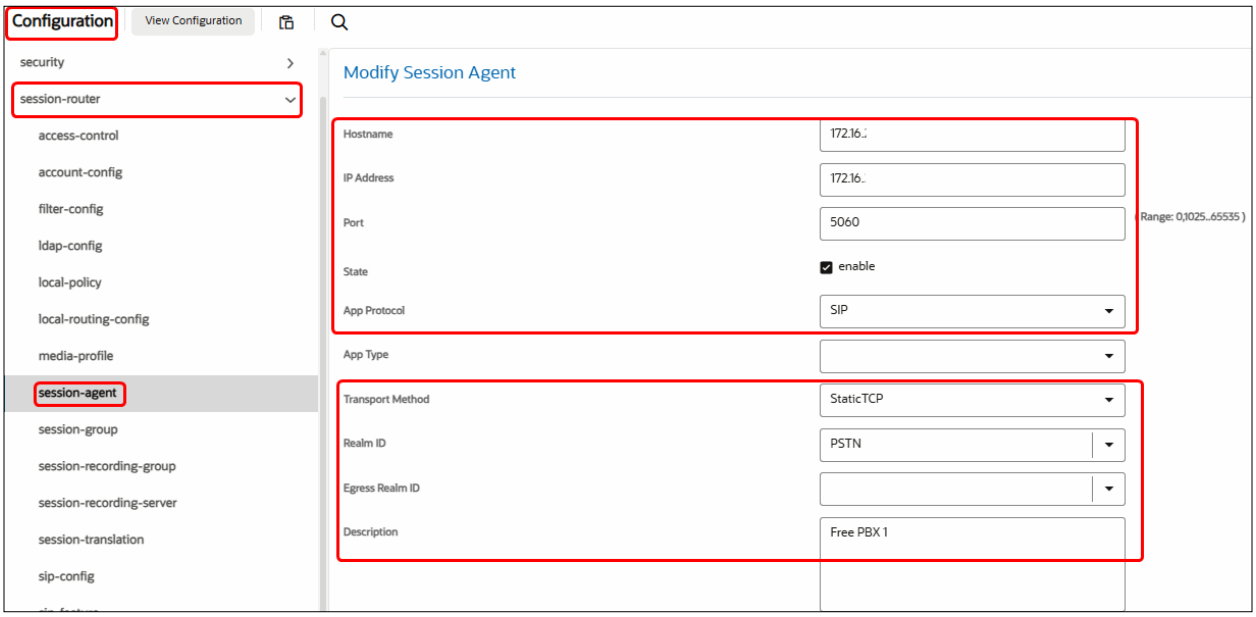

Figure 24: Session Agent for OnPrem PBX

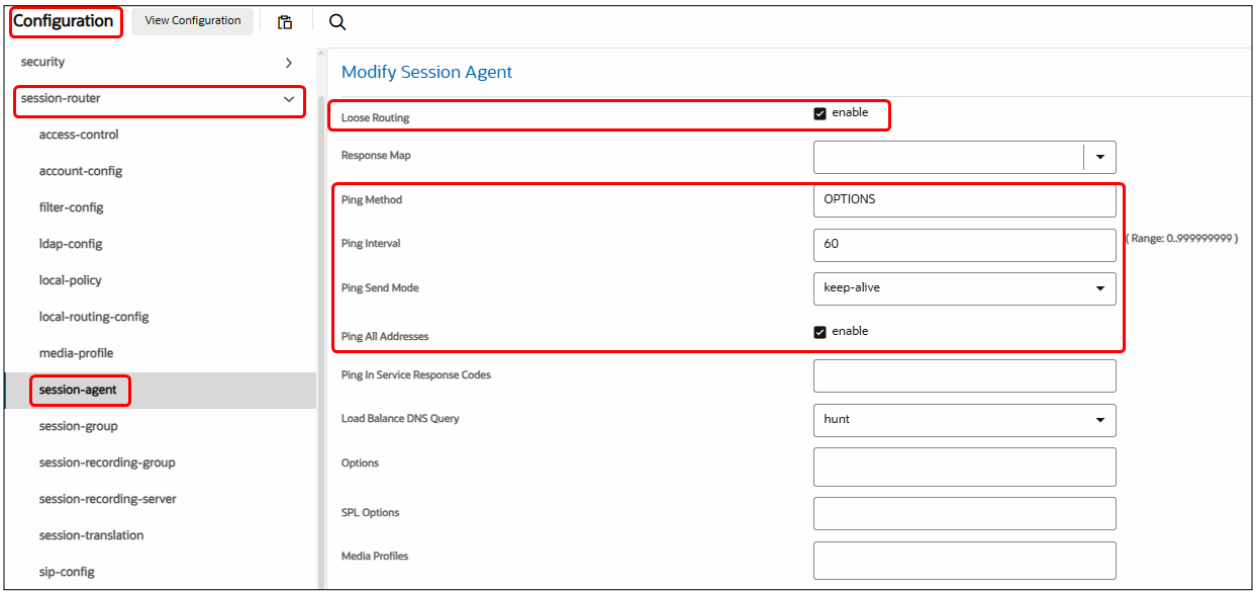

Figure 24.1: Session Agent for OnPrem PBX (Cont.)

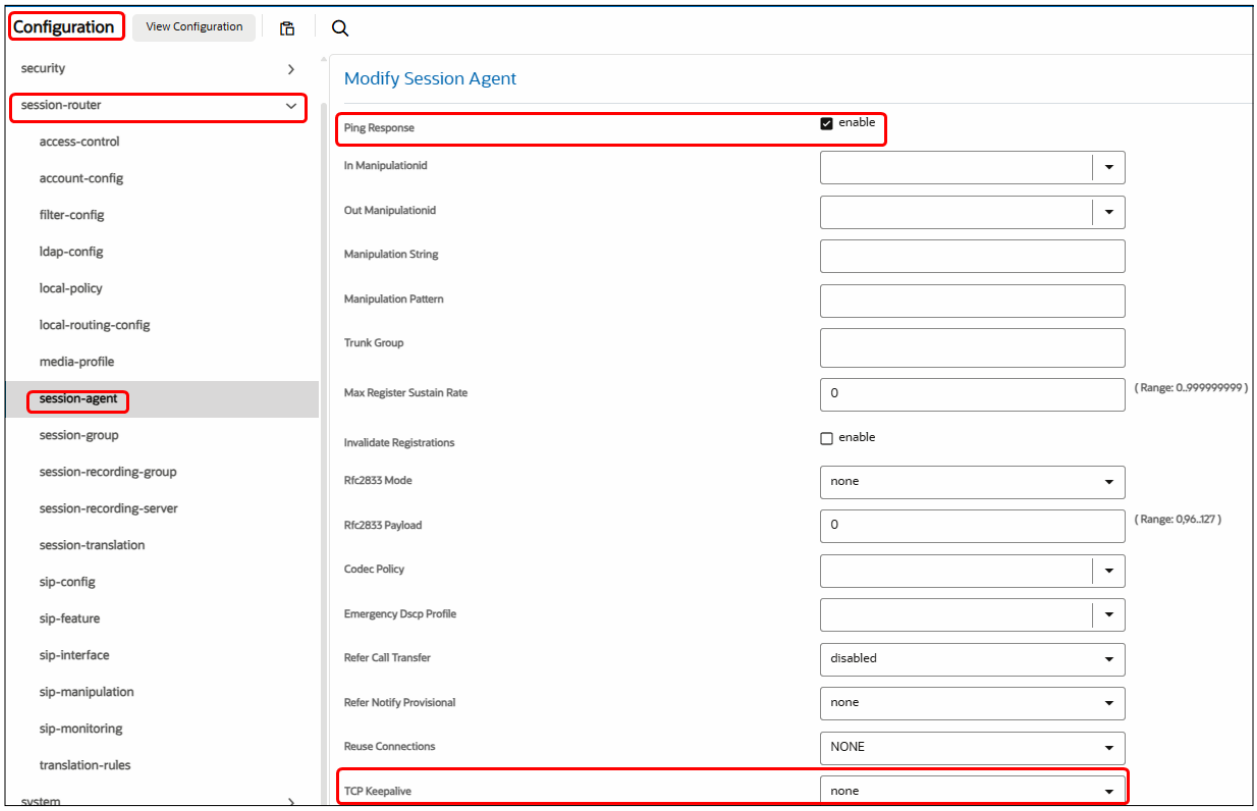

Figure 24.2: Session Agent for OnPrem PBX (Cont.)

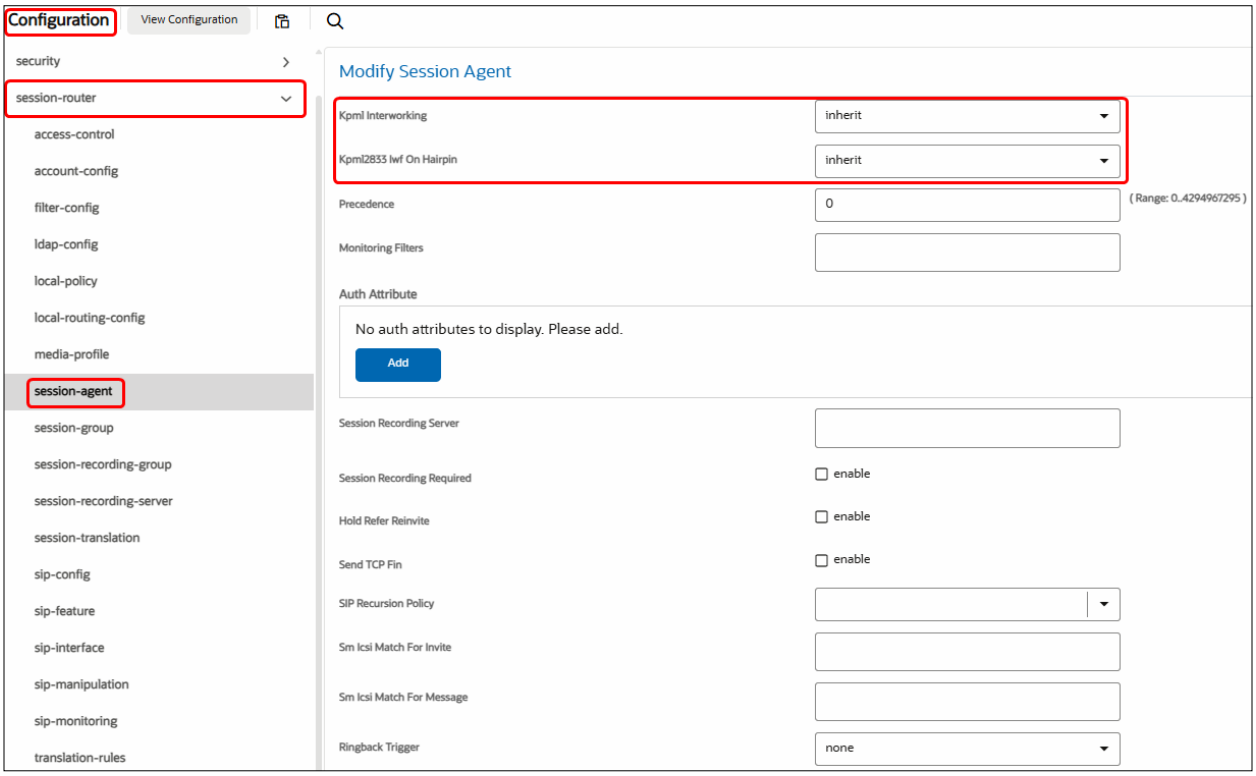

Figure 24.3: Session Agent for OnPrem PBX (Cont.)

- Configure the Session agent for PSTN Gateway as shown below.

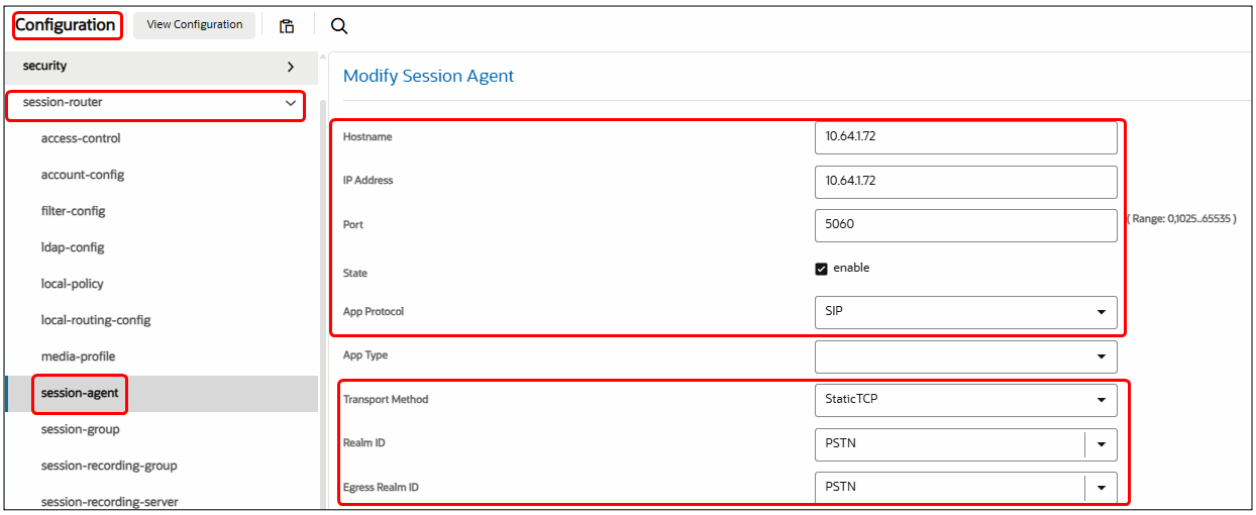

Figure 25: Session Agent for PSTN Gateway.

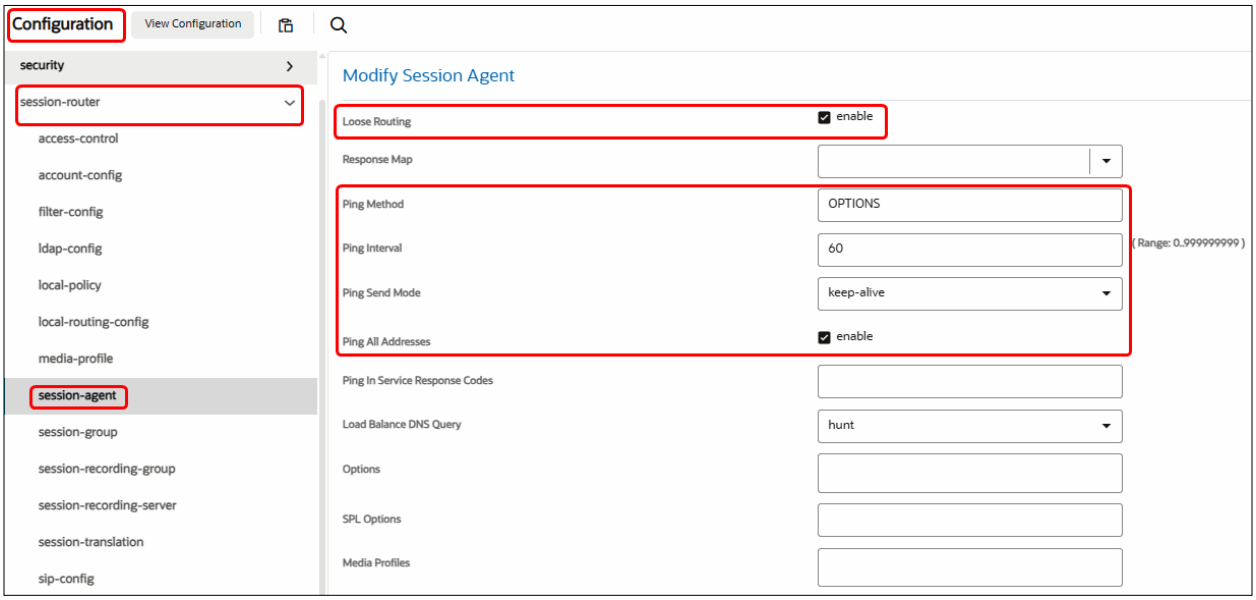

Figure 25.1: Session Agent for PSTN Gateway (Cont.)

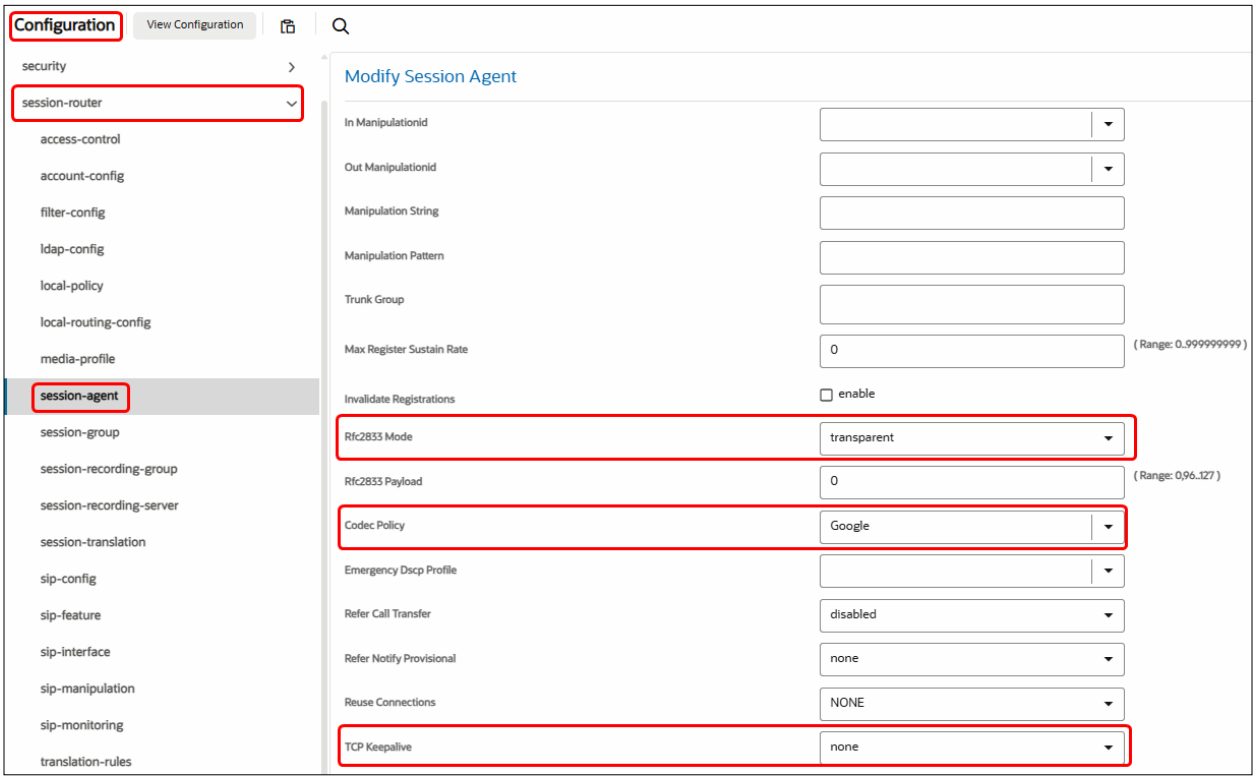

Figure 25.2: Session Agent for PSTN Gateway (Cont.)

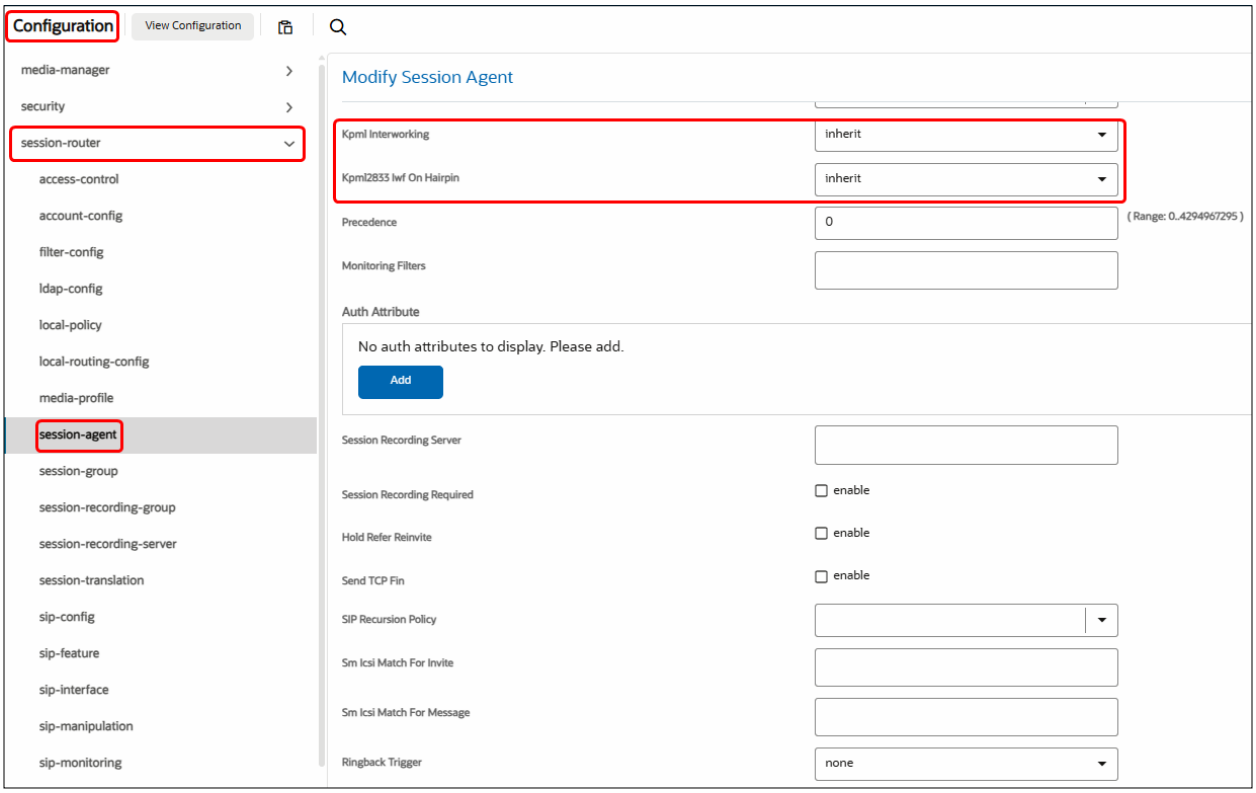

Figure 25.3: Session Agent for PSTN Gateway (Cont.)

7.4.21 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 CES, OnPrem PBX and PSTN Gateway as shown below.

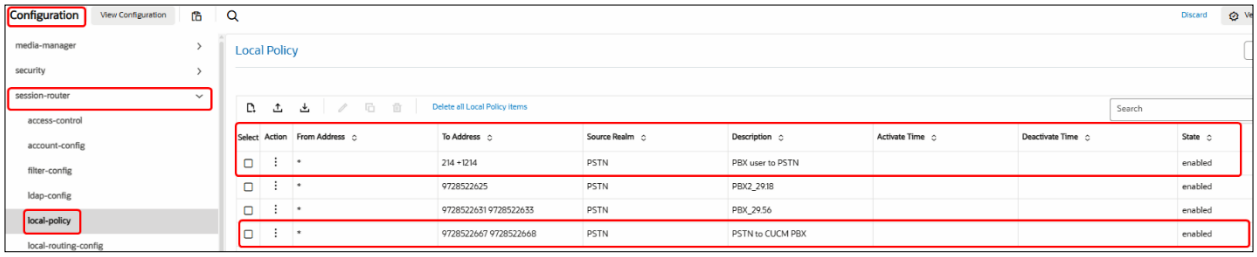

Figure 26: Local Policy

- Below Local Policy is used to route calls from OnPrem PBX to PSTN Gateway.

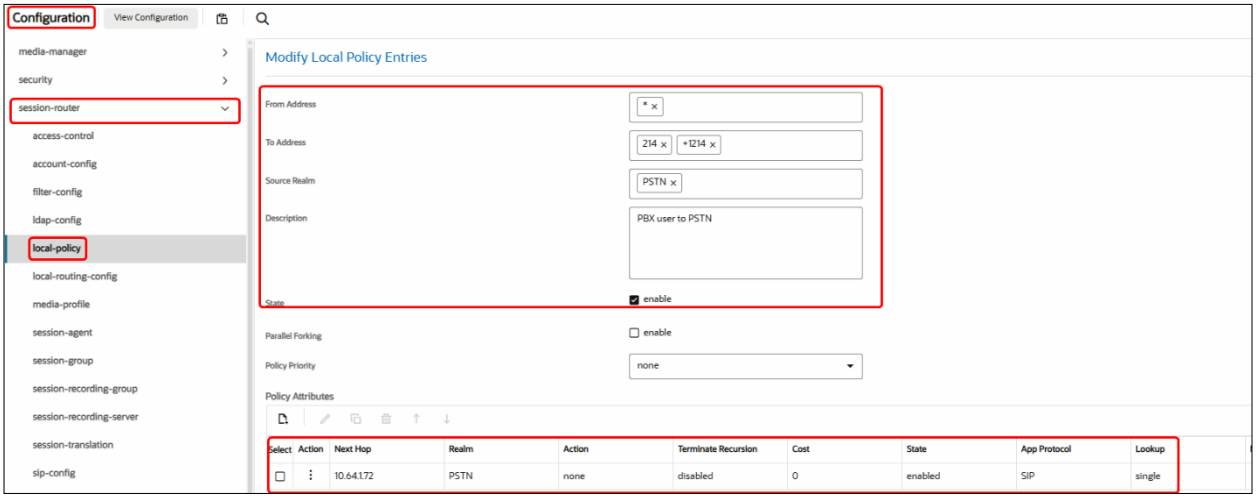

Figure 26.1: Local Policy routing from PBX to PSTN

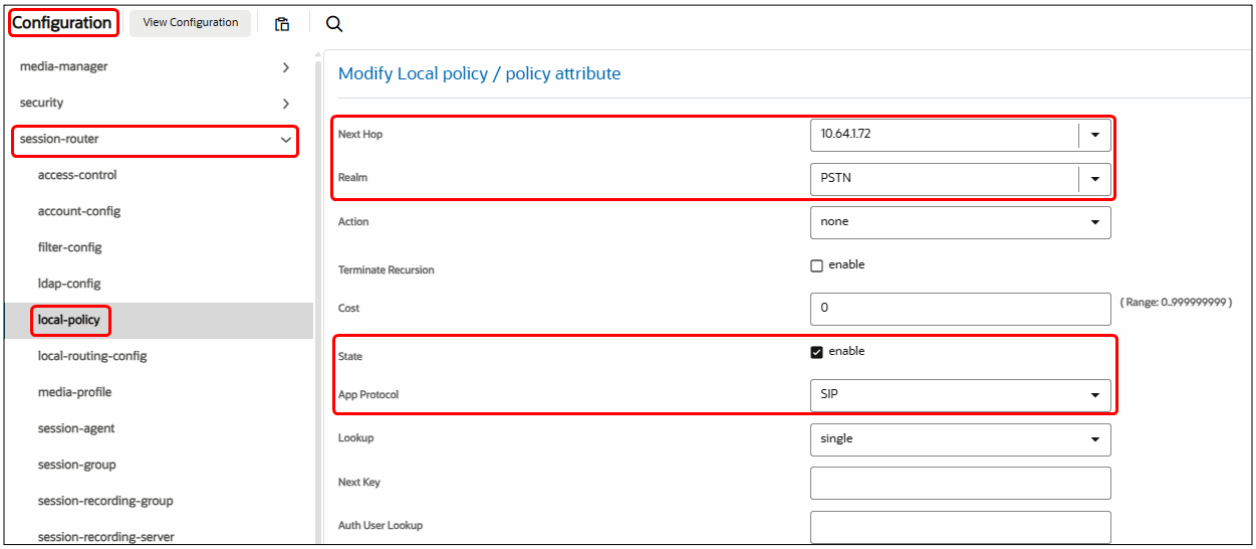

Figure 26.2: Local Policy routing Towards PSTN Gateway.

- Below Local Policy is used to route calls from PSTN Gateway to OnPrem PBX.

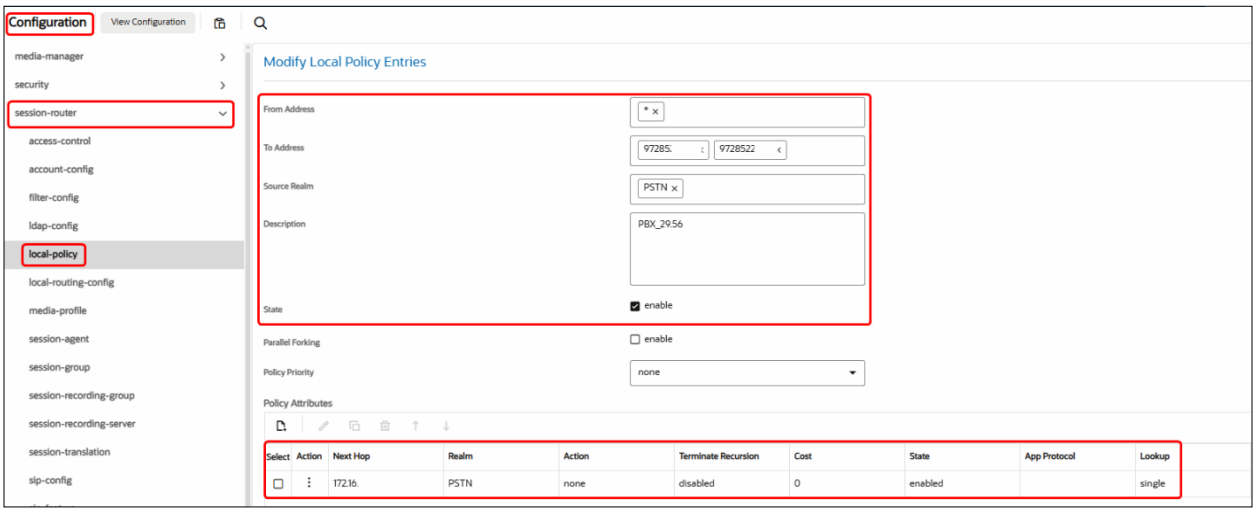

Figure 26.3: Local Policy routing Towards PBX Gateway.

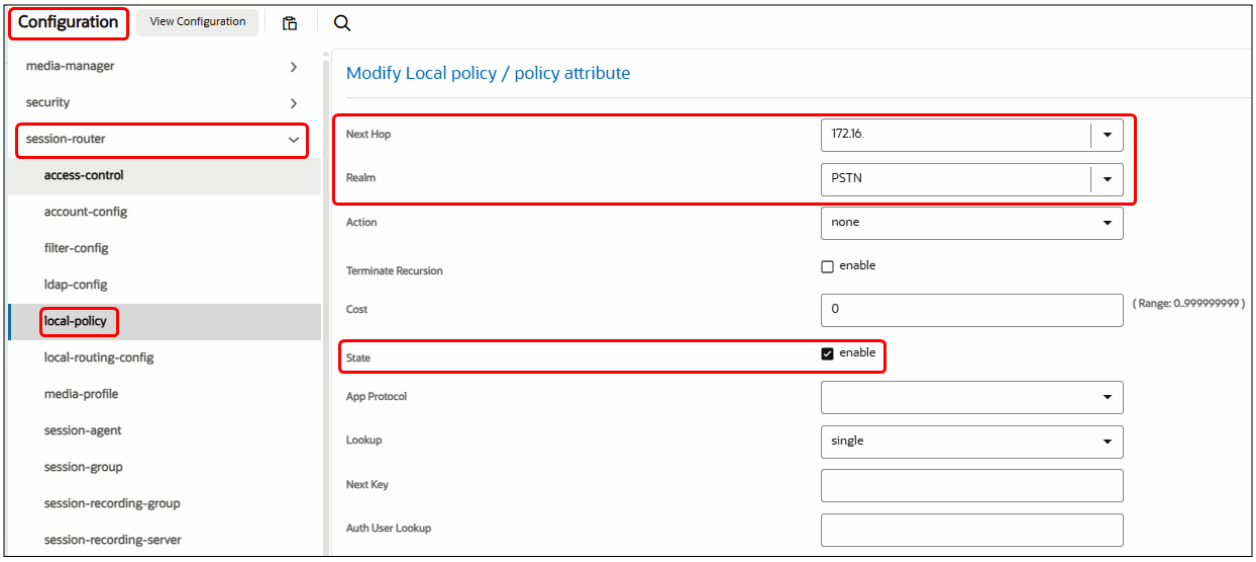

Figure 26.4: Local Policy routing Towards PBX Gateway.

7.4.22 SIP Manipulation

- Navigate to **Configuration → session-router → sip-manipulation**.
- Configure SIP manipulation towards Google CES as shown below.

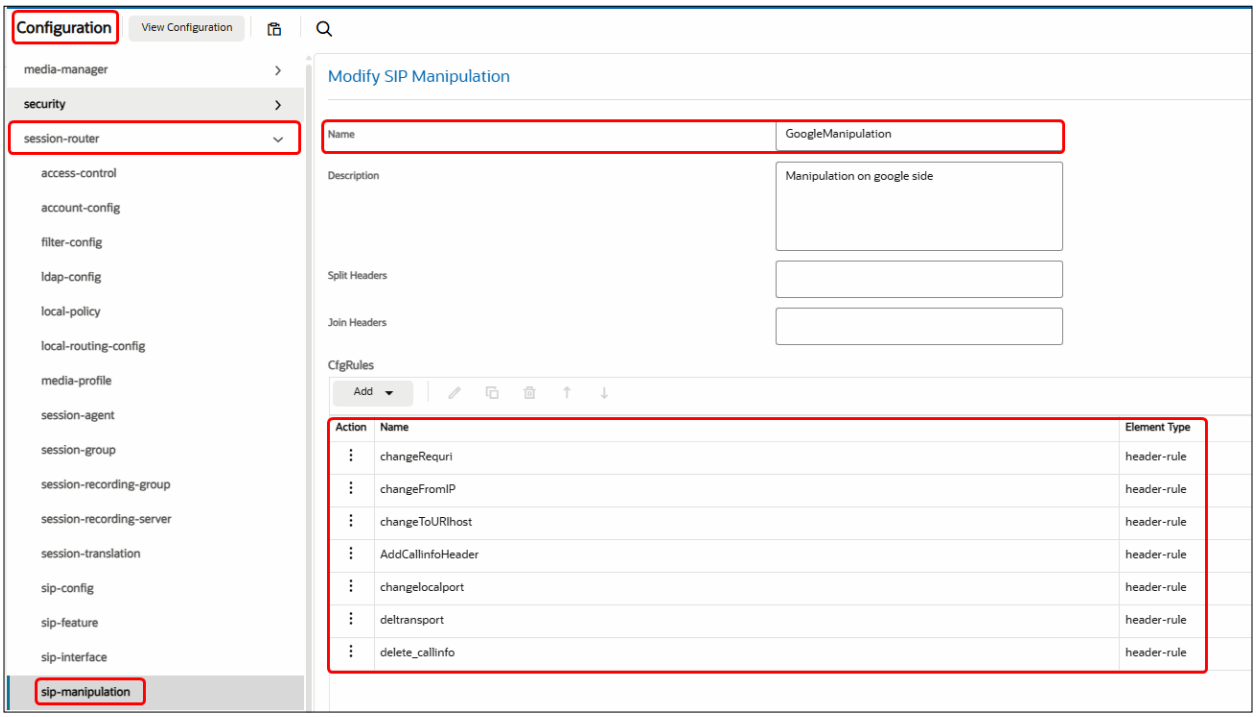

Figure 27: SIP Manipulation towards Google CES.

- Add a header-rule for Google CES

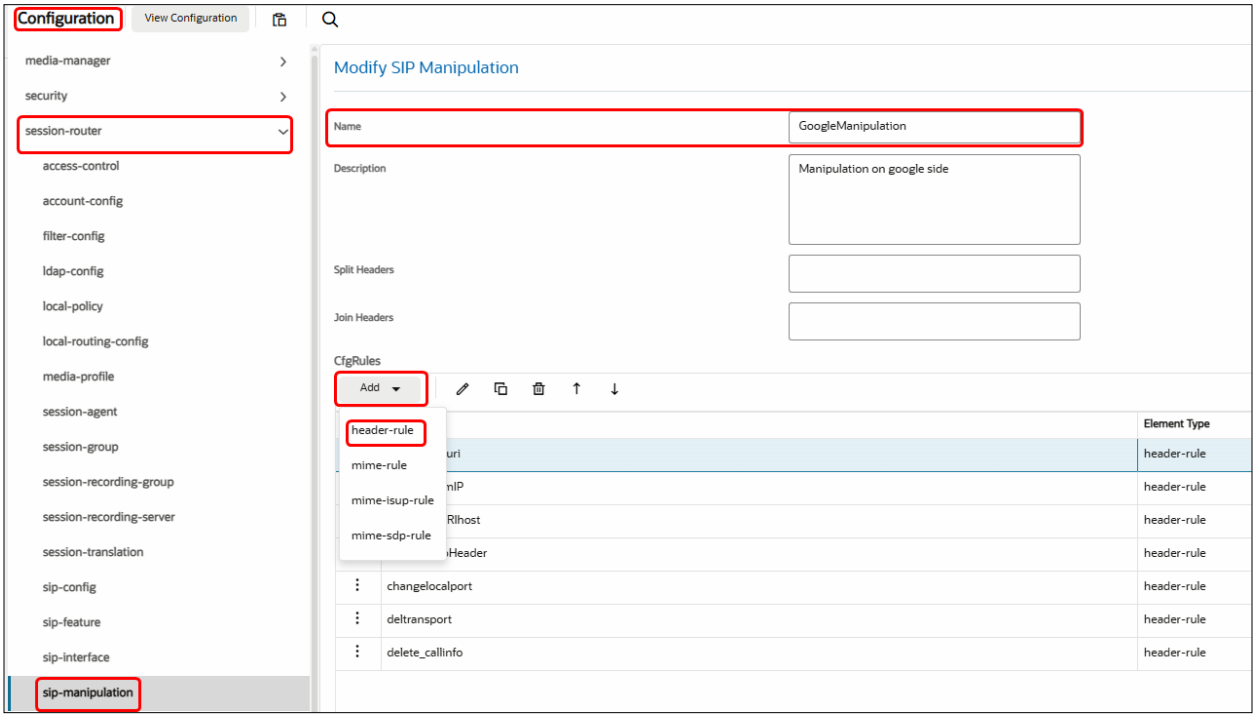

Figure 27.1: SIP Manipulation towards Google CES.

- Below header rule is created to change Request-URI host and user parts towards Google CES to **us.telephony.goog:5672** and **+1361880XXXX**.

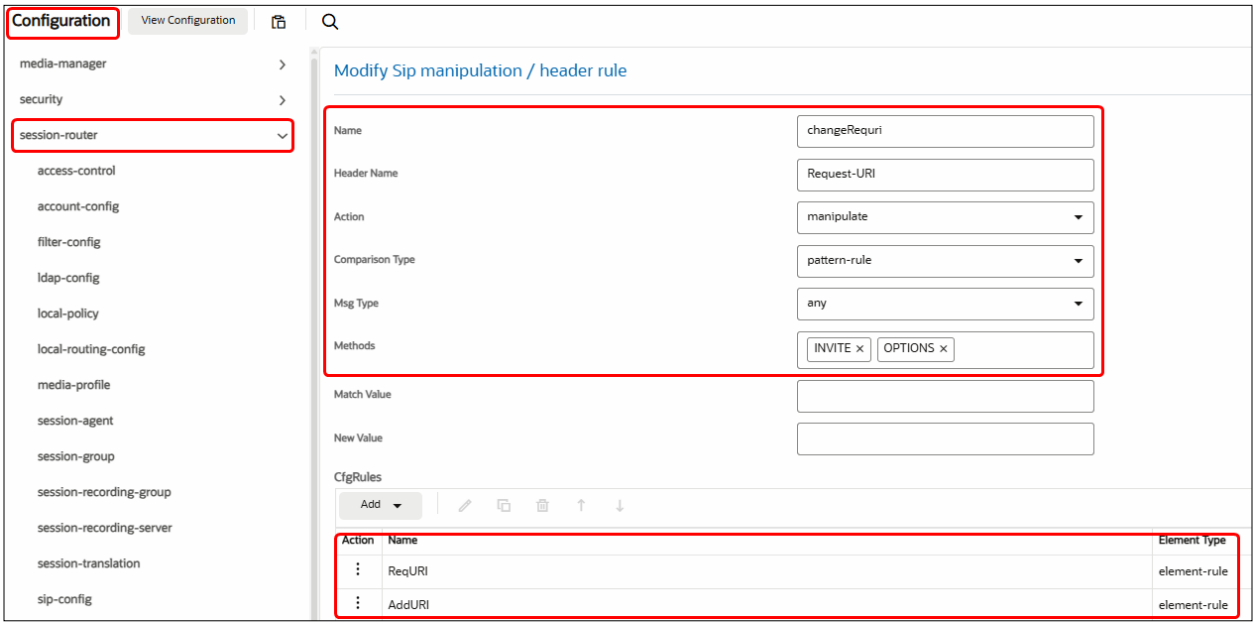

Figure 27.2: SIP Manipulation towards Google CES- To change Request URI

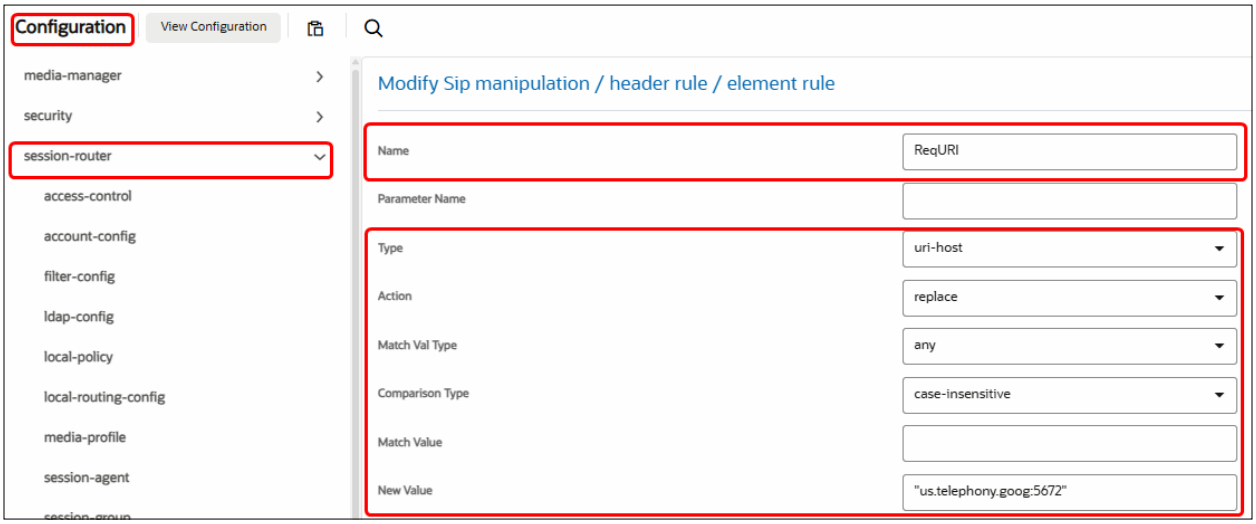

Figure 27.3: SIP Manipulation towards Google CES – To change Request URI-host.

Figure 27.4: SIP Manipulation towards Google CES – Change Request URI-user.

- Below header rule is created to change FROM header IP address towards Google CES to IP address of Oracle SBC.

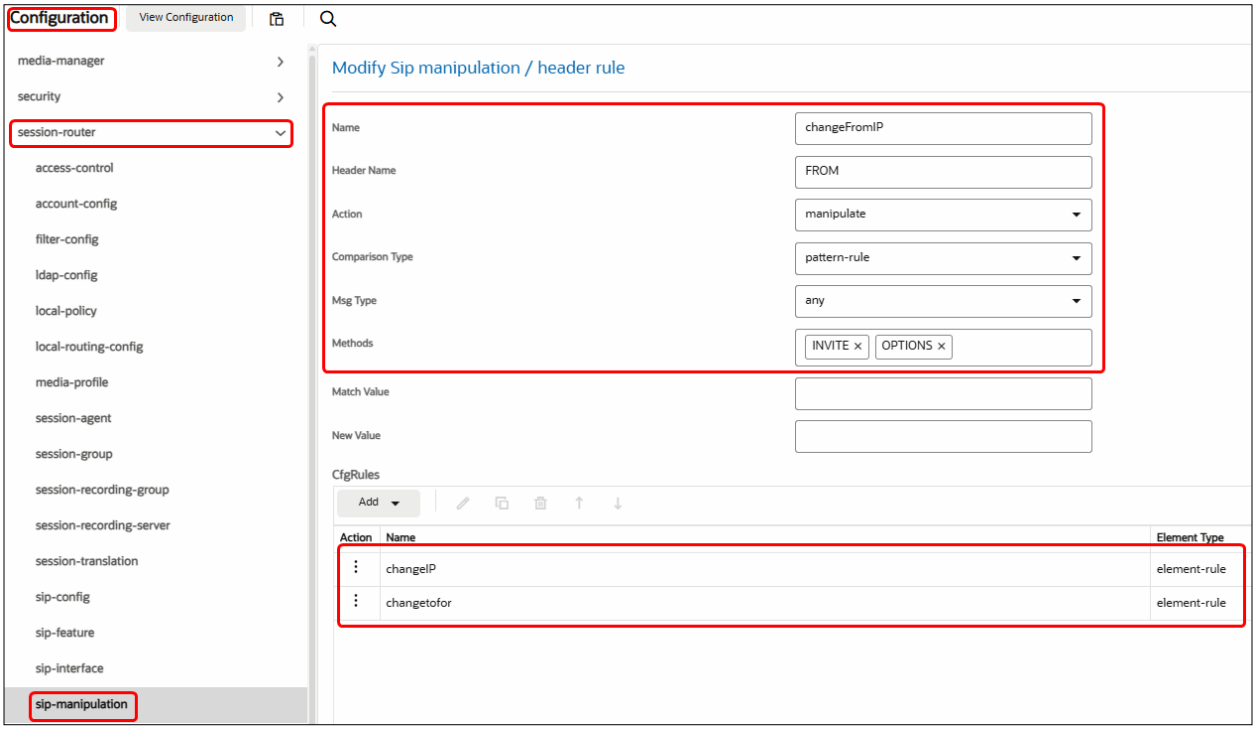

Figure 27.5: SIP Manipulation towards Google CES – Change FROM header.

Figure 27.6: SIP Manipulation towards Google CES – Change FROM header uri-host.

- Match value : To:sips:"us.telephony.goog":5672;trasnaport=tls
- New Value : To:<sips:"+1361880XXXX@us.telephony.goog":5672;trasnaport=tls>

Figure 27.7: SIP Manipulation towards Google CES – Change FROM header uri-user.

- Below header rule is created to change TO header host part towards Google CES to IP address of Google CES and user part with Google CES DID

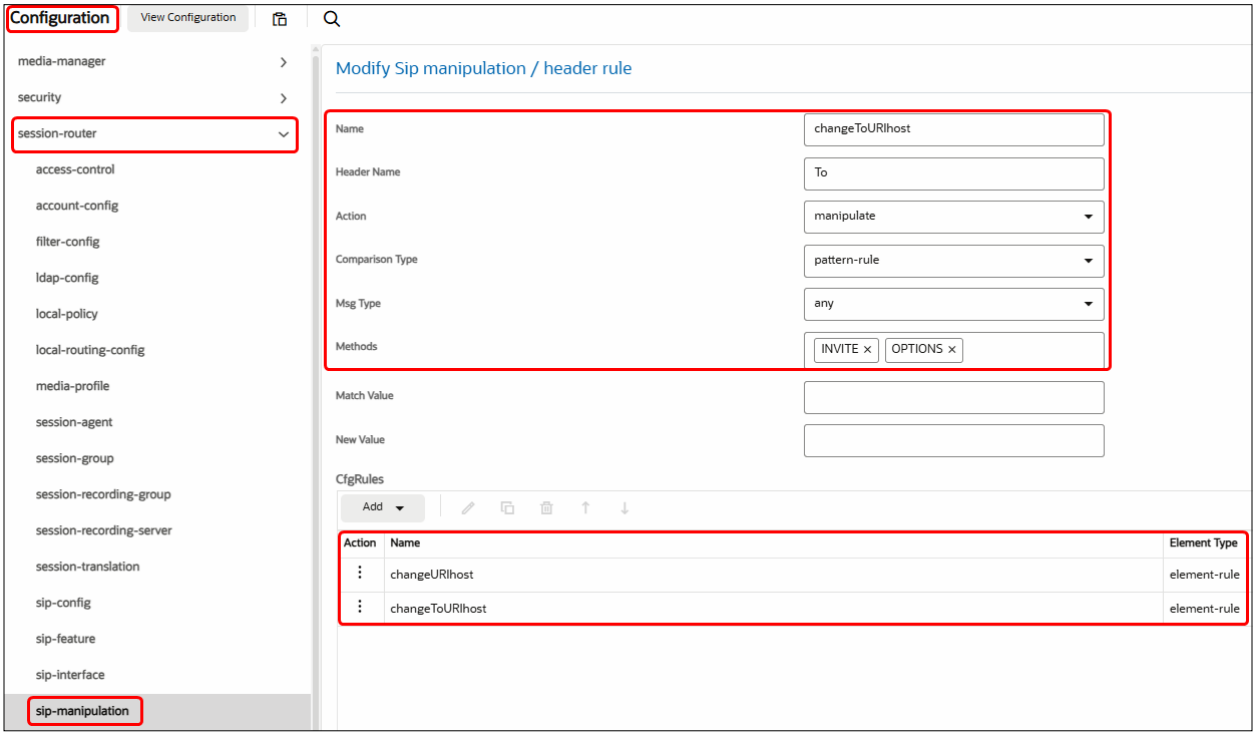

Figure 27.8: SIP Manipulation towards Google CES – Change TO header.

Figure 27.9: SIP Manipulation towards Google CES – Change TO header uri-host.

Figure 27.10: SIP Manipulation towards Google CES – Change TO header uri-user.

- Below header rule is created to add Call-Info header towards Google CES with the Dialog Flow API request along with the Conversation ID.
- **Conversation on the Fly** is set to True in Google CES using REST API. Conversation ID is randomly generated by Oracle SBC for each call.
- New Value is set to **"<**http://dialogflow.googleapis.com/v2beta1/projects/CES-389811/conversations/OR_"+$CALL_ID.$0+">;purpose=Goog-ContactCenter-Conversation**"**

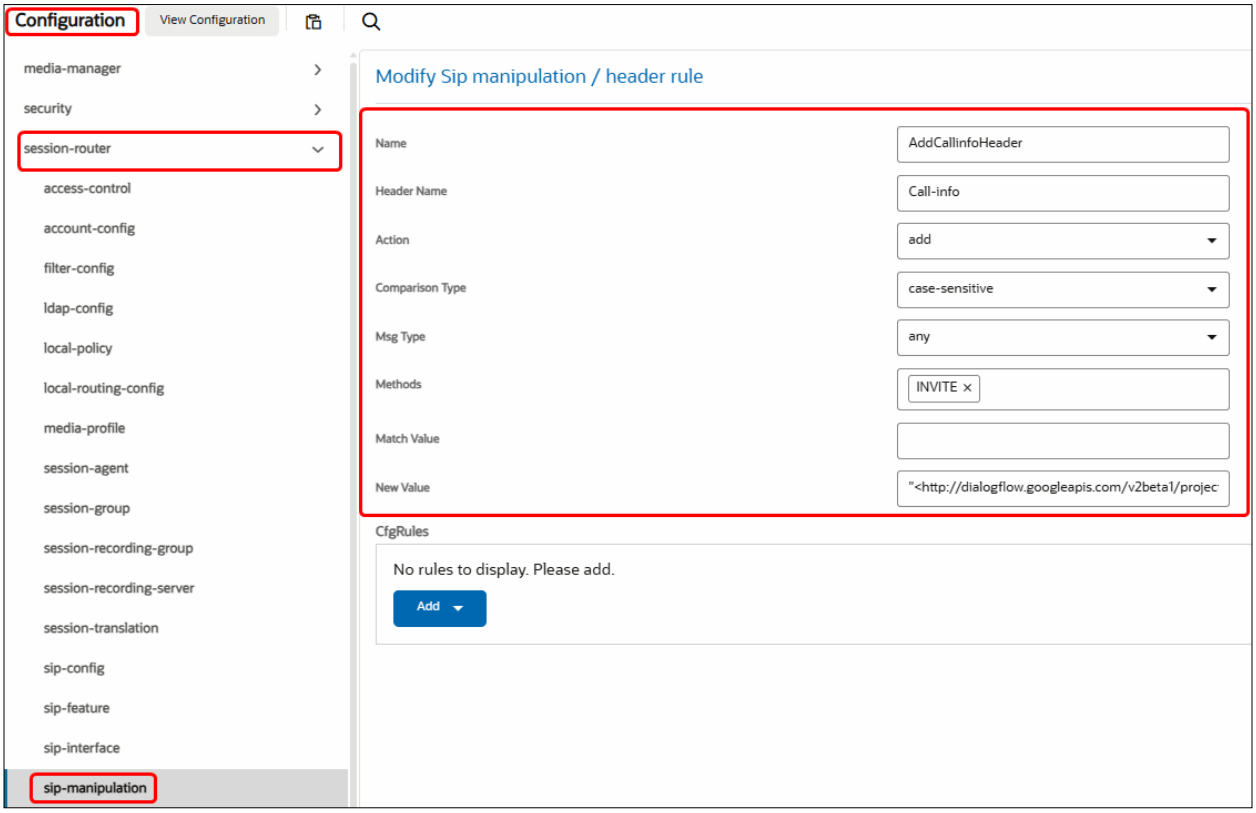

Figure 27.11: SIP Manipulation towards Google CES – Add Call-Info.

- **Participation Label:**

- The transcript recording files stored in the Google CES bucket include two participant roles "HUMAN_AGENT" and "END_USER".
- To map the participant roles to the transcripts generated, Google uses the participant labels provided in the call-info header. Use the below rule only if Participant labels are required in your setup.
- Sample call-info header with participant roles:
  - Call-info: "<http://dialogflow.googleapis.com/v2beta1/projects/CES-389811/conversations/OR_"+$CALL_ID.$0+"**?roles=HUMAN_AGENT,END_USER**>;purpose=Goog-ContactCenter-Conversation"

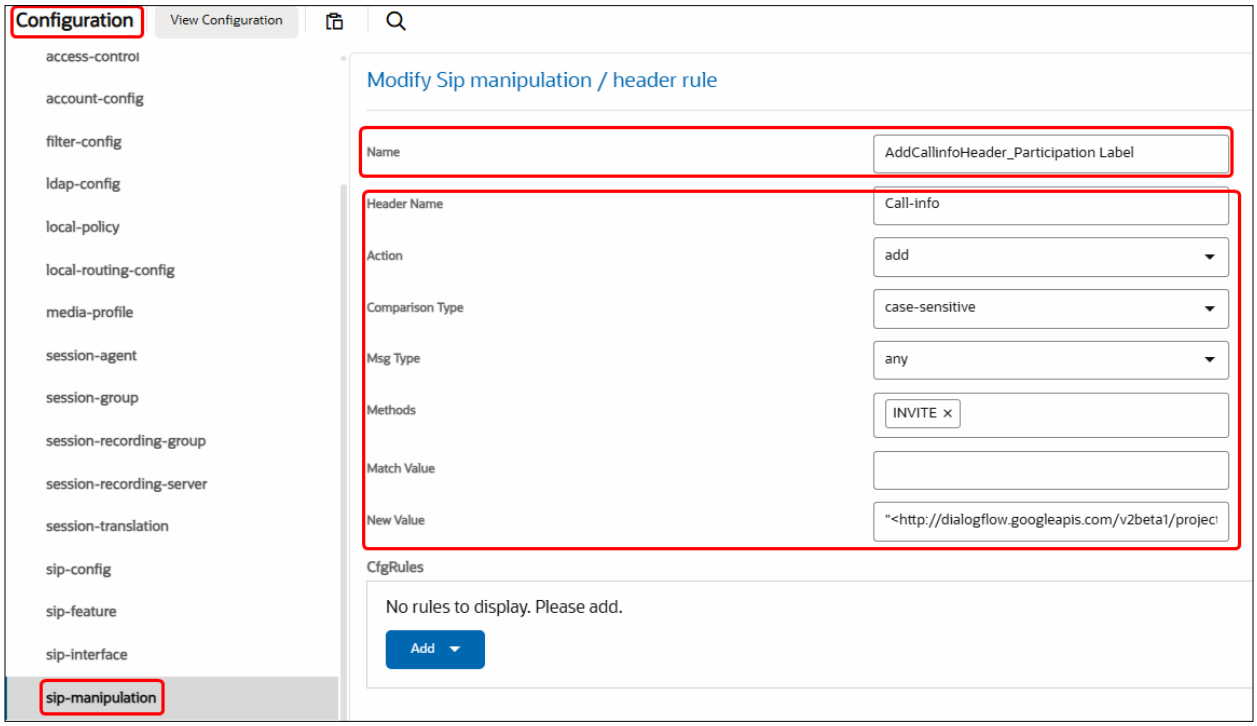

Figure 27.12: SIP Manipulation towards Google CES – Add Call-Info for Participation Label.

- Below header rule is created to delete the Google CES FQDN generated by Oracle SBC during the creation of Conversation ID (this rule is applied only when Conversation on the Fly is set to True in Google CES).

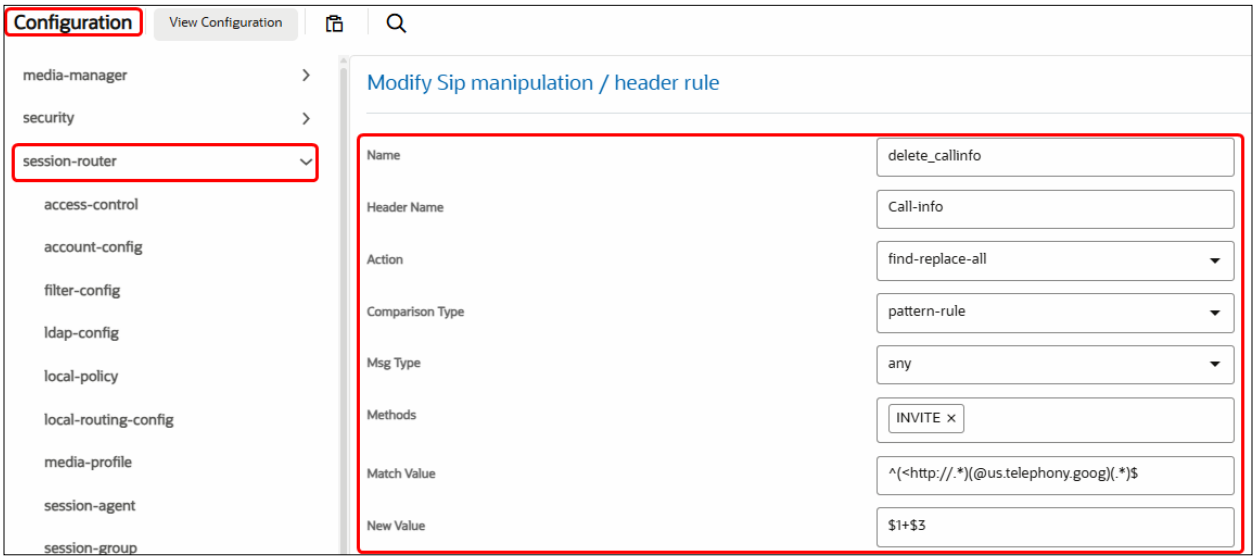

Figure 27.13: SIP Manipulation towards Google CES – Delete Call-Info host IP address.

- Below header rule is created to change the port number in the Request URI towards Google CES.

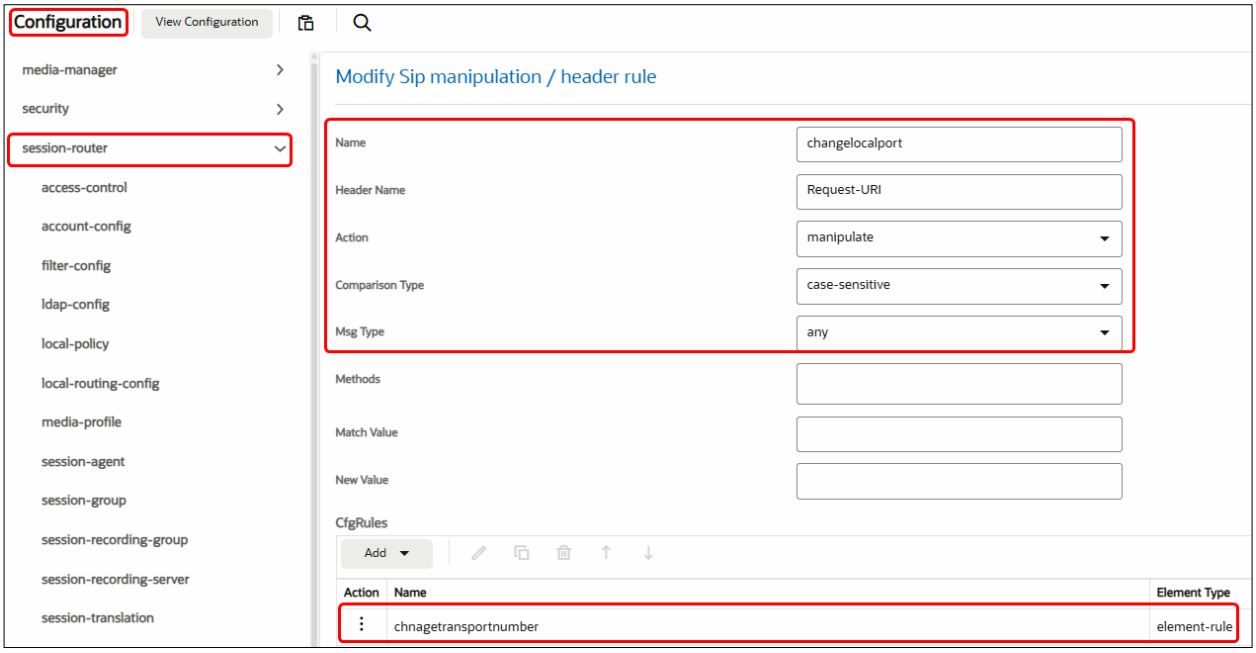

Figure 27.14: SIP Manipulation towards Google CES – Change Request URI Port number.

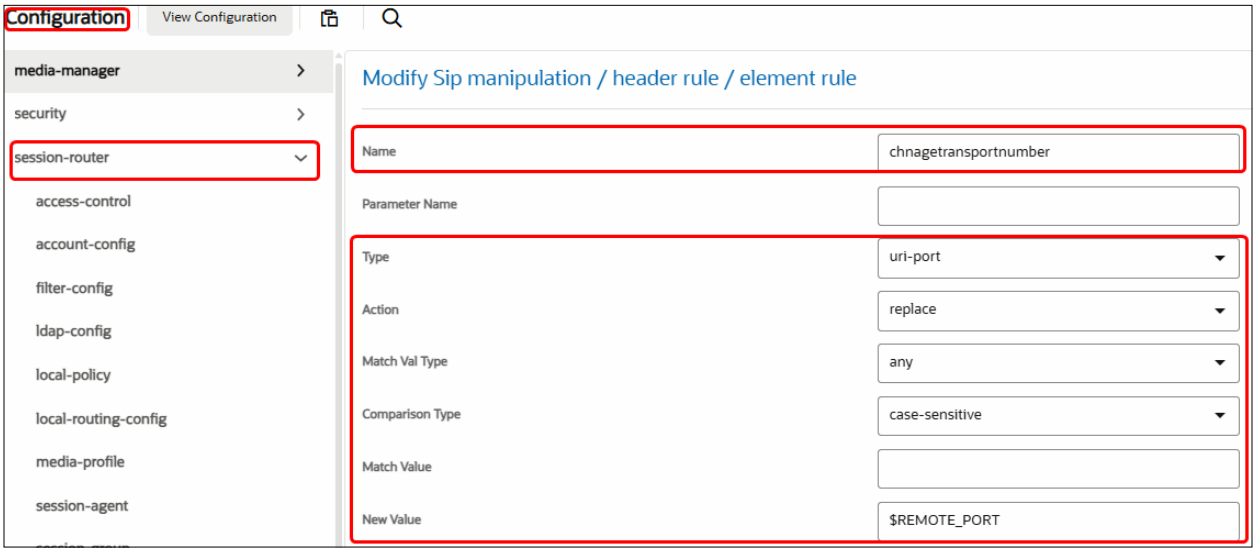

Figure 27.15: SIP Manipulation towards Google CES – Change Request URI Port number.

- Below header rule is created to delete the transport parameter in the Request URI towards Google CES.

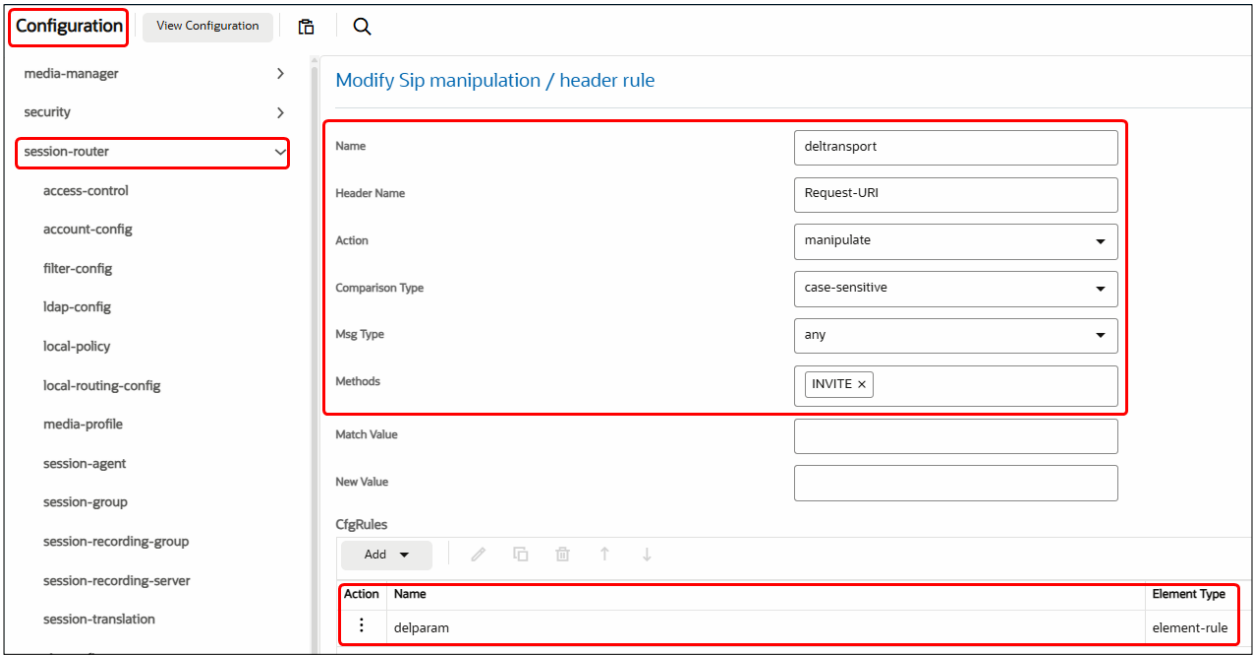

Figure 27.16: SIP Manipulation towards Google CES – Delete Transport parameter.

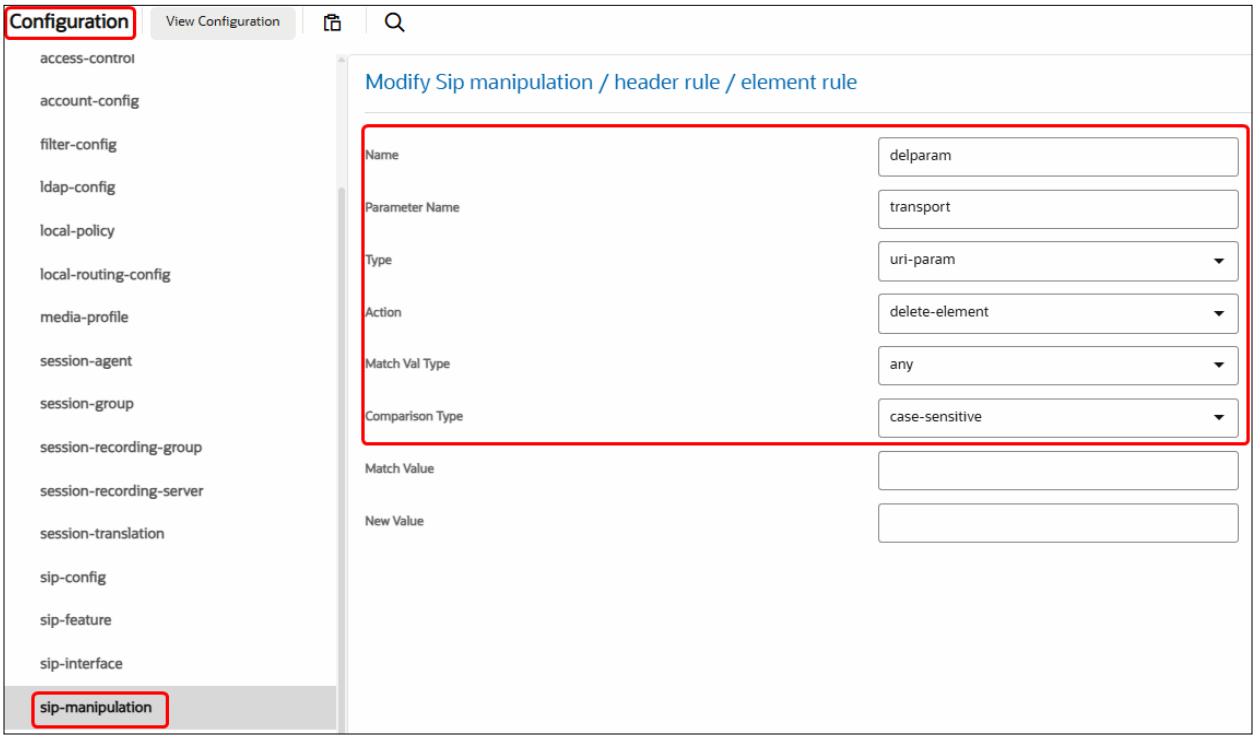

Figure 27.17: SIP Manipulation towards Google CES Cont. – Delete Transport parameter.

8 SIP INVITE To Google CES

8.1 SIP INVITE for SIPREC call

```
INVITE sip:+13618809831@us.telephony.goog:5672 SIP/2.0
Via: SIP/2.0/TLS 192.65.79.223:5061;branch=z9hG4bKqtf9nq103ov0tbubt2n0
From: sip:acmeSrc@192.65.79.223;tag=fc4e824a0f617ddcbf1e98c90f7ead51
To: <sips:+13618809831@us.telephony.goog:5672;transport=tcp>
Call-ID: 44dd70a5480470e9f5932cb3f578484d020@us.telephony.goog
CSeq: 1080 INVITE
Contact: <sips:acmeSrc@192.65.79.223:5061;transport=tcp>;+sip.src
Max-Forwards: 70
Require: siprec
Content-Type: multipart/mixed; boundary=unique-boundary-1
Content-Length: 2237
MIME-Version: 1.0
Call-Info: <http://dialogflow.googleapis.com/v2beta1/projects/ccai-389811/conversations/
OR_44dd70a5480470e9f5932cb3f578484d020>;purpose=Goog-ContactCenter-Conversation

--unique-boundary-1
Content-Type: application/sdp

v=0
o=- 3562256 226213 IN IP4 192.65.79.223
s=-
c=IN IP4 192.65.79.223
t=0 0
m=audio 20910 RTP/SAVP 0 101
c=IN IP4 192.65.79.223
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=label:234881075
a=sendonly
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:1ZbR9lHLKuYJ8T4ejRWElrWkdg4Zket0XmrtM8Jk
m=audio 20914 RTP/SAVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=maxptime:150
a=label:234881076
a=sendonly
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:aUYfqWeUmqFDSeCQ65SVEu4muW9bdvNCmuqpaIiD
```

The INVITE Request-URI should include e.164 number obtained from Google and it should have respective regional host name with SIP Signaling port :5672

Google requires the Call-Info header, and it must contain a conversation ID. The conversation ID is unique, and the format of the conversation ID follows the regex "[a-zA-Z][a-zA-Z0-9_-]*" and is assigned for each call.

dialogflow.googleapis.com/v2beta1 - API endpoint
projects/ccai-389811 - Google Cloud CCAI project ID
conversations/OR_xxxx - The unique conversation session ID that is assigned for that each call

The connection IP toward Google CCAI must be a public IP, not a private one.

For SIPREC, there can be multiple media lines with *a=sendonly*, for SIP there will be a single media line with *a=sendrecv*
Encrypted SRTP and the allocated port range should be used (Port: 16384-32767) else CES will not receive audio.

It must be a supported crypto suite by Google.

Figure 28: SIPREC call

8.2 SIP INVITE for GTP call

```
INVITE sip:+13149445469@us.telephony.goog:5672 SIP/2.0
Via: SIP/2.0/TLS 192.65.79.223:5061;branch=z9hG4bKib6trn0000el0aih
From: <sip:+19728522631@192.65.79.223>;tag=d6de37c9-7561-4937-a8b9-0ca72644801a
To: <sip:+13149445469@us.telephony.goog>
Contact: <sip:9728522631@192.65.79.223:5061;transport=tls>
Call-ID: 02e58ba8-b22e-41a5-aa3d-d3e88a671c0c
CSeq: 21261 INVITE
Allow: OPTIONS, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, REGISTER, SUBSCRIBE, NOTIFY, PUBLISH, MESSAGE, REFER
Supported: 100rel, timer, replaces, norefersub, histinfo
Session-Expires: 1800; refresher=uac
Min-SE: 90
Max-Forwards: 69
User-Agent: FPBX-16.0.40.13(20.4.0)
Content-Type: application/sdp
Content-Length: 405
Call-Info: <http://dialogflow.googleapis.com/v2beta1/projects/ccai-389811/conversations
/OR_02e58ba8-b22e-41a5-aa3d-d3e88a671c0c>;purpose=Goog-ContactCenter-Conversation

v=0
o=- 1738828058 1738828058 IN IP4 192.65.79.223
s=Asterisk
c=IN IP4 192.65.79.223
t=0 0
m=audio 21204 RTP/SAVP 0 8 101 107
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=rtpmap:107 opus/48000/2
a=fmtp:107 useinbandfec=1
a=maxptime:20
a=sendrecv
a=ptime:20
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:ydaFiGf4WVGacrEmG0HNx9d1V8cWoG0fdwG0x0PS
```

The INVITE Request-URI should include e.164 number obtained from Google and it should have respective regional host name with SIP Signaling port :5672

Google requires the Call-Info header, and it must contain a conversation ID. The conversation ID is unique, and the format of the conversation ID follows the regex "[a-zA-Z][a-zA-Z0-9_-]*" and is assigned for each call.

dialogflow.googleapis.com/v2beta1 - API endpoint
projects/ccai-389811 - Google Cloud CCAI project ID
conversations/OR_xxxx - The unique conversation session ID that is assigned for that each call

The connection IP toward Google CCAI must be a public IP, not a private one.

For GTP, there can be single media lines with *a=sendrecv*, for SIPREC there will be a multiple media line with *a=sendonly*
Encrypted SRTP and the allocated port range should be used (Port: 16384-32767) else CES will not receive audio.

It must be a supported crypto suite by Google.

Figure 29: GTP call

9 Oracle E-SBC Running configuration

Attached is the Oracle E-SBC running configuration.

Google_SIPREC_TLS_New1_2025_11_18_15_13_30.gz

10Summary of Tests and Results

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
SBC Configuration Verification					
1	SBC Configuration Verification	TLS connection setup. SBC initiates TLS connection with CES	Successful 4way handshake with Google CES. Validate the right certificates are being negotiated. SBC should be loaded with GTSR1 cert for Google. SBC should also send the certificate chain when sending its cert.	PASSED	TLS handshake is verified
2	SBC Configuration Verification	TCP Keep Alive. SBC will perform monitoring checks by attempting TCP Keep Alive to ensure Network Connectivity	Successful 3way handshake and thereafter termination	PASSED	
3	SBC Configuration Verification	TCP link is persistent. Establish calls, send multiple calls that should all use the same TCP transport connection	Persistent TCP connection, we should establish a single connection and multiplex all calls over that connection.	PASSED	
4	SBC Configuration Verification	Session Timer support. SBC should be initiator for the Session Refresh timer using Update or Re-Invite	every 900 secs the SBC should refresh the SIP session.	PASSED	Re-INVITE is sent for Session refresh
5	SBC Configuration Verification	SIP Header Manipulation (call-info header)	Validate if the Google requested header manipulation is present in the SIP INVITE. Ensure every SDP media has a label.	PASSED	

6	SBC Configuration Verification	*SBCs may need further Header manipulations based on SIP stack constraints. Verify required manipulation are added in SBC to support Google CES Example: FROM, TO header manipulations HOST part change in headers etc..,	All signaling in e.164 format	PASSED	
7	SBC Configuration Verification	SDES for SRTP. Configure the SDES parameters for crypto negotiation for the BYOT trunk	Validate the crypto is successfully negotiated and media is encrypted. All SBCs should support SDES for media encryption.	PASSED	
8	SBC Configuration Verification	DTLS for Media Encryption. Configure the DTLS parameters for crypto negotiation for the BYOT trunk, certificate for DTLS must be self-signed by the SBC.	Validate the crypto is successfully negotiated and media is encrypted. DTLS is not supported by Oracle and can be skipped.	NOT SUPPOR TED	
Inbound					
9	Inbound	SIP OPTIONS. SBC send SIP options every 60 seconds	Verify SBC sends SIP OPTIONS every 60 seconds and responds with 200 OK	PASSED	
10	Inbound	Inbound call: Calling Party disconnects the call. Inbound siprec call, ensure recordings are present, disconnect call from calling party and confirm proper disconnect	Verify Call is established with audio and transcripts from both participants Verify call is disconnected properly	PASSED	

11	Inbound	Inbound call: Called Party disconnects the call. Inbound siprec call, ensure recording are present, disconnect call from called party and confirm proper disconnect	Verify Call is established with audio and transcripts from both participants Verify call is disconnected properly	PASSED	
12	Inbound	Long duration call- Outbound Call- 1 hour max. Long duration siprec call	Ensure siprec calls stay up for an hour, confirm transcripts are present for entire duration	PASSED	
13	Inbound	Long duration hold and resume (wait until session audit\session refresh occurs from DUT). Long duration siprec call, have the call placed on hold by agent, have call resume. Have customer place on hold then have call resume.	Call is connected, we have two active streams, confirm once a stream goes on hold, we receive corresponding signaling events, and that we no longer record transcripts for the participant on hold.	PASSED	
14	Inbound	Handling Error codes 603 decline. User A Calls PSTN A PSTN A rejects the incoming call	Verify SBC handles Call rejected properly	PASSED	
15	Inbound	Inbound call hold	Validate if media is	PASSED	Call recording

		scenarios. Call starts out inactive for both participants; session moves to active	present when expected, confirm signaling events modify sdp properly, once call is move to active validate media and transcripts		is deactivated using API command. Audio during the inactive state is not recorded.

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
16	Inbound	Inbound call hold scenarios. call starts out as active for both participants, session move to inactive, and transitions back to active	Validate if media is present when expected, confirm signaling events modify sdp properly, once call is moved to active validate media and transcripts	PASSED	Recording was not present after deactivating conversation and recording resumed after activating conversation via API
17	Inbound	Update. Validate that update sent prior to call establishment do not contain SDP	Validate that update prior to call establishment do not contain SDP as expected	PASSED	REINVITE message is sent from SBC every 900 seconds without SDP
18	Inbound	Update. Validate that updates post call establishment contain SDP to modify session	If SBC uses update to modify sessions, ensure SDP is included	NOT APPLICABLE	REINVITE message is sent from SBC every 900 seconds without SDP
19	Inbound	re-invites. Ensure re-invites that modify session include SDP	Ensure re-invites that modify session include SDP	PASSED	REINVITE is sent to Google CES as part of session refresh, hold scenarios
20	Inbound	Codec negotiation. Ensure that g711 u-law is preferred codec	Ensure we can prioritize g711 as preferred codec, note where SBC configures preferred codec	PASSED	
21	Inbound	3 way conference. Determine	Determine requirements,	PASSED	

		requirements, record all leg.	record all legs		
22	Inbound	CES cloud project setup. Establish CES cloud project, provision the project with a GTP phone number for access (Create conversations/participants on the fly through SIP headers)	Verify project is setup, functional test to confirm you can connect to the GTP access phone number	PASSED	
23	Inbound	CES cloud project setup. Establish CES cloud project, provision the project with a GTP phone number for access (Pre-creation of conversations/participants)	Verify project is setup, functional test to confirm you can connect to the GTP access phone number	NOT APPLICABLE	This test case is not applicable for call recording
24	Inbound	Consultative transfer. Consultative transfer from 1. PSTN > User1 > User2 2. PSTN > User1 > PSTN user2		PASSED	
25	Inbound	Blind transfer. Blind transfer from 1. PSTN > User1 > User2 2. PSTN > User1 > PSTN user2		PASSED	
26	Use documentation to build trunk using self service model			PASSED	
27	Inbound call	Call starts out	Inbound call hold	PASSED	

	hold scenarios using A-law as codec	inactive for both participants; session moves to active	scenarios using A-law as codec		
28	Inbound call: Called Party disconnects the call. using a a-law codec	Inbound siprec call, ensure recording are present , disconnect call from called party and confirm proper disconnect	Inbound call: Called Party disconnects the call. using a a-law codec	PASSED	
29	Long duration call- Outbound Call- 1 hour max using a-law codec	Long duration siprec call	Long duration call- Outbound Call- 1 hour max using a-law codec	PASSED	REINVITE messages are sent from SBC to Google CES every 15min (900 seconds)
30	Inbound call: Configure trunk in non-default region,	Confirm call is processed within the region for signaling and media that corresponds to the region trunk was provisioned in	"Verify Call is established with audio and transcripts from both participants	PASSED	Testing conducted in US region
31	Participant Labels test	Configure call info header to specify roles, ensure the media streams align	"Frist media stream HUMAN_AGENT role and	PASSED	When the roles are set to "HUMAN AGENT" and "END USER," Call-Info: <http://dialogflow.googleapis.com/v2beta1/projects/CES-389811/conversations/OR_0e39f42930cb0a5b0aba6241e11d8346?roles=HUMAN_AGENT,END_USER

					>;purpose=Goog-ContactCenter-Conversation the transcript shows the first media stream with the participation role as "HUMAN AGENT," followed by "END USER." It showed 5/10 attempts. The call-id in the call-info header is sent with hyphen sign
32	DTLS test			Not Supported	
33	Conference TEST	Conference call between PSTN and PBX users	Validate both-way audio	PASSED	Both-way audio for all users was present.
34	Validate Call recording			PASSED	