Configuration Guide for Google CES Agent Handoff Using Cisco Unified Border Element (Cisco UBE) V17.15.4

Table of Contents

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 Agent handoff** using **Cisco Unified Border Element (Cisco UBE) V17.15.4**

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 Agent Handoff with Cisco UBE V17.15.4 configuration.

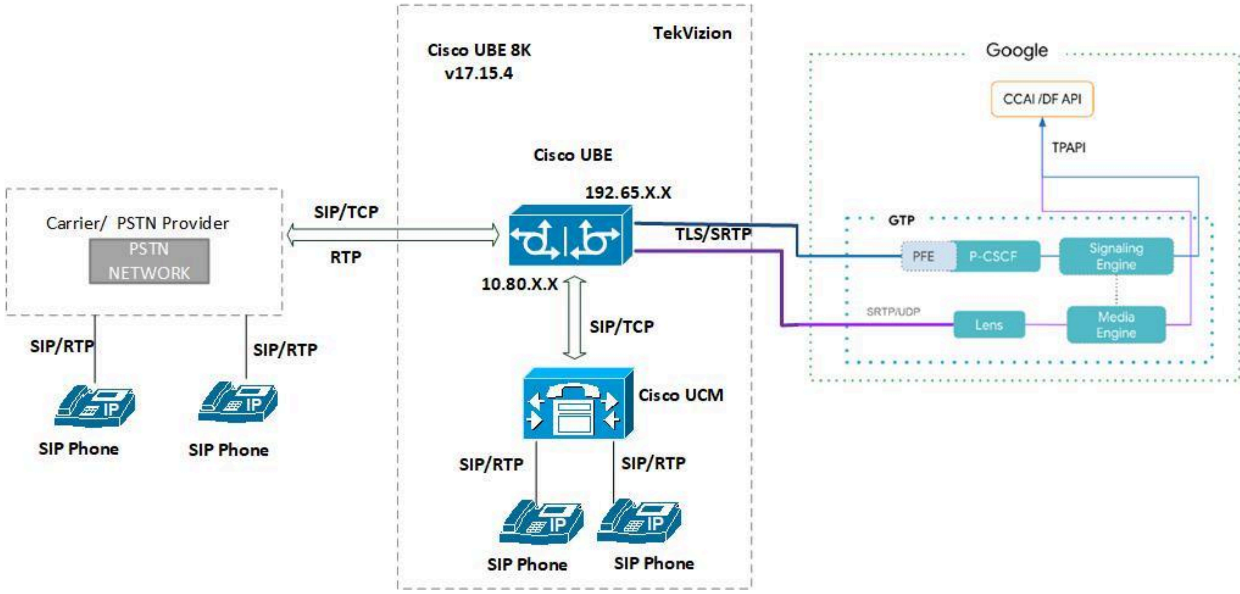

Figure 1: SIP Trunk Lab Reference Network

The lab network consists of the following components.

- Google CES Cloud Environment
- Cisco UBE V17.15.4
- PSTN Gateway
- OnPrem PBX (Cisco Unified Communications Manager V15.0.1.11901-2)

3 Hardware Components

- Cisco UBE C8000V platform

4 Software Requirements

- Cisco UBE-Version: 14.10 running IOS XE 17.15.4
- OnPrem PBX (Cisco Unified Communications Manager V15.0.1.11901-2)

5 Google CES Certified Cisco UBE Version

Table 1 – Google CES Certified Cisco UBE Version

Google CES Certified Cisco UBE Version	
Cisco UBE SBC	17.15.4

6 Features

6.1 Caveats and Limitations

SIP profile restriction	Cisco UBE have character limitation where the header length (including header name) after modification should not exceed 300 characters. Max header length for add value is approximately 220 characters. Max SDP length is 2048 characters. If any header length exceeds this maximum value after applying SIP profiles, then the profile cannot apply.

7 Configuration

7.1 Configuration Checklist

Below are the steps that are required to configure Cisco UBE.

Table 2 – Cisco UBE Configuration Steps

Step	Description	Reference
Cisco UBE		
Step 1	IP Networking	Section 7.4.1
Step 2	Routing	Section 7.4.2
Step 3	DNS Servers	Section 7.4.3
Step 4	Certificates	Section 7.4.4
Step 5	Import Signed Host Certificate	Section 7.4.5
Step 6	Trusted CA Trust point for Google CES	Section 7.4.6
Step 7	Default Trust point and TLS Version	Section 7.4.7
Step 8	Global Cisco UBE Settings	Section 7.4.8
Step 9	Message Handling Rules	Section 7.4.9
Step 10	Message Handling Rules for Inbound messages from Google CES	Section 7.4.10
Step 11	SRTP Crypto	Section 7.4.11
Step 12	Translation Rule	Section 7.4.12
Step 13	Dial Peers	Section 7.4.13
Step 14	Running Configurations	Section 7.4.14

7.2 IP Address Worksheet

The specific values listed in the table below and in subsequent sections that 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	10.80.X.X
Cisco UBE	
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 for troubleshooting Google CES API configuration for Agent Handoff.

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

7.4 Cisco UBE Configuration

7.4.1 IP Networking

Below are the interface configurations toward the PSTN Gateway and Google CES.

```
interface GigabitEthernet1
 description to PSTN Gateway
 ip address 10.80.X.X 255.255.255.0
 negotiation auto
!
Interface GigabitEthernet2
 description to Google CES
 ip address 192.65.X.X 255.255.255.128
 negotiation auto
!
```

7.4.2 Routing

Below are the static routes configured for Google CES and PSTN Gateway.

```
ip route 0.0.0.0 0.0.0.0 192.65.X.X
ip route 0.0.0.0 0.0.0.0 10.80.X.X
```

7.4.3 DNS Servers

Below is the DNS server configured to resolve Google FQDN.

```
ip name-server 8.8.8.8
```

7.4.4 Certificates

Below are the steps to create and install a certificate in Cisco UBE
Enter config mode and the command below generates **RSA Key Pair.**

```
crypto key generate rsa general-keys label sbc8 exportable redundancy modulus 2048
The name for the keys will be: sbc8

% The key modulus size is 2048 bits
% Generating 2048 bit RSA keys, keys will be exportable with redundancy...
[OK] (elapsed time was 1 seconds)
```

Below command creates **Trust point** for Cisco UBE. This trust point is used in *Section 7.4.7 – Default Trust point and TLS version*

```
crypto pki trustpoint sbc8
 enrollment terminal
 fqdn sbc8.xxx.com
 subject-name cn=sbc8.tekvizonlabs.com
 subject-alt-name sbc8.tekvizonlabs.com
 revocation-check none
 rsakeypair sbc8
 hash sha256
!
```

Generate Certificate Signing Request (CSR)

Below command generates Certificate Signing Request (CSR). This CSR can be used to request a certificate from one of the supported Certificate Authorities

crypto pki enroll sbc8

```
% Start certificate enrollment ..

% The subject name in the certificate will include: cn=sbc8.tekvizonlabs.com
% The subject name in the certificate will include: sbc8.tekvizonlabs.com
% Include the router serial number in the subject name? [yes/no]: no
% Include an IP address in the subject name? [no]: no
Display Certificate Request to terminal? [yes/no]: yes
Certificate Request follows:
```

Authenticate CA Certificate

Enter the following command in config mode, then paste the CA certificate that verifies the host certificate into the Trust point (usually the intermediate certificates). Open the base 64 CER/PEM file with notepad, copy the text, and paste it, having secure CA followed by Root CA into the terminal when prompted.

```
crypto pki authenticate sbc8

Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
<Certificate>
```

7.4.5 Import Signed Host Certificate

Enter the following command then paste the host certificate into the trust point. Open the base 64 CER/PEM file with notepad, copy the text, and paste it into the terminal when prompted.

```
crypto pki import sbc8 certificate

Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
<Certificate>
```

7.4.6 Trusted CA Trust point for Google CES

Download the Google certificate via link https://pki.goog/roots.pem and select GTS Root R1 certificate.
Below are the configurations to create the CA certificate trust points to validate Google CES TLS messages

```
crypto pki trustpoint GoogleCA_1
 enrollment terminal
 revocation-check none
 hash sha256
!
```

Enter the following command then paste the CA certificate into the Trust point. Open the base 64 CER/PEM file with notepad, copy the text, and paste it into the terminal when prompted.

```
crypto pki authenticate GoogleCA_1

Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
< GTS Root R1 certificate>
```

7.4.7 Default Trust point and TLS version

Below are the SIP user agent details, with TLS version 1.3 used to establish the connection with Google CES.

```
sip-ua
 transport tcp tls v1.3
  crypto signaling default trustpoint sbc8
!
```

7.4.8 Global Cisco UBE Settings

Below are the Global VoIP and SIP settings configured in the Cisco UBE

```
voice service voip
ip address trusted list
  ipv4 10.80.X.X
  ipv4 74.125.X.X
  ipv4 216.239.X.X
 address-hiding
 mode border-element
 srtp fallback
 allow-connections sip to sip
 trace
 sip
  session refresh
  midcall-signaling passthru
  pass-thru headers unsupp
  sip-profiles inbound
!
```

Explanation:

Command	**Description**
ip address trusted list	To allow all traffic between Google CES and Cisco UBE
allow-connections sip to sip	Allows back-to-back user agent connections between two SIP call legs
srtp fallback	Allows a call to automatically fall back to unencrypted RTP when SRTP (Secure Real-time Transport Protocol) negotiation fails, while maintaining the signaling path intact
pass-thru headers unsupp	The pass-through functionality includes all or only a configured list of unsupported or non-mandatory SIP headers and all unsupported content.
sip-profiles inbound	Enables inbound SIP profiles feature
midcall-signaling passthru	Enables the transparent forwarding of in-call SIP messages between endpoints, maintaining end-to-end signaling for mid-call modifications while the Cisco UBE continues to handle only the media path and basic call control

7.4.9 Message Handling Rules

Manipulations for Outbound messages to Google CES

The following SIP profile is used to send the Call-Info header, which is a mandatory requirement for establishing connectivity with Google CES Agent Handoff. This rule is applied to outbound SIP messages to Google CES. This rule is used in *Section 7.4.13 - dial peer 9000*

The below manipulation,

- The Call-ID header in the INVITE message as input. e.g. Call-ID: (.*)-(.*)-(.*)-(.*)@(.*) and modifies as Call-ID:\1-\2-\3-\4@\5
- Adds the Call-Info header with the static string of <http://dialogflow.googleapis.com/v2beta1/projects/CES-38XXX/conversations/Sr_\1\2\3\4>;purpose=Goog-ContactCenter-Conversation".
- Has the alpha characters "Sr" which indicates Cisco since the Conversation ID in the Call-Info header must be in the format of **"[a-zA-Z][a-zA-Z0-9_-]*"**

```
voice class sip-profiles 9000
 rule 5 request ANY sip-header Call-ID modify "Call-ID: (.*)-(.*)-(.*)-(.*)@(.*)"
"Call-ID:\1-\2-\3-\4@\5\x0D\x0ACall-Info:<http://dialogflow.googleapis.com/v2beta1/projects/CES-38XXX/conversations/Sr_\1\2\3\4>;purpose=Goog-ContactCenter-Conversation"
!
```

Options Keepalive

The following profile modifies the SIP Request URI and the TO headers towards Google CES with Fully Qualified Domain Name. This profile is applied to SIP Options Keepalive message towards Google CES as shown below.

This Options Keepalive is used in *Section 7.4.13 – dial peer 9000*

```
Rule 1: To modify SIP-Req-URI header to us.telephony.goog:5672
Rule 2: To modify "TO" header to us.telephony.goog:5672

voice class sip-profiles 201

rule 1 request OPTIONS sip-header SIP-Req-URI modify "sip:74.125.X.X:5672"
"sip:us.telephony.goog:5672"
 rule 2 request OPTIONS sip-header To modify "<sip:74.125.X.X" "sip:us.telephony.goog"
!
voice class sip-options-keepalive 9000
 description towards Google
 up-interval 30
 transport tcp tls
 sip-profiles 201
!
```

Manipulations for Outbound messages to Google CES via UUI header

The SIP profile instructs Cisco UBE to insert a specific User-to-User header required by Google CES for exchanging session-related information, enabling connectivity with the Google CES Agent Handoff.

```
voice class sip-profiles 9001
 rule 1 request ANY sip-header User-to-User add "User-to-User:
XXXX56532;encoding=hex;purpose=Goog-Session-Param"
!
```

7.4.10 Message Handling Rules for Inbound messages from Google CES

Below is the manipulation to modify Request-URI, and To headers during call hand-off to an agent.

The rule below is used in *Section* *7.4.13 - Dial Peers*

```
voice class sip-profiles 801
 request INVITE sip-header SIP-Req-URI modify "sip:(.*)@.*" "sip:\1@10.80.X.X:5060 SIP/2.0"
 request INVITE sip-header To modify "sip:(.*)@.*" sip:\1@10.80.X.X:5060 SIP/2.0
!
```

7.4.11 SRTP Crypto

Below is the crypto cipher profile used for Google CES. The rule below is applied to *Section 7.4.13 - Dial peers*

```
voice class srtp-crypto 9000
 crypto 1 AES_CM_128_HMAC_SHA1_80
```

7.4.12 Translation Rule

Below is the Translation rule and Translation profile applied towards Google CES. This translates the incoming number pattern 972852XXXX to Google CES DID +13149XXXXXX

The rule below is used in *Section 7.4.13 - Dial Peers*

```
voice translation-rule 9000
 rule 1 /972852XXXX/ /+13149XXXXXX/
!
voice translation-profile 9000
  translate called 9000
!
```

Below is the Translation rule and Translation profile applied towards Agent. This translates the incoming number pattern +1972852XXXX to Agent DID 972852XXXX

The rule below is used in *Section 7.4.13 - Dial Peers*

```
voice translation-rule 10
 rule 1 /^\+1972852XXXX$/ /972852XXXX/
voice translation-profile FROM-GOOGLE
 translate called 10
!
```

7.4.13 Dial Peers

Below are the Inbound dial-peers configured to route the calls from PSTN Gateway and Google CES.

```
voice class uri 401 sip
 host ipv4:10.64.X.X
!
dial-peer voice 301 voip
 description inbound from PSTN Gateway
 translation-profile incoming 9000
 session protocol sipv2
 session transport tcp
 incoming uri from 401
 voice-class codec 1 offer-all
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet1
 voice-class sip bind media source-interface GigabitEthernet1
 dtmf-relay rtp-nte
 no vad
!
voice class uri 601 sip
 host ipv4:216.239.36.X
 host ipv4:74.125.X.X
!
dial-peer voice 5000 voip
 description *** Incoming from Google (TLS) ***
 translation-profile incoming FROM-GOOGLE
 session protocol sipv2
 session transport tcp tls
 incoming uri via 601
 voice-class sip profiles 801
 voice-class sip bind control source-interface GigabitEthernet2
 voice-class sip bind media source-interface GigabitEthernet2
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad!
```

Below is the outbound dial peer configuration towards Google CES via TLS

```
dial-peer voice 9000 voip
 description GoogleCES
 destination-pattern +1314XXXXXXX
 session protocol sipv2
 session target dns:us.telephony.goog:5672
 session transport tcp tls
 voice-class sip profiles 9000
 voice-class sip srtp-crypto 9000
 voice-class sip options-keepalive profile 9000
 voice-class sip bind control source-interface GigabitEthernet2
 voice-class sip bind media source-interface GigabitEthernet2
 srtp
 codec g711ulaw
!
```

Below is the Outbound dial-peer configured to route the calls towards Agent PBX

```
dial-peer voice 5001 voip
description *** Outgoing to Agent (TCP) ***
 destination-pattern .T
 session protocol sipv2
 session target ipv4:10.80.X.X
 session transport tcp
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet1
 voice-class sip bind media source-interface GigabitEthernet1
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
```

7.4.14 Running Configurations for Cisco UBE

```
version 17.15
service timestamps debug datetime msec
service timestamps log datetime msec
service internal
platform qfp utilization monitor load 80
platform sslvpn use-pd
platform console virtual
!
hostname 8k_2
!
boot-start-marker
boot system bootflash:packages.conf
boot-end-marker
!
!
logging buffered 50000000
no aaa new-model
!
!
ip name-server 8.8.8.8
ip domain name tekvizionlabs.com
!
!
login on-success log
!
!
subscriber templating
!
!
!
crypto pki trustpoint GoogleCA_1
 enrollment terminal
 revocation-check none
 hash sha512
!
crypto pki trustpoint sbc8
 enrollment terminal
 fqdn sbc8.tekvizionlabs.com
 subject-name cn=sbc8.tekvizionlabs.com
 subject-alt-name sbc8.tekvizionlabs.com
 revocation-check none
 rsakeypair sbc8
 hash sha512
!
crypto pki certificate chain GoogleCA_1
```

```
 certificate ca 0203E5936F31B01349886BA217
        quit
!

crypto pki certificate chain sbc8
certificate 00E314DB9F75151D8
certificate ca 07
        quit
!
!
!
voice service voip
 ip address trusted list
  ipv4 10.80.X.X
  ipv4 74.125.X.X
  ipv4 216.239.X.X
address-hiding
 mode border-element
 srtp fallback
 allow-connections sip to sip
trace
 sip
  session refresh
  midcall-signaling passthru
  pass-thru headers unsupp
  sip-profiles inbound
!
!
voice class uri 401 sip
 host ipv4:10.64.X.X
!
voice class uri 501 sip
 host ipv4:10.80.X.X
!
voice class uri 601 sip
 host ipv4:216.239.X.X
 host ipv4:74.125.X.X
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g711alaw
!
voice class sip-profiles 201
 rule 1 request OPTIONS sip-header SIP-Req-URI modify "sip:74.125.X.X:5672"
"sip:us.telephony.goog:5672"
 rule 2 request OPTIONS sip-header To modify "<sip:74.125.X.X" "sip:us.telephony.goog"
!
voice class sip-profiles 9000
```

```
 rule 5 request ANY sip-header Call-ID modify "Call-ID: (.*)-(.*)-(.*)-(.*)@(.*)"
"Call-ID:\1-\2-\3-\4@\5\x0D\x0ACall-Info:<http://dialogflow.googleapis.com/v2beta1/projects/CE
S-3898XX/conversations/Sr_\1\2\3\4>;purpose=Goog-ContactCenter-Conversation"
!
voice class sip-profiles 9001
 rule 1 request ANY sip-header User-to-User add "User-to-User:
XXXX;encoding=hex;purpose=Goog-Session-Param"
!
voice class sip-profiles 801
 request INVITE sip-header SIP-Req-URI modify "sip:(.*)@.*" "sip:\1@10.80.X.X:5060 SIP/2.0"
 request INVITE sip-header To modify "sip:(.*)@.*" "sip:\1@10.80.X.X:5060 SIP/2.0"
!
!
voice class sip-options-keepalive 9000
 description towards Google
 up-interval 30
 transport tcp tls
 sip-profiles 201
!
voice class srtp-crypto 9000
 crypto 1 AES_CM_128_HMAC_SHA1_80
!
!
voice translation-rule 10
 rule 1 /^\+1972852XXXX$/ /972852XXXX/
!
voice translation-rule 9000
 rule 1 /972852XXXX/ /+1314XXXXXXX/
!
!
voice translation-profile 9000
 translate called 9000
!
voice translation-profile FROM-GOOGLE
 translate called 10
!
license udi pid C8000V sn 9XXX
license boot level network-essentials addon dna-essentials
memory free low-watermark processor 165741
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
!
redundancy
!
interface GigabitEthernet1
```

```
 description Interface to PBX and PSTN Gateway
 ip address 10.80.X.X 255.255.255.0
 negotiation auto
!
interface GigabitEthernet2
 description to Google CES
 ip address 192.65.X.X 255.255.255.0
 negotiation auto
!
!
ip http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet1
ip tftp source-interface GigabitEthernet1
ip route 0.0.0.0 0.0.0.0 10.80.X.X
ip route 0.0.0.0 0.0.0.0 192.65.X.X
ip ssh bulk-mode 131072
!
control-plane
!
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!

dial-peer voice 9000 voip
 description GoogleCES_SIPTest
 destination-pattern +1314XXXXXXX
 session protocol sipv2
 session target dns:us.telephony.goog:5672
 session transport tcp tls
 voice-class sip profiles 9000
 voice-class sip srtp-crypto 9000
 voice-class sip options-keepalive profile 9000
 voice-class sip bind control source-interface GigabitEthernet2
 voice-class sip bind media source-interface GigabitEthernet2
 srtp
 codec g711ulaw
!
dial-peer voice 301 voip
 description inbound from PSTN Gateway
 translation-profile incoming 9000
```

```
 session protocol sipv2
 session transport tcp
 incoming uri via 401
 voice-class codec 1 offer-all
 voice-class sip options-keepalive
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet1
 voice-class sip bind media source-interface GigabitEthernet1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 5000 voip
 description *** Incoming from Google (TLS) ***
 translation-profile incoming FROM-GOOGLE
 session protocol sipv2
 session transport tcp tls
 incoming uri via 601
 voice-class sip profiles 801
 voice-class sip bind control source-interface GigabitEthernet2
 voice-class sip bind media source-interface GigabitEthernet2
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
dial-peer voice 5001 voip
 description *** Outgoing to Agent (TCP) ***
 destination-pattern .T
 session protocol sipv2
 session target ipv4:10.80.X.X
 session transport tcp
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet1
 voice-class sip bind media source-interface GigabitEthernet1
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
!
sip-ua
 transport tcp tls v1.3
  crypto signaling default trustpoint sbc8
!
line con 0
 exec-timeout 0 0
 stopbits 1
line aux 0
line vty 0
```

```
 password admin
 login local
 transport input ssh
line vty 1 4
 password admin
 login local
 transport preferred ssh
 transport input ssh
!
end
```

8 SIP INVITE To GOOGLE CES

8.1 SIP INVITE for SIPREC call

```
INVITE sip:+1361        @us.telephony.goog:5672 SIP/2.0
Via: SIP/2.0/TLS 192.65.       :5061;branch=z9hG4bK7C818B7
From: <sip:192.65.       >;tag=36A15FA-1F91
To: <sip:+13614(      @us.telephony.goog>
Date: Thu, 09 Oct 2025 07:14:33 GMT
Call-ID:706C7FB8-A41611F0-8B7FCDC6-63FA2931@192.65.
Call-Info:<http://dialogflow.googleapis.com/v2beta1/projects/ccai-389   /conversations/
Sr_70                              31>;purpose=Goog-ContactCenter-Conversation
Supported: 100rel,timer,resource-priority,replaces
Require: siprec
Min-SE:  1800
Cisco-Guid: 1883195156-2752909808-2228249622-2368317232
User-Agent: Cisco-SIPGateway/IOS-17.15.4
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1759994073
Contact: <sip:192.65.       :5061;transport=tls>;+sip.src
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 69
Session-ID: 6bcdbe58441059e59a717f5e71514b53;remote=00000000000000000000000000000000
Session-Expires:  1800
Content-Type: multipart/mixed;boundary=uniqueBoundary
Mime-Version: 1.0
Content-Length: 2544
--uniqueBoundary
Content-Type: application/sdp
Content-Disposition: session;handling=required
v=0
o=CiscoSystemsSIP-GW-UserAgent 3363 1345 IN IP4 192.65
s=SIP Call
c=IN IP4 192.65.
t=0 0
m=audio 8100 RTP/SAVP 0 19
c=IN IP4 192.65.
a=rtpmap:0 PCMU/8000
a=rtpmap:19 CN/8000
a=ptime:20
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
a=sendonly
a=label:1
m=audio 8102 RTP/SAVP 0 19
c=IN IP4 192.65.
a=rtpmap:0 PCMU/8000
a=rtpmap:19 CN/8000
a=ptime:20
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
a=sendonly
a=label:2
```

Figure 2: SIPREC Call

8.2 SIP INVITE for GTP call

```
INVITE sip:+13149 9@us.telephony.goog:5672 SIP/2.0
Via: SIP/2.0/TLS 192.65. 3:5061;branch=z9hG4bK1FA25CA
From: "214 242 " <sip:214242 @192.65. >;tag=82D940-A3B
 --More-- Supported: 100rel,timer,resource-priority,replaces
Content-Type: application/sdp
Content-Length: 168
v=0
o=CiscoSystemsSIP-GW-UserAgent 1871 238 IN IP4 10.80
s=SIP Call
c=IN IP4 10.80.
t=0 0
m=audio 0 RTP/AVP 18 0 8 9 4 2 15 3
c=IN IP4 10.80.
To: <sip:+13149 @us.telephony.goog>
Date: Tue, 14 Oct 2025 13:32:26 GMT
Call-ID:EC613C5-A83911F0-8427E476-A0CAA8EE@192.65.
Call-Info:<http://dialogflow.googleapis.com/v2beta1/projects/ccai-3898 /conversations/
Sr_EC EE>;purpose=Goog-ContactCenter-Conversation
Supported: 100rel,timer,resource-priority,replaces
Min-SE:  1800
Cisco-Guid: 0253948781-2822312432-2289984534-2368317232
User-Agent: Cisco-SIPGateway/IOS-17.15.4
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1760448746
Contact: <sip:2142425989@192.65. :5061;transport=tls>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 69
Session-ID: 23fc8e7bd8365247a8f2b880b74b46e1;remote=00000000000000000000000000000000
Session-Expires:  1800
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 560
v=0
o=CiscoSystemsSIP-GW-UserAgent 5066 9061 IN IP4 192.65.
s=SIP Call
c=IN IP4 192.65.7
t=0 0
m=audio 8150 RTP/SAVP 0 19
c=IN IP4 192.65. 3
a=rtpmap:0 PCMU/8000
a=rtpmap:19 CN/8000
a=ptime:20
a=crypto:1 AEAD_AES_256_GCM inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
a=crypto:2 AEAD_AES_128_GCM inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
a=crypto:3 AES_CM_128_HMAC_SHA1_80 inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
a=crypto:4 AES_CM_128_HMAC_SHA1_32 inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
```

Figure 3: GTP Call

9 Summary of Tests and Results

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
SBC Configuration Verification					
35	UUI header test	Use the UUI header as opposed to call-info header to send conversation id	Call should process as normal, and recording under conversation ID derived from UUI header as opposed to call-info	PASSED	Calls are successfully established with the live agent when using UUI header instead of call-info header.
36	Keep_conversation_running=TRUE test	ConversationProfile needs to have SipConfig set with keepConversationRunning = TRUE. Send first call with a Call-Info header and have a call for 2 turns. End the call. Send second call with the SAME Call-Info header as above and have a call for 3 turns. End the call.	Two calls having the same call-info has both conversation details.	PASSED	Both call transcripts are present for the same conversation session id.
37	Live Agent Transfer	Call goes to virtual agent, initial live agent handoff and verify outgoing SIP INVITE, call connection and disconnection		PASSED	Calls to virtual agent is successfully handed off to live Agent.
38	Live Agent Transfer	Call goes to virtual agent, initial live agent handoff and verify outgoing SIP INVITE, call connection and disconnection.		PASSED	Calls to virtual agent is successfully handed off to live Agent

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		Make calls to +13149XXXXXX, "speak to an agent", +1-972-852-XXXX should then ring and get connected as the agent			
39	UUI headers	Call goes to virtual agent, say "end the call", validate that the SIP BYE has a UUI header Make calls to +13149XXXXXX, "end the call", check SIP BYE and ensure there is one or more (identify if there are 3 or 1) UUI headers with purpose Goog-Session-Param		PASSED	Call gets connected successfully to live agent and when performing the conversation "End the call", the call gets disconnected successfully with 3 UUI headers.
40	SIP REFER	Call goes to virtual agent, say "send a sip refer", validate that a SIP REFER is received to 972-852-XXXX. Make calls to +1314XXXXXX, "send a sip refer", SIP REFER should be received with refer to set to 972-852-XXXX		PASSED	REFER request is received and the transfer is successful.