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

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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 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.

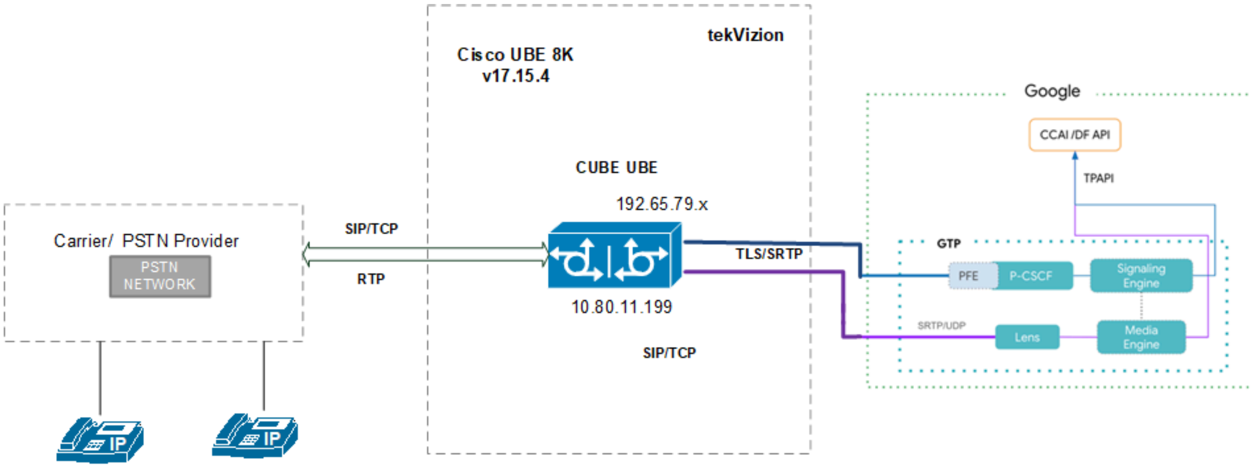


Figure 1: SIP Trunk Lab Reference Network

The lab network consists of the following components.

- Google CES Cloud Environment
- Cisco UBE v17.15.4

3 Hardware Components

- Cisco UBE C8300

4 Software Requirements

- Cisco UBE C8300V Edge software v17.15.4

5 Google CES Certified Cisco UBE Version

Table 1 – Google CES Certified Cisco UBE Version

Google CES Certified Cisco UBE Version	
Cisco UBE Virtual SBC	17.15.4

-

6 Features

6.1 Caveats and Limitations

Conversation deactivation using Google CES API (Google CES sends mid-call SIP INVITE with SDP INACTIVE)	Cisco UBE does not support mid-call renegotiations through Hold/Resume sent from the recording solutions (Doc)

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 Media Proxy 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 Peer	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 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 to configure Google CES API configuration for Call Agent Handoff.

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

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

7.4 Cisco UBE Configuration

7.4.1 IP Networking

Below are the interface configuration towards PSTN & Google CES.

```
interface GigabitEthernet1
 description to PSTN
 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.

```
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
ip route 216.0.0.0 255.0.0.0 192.65.X.X
```

7.4.3 DNS Servers

Below is the DNS server configured in the lab topology 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 Media Proxy

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 Media Proxy. This trust point is used in *Section 6.4.7 – Default Trust point and TLS version*

```
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 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.tekvizionlabs.com
% The subject name in the certificate will include: sbc8.tekvizionlabs.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 only select the 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 and the TLS version v1.3 is used to connect with Google CES

```
sip-ua
 no remote-party-id
 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
  no call service stop
!
```

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
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 CUBE 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 header to connect 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 UBE 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/CE
S-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 (Sample)

This SIP profile tells CUBE to insert a special User-to-User header required by Google CES to exchange session-related info to connect with Google CES Agent-Handoff.

The rule below is used instead of call-info header in *Section 7.4.13 - dial peer 9000 (UUI header testing).*

```
voice class sip-profiles 9001
 rule 1 request ANY sip-header User-to-User add "User-to-User: 6b6579313D76616C7565313B6B6579323D76616C756532;encoding=hex;purpose=Goog-Session-Param"
!
```

7.4.10 Message Handling Rules for Inbound messages from Google CES

The SIP profile makes sure that when a call from Google (or another provider) is sent toward PSTN both the Request-URI and To headers are rewritten so that Agent sees the request as being directly addressed to it.

The rule below is used in *Section 7.4.13 - dial peer 5000*

```
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 peer 9000.*

```
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 9728522626 to Google CES DID +13149445469

The rule below is used in *Section 7.4.13 - dial peer 9000*

```
voice translation-rule 9000
 rule 1 /9728522626/ /+13149445469/
!
voice translation-profile 9000
  translate called 9000
!
```

Below is the Translation rule and Translation profile applied from Google CES. This translates the incoming number pattern +19728522626 to Agent DID 9728522667

The rule below is used in *Section 7.4.13 - dial peer 5000*

```
voice translation-rule 10
 rule 1 /^\+19728522626$/ /9728522667/
!
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 from Google CES along with voice uri.

```
voice class uri 401 sip
 host ipv4:10.64.X.X
!
dial-peer voice 301 voip
 description inbound from PSTN
 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 towards Google CES via TLS to connect to Agent handoff

```
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 are the Outbound dial-peers configured to route the calls towards On-prem PBX.

```
dial-peer voice 5001 voip
 description *** Outgoing to Agent ***
 destination-pattern .T
 session protocol sipv2
 session target ipv4:10.80.X.X
 session transport tcp
 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

```
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 TP-self-signed-3322804963
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-3322804963
 revocation-check none
 rsakeypair TP-self-signed-3322804963
 hash sha512
!
crypto pki trustpoint SLA-TrustPoint
 enrollment pkcs12
 revocation-check crl
 hash sha512
!
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 TP-self-signed-3322804963
 certificate self-signed 01
XXX
        quit
crypto pki certificate chain SLA-TrustPoint
 certificate ca 01
XX
        quit
crypto pki certificate chain GoogleCA_1
 certificate ca 0203E5936F31B01349886BA217
XXX
        quit
crypto pki certificate chain sbc8
XXX
        quit
 certificate ca 07
  XX
        quit
!
!
!
voice service voip
 ip address trusted list
  ipv4 10.80.X.X
  ipv4 74.125.X.X
  ipv4 216.239.X.X
  ipv4 216.239.X.X
  ipv4 74.125.X.X
 address-hiding
 mode border-element
 srtp fallback
 allow-connections sip to sip
 no supplementary-service sip refer
 trace
 sip
  session refresh
  midcall-signaling passthru
  pass-thru headers unsupp
  sip-profiles inbound
  no call service stop
!
!
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-389811/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:
6b6579313D76616C7565313B6B6579323D76616C756532;encoding=hex;purpose=Goog-Sessio
n-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
!
```

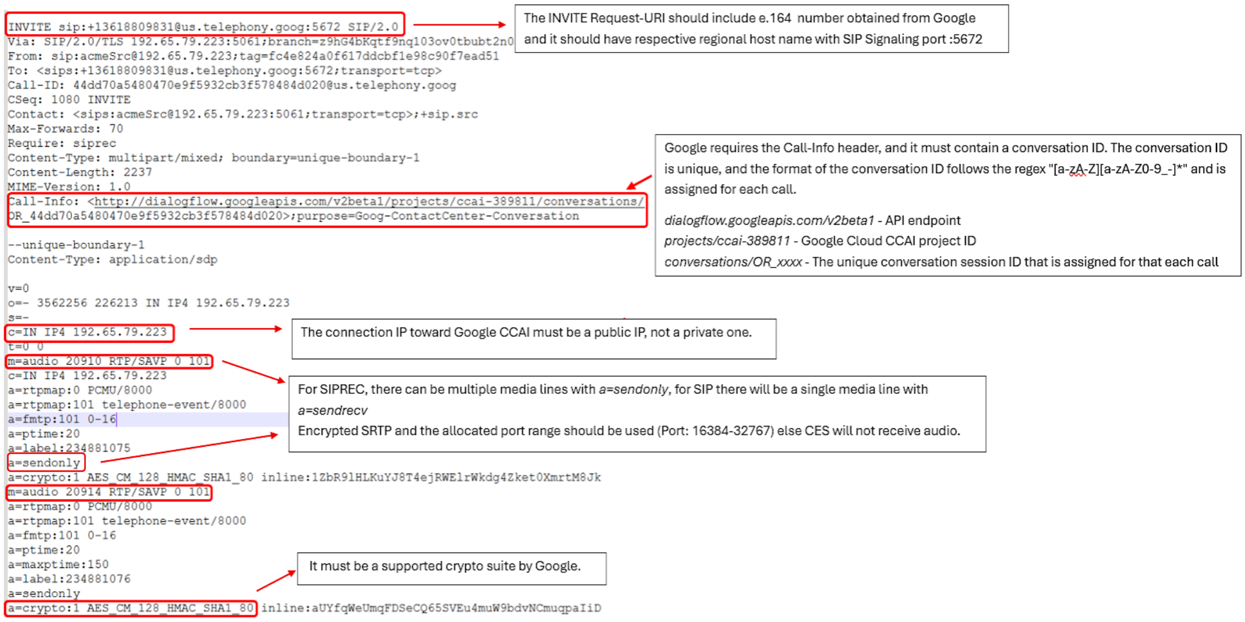
```
!
voice translation-rule 10
 rule 1 /^\+19728522626$/ /9728522667/
!
voice translation-rule 9000
 rule 1 /9728522626/ /+1314XXXXXXX/
!
!
voice translation-profile 9000
 translate called 9000
!
voice translation-profile FROM-GOOGLE
 translate called 10
!
license udi pid C8000V sn 9FQTYW5MY8J
license boot level network-essentials addon dna-essentials
memory free low-watermark processor 165741
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
enable secret 9 $9$Tsw1wxQOc7H6f.$mRFCtcyErPp/a8B5sXeXzbvVXnedvUCr5GkNlYOqUvl
!
username admin privilege 15 secret 9
$9$YsVxQ3s5SS7CEk$z7C9Zd5/gpyJtMdlYmCNGSE.mOA3oB1palQE2xjbAZ2
!
redundancy
!
interface GigabitEthernet1
 description Interface to PBX & PSTN
 ip address 10.80.X.X 255.255.255.0
 negotiation auto
!
interface GigabitEthernet2
 description to Google CES
 ip address 192.65.79.X 255.255.255.0
 negotiation auto
!
interface GigabitEthernet3
 no ip address
 negotiation auto
!
interface GigabitEthernet4
 no ip address
 negotiation auto
!
ip forward-protocol nd
```

```
no ip forward-protocol udp
!
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 0.0.0.0 192.65.79.X
ip route 216.0.0.0 255.0.0.0 192.65.79.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
 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
!
dial-peer voice 1000 voip
 description outbound to pstn
 destination-pattern 214.......
 session protocol sipv2
 session target ipv4:10.64.X.X:5060
 session transport tcp
 voice-class codec 1 offer-all
 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
 no vad
!
!
sip-ua
 no remote-party-id
 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:+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

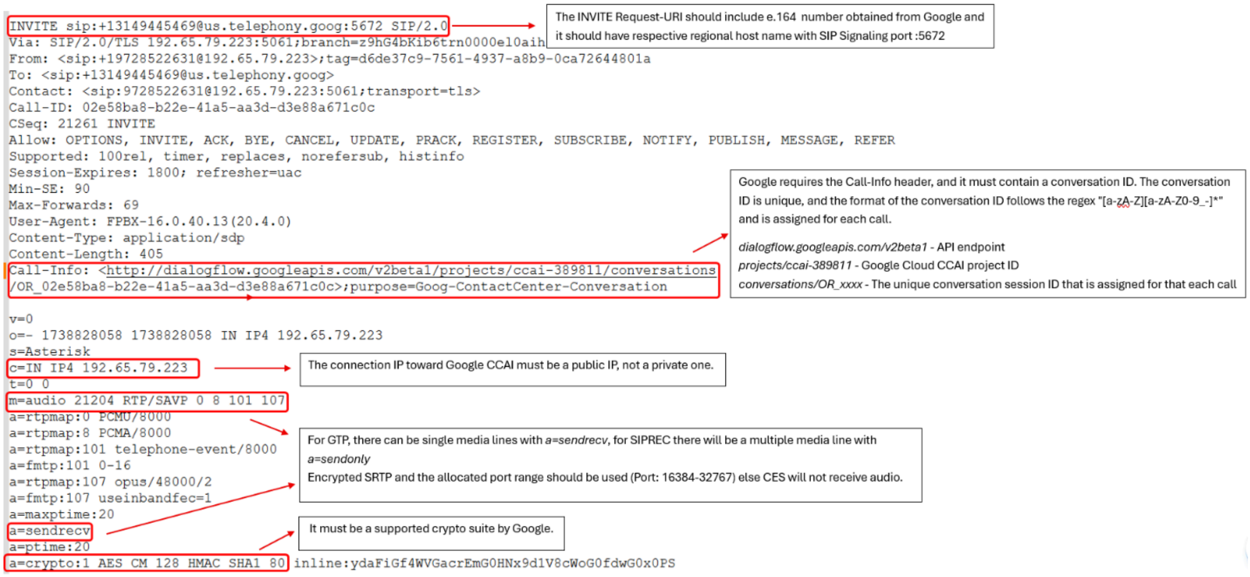
The connection IP toward Google CCAI must be a public IP, not a private one.

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 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 gets successfully connected to 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	Two calls having the same call-info has both conversation details.	PASSED	Both call transcripts are present for the same conversation session id.

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		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.			
37	Live Agent Transfer	Call goes to virtual agent, initial live agent handoff and verify outgoing SIP INVITE, call connection and disconnection		PASSED	Call gets connected successfully to BOT and INVITE sent successfully to connect with Agent.
38	Live Agent Transfer	Call goes to virtual agent, initial live agent handoff and verify outgoing SIP INVITE, call connection and disconnection. Make calls to +13149445469, "speak to an agent", +1-972-852-2626 should then ring and get connected as the agent		PASSED	Call was connected successfully with live agent and when performing conversation "Speak to an agent", a new INVITE was sent from Google to transfer to an agent and call gets connected successfully with both-way audio.
39	UUI headers	Call goes to virtual agent, say "end the call", validate that the SIP BYE has a UUI header Make calls to +13149445469, "end the call", check SIP BYE and ensure there is one or more (identify if there are 3 or 1) UUI		PASSED	Call gets connected successfully to live agent and when performing conversation "End the call", the call gets disconnected normally with 3 UUI header.

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		headers with purpose Goog-Session-Param			
4 0	SIP REFER	Call goes to virtual agent, say "send a sip refer", validate that a SIP REFER is received to 972-852-2626. Make calls to +13149445469, "send a sip refer", SIP REFER should be received with refer to set to 972-852-2626		PASSED	REFER request & transfer is successful.