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

TekVizion

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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 Call Recording** using **Cisco Unified Border Element (Cisco UBE) v17.15.4 and Cisco UBE Media Proxy 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 Call Recording with Cisco UBE v17.15.4 and Cisco UBE Media Proxy v17.15.4 configuration.

- Google CES SIPREC solution supports only TLS/SRTP.
- Hence as per the recommendations from Cisco, Cisco UBE Media Proxy is included in the topology to fork the RTP streams sent from the Cisco UBE to SRTP streams towards Google CES.

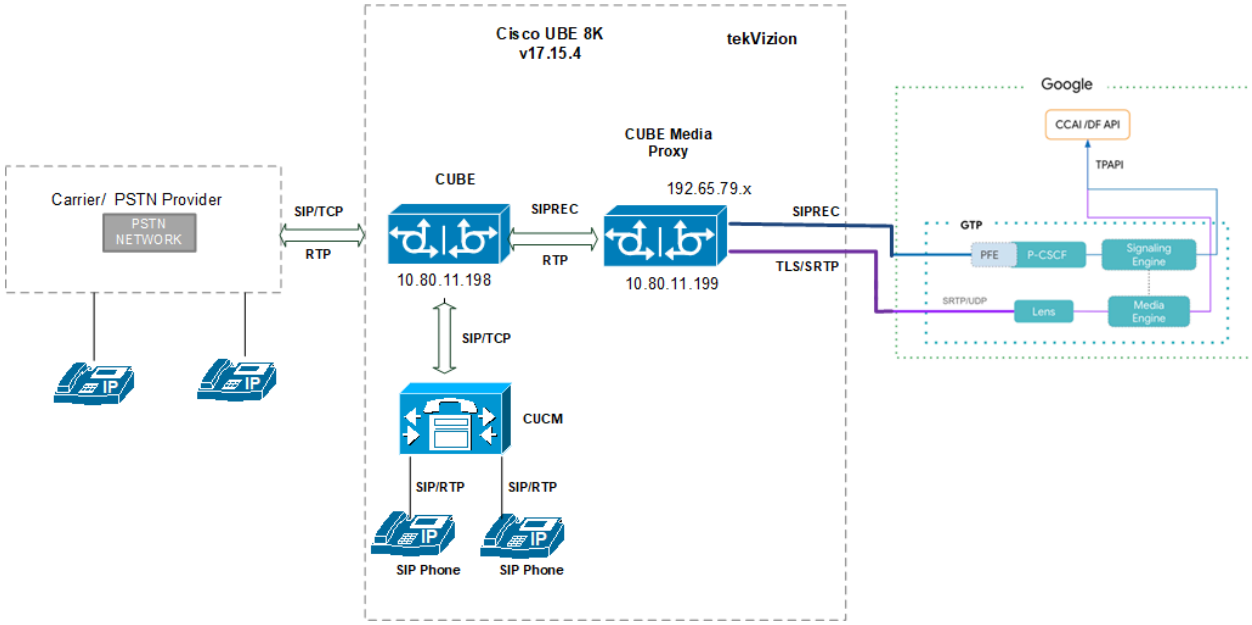

Figure 1: SIP Trunk Lab Reference Network

The lab network consists of the following components.

- Google CES Cloud Environment
- Cisco UBE v17.15.4
- Cisco UBE Media Proxy v17.15.4
- OnPrem PBX (Cisco Unified Communications Manager v15.0.1.11901-2)

3 Hardware Components

- Cisco UBE C8300
- Cisco UBE Media Proxy C8300

4 Software Requirements

- Cisco UBE C8300 V17.15.4
- Cisco UBE Media Proxy C8300 V17.15.4
- Cisco Unified Communications Manager v15.0.1.12900-234 (OnPrem PBX)

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
Cisco UBE Virtual SBC	17.15.03a
Cisco UBE Virtual SBC	17.15.01a

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

6.4 Failed Testcase

- None

7 Configuration

7.1 Configuration Checklist

Below are the steps that are required to configure Cisco UBE and Cisco UBE Media Proxy.

Table 2 – Cisco UBE and Cisco UBE Media Proxy Configuration Steps

Step	Description	Reference
Cisco UBE		
Step 1	IP Networking	Section 7.4.1
Step 2	Routing	Section 7.4.2
Step 3	Global Cisco UBE Settings	Section 7.4.3
Step 4	Codecs	Section 7.4.4
Step 5	Dial-peer Groups	Section 7.4.5
Step 6	Tenant	Section 7.4.6
Step 7	SIPREC Configuration	Section 7.4.7
Step 8	Dial Peer	Section 7.4.8
Step 9	Running configurations	Section 7.4.9
Cisco UBE Media Proxy		
Step 1	IP Networking	Section 7.5.1
Step 2	Routing	Section 7.5.2
Step 3	DNS Servers	Section 7.5.3
Step 4	Certificates	Section 7.5.4
Step 5	Import Signed Host Certificate	Section 7.5.5
Step 6	Trusted CA Trust point for Google CES	Section 7.5.6
Step 7	Default Trust point and TLS Version	Section 7.5.7
Step 8	Global Cisco UBE Media Proxy Settings	Section 7.5.8
Step 9	Message Handling Rules	Section 7.5.9
Step 10	SRTP Crypto	Section 7.5.10
Step 11	Translation Rule	Section 7.5.11
Step 12	SIPREC Configuration	Section 7.5.12

Step 13	Dial Peer	Section 7.5.13
Step 14	Message Handling Rules for Participation Label.	Section 7.5.14
Step 15	Running Configurations	Section 7.5.15

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 Media Proxy	
LAN IP Address	10.80.X.X
WAN IP Address	192.65.X.X
Cisco UBE	
LAN IP Address	10.80.X.X

7.3 Google CES API Configuration

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

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

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

7.4 Cisco UBE Configuration

The following is the example configuration of Cisco UBE for Google CES Call Recording

7.4.1 IP Networking

Below is the interface configuration towards OnPrem PBX and PSTN & Media Proxy

```
interface GigabitEthernet2
  description Interface to PBX,PSTN & Media Proxy
   ip address 10.80.X.X 255.255.255.0
   negotiation auto
!
```

7.4.2 Routing

Below is the static route configured towards Cisco UBE Media proxy, PSTN Gateway & OnPrem PBX.

```
ip route 0.0.0.0 0.0.0.0 10.80.X.X
ip route 10.64.X.X 255.255.0.0 10.80.X.X
!
```

7.4.3 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 10.64.X.X 255.255.255.0
  ipv4 10.80.X.X 255.255.255.0
 address-hiding
 mode border-element
 allow-connections sip to sip
 trace
 sip
  session refresh
  error-passthru
  early-offer forced
  sip-profiles inbound
!
```

Explanation

Command	Description
ip address trusted list	To allow all traffic between PSTN Gateway and CUCM
allow-connections sip to sip	Allows back-to-back user agent connections between two SIP call legs

session refresh	Ensures that a SIP session remains active
sip-profiles inbound	The set of rules or configurations that modify incoming Session Initiation Protocol (SIP) messages

7.4.4 Codecs

The below Voice class codec is used between PSTN gateway and OnPrem PBX. This dial peer group is used in *Section 7.4.8 - dial peer*

```
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g711alaw
!
```

7.4.5 Dial-peer Groups

Below is the dial-peer group configured to route the call from PSTN Gateway to OnPrem PBX. This dial peer group is used in *Section 7.4.8 - dial peer 301*

```
voice class dpg 101
 description incoming PSTNGW to PBX
 dial-peer 400 preference 1
!
```

Below is the dial-peer group configured to route the call from OnPrem PBX to PSTN Gateway. This dial peer group is used in *Section 7.4.8 - dial peer 302*

```
voice class dpg 103
 description incoming PBX to PSTNgw
 dial-peer 1000 preference 1
!
```

7.4.6 Tenant

Below is the tenant configuration towards PSTN Gateway. This is used in *Section 7.4.8 - dial peer 1000*

```
voice class tenant 100
 session transport tcp
 bind control source-interface GigabitEthernet2
 bind media source-interface GigabitEthernet2
!
```

7.4.7 SIPREC Configuration

Below is the Media class configured in Cisco UBE to enable SIPREC recording. This is used in *Section 7.4.8 – dial peer 1000 and 400*

```
media class 300
 recorder parameter siprec
```

```
 media-recording 900
!
```

7.4.8 Dial Peer

Below are the outbound dial-peers configured to route the calls towards PSTN Gateway, PBX and Cisco UBE Media Proxy

```
dial-peer voice 1000 voip
 description outbound to pstn
 destination-pattern 214.......
 session protocol sipv2
 session target ipv4:10.64.X.X:5060
 voice-class codec 1 offer-all
 voice-class sip tenant 100
 voice-class sip session refresh
 media-class 300
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 400 voip
 description outbound to PBX
 destination-pattern 972852266.
 session protocol sipv2
 session target ipv4:10.80.X.X:5060
  voice-class codec 1 offer-all
 voice-class sip tenant 100
 voice-class sip session refresh
 media-class 300
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 900 voip
 description outbound to ProxyCUBE
 destination-pattern 900
 session protocol sipv2
 session target ipv4:10.80.X.X:5060
 session transport udp
 voice-class codec 1 offer-all
 voice-class sip options-keepalive
 voice-class sip tenant 100
 dtmf-relay rtp-nte
 no vad
!
```

Below are the inbound dial-peers configured to receive calls from OnPrem PBX and PSTN Gateway. Dial peers are matched based on the IP configured in the Uri SIP profile

```
voice class uri 401 sip
 host ipv4:10.64.X.X
!
voice class uri 501 sip
 host ipv4:10.80.X.X
!
```

```
dial-peer voice 302 voip
 description inbound from cucm
 session protocol sipv2
 session transport tcp
 destination dpg 103
 incoming uri via 501
 voice-class sip tenant 100
 voice-class codec 1 offer-all
 voice-class sip session refresh
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 301 voip
 description inbound from PSTN
 translation-profile incoming 100
 session protocol sipv2
 destination dpg 101
 incoming uri via 401
 voice-class codec 1 offer-all
 voice-class sip tenant 100
 voice-class sip session refresh
 dtmf-relay rtp-nte
 no vad
!
```

7.4.9 Running Configurations

Building configuration...

```
version 17.15

service tcp-keepalives-in
service tcp-keepalives-out
service timestamps debug datetime msec
service timestamps log datetime msec
service call-home
platform qfp utilization monitor load 80
platform sslvpn use-pd
```

```
platform console virtual
!
hostname Cube_8k1
!
boot-start-marker
boot system bootflash:packages.conf
boot-end-marker
!
logging buffered 512000
no aaa new-model
!
ip domain name tekvizion.com
!
login on-success log
!
subscriber templating
!
!
!
!
voice service voip
 ip address trusted list
  ipv4 10.64.X.X 255.255.255.0
  ipv4 10.80.X.X 255.255.255.0
 address-hiding
 mode border-element
 allow-connections sip to sip
 trace
 sip
  session refresh
  error-passthru
  early-offer forced
  sip-profiles inbound
  no call service stop
!
voice class uri 501 sip
 host ipv4:10.80.X.X
!
voice class uri 401 sip
 host ipv4:10.64.X.X
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g711alaw
!
voice class codec 2
 codec preference 2 g711alaw
!
```

```
voice class dpg 101
 description Incoming call from PSTN to PBX
 dial-peer 400 preference 1
!
voice class dpg 103
 description incoming PBX to PSTNgw
 dial-peer 1000 preference 1
!
voice class tenant 100
 session transport tcp
 bind control source-interface GigabitEthernet2
 bind media source-interface GigabitEthernet2
 no early-offer forced
!
media class 300
 recorder parameter siprec
  media-recording 900
!
license udi pid C8000V sn 99Y8247FY8W
license boot level network-essentials addon dna-essentials
memory free low-watermark processor 68445
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
username admin privilege 15 secret 9
$9$PSFyyqxv3ffDtU$FAK1qjCSYV0JAmqxfxuEci8bll9GMqag5jeN38GyUJI
!
redundancy
!
interface GigabitEthernet1
 description Mangement IP
 ip address 10.80.X.X 255.255.255.0
 negotiation auto
!
interface GigabitEthernet2
 description Interface to PBX,PSTN & Media Proxy
 ip address 10.80.X.X 255.255.255.0
 negotiation auto
!
interface GigabitEthernet3
 no ip address
 negotiation auto
!
ip forward-protocol nd
!
ip http server
```

```
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet1
ip tftp source-interface GigabitEthernet1
ip route profile
ip route 0.0.0.0 0.0.0.0 10.80.X.X
ip route 10.64.X.X 255.255.0.0 10.80.X.X
ip route 172.16.X.X 255.255.255.0 10.80.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 400 voip
 description outbound to PBX
 destination-pattern 972852266.
 session protocol sipv2
 session target ipv4:10.80.X.X:5060
 session transport tcp
 voice-class codec 1 offer-all
 voice-class sip tenant 100
 voice-class sip session refresh
 media-class 300
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 900 voip
 description outbound to ProxyCUBE
 destination-pattern 900
 session protocol sipv2
 session target ipv4:10.80.X.X:5060
 session transport udp
 voice-class sip tenant 100
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet2
 voice-class sip bind media source-interface GigabitEthernet2
 dtmf-relay rtp-nte
 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
 voice-class codec 1 offer-all
 voice-class sip tenant 100
 voice-class sip session refresh
 media-class 300
 dtmf-relay rtp-nte
 no vad
!


dial-peer voice 302 voip
 description inbound from cucm
 session protocol sipv2
 session transport tcp
 destination dpg 103
 incoming uri via 501
 voice-class codec 1 offer-all
 voice-class sip tenant 100
 voice-class sip session refresh
dtmf-relay rtp-nte
 no vad
!
dial-peer voice 301 voip
 description inbound from PSTN
 translation-profile incoming 100
 session protocol sipv2
 session transport tcp
 destination dpg 101
 incoming uri via 401
 voice-class codec 1 offer-all
 voice-class sip tenant 100
 voice-class sip session refresh
 dtmf-relay rtp-nte
 no vad
!
sip-ua
 no remote-party-id
!
line con 0
 exec-timeout 0 0
 stopbits 1
line aux 0
line vty 0
 login local
 transport input ssh
```

```
line vty 1 4
 logging synchronous
 login
 transport preferred ssh
 transport input ssh
!
call-home
 ! If contact email address in call-home is configured as sch-smart-licensing@cisco.com
 ! the email address configured in Cisco Smart License Portal will be used as contact email
address to send SCH notifications.
 contact-email-addr sch-smart-licensing@cisco.com
 profile "CiscoTAC-1"
  destination transport-method http
ntp server X.X.X.X
end
```

7.5 Cisco UBE Media Proxy Configuration

7.5.1 IP Networking

Below is the Cisco UBE Media Proxy IP and interface IP towards Google CES

```
interface GigabitEthernet1
 description to CUBEProxy
 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
!
```

7.5.2 Routing

Below is the static route configured towards Google CES

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

7.5.3 DNS Servers

Below is the DNS server configured in the lab topology to resolve Google FQDN

```
ip name-server 8.8.8.8
```

7.5.4 Certificates

Below are the steps to create and install a certificate in Cisco UBE Media Proxy

Enter config mode and the below command 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 6.5.7 – 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>
```

7.5.5 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>
```

7.5.6 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>
```

7.5.7 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
!
```

7.5.8 Global Cisco UBE Media Proxy Settings

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

```
voice service voip
ip address trusted list
  ipv4 10.80.X.X
  ipv4 74.125.X.X
 address-hiding
 mode border-element
 allow-connections sip to sip
 trace
 sip
```

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.

7.5.9 Message Handling Rules

Manipulations for Outbound messages to Google CES

The following SIP profile is used to send the Call-Info header towards Google CES. This rule is applied to outbound SIP messages to Google CES. This rule is used in *Section 7.5.13 - 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 Media Proxy 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"
!
```

Manipulations for inbound message from Cisco UBE Media Proxy

The following SIP profile is applied towards Cisco UBE. This rule is used in *Section 6.5.13 - dial peer 1000*

- During Session refresh, Google CES sends UPDATE message towards Cisco UBE Media Proxy which in turn sends UPDATE message with Require:Siprec header towards Cisco UBE. Cisco UBE responds with 420 Bad Extension with the reason "Unsupported SIPREC".

- The below manipulation removes Require header in the UPDATE message sent from Cisco UBE Media Proxy to Cisco UBE

```
voice class sip-profiles 202
 rule 1 request UPDATE sip-header Require remove
!
```

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 7.5.13 – 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
!
```

7.5.10 SRTP Crypto

Below is the crypto cipher profile used for Google CES. The below rule is applied to *Section 7.5.13 - dial peer 9000*.

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

7.5.11 Translation Rule

Below is the Translation rule and Translation profile applied towards Google CES. This translates the incoming number pattern 900 (*Refer Section 7.4.8 – Dial peer 900*) to Google CES DID

The below rule is used in *Section 7.5.13 - dial peer 9000*

```
voice translation-rule 9000
 rule 1 /900/ /+ 1361XXXXXX/
!
!
voice translation-profile 9000
 translate calling 9000
 translate called 9000
!
```

7.5.12 SIPREC Configuration

Below Media profile is configured for secured media forking and it is associated with the Media Class. This profile is used in *Section 7.5.13 - dial peer 1000*

```
media profile recorder 9000
 media-recording proxy secure 9000
 proxy policy mandatory 9000
```

```
media class 9000
 recorder profile 9000
```

7.5.13 Dial Peers

Below is the inbound dial peer from Cisco UBE to Cisco UBE Media Proxy via UDP. Dial peer to Cisco UBE Media Proxy are matched based on the Cisco UBE IP configured in the Voice class uri profile 1000.

```
voice class uri 1000 sip
 host ipv4:10.80.X.X

dial-peer voice 1000 voip
 description inbound from cubeSBC
 translation-profile incoming 9000
 session protocol sipv2
 session transport udp
 incoming uri from 1000
 voice-class sip profiles 202
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet1
 voice-class sip bind media source-interface GigabitEthernet1
 media-class 9000
 codec g711ulaw
!
```

Below is the outbound dial peer towards Google CES via TLS

```
dial-peer voice 9000 voip
 description GoogleCES
 destination-pattern +1361XXXXXX
 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
!
```

7.5.14 Message Handling Rules for 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/Sr_XXXX?roles=HUMAN_AGENT,END_USER>;purpose=Goog-ContactCenter-Conversation.

```
voice class sip-profiles 9000
 rule 5 request ANY sip-header Call-ID modify "Call-ID: (.*)@(.*)"
"Call-ID:\1@\2\x0D\x0ACall-Info:<http://dialogflow.googleapis.com/v2beta1/projects/CES-389XXX/conversations/Sr_\1?roles=HUMAN_AGENT,END_USER>;purpose=Goog-ContactCenter-Conversation"
!
```

Note: Make sure to remove the old rule 5 and replace with this new rule 5 to perform the participation label testing.

7.5.15 Running Configurations

```
version 17.15
service timestamps debug datetime msec
service timestamps log datetime msec
service call-home
platform qfp utilization monitor load 80
platform sslvpn use-pd
platform console virtual
!
hostname Cisco_8k2_MP
!
boot-start-marker
boot system bootflash:packages.conf
boot-end-marker
!
```

```
!
logging buffered 512000
no aaa new-model
!
!
ip name-server 8.8.8.8
ip domain name tekvizion.com
!
login on-success log
!
!
subscriber templating
!
!
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
!
crypto pki trustpoint GoogleCA_1
 enrollment terminal
 revocation-check none
 hash sha256
!
!
crypto pki certificate chain sbc8
XX
      quit
crypto pki certificate chain GoogleCA_1
XX
      quit
!
!
voice service voip
 ip address trusted list
  ipv4 10.80.X.X
  ipv4 74.125.X.X
 address-hiding
 mode border-element
 allow-connections sip to sip
 trace
 sip
  early-offer forced
```

```
!
!
voice class uri 1000 sip
 host ipv4:10.80.X.X
voice class codec 1
 codec preference 1 g711ulaw
!
!
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 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 202
 rule 1 request UPDATE sip-header Require remove
!
!
voice class dpg 9000
 dial-peer 9000 preference 1
!
!
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 9000
 rule 1 /900/ /+ 1361XXXXXXX/
!
!
voice translation-profile 9000
 translate calling 9000
 translate called 9000
!
!
!
media profile recorder 9000
```

```
 media-recording proxy secure 9000
 proxy policy mandatory 9000
!
media class 9000
 recorder profile 9000
!
license udi pid C8000V sn 9FQTYW5MY8J
license boot level network-essentials addon dna-essentials
memory free low-watermark processor 68445
diagnostic bootup level minimal
!
!
spanning-tree extend system-id
!
!
username admin privilege 15 secret 9
$9$hiqJhQhhJ7mEmE$hW.ZhYTTPD8GPaayCHop/HmmFOUgXvy1GBWn89TjZmw
!
redundancy
!
!
track 1 interface GigabitEthernet1 line-protocol
!
track 2 interface GigabitEthernet2 line-protocol
!
!
!
interface GigabitEthernet1
 description to CUBEProxy
 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
!
interface GigabitEthernet3
 description Managemnet IP
 ip address 10.80.X.X 255.255.255.0
 negotiation auto
!
ip forward-protocol nd
!
ip http server
ip http authentication local
ip http secure-server
```

```
ip http secure-trustpoint sbc8
ip http client source-interface GigabitEthernet1
ip tftp source-interface GigabitEthernet1
ip route profile
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 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 1000 voip
 description inbound from cubeSBC
 translation-profile incoming 9000
 session protocol sipv2
 session transport udp
 incoming uri from 1000
 voice-class sip profiles 202
 voice-class sip bind control source-interface GigabitEthernet1
 voice-class sip bind media source-interface GigabitEthernet1
 media-class 9000
 codec g711ulaw
!
dial-peer voice 9000 voip
 description GoogleCES
 destination-pattern + 1361XXXXXXX
 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
!
!
```

```
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 4
 exec-timeout 15 0
 password xxx
 login local
 transport input ssh
!
call-home
 ! If contact email address in call-home is configured as sch-smart-licensing@cisco.com
 ! the email address configured in Cisco Smart License Portal will be used as contact email
address to send SCH notifications.
 contact-email-addr sch-smart-licensing@cisco.com
 profile "CiscoTAC-1"
  active
  destination transport-method http
ntp server 10.10.10.5
!
!
end
```

8k_1_siprecv3_backup.txt

8k_2_siprecv3_backup.txt

8 SIP INVITE to GOOGLE CES

8.1 SIP INVITE for SIPREC call

Figure 2: SIPREC call

8.2 SIP INVITE for GTP call

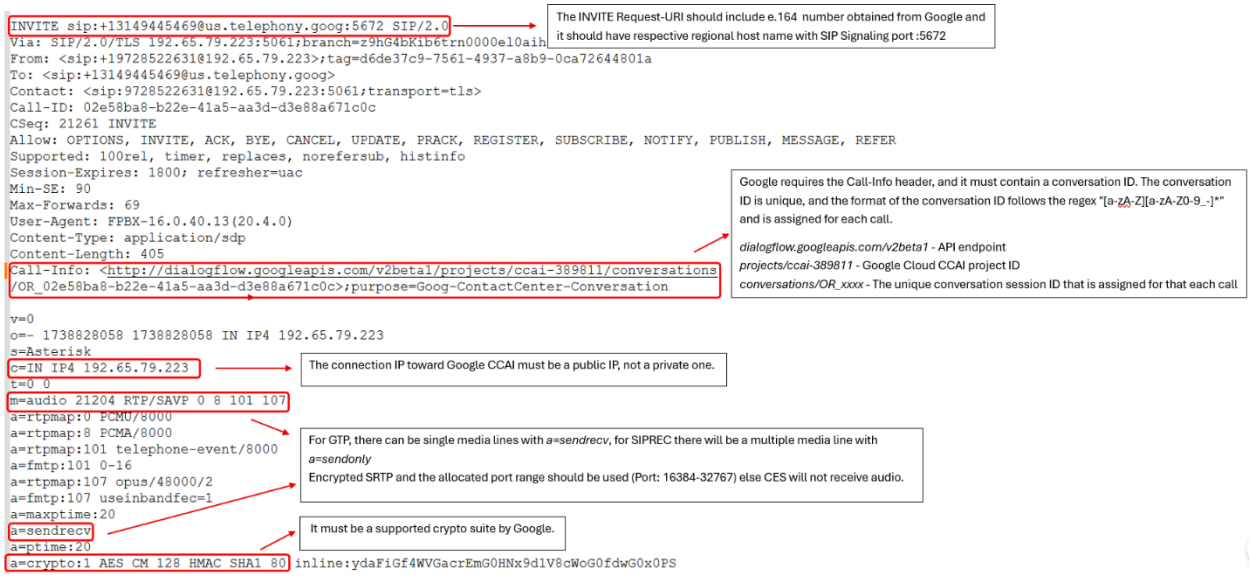

Figure 3: GTP call

9 Summary 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	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
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	TCP Keep-alive packets are sent to the Google CES
3	SBC Configuration Verification	TCP link is persistent. Establish call, 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	Cisco UBE does not send session refresh RE-INVITE. Google CES sends refreshing sessions every 15 minutes using UPDATE message
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	All signaling in e.164 format	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		HOST part change in headers etc..,			
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.	NOT SUPPORTED	Cisco UBE does not support DTLS
Inbound					
9	SIP OPTIONS	SBC send SIP options every 60 seconds	Verify SBC sends SIP OPTIONS every 60 seconds and responded with 200 OK	PASSED	
10	Inbound	Inbound call: Calling Party disconnects the call. Inbound siprec call, ensure recording 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	Verify Call is established with audio and transcripts	PASSED	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		are present, disconnect call from called party and confirm proper disconnect	from both participants Verify call is disconnected properly		
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	Cisco UBE does not send session refresh RE-INVITE. Google CES sends session refresh every 15 minutes using UPDATE message
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	Cisco UBE does not send session refresh RE-INVITE. Google CES sends session refresh every 15 minutes using UPDATE
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	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
15	Inbound	Inbound call hold scenarios. Call starts out inactive for both participants; session moves to active	Validate if media is present when expected, confirm signaling events modify sdp properly, once call is move to active validate media and transcripts	PASSED	No audio was heard when the call started as inactive and after activating conversion from Google via API, recording started.
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	NOT SUPPORTED	Audio was heard when the call starts as active and when call moved to inactive , recording is still present. When de-activating the conversation via API, Google sends a REINVITE a=inactive in SDP, for which Cube responds with a=sendonly.
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	

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
18	Inbound	Update. Validate that updates post call establishment contain SDP to modify session	If SBC uses update to modify session, ensure SDP is included	NOT SUPPORTED	Update with SDP is not supported at SBC end
19	Inbound	re-invites. Ensure re-invites that modify session include SDP	Ensure re-invites that modify session include SDP	PASSED	
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 requirements, record all leg.	Determine requirements, record all legs	PASSED	
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/particip	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 from sbc end.

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
		ants)			
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 hold scenarios using A-law as codec	Call starts out inactive for both participants; session moves to active	Inbound call hold scenarios using A-law as codec	PASSED	
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	UPDATE messages are sent from SBC to Google CES every 15min (900 seconds)

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
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/Sr_1097F06C-A4D811F0-9CC7CDC6-63FA2931?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 10/10 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.

ID	Title	Description	Expected Results	Status (Passed or Failed etc)	Observations
34	Validate Call recording			PASSED	