Connecting Google Voice SIP Link with Cisco Unified Border Element (CUBE v14.4) IOS-XE 17.6.2

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Introduction

Customers using Google Voice SIP Link have the option of connecting to the public telephone network (PSTN) using a Cisco Unified Border Element (CUBE) session border controller (SBC).

This application note describes a tested CUBE configuration for connecting Google Voice SIP Link to the PSTN or a PBX using a SIP trunk. CUBE can be configured to connect with many service providers offering SIP trunking services. Please refer to provider documentation and the content provided at www.cisco.com/go/interoperability for guidance on how to adjust this tested configuration to meet the specific requirements of your trunking service.

This document assumes the reader is knowledgeable with the terminology and configuration of Google Voice admin portal. Only CUBE configurations required for this tested solution are presented. Feature configuration and most importantly the dial plan, are customer specific so must be customized accordingly.

- This application note describes how to configure Google Voice SIP Link to the PSTN using CUBE v14.4 [IOS-XE 17.6.2].

- Testing was performed in accordance with Google Voice SIP Link test plan methodology which includes validation of basic calls, RFC2833 DTMF, call transfer, call forward, Ring Group, Auto attendant and hold/resume.

- The CUBE configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between the PSTN network and Google Voice SIP Link. The configuration described in this document details the important settings required for successful interoperability.

Network Topology
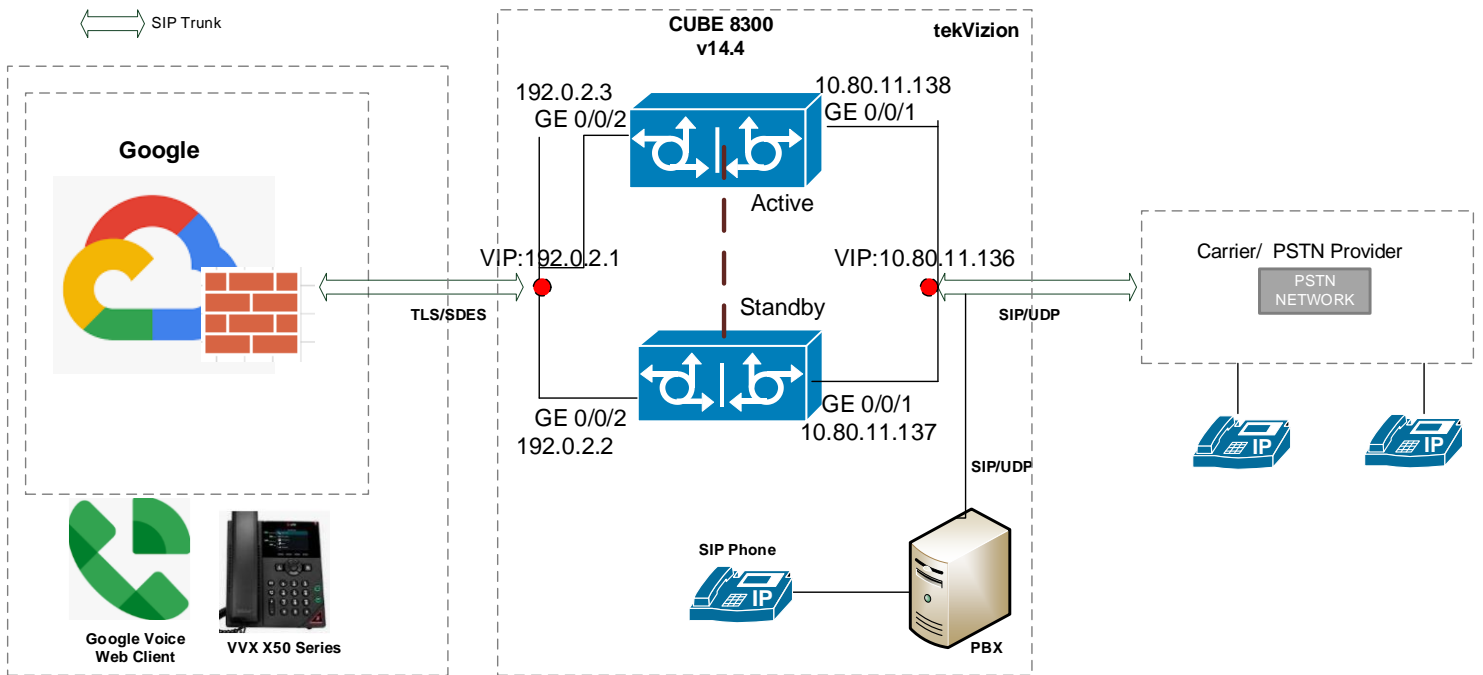
Figure 1: Network Topology

Google Voice SIP Link and CUBE Settings:

Setting	Value
Transport from CUBE to Google Voice SIP link	TLS with SRTP
Transport from CUBE and PSTN/PBX	UDP with RTP

Tested System Components

The following components were used in the testing of this solution. Please refer to product documentation for details of other supported options.

Hardware

A Cisco Catalyst Edge 8300 router was used for this tested solution. Any CUBE platform may be used though, (refer to https://www.cisco.com/go/cube) for more information.

Software Used

- CUBE v14.4 IOS-XE 17.6.2
- Poly VVX 250 OBI Edition V6.4.3.10318
- OnPrem PBX (Asterisk PBX) V16.0.17

Features

Features Supported

- Basic calls
- Call Hold and Resume
- Call Transfer
- DTMF RFC 2833
- Short Code calls
- Calling Party Number Presentation
- Calling Party Number Restricted
- Ring Group
- Auto Attendant
- Voicemail
- Call Recording
- E164 and Non-E164 dialing

Features Not Supported

- Call conference
- Linked Phone Numbers
- Call Forward
- STIR-Shaken

Caveats

The following are the observations from CUBE.

- After High Availability Switchover, if a call is immediately disconnected from Google Voice user client, the PSTN leg remains connected until the standby SBC establishes a SIP TLS connection to Google Voice.
- CUBE does not support codec preference list in SRTP to RTP
- By default, CUBE processes TCP keepalive for every 1 min when the TLS SIP OPTIONs is down/disabled.

The following are the limitations for Google Voice users

- No DTMF options are presented in the Google Voice voicemail system. A PSTN user can leave a voice message after the tone. Once the voice message is left, the caller has to disconnect the call manually and no options/announcement are given to navigate further options.
- Google Voice supports only UPDATE as a session refresh mechanism.

Google Voice SIP Link Configuration

Refer to the following for further information on how to configure Google Voice SIP Link:
support.google.com/a?p=siplink

Configuring Cisco Unified Border Element for Google Voice SIP Link

The following configuration involves the CUBE High Availability (active/standby CUBEs for stateful failover of active calls).

Licensing

Ensure that the appropriate licenses are enabled for using CUBE and TLS for the platform you are using. You will need to save your configuration and reload the platform when changing feature licenses.

For Cisco ISR 1000 Series and Cisco 4000 Series routers, use the following commands:

```
license boot level uck9
license boot level securityk9
```

For Cisco ASR 1000 Series routers, use *either* the Advanced IP services or Advanced Enterprise services with one of the following commands:

```
license boot level advipservices
license boot level adventerprise
```

For Cisco Catalyst 8300 and 8200 Series Edge Platforms, use the DNA Network Essentials feature license, or better and the required throughput level. The following example uses 25Mbps bidirectional crypto throughput, select the appropriate level for the number of calls anticipated.

```
license boot level network-essentials
platform hardware throughput crypto 25M
```

For Cisco Catalyst 8000V Edge Software, use the DNA Network Essentials feature license, or better and the required throughput level. The following example uses 1Gbps throughput, select the appropriate level for the number of calls anticipated.

```
license boot level network-essentials addon dna-essentials
platform hardware throughput level MB 1000
```

IP Networking

Note: CUBE and service provider addresses used in this guide are fictional and provided for illustration purposes only.

```
interface GigabitEthernet0/0/0
 description HA interface
 ip address 10.64.5.235 255.255.0.0
 negotiation auto

interface GigabitEthernet0/0/1
 description To PSTN and PBX
 ip address 10.80.11.137 255.255.255.0
 negotiation auto
 redundancy rii 16
 redundancy group 1 ip 10.80.11.136 exclusive
!
interface GigabitEthernet0/0/2
 description To Google Voice
 ip address 192.65.79.x 255.255.255.x
 negotiation auto
 redundancy rii 15
 redundancy group 1 ip 192.65.79.x exclusive
```

Explanation

Command	Description
redundancy rii id	Redundant interface identifier to generate virtual MAC address for HA interface. RII must be unique across networks and configured the same for equivalent interfaces in active and standby platforms.
redundancy group 1 ip 192.65.79.x exclusive	Enable Redundancy group in physical interface with virtual IP towards Google Voice.

Route To Google Voice SIP link & Internet

```
ip route 216.239.36.0 255.255.255.0 192.65.79.x
```

Route To PSTN-PBX

```
ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 172.16.0.0 255.255.0.0 10.80.11.1
```

Domain Name

Use the same domain name for the router as used for the Microsoft 365 tenant.

```
ip domain name example.com
```

DNS Servers

DNS must be configured to resolve addresses for Google trunk.

```
ip name-server 8.8.8.8
```

NTP Servers

Configure a suitable NTP source to ensure that the correct time is used by the platform.

```
ntp server 10.10.10.5
```

Certificates

The following steps describe how to create and install a certificate. The SBC TLS certificate must contain its fully qualified domain name (FQDN) as common name (CN), be 2,048 bits in size, and use RSA or ECDSA encryption. Wildcard certificates are not supported.

Generate RSA key

```
crypto key generate rsa general-keys label sbc6 exportable redundancy modulus 2048
The name for the keys will be: sbc6

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

Create SBC Trustpoint

```
crypto pki trustpoint sbc6
 enrollment terminal
 fqdn sbc6.tekvizionlabs.com
 subject-name cn=sbc6.tekvizionlabs.com
 subject-alt-name sbc6.tekvizionlabs.com
 revocation-check crl
 rsakeypair sbc6
```

Generate Certificate Signing Request (CSR)

Use this CSR to request a certificate from one of the supported Certificate authorities.

```
crypto pki enroll sbc6
% Start certificate enrollment ..

% The subject name in the certificate will include: cn=sbc6.tekvizionlabs.com
% The subject name in the certificate will include: sbc6.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, then paste the CA certificate that verifies the host certificate into the trust point (usually the intermediate certificate). Open the base 64 CER/PEM file with notepad, copy the text, and paste it into the terminal when prompted.

```
crypto pki authenticate sbc6


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

Note: Refer the running configuration for the trust point of Root CA.

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 sbc certificate


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

Specify the default trust point and TLS version to use

```
sip-ua
 transport tcp tls v1.2
 crypto signaling default trustpoint sbc6
```

Install Trusted Root Certificate Authority Bundle

To validate certificates used by Google servers, a Cisco Trusted Certificate Authority bundle must be installed as follows:

```
crypto pki trustpool import url http://www.cisco.com/security/pki/trs/ios.p7b
```

Trusted CA trust point for Google CA

As an alternative to using the Cisco CA bundle, which includes the Google CAs, individual trust points for these certificates may be created as follows.

```
crypto pki trustpoint GoogleCA1
 enrollment terminal
 revocation-check none

crypto pki trustpoint GoogleCA2
 enrollment terminal
 revocation-check none
```

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

```
crypto pki authenticate GoogleCA1
```

```
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>


crypto pki authenticate GoogleCA2


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

Exporting RSA key and certificate from CUBE 1 for High Availability

```
crypto pki export sbc6 pkcs12 ftp://<username>@x.x.x.x/ password xxxxx
Address or name of remote host [x.x.x.x]?
Destination filename [sbc6]?
Writing sbc Writing pkcs12 file to ftp://<username>@x.x.x.x/sbc6
!
CRYPTO_PKI: Exported PKCS12 file successfully.
```

Import RSA key and certificate in CUBE 2 for High Availability

Using the below command, import the certificate to CUBE 2. This will automatically create the trustpoint "sbc"

```
crypto pki import sbc6 pkcs12 ftp://<username>@x.x.x.x/sbc6 password xxxx
% Importing pkcs12...
Address or name of remote host [x.x.x.x]?
Source filename [sbc6]?
Reading file from ftp://<username>@x.x.x.x/sbc6!
[OK - 4931/4096 bytes]

CRYPTO_PKI: Imported PKCS12 file successfully.
```

Global CUBE settings

In order to enable CUBE with settings required to interwork with Google Voice, the following commands must be entered:

```
voice service voip
 ip address trusted list
  ipv4 216.239.36.0 255.255.255.255
  ipv4 10.64.1.0 255.255.255.0
  ipv4 172.16.0.0 255.255.0.0
 address-hiding
 mode border-element
 allow-connections sip to sip
 redundancy-group 1
 fax protocol pass-through g711alaw
 trace
 sip
  error-passthru
  asserted-id pai
  privacy pstn
  early-offer forced
  sip-profiles inbound
sip-ua
 transport tcp tls v1.2
 crypto signaling default trustpoint sbc6
```

Explanation

Command	Description
ip address trusted list	To allow all traffic from a peer trunk to CUBE.
allow-connections sip to sip	Allow back to back user agent connections between two SIP call legs
redundancy-group 1	Enable Redundancy group

Configure Redundancy group

```
redundancy
 mode none
 application redundancy
  group 1
    priority 150 failover threshold 75
    timers delay 30 reload 60
    control GigabitEthernet0/0/0 protocol 1
    data GigabitEthernet0/0/0
    track 1 shutdown
    track 2 shutdown
!
track 1 interface GigabitEthernet0/0/1 line-protocol
!
track 2 interface GigabitEthernet0/0/2 line-protocol
```

Explanation

Command	Description
priority 150 failover threshold 75	Set priority weight for CUBE 1 and CUBE 2. High priority CUBE turns Active and other Standby
timers delay 30 reload 60	the amount of time to delay RG group's initialization and role negotiation after the interface comes up and reload
control GigabitEthernet0/0/0 protocol 1	interface used to exchange keepalive
data GigabitEthernet0/0/0	interface used for checkpointing of data traffic

Message Handling Rules

SIP Profiles: Manipulations for outbound messages to Google Voice SIP link

The following sip profile is required to:

Rule 1 and 3: To add X-Google-PBX-Trunk-secret-key header in request and response

Rule 2: To modify SIP-Req-URI header to trunk.sip.voice.google.com:5672

Rule 4: To modify "TO" header to trunk.sip.voice.google.com:5672

Rule 5: Modify Contact header with IP to SBC FQDN

```
voice class sip-profiles 200
 rule 1 request ANY sip-header User-Agent modify "(IOS.*)" "\1\x0D\x0AX-Google-Pbx-
Trunk-Secret-Key:xxxxxxxxxxx"
 rule 2 request ANY sip-header SIP-Req-URI modify
"sip:(.*)@siplink.telephony.goog:5672" "sip:\1@trunk.sip.voice.google.com:5672"
 rule 3 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-Google-Pbx-
Trunk-Secret-Key:xxxxxxx"
 rule 4 request ANY sip-header To modify "<sip:(.*)@siplink.telephony.goog>"
"<sip:\1@trunk.sip.voice.google.com>"
 rule 5 request ANY sip-header Contact modify "<sip:(.*)@192.65.79.x:"
"<sip:\1@sbc6.tekvizionlabs.com:"
!
```

SIP Profiles: Manipulations for inbound messages from Google Voice SIP link

The following sip profile is required to:

Rule 1: Remove transport "grpc" received from Google, CUBE does not handle this transport.

Rule 2: Remove candidate attributes received from Google.

```
voice class sip-profiles 2
 rule 100 request ANY sdp-header Audio-Attribute modify "a=candidate.*"
"a=label:main-audio"
 rule 110 response ANY sdp-header Audio-Attribute modify "a=candidate.*"
"a=label:main-audio"
!
```

Options Keepalive

To ensure that contact and from headers include the SBC fully qualified domain name, the following profile is used.

```
voice class sip-profiles 201
 rule 1 request OPTIONS sip-header SIP-Req-URI modify
"sip:siplink.telephony.goog:5672" "sip:trunk.sip.voice.google.com:5672"
```

```
 rule 2 request OPTIONS sip-header To modify "<sip:siplink.telephony.goog>"
"<sip:trunk.sip.voice.google.com>"
 rule 3 request OPTIONS sip-header Contact modify "<sip:192.65.79.x:"
"<sip:sbc6.tekvizionlabs.com:"
 rule 4 request OPTIONS sip-header From modify "<sip:192.65.79.x>"
"<sip:sbc6.tekvizionlabs.com>"
 !
voice class sip-options-keepalive 200
 description OPTIONS towards Google
 transport tcp tls
 sip-profiles 201
```

SRTP crypto

Used to set the crypto cipher for the Google Voice trunk.

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

Tenant

Tenant for Google Trunk:

```
voice class tenant 200
  srtp-crypto 1
  localhost dns: sbc6.tekvizionlabs.com
  session transport tcp tls
  bind control source-interface GigabitEthernet0/0/2
  bind media source-interface GigabitEthernet0/0/2
  sip-profiles 200
  sip-profiles 2 inbound
  early-offer forced
```

Tenant to PSTN/PBX:

```
voice class tenant 100
  options-ping 60
  session transport udp
  bind control source-interface GigabitEthernet0/0/1
  bind media source-interface GigabitEthernet0/0/1
  early-offer forced
```

Number translation rules

The following translation rule applies for Google Voice SIP link sort code dialing and non +E164 from PSTN/PBX to Google Voice SIP Link in E164.

From PSTN/PBX translation rule with non +E164

```
voice translation-rule 100
 rule 1 /^\([2-9].........\)/ /+1\1/
!
voice translation-profile 100
 translate calling 100
 translate called 100
```

From Google Voice translation rule for 3-digit dialing

```
voice translation-rule 200
 rule 1 /1\(...\)/ /\1/
!
voice translation-profile 200
 translate called 200
```

Codecs

Codecs towards Google Voice

```
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
 codec preference 3 opus
 codec preference 4 g722-64
```

Codecs towards PSTN

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

Dial peers

Outbound Dial-peer to the PSTN and PBX using UDP with RTP:

```
dial-peer voice 100 voip
 description outbound to PSTN
 destination-pattern .T
 translation-profile incoming 200
 session protocol sipv2
 session target ipv4:10.64.1.x:5060
 session transport udp
 voice-class codec 2 offer-all
 voice-class sip tenant 100
 voice-class sip options-keepalive
 no voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
voice class e164-pattern-map 300
  e164 +197259801xx
  e164 +1972598011x
!
dial-peer voice 300 voip
 description outbound to PBX
```

```
  session protocol sipv2
  session target ipv4:172.16.29.18:5060
  session transport udp
  destination e164-pattern-map 300
  voice-class codec 2
  voice-class sip tenant 100
  voice-class sip options-keepalive
  no voice-class sip session refresh
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  dtmf-relay rtp-nte
  no vad
!
```

Inbound Dial-peer from the PSTN and PBX using UDP with RTP:

```
voice class uri 100 sip
 host ipv4:10.64.1.x
!
dial-peer voice 110 voip
 description inbound from PSTN
 translation-profile incoming 100
 session protocol sipv2
 session transport udp
 incoming uri via 100
 voice-class codec 2 offer-all
 voice-class sip tenant 100
 no voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
voice class uri 300 sip
 host ipv4:172.16.29.18
!
dial-peer voice 310 voip
 description inbound from PBX
 translation-profile incoming 100
 session protocol sipv2
```

```
 session transport udp
 incoming uri via 300
 voice-class codec 1
 voice-class sip tenant 100
 no voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
```

Outbound Dial-peers to Google Voice SIP Link using TLS with SRTP:

```
voice class e164-pattern-map 200
  e164 +197259800xx
  e164 +1972598010x
!
dial-peer voice 200 voip
 description outbound to Google
 session protocol sipv2
 session target dns:siplink.telephony.goog:5672
 session transport tcp tls
 destination e164-pattern-map 200
 voice-class codec 1 offer-all
 voice-class sip tenant 200
 voice-class sip localhost dns:sbc6.tekvizionlabs.com
 voice-class sip privacy-policy passthru
 voice-class sip options-keepalive profile 200
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 srtp
 no vad
!
```

Inbound Dial-peer from Google Voice using TLS with SRTP:

```
voice class uri 200 sip
 host pcscf.sip.voice.google.com
!
dial-peer voice 210 voip
 description inbound from Google
 session protocol sipv2
 session transport tcp tls
 incoming uri request 200
 voice-class codec 1
 voice-class sip tenant 200
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
```

```
 dtmf-relay rtp-nte
 srtp
 no vad
!
```

Configuration Example

The following configuration contains a sample configuration of CUBE with all parameters detailed above.

CUBE 1 (Active):

```
version 17.6
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
service call-home
platform qfp utilization monitor load 80
platform punt-keepalive disable-kernel-core
!
hostname 8K_CUBE
!
boot-start-marker
boot system bootflash:c8000be-universalk9.17.06.02.SPA.bin
boot-end-marker
!
logging buffered 2147483
!
no aaa new-model
clock timezone UTC -5 0
clock calendar-valid
!
ip name-server 8.8.8.8
ip domain name tekvizionlabs.com
!
login on-success log
!
subscriber templating
multilink bundle-name authenticated
!
crypto pki trustpoint SLA-TrustPoint
 enrollment pkcs12
 revocation-check crl
!
crypto pki trustpoint sbc6
 enrollment terminal
 fqdn sbc6.tekvizionlabs.com
 subject-name cn=sbc6.tekvizionlabs.com
 subject-alt-name sbc6.tekvizionlabs.com
 revocation-check crl
 rsakeypair sbc6
```

```
!
crypto pki trustpoint GoogleCA1
 enrollment terminal
 revocation-check none
!
crypto pki trustpoint GoogleCA2
 enrollment terminal
 revocation-check none
!
crypto pki certificate chain SLA-TrustPoint
 certificate ca 01
crypto pki certificate chain sbc6
 certificate 00A76F21D0D0E2906D
 certificate ca 07
crypto pki certificate chain GoogleCA1
 certificate ca 0203E5936F31B01349886BA217
crypto pki certificate chain GoogleCA2
 certificate ca 040000000001154B5AC394
   !
crypto pki certificate pool
 ! ('certificate ca' cmd has been deprecated. Downloaded
 !  Trustpool certificates should be re-downloaded
 !  using 'crypro pki trustpool import url <url>')!
!
voice service voip
 ip address trusted list
  ipv4 216.239.36.0 255.255.255.0
  ipv4 10.64.1.0 255.255.255.0
  ipv4 172.16.0.0 255.255.0.0
 address-hiding
 mode border-element
 allow-connections sip to sip
 redundancy-group 1
 fax protocol pass-through g711alaw
 trace
 sip
  error-passthru
  asserted-id pai
 privacy pstn
 early-offer forced
  sip-profiles inbound
!
!
```

```
voice class uri 200 sip
 host pcscf.sip.voice.google.com
!
voice class uri 300 sip
 host ipv4:172.16.29.18
!
voice class uri 100 sip
 host ipv4:10.64.1.72
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
 codec preference 3 opus
 codec preference 4 g722-64
!
voice class codec 2
 codec preference 1 g711ulaw
 codec preference 2 g711alaw
!
voice class sip-profiles 200
 rule 1 request ANY sip-header User-Agent modify "(IOS.*)" "\1\x0D\x0AX-Google-Pbx-
Trunk-Secret-Key:xxxxxxxxx"
 rule 2 request ANY sip-header SIP-Req-URI modify
"sip:(.*)@siplink.telephony.goog:5672" "sip:\1@trunk.sip.voice.google.com:5672"
 rule 3 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-Google-Pbx-
Trunk-Secret-Key:xxxxx"
 rule4 request ANY sip-header To modify"<sip:(.*)@siplink.telephony.goog>"
"<sip:\1@trunk.sip.voice.google.com>"
 rule 5 request ANY sip-header Contact modify "<sip:(.*)@192.65.79.x:"
"<sip:\1@sbc6.tekvizionlabs.com:"
!
voice class sip-profiles 2
rule 100 request ANY sdp-header Audio-Attribute modify "a=candidate.*" "a=label:main-
audio"
 rule 110 response ANY sdp-header Audio-Attribute modify "a=candidate.*"
"a=label:main-audio"
!
voice class sip-profiles 201
 rule 1 request OPTIONS sip-header SIP-Req-URI modify
"sip:siplink.telephony.goog:5672" "sip:trunk.sip.voice.google.com:5672"
 rule 2 request OPTIONS sip-header To modify "<sip:siplink.telephony.goog>"
"<sip:trunk.sip.voice.google.com>"
 rule 3 request OPTIONS sip-header Contact modify "<sip:192.65.79.x:"
"<sip:sbc6.tekvizionlabs.com:"
 rule 4 request OPTIONS sip-header From modify "<sip:192.65.79.x>"
"<sip:sbc6.tekvizionlabs.com>"
!
```

```
voice class e164-pattern-map 200
  e164 +197259800xx
  e164 +1972598010x
 !
!
voice class e164-pattern-map 300
  e164 +1972598011x
  e164 +197259801xx
 !
!
voice class sip-options-keepalive 200
 description OPTIONS towards Google
 transport tcp tls
 sip-profiles 201
!
voice class tenant 200
  srtp-crypto 1
  localhost dns:sbc6.tekvizionlabs.com
  session transport tcp tls
  bind control source-interface GigabitEthernet0/0/2
  bind media source-interface GigabitEthernet0/0/2
  sip-profiles 200
  sip-profiles 2 inbound
  early-offer forced
!
voice class tenant 100
  options-ping 60
  session transport udp
  bind control source-interface GigabitEthernet0/0/1
  bind media source-interface GigabitEthernet0/0/1
  early-offer forced
!
voice class srtp-crypto 1
 crypto 1 AES_CM_128_HMAC_SHA1_80
!
voice translation-rule 100
 rule 1 /^\([2-9].........\)/ /+1\1/
!
voice translation-rule 200
 rule 1 /1\(...\)/ /\1/
!
!
voice translation-profile 100
```

```
 translate calling 100
 translate called 100
!
voice translation-profile 200
 translate called 200
!
!
voice-card 0/1
 dsp services dspfarm
 no watchdog
!
no license feature hseck9
license udi pid C8300-1N1S-6T sn XXXX
license boot level network-essentials addon dna-essentials
memory free low-watermark processor 69096
!
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
!
redundancy
 mode none
 application redundancy
  group 1
    name cube-ha
    priority 100 failover threshold 75
    timers delay 30 reload 60
    control GigabitEthernet0/0/0 protocol 1
    data GigabitEthernet0/0/0
    track 1 shutdown
    track 2 shutdown
!
track 1 interface GigabitEthernet0/0/1 line-protocol
!
track 2 interface GigabitEthernet0/0/2 line-protocol
!
interface GigabitEthernet0/0/0
 description To HAinterface
 ip address 10.64.5.234 255.255.0.0
 negotiation auto
!
interface GigabitEthernet0/0/1
```

```
 description To PBX and PSTN
 ip address 10.80.11.137 255.255.255.0
 negotiation auto
 redundancy rii 16
 redundancy group 1 ip 10.80.11.136 exclusive
!
interface GigabitEthernet0/0/2
 description To Google
 ip address 192.65.79.x 255.255.255.x
 negotiation auto
 redundancy rii 15
 redundancy group 1 ip 192.65.79.x exclusive
!
interface GigabitEthernet0/0/3
 no ip address
 negotiation auto
!
interface GigabitEthernet0/0/4
 no ip address
 negotiation auto
!
interface GigabitEthernet0/0/5
 no ip address
 negotiation auto
!
interface Service-Engine0/1/0
!
ip tcp keepalive retries 5
ip tcp keepalive interval 7
ip http server
ip http secure-server
ip forward-protocol nd
ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 172.16.0.0 255.255.0.0 10.80.11.1
ip route 216.239.36.0 255.255.255.0 192.65.79.129
!
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
!
!
dspfarm profile 1 transcode
 codec g729abr8
 codec g729ar8
 codec g711alaw
 codec g711ulaw
 codec g722-64
 codec opus
 maximum sessions 9
 associate application CUBE
!
dial-peer voice 200 voip
 description outbound to Google
 session protocol sipv2
 session target dns:siplink.telephony.goog:5672
 session transport tcp tls
 destination e164-pattern-map 200
 voice-class codec 1 offer-all
 voice-class sip tenant 200
 voice-class sip privacy-policy passthru
 voice-class sip options-keepalive profile 200
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 srtp
 no vad
!
dial-peer voice 210 voip
 description inbound from Google
 session protocol sipv2
 session transport tcp tls
 incoming uri request 200
 voice-class codec 1
 voice-class sip tenant 200
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 srtp
```

```
 no vad
!
dial-peer voice 100 voip
 description outbound to PSTN
 destination-pattern .T
 translation-profile incoming 200
 session protocol sipv2
 session target ipv4:10.64.1.72:5060
 session transport udp
 voice-class codec 2 offer-all
voice-class sip tenant 100
 voice-class sip options-keepalive
 no voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 110 voip
 description inbound from PSTN
 translation-profile incoming 100
 session protocol sipv2
 session transport udp
 incoming uri via 100
 voice-class codec 2 offer-all
 voice-class sip tenant 100
 no voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 300 voip
 description outbound to PBX
 session protocol sipv2
 session target ipv4:172.16.29.18:5060
 session transport udp
 destination e164-pattern-map 300
 voice-class codec 2
 voice-class sip tenant 100
 voice-class sip options-keepalive
 no voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/1
```

```
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 310 voip
 description inbound from PBX
 translation-profile incoming 100
 session protocol sipv2
 session transport udp
 incoming uri via 300
 voice-class codec 2
 voice-class sip tenant 100
 no voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
sip-ua
 transport tcp tls v1.2
 crypto signaling default trustpoint sbc6
!
line con 0
 exec-timeout 5 0
 password 7 06120A2
 logging synchronous
 login
 stopbits 1
line aux 0
line vty 0 4
 exec-timeout 60 0
 password 7 15060E
 logging synchronous
 login
 transport input telnet
line vty 5 14
 login
 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
```

CUBE2 (Standby):

```
version 17.6
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
service call-home
platform qfp utilization monitor load 80
platform punt-keepalive disable-kernel-core
!
hostname 8K_Cube2
!
boot-start-marker
boot system bootflash:c8000be-universalk9.17.06.02.SPA.bin
boot-end-marker
!
logging queue-limit 2000000
logging buffered 2147483
!
no aaa new-model
clock timezone UTC -5 0
clock calendar-valid
!
ip name-server 8.8.8.8
ip domain name tekvizionlabs.com
!
login on-success log
!
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-2307055185
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-2307055185
 revocation-check none
 rsakeypair TP-self-signed-2307055185
!
crypto pki trustpoint SLA-TrustPoint
 enrollment pkcs12
 revocation-check crl
!
crypto pki trustpoint sbc6
```

```
 enrollment pkcs12
 revocation-check crl
 rsakeypair sbc6
!
crypto pki trustpoint GoogleCA1
 enrollment terminal
 revocation-check none
!
!
crypto pki trustpoint GoogleCA2
 enrollment terminal
 revocation-check none
!
crypto pki certificate chain TP-self-signed-2307055185
 certificate self-signed 01
crypto pki certificate chain SLA-TrustPoint
 certificate ca 01
crypto pki certificate chain sbc6
 certificate 00A76F21D0D0E2906D
 certificate ca 07
crypto pki certificate chain GoogleCA1
 certificate ca 0203E5936F31B01349886BA217
crypto pki certificate chain GoogleCA2
 certificate ca 040000000001154B5AC394
!
crypto pki certificate pool
 ! ('certificate ca' cmd has been deprecated. Downloaded
 !  Trustpool certificates should be re-downloaded
 !  using 'crypro pki trustpool import url <url>')!
!
voice service voip
 ip address trusted list
  ipv4 216.239.36.0 255.255.255.0
  ipv4 10.64.1.0 255.255.255.0
  ipv4 172.16.0.0 255.255.0.0
 address-hiding
 mode border-element
 allow-connections sip to sip
 redundancy-group 1
 fax protocol pass-through g711alaw
 trace
 sip
  error-passthru
```

```
   asserted-id pai
   privacy pstn
   early-offer forced
   sip-profiles inbound
!
voice class uri 200 sip
 host pcscf.sip.voice.google.com
!
voice class uri 300 sip
 host ipv4:172.16.29.18
!
voice class uri 100 sip
 host ipv4:10.64.1.0
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
 codec preference 3 opus
 codec preference 4 g722-64
!
voice class codec 2
 codec preference 1 g711ulaw
 codec preference 2 g711alaw
!
voice class sip-profiles 200
 rule 1 request ANY sip-header User-Agent modify "(IOS.*)" "\1\x0D\x0AX-Google-Pbx-
Trunk-Secret-Key:xxxxxxxx"
 rule 2 request ANY sip-header SIP-Req-URI modify
"sip:(.*)@siplink.telephony.goog:5672" "sip:\1@trunk.sip.voice.google.com:5672"
 rule 3 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-Google-Pbx-
Trunk-Secret-Key:xxxx"
 rule 4 request ANY sip-header To modify "<sip:(.*)@siplink.telephony.goog>"
"<sip:\1@trunk.sip.voice.google.com>"
 rule 5 request ANY sip-header Contact modify "<sip:(.*)@192.65.79.x:"
"<sip:\1@sbc6.tekvizionlabs.com:"
!
voice class sip-profiles 2
 rule 100 request ANY sdp-header Audio-Attribute modify "a=candidate.*"
"a=label:main-audio"
 rule 110 response ANY sdp-header Audio-Attribute modify "a=candidate.*"
"a=label:main-audio"
!
voice class sip-profiles 201
 rule 1 request OPTIONS sip-header SIP-Req-URI modify
"sip:siplink.telephony.goog:5672" "sip:trunk.sip.voice.google.com:5672"
```

```
 rule 2 request OPTIONS sip-header To modify "<sip:siplink.telephony.goog>"
"<sip:trunk.sip.voice.google.com>"
 rule 3 request OPTIONS sip-header Contact modify "<sip:192.65.79.x:"
"<sip:sbc6.tekvizionlabs.com:"
 rule 4 request OPTIONS sip-header From modify "<sip:192.65.79.x>"
"<sip:sbc6.tekvizionlabs.com>"
!
voice class e164-pattern-map 200
  e164 +197259800xx
  e164 +1972598010x
 !
!
voice class e164-pattern-map 300
  e164 +1972598011x
  e164 +197259801xx
 !
!
voice class sip-options-keepalive 200
 description OPTIONS towards Google
 transport tcp tls
 sip-profiles 201
!
voice class tenant 200
  srtp-crypto 1
  localhost dns:sbc6.tekvizionlabs.com
  session transport tcp tls
  bind control source-interface GigabitEthernet0/0/2
  bind media source-interface GigabitEthernet0/0/2
  sip-profiles 200
  sip-profiles 2 inbound
  early-offer forced
!
voice class tenant 100
  options-ping 60
  session transport udp
  bind control source-interface GigabitEthernet0/0/1
  bind media source-interface GigabitEthernet0/0/1
  early-offer forced
!
voice class srtp-crypto 1
 crypto 1 AES_CM_128_HMAC_SHA1_80
!
voice translation-rule 100
 rule 1 /^\([2-9].........\)/ /+1\1/
```

```
!
voice translation-rule 200
 rule 1 /1\(...\)/ /\1/
!
!
voice translation-profile 100
 translate calling 100
 translate called 100
!
voice translation-profile 200
 translate called 200
!
voice-card 0/1
 dsp services dspfarm
 no watchdog
!
no license feature hseck9
license udi pid C8300-1N1S-6T sn xxx
license boot level network-essentials addon dna-essentials
memory free low-watermark processor 69096
!
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 1
   priority 150 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/0/0 protocol 1
   data GigabitEthernet0/0/0
   track 1 shutdown
   track 2 shutdown
!
track 1 interface GigabitEthernet0/0/1 line-protocol
!
track 2 interface GigabitEthernet0/0/2 line-protocol
!
interface GigabitEthernet0/0/0
 description To HA interface
 ip address 10.64.5.235 255.255.0.0
```

```
 negotiation auto
!
interface GigabitEthernet0/0/1
 description To PSTN and PBX
 ip address 10.80.11.138 255.255.255.0
 negotiation auto
 redundancy rii 16
 redundancy group 1 ip 10.80.11.136 exclusive
!
interface GigabitEthernet0/0/2
 description To Google
 ip address 192.65.79.x 255.255.255.128
 negotiation auto
 redundancy rii 15
 redundancy group 1 ip 192.65.79.x exclusive
!
interface GigabitEthernet0/0/3
 no ip address
 shutdown
 negotiation auto
!
interface GigabitEthernet0/0/4
 no ip address
 shutdown
 negotiation auto
!
interface GigabitEthernet0/0/5
 no ip address
 shutdown
 negotiation auto
!
interface Service-Engine0/1/0
!
ip tcp keepalive retries 5
ip tcp keepalive interval 7
ip http server
ip http authentication local
ip http secure-server
ip forward-protocol nd
ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 172.16.0.0 255.255.0.0 10.80.11.1
ip route 216.239.36.0 255.255.255.0 192.65.79.129
!
```

```
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
!
dspfarm profile 1 transcode
 codec g729abr8
 codec g729ar8
 codec g711alaw
 codec g711ulaw
 codec g722-64
 codec opus
 maximum sessions 9
 associate application CUBE
!
dial-peer voice 200 voip
 description outbound to Google
 session protocol sipv2
 session target dns:siplink.telephony.goog:5672
 session transport tcp tls
 destination e164-pattern-map 200
 voice-class codec 1 offer-all
 voice-class sip tenant 200

 voice-class sip privacy-policy passthru
 voice-class sip options-keepalive profile 200
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 srtp
 no vad
!
dial-peer voice 210 voip
 description inbound from Google
 session protocol sipv2
 session transport tcp tls
 incoming uri request 200
 voice-class codec 1 offer-all
```

```
 voice-class sip tenant 200
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 srtp
 no vad
!
dial-peer voice 100 voip
 description outbound to PSTN
 translation-profile outgoing 200
 destination-pattern .T
 session protocol sipv2
 session target ipv4:10.64.1.0:5060
 session transport udp
 voice-class codec 1
 voice-class sip tenant 100
 voice-class sip options-keepalive
 no voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 110 voip
 description inbound from PSTN
 translation-profile incoming 100
 session protocol sipv2
 session transport udp
 incoming uri via 100
 voice-class codec 1
 voice-class sip tenant 100
 voice-class sip profiles 100
 no voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 300 voip
 description outbound to PBX
 session protocol sipv2
 session target ipv4:172.16.29.18:5060
```

```
 session transport udp
 destination e164-pattern-map 300
 voice-class codec 1
 voice-class sip tenant 100
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 310 voip
 description inbound from PBX
 translation-profile incoming 100
 session protocol sipv2
 session transport udp
 incoming uri via 300
 voice-class codec 1
 voice-class sip tenant 100
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
sip-ua
 transport tcp tls v1.2
 crypto signaling default trustpoint sbc6
!
line con 0
 exec-timeout 5 0
 password 7 06120A
 logging synchronous
 login
 stopbits 1
line aux 0
line vty 0 4
 exec-timeout 60 0
 password 7 0212015
 logging synchronous
 login
 transport input telnet
line vty 5 14
 login
 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
```

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