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

June 13, 2023
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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

Google Voice SIP Link and CUBE Settings:

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transport from CUBE to Google Voice SIP link</td>
<td>TLS with SRTP</td>
</tr>
<tr>
<td>Transport from CUBE and PSTN/PBX</td>
<td>UDP with RTP</td>
</tr>
</tbody>
</table>

Figure 1: Network Topology
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:

```plaintext
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:

```plaintext
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.

```plaintext
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.

```plaintext
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

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>redundancy rii id</td>
<td>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.</td>
</tr>
<tr>
<td>redundancy group 1 ip 192.65.79.x exclusive</td>
<td>Enable Redundancy group in physical interface with virtual IP towards Google Voice.</td>
</tr>
</tbody>
</table>
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:


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 xxxxx
% 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
```

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ip address trusted list</td>
<td>To allow all traffic from a peer trunk to CUBE.</td>
</tr>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow back to back user agent connections between two SIP call legs</td>
</tr>
<tr>
<td>redundancy-group 1</td>
<td>Enable Redundancy group</td>
</tr>
</tbody>
</table>
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

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

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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

<table>
<thead>
<tr>
<th>voice class codec 1</th>
</tr>
</thead>
<tbody>
<tr>
<td>codec preference 1 g711alaw</td>
</tr>
<tr>
<td>codec preference 2 g711ulaw</td>
</tr>
<tr>
<td>codec preference 3 opus</td>
</tr>
<tr>
<td>codec preference 4 g722-64</td>
</tr>
</tbody>
</table>

Codecs towards PSTN

<table>
<thead>
<tr>
<th>voice class codec 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>codec preference 1 g711ulaw</td>
</tr>
<tr>
<td>codec preference 2 g711alaw</td>
</tr>
</tbody>
</table>

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-nre
  no vad

! voice class e164-pattern-map 300
  e164 +197259801xx
  e164 +1972598011x

! dial-peer voice 300 voip
  description outbound to PBX

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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-n-te
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-n-te
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-nrt
no vad
!
Outbound Dial-peers to Google Voice SIP Link using TLS with SRTP:

```plaintext
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-nre
srtp
no vad
!
```

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

```plaintext
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-nce
srtpt
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 pcsf.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:xxxxxxx"
    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:"
!
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-pong 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-n-te
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-n-te
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-n-te
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-n-te
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-nre
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-nre
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
rsa keypair 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 04000000001154B5AC394
!

crypto pki certificate pool
! ('certificate ca' cmd has been deprecated. Downloaded
! Trustpool certificates should be re-downloaded
! using 'crypto 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\x0A-X-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\x0A-X-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.google.com>"
"<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/
  !
! 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-npe
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-nce
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-nce
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-nce
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-npe
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-npe
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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