# Let's go cisco live!



# High-Capacity Premises-based PSTN Option for Webex Calling

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# Additional sessions on IOS-XE UC (CUBE, Local Gateway, Survivability Gateway)

- BROCOL-2314 CUBE v14 Updates
- Session Room A4 Tuesday 1:45PM 2:45PM



- BRKCOL-2312 High-Capacity Premises-based PSTN Option for Webex Calling
  - Session Room A1 Wednesday 2:30PM 3:30PM
- Walk-in-Lab: LABCOL-2417 Local Gateway for Webex Calling



- Session Room A9 Thursday 10:30AM 11:30AM
- Walk-in-Lab: LABCOL-2416 Site Survivability for Webex Calling





# Agenda

- Local Gateway (LGW) overview and sizing
- Multiple Registration-based LGWs on a single CUBE
- Validate Registration-based LGW Configuration through Control Hub
- Introducing Certificate-based Local Gateway
- Configuring a Certificate-based Local Gateway
- 3<sup>rd</sup> Party SBC as a Local Gateway
- Resources

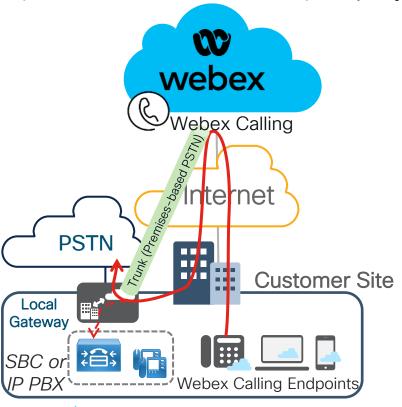


# Local Gateway (LGW) Overview and Sizing



# Webex Calling Trunk - Local Gateway

(Premises-based PSTN) Deployment



- Provides connectivity to a customerowned premises-based PSTN service
- May also provide connectivity to an onpremises IP PBX or dedicated SBC/PSTN GW
- Enables on-prem to Webex Calling transition
- Endpoint registration is NOT proxied through Local Gateway, unlike CUBE Lineside. Endpoints directly register to Webex Calling over the Internet.

**IOS-XE Software Release Mapping** 

CUBE Version	Initial IOS-XE Release for this CUBE version and Release date		Subsequent IOS-XE Release for this CUBE version	
14.1	17.3.2 <b>*</b>	Oct 2020	17.3.8a	
14.2	17.4.1a	Nov 2020	17.4.2	
14.3	17.5.1	March 2021	17.5.1a	
14.4	17.6.1a	July 2021	17.6.6a	
14.4	17.7.1a	Nov 2021	17.7.2	
14.5	17.8.1a	March 2022		
14.6	17.9.1a	July 2022	17.9.4a	
14.6	17.10.1a	Nov 2022	Last relea	
14.6	17.11.1a	March 2023	ISR44	
14.7	17.12.1a	July 2023	17.12.2	
14.8	17.13.1a	Nov 2023		
14.9	17.14.1a	March 2024		
	Version 14.1 14.2 14.3 14.4 14.4 14.5 14.6 14.6 14.6 14.7 14.8	Version       version and         14.1       17.3.2*         14.2       17.4.1a         14.3       17.5.1         14.4       17.6.1a         14.5       17.7.1a         14.6       17.9.1a         14.6       17.10.1a         14.6       17.11.1a         14.7       17.12.1a         14.8       17.13.1a	Version         version and Release date           14.1         17.3.2*         Oct 2020           14.2         17.4.1a         Nov 2020           14.3         17.5.1         March 2021           14.4         17.6.1a         July 2021           14.4         17.7.1a         Nov 2021           14.5         17.8.1a         March 2022           14.6         17.9.1a         July 2022           14.6         17.10.1a         Nov 2022           14.6         17.11.1a         March 2023           14.7         17.12.1a         July 2023           14.8         17.13.1a         Nov 2023	Version         version and Release date         CUBE version           14.1         17.3.2*         Oct 2020         17.3.8a           14.2         17.4.1a         Nov 2020         17.4.2           14.3         17.5.1         March 2021         17.5.1a           14.4         17.6.1a         July 2021         17.6.6a           14.4         17.7.1a         Nov 2021         17.7.2           14.5         17.8.1a         March 2022         17.9.4a           14.6         17.9.1a         July 2022         17.9.4a           14.6         17.10.1a         Nov 2022         17.9.4a           14.7         17.12.1a         July 2023         17.12.2           14.8         17.13.1a         Nov 2023         17.12.2

# Local Gateway Trunking models



# Local Gateway Trunking Models

- There are two types of Local Gateway trunking models:
  - Registration-based trunks
  - Certificate-based trunks
- Both models provide similar functionality, but they differ in scale and device support



# Comparing Local Gateway trunking models

Functionality	Registration-based	Certificate-based
Concurrent Calls	Concurrent calls of up to 250 per trunk (OTT Internet)	Greater than 250 concurrent calls per trunk
Device Type	Supports only CUBE (except ASR1000 series)	Supports all CUBE and 3 <sup>rd</sup> party SBCs
Authentication model	Digest-based authentication model, which relies on a shared username and password used to authenticate registration and calls.	Certificate-based authentication model
Public DNS service requirements	None	Domain claims required.  A DNS A or SRV record
cisco Lillar		must be configured in public DNS server

## Network, firewall, and NAT requirements

#### Registration-based

Any NAT or Public IP is supported.

 Dynamic NAT is preferred since it's easier for setup and requires less firewall configs

For ingress traffic, inbound pinholes(from WxC to LGW) are opened by the firewall based on outbound registration messages

Pinhole opening is recommended for all Webex Calling IP address and ports.

#### Certificate-based

Public internet-facing network including a public IP or Static NAT.

Both requires firewall to allow both ingress and egress traffic (Webex calling to Local Gateway and vice versa).

# CA and certificate requirements

Registration-based	Certificate-based
	Local gateway must have a signed certificate using one of the certificate authorities listed in Root Certificate Authorities.  • Wild-card certificates are not supported • Certificates must be signed per guidelines as mentioned in Configure Trunks, Route Groups, and Dial Plans for Webex Calling

CA bundle that signed the Webex service's certificate has to be uploaded to the Local Gateway.



# **Local Gateway**

Platform Support

Only Certificate-based supported





Audiocodes



Ribbon



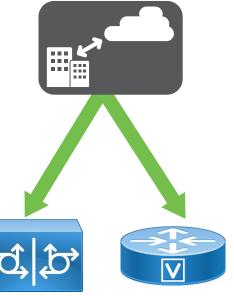
Local Gateway (LGW) **IOS-XE VGW** CUBE

Both Registration-based and Certificate-based supported

## CUBE as Local Gateway

Platform Support

Local Gateway (LGW)



IOS-XE voice GW

- Cisco CUBE (for IP-based connectivity) or Cisco IOS Gateway (for TDM-based connectivity)
- Hardware and software requirements:
  - ISR 4321, 4331, 4351, 4431, 4451, 4461 (IOS-XE 17.9.4a / 17.12.2)
  - vCUBE in AWS, Azure
  - Catalyst 8200/8300 series (IOS-XE 17.9.4a / 17.12.2)

td.15

- CSR 1000v (vCUBE) (IOS-XE 17.3.8a)
- Catalyst 8000v Edge (vCUBE) (IOS-XE 17.9.4a / 17.12.2)
  - C8000v/CSR 1000v licenses are not included in Webex Calling Flex and need to be purchased separately
  - Estimate 200 kbps total data throughput for every audio call
- ISR 1100 (IOS-XE 17.9.4a / 17.12.2)

CUBE

# Calling Capacity requirements

 Registration-based and Certificated-based trunking models have different concurrent call capacities as shown below

Concurrent calls per local gateway / trunk	Trunk type Preference	Minimum Link Quality
~ 2000-6500	Certificate-based	Interconnect
250 to ~ 2000	Certificate-based	Over the top Internet (OTT)
up to 250	Registration-based	OTT



# Connection qualifications

 Over the top (OTT) Internet and interconnect (e.g. Webex Edge Connect) must meet the following link quality conditions

Connection Type	Latency	Jitter	Packet loss
OTT	100 ms (max)	100 ms (max)	0.2%
Interconnect	30 ms	5 ms	Zero packet loss



Multiple Registration-based LGWs on a single CUBE



# Registration-based Local Gateway

Rapid deployment on an internal network behind a NAT/firewall

Internet

LGW and cloud

Security w/o certificates

Use any supported CUBE platform

Local GW registers over SIP TLS using conn. parameters from Control Hub

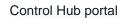
PSTN

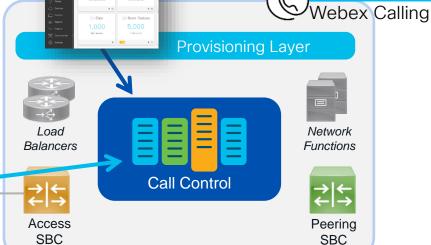
Local

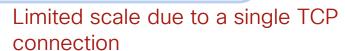


**Customer Site** 

REGISTER







webex

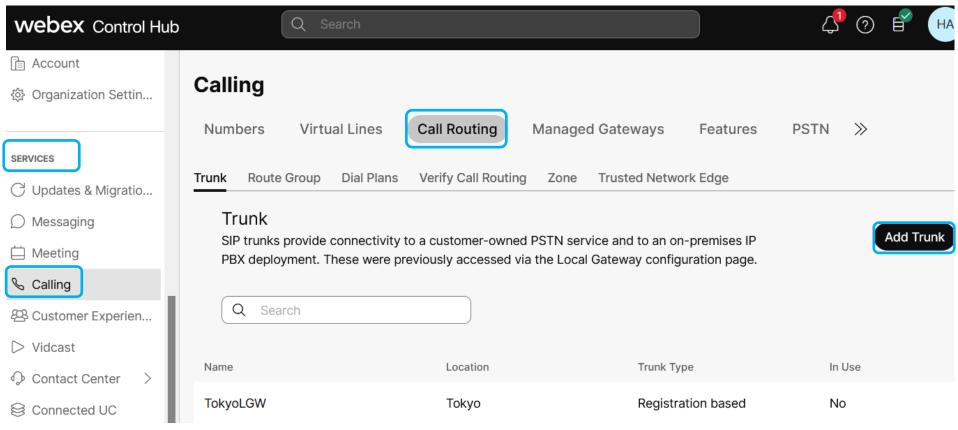
 Sensitive to network impairments (TCP throughput  $\propto$  latency/loss)



# Onboarding Process Webex Calling Trunk



# Log in to Control Hub. Navigate to Services - Click Calling and then go to the Call Routing Tab. Click Add Trunk.



# Add a new Trunk for the desired Location

#### Location

This location is where the trunk is physically connected. To create a new location, visit the Locations page.

Atlanta

#### Name



#### Trunk Type

Choose the right trunk type for this local gateway. Learn more on trunk type

Registration based  $\ \lor$ 

#### **Device Type**



 Trunk name is limited to 24 characters

**Dual Identity Support** 





# Save the Trunk parameters to build the CUBE CLI for I GW

Parameters on this display required for building LGW CLI

#### Add Trunk



#### Hussain Successfully Created.

Visit Route Group page to add trunk(s) to a route group.

Visit Locations page to configure PSTN connection to individual locations.

Visit Dial Plans page to use this trunk as the routing choice for a dial plan.

#### Trunk Info

#### Status

unknown

Trunk Group OTG/DTG hussain2572 lgu

Outbound Proxy Address la01.sipconnect-us10.cisco-bcld.com

BRKCOL-2312

Registrar Domain 40462196.cisco-bcld.com Line/Port

Hussain6346 LGU @40462196.cisco-bcld.com

#### Authentication Information

Record the username and password below. If you lose this information, you need to retrieve the username and reset the password.

Username: Hussain2572\_LGU

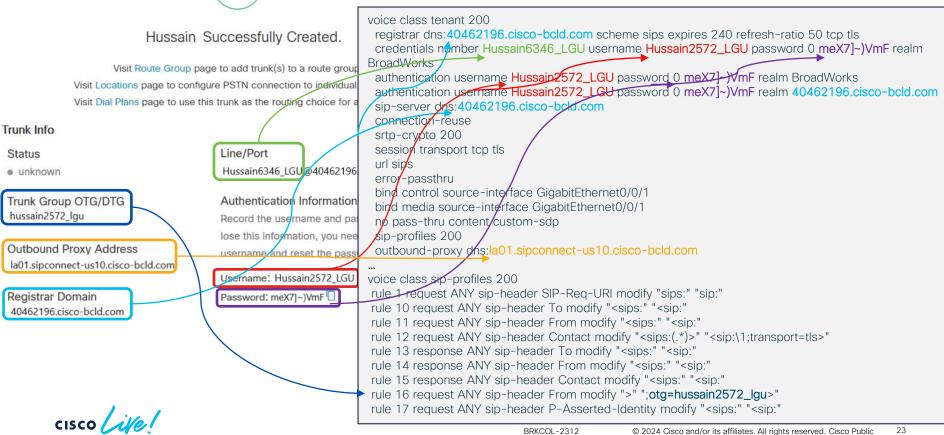
Password: meX7]~)VmF



#### Add Trunk



# Control Hub Trunk Info Connection Parameters -> LGW CLI Config



Establishing Secure Connectivity b/w LGW and Webex Calling voice class tenant 200 registrar dns:40462196.cisco-bcld.com scheme sips expires 240 refresh-ratio 50 tcp tls session transport top tls url sips webex outbound-proxy dns:la01.sipconnect-us10.cisco-bcld.com SRV Querv DNS ₩ebex Calling \_sips\_tcp.la01.sipconnect-us10.cisco-bcld.com Query Response sipconnect03a-us10.cisco-bcld.com Priority 5 Weight 50 Port 8934 sipconnect04a-us10.cisco-bcld.com Priority 10 Weight 50 Port 8934 A Query sipconnect03a-us10.cisco-bcld.com Query Response IP Address of sipconnect03a-us10.cisco-bcld.com is: (199.59.70.30) Access SBC TLS Connection to 199,59,70,30:8934 REGISTER sip:40462196.cisco-bcld.com:5061 SIP/2.0 401 REGISTER (w/credentials) 200 OK

```
pattern dtg=hussain2572 lgu
                                                            description Outgoing dial-peer to IP PSTN
                             Dial-peer structure
                                                            destination-pattern BAD.BAD
voice class dpg 100
                                                            session protocol sipv2
 description Incoming WxC(DP200201) to IP PSTN(DP101)
                                                             session target ipv4:198.18.133.3
 dial-peer 101 preference 1
                                                            voice-class codec 99
                                                            voice-class sip tenant 100
voice class dpg 200
                                                            dtmf-relay rtp-nte
description Incoming IP PSTN(DP100) to WxC(DP200201)
                                                            no vad
dial-peer 200201 preference 1
                                      Outbound PSTN Cal
                                                               Outbound IP PSTN Dial-Peer
           Cisco Webex
              Calling
                                                            ITSP SIP Trunk
                          Webex Calling
                                                                                    IP PSTN
          sip-server
                            SIP Trunk
                                      G0/0/1
                                                        G0/0/0
                                                                                  198.18.133.3
dial-peer voice 200201 voip
  description In/Out WxC
                                                               Inbound IP PSTN Dial-Peer
  max-conn 250
                                        Inbound PSTN Call
  destination-pattern BAD.BAD
                               WxC Dial-Peer
  session protocol sipv2
                                                        voice class uri 100 sip
  session target sip-server Inbound/Outbound
                                                          host ipv4:198.18.133.3
  destination dpg 100
                                                        dial-peer voice 100 voip
  incoming uri request 200
                                                          description Incoming dial-peer from IP PSTN
  voice-class codec 99
                                                          incoming uri via 100
  voice-class stun-usage 200
                                                          session protocol sipv2
  no voice-class sip localhost
                                                          destination dpg 200
  voice-class sip tenant 200
                                                          voice-class codec 99
  dtmf-relay rtp-nte
                                                          voice-class sip tenant 300
  srtp
                                                          dtmf-relay rtp-nte
  no vad
                                                          no vad
```

dial-peer voice 101 voip

voice class uri 200 sip

Local Gateway call routing based on Trunk Group Local Gateway Cisco Webex Calling voice class uri 200 sip **IP PSTN** pattern dtg=hussain2572\_lgu pattern uniquely identifies a Local gateway site ! within an Enterprise, Trunk Group OTG/DTG from ! Control Hub dial-peer voice 200201 voip description Webex Calling incoming uri request 200

INVITE Received by Local Gateway from Webex Calling

Received:

INVITE sip:+16785551234@198.18.1.226:5061;transport=tls;**dtg=hussain2572\_lgu** SIP/2.0 Via: SIP/2.0/TLS 199.59.70.30:8934;branch=z9hG4bK2hokad30fg14d0358060.1

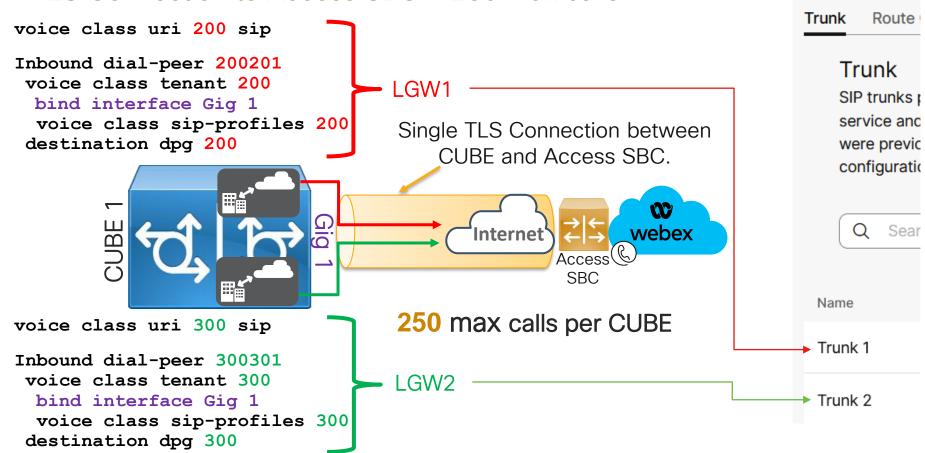


destination dpg 100

What constitutes a Registration-based LGW within a CUBE platform?

voice class sip-profiles 200 rule 20 request ANY sip-header From modify ">" ";otg= hussain2572 lqu >" voice class tenant 200 registrar dns:XXXXXX scheme sips expires 240 refresh-ratio 50 tcp tls credentials number XXXXXX username XXXXXX password 0 XXXXXX realm BroadWorks authentication username XXXXXX password 0 XXXXXX realm BroadWorks authentication username XXXXXX password 0 XXXXXX realm XXXXXX sip-server dns:XXXXXX Calling session transport tcp tls url sips bind control source-interface GigabitEthernet1 Route Group bind media source-interface GigabitEthernet1 Trunk sip-profiles 200 outbound-proxy dns:XXXXXX Trunk voice class uri 200 sip SIP trunks provide connec pattern dtg=hussain2572 lgu deployment. These were dial-peer voice 200201 voip description In/Out WxC Q Search max-conn 250 destination-pattern BAD.BAD session protocol sipv2 session target sip-server Name destination dpg 100 incoming uri request 200 voice-class sip tenant 200 Atlanta

# Single CUBE platform with two LGWs 1 TLS Connection to Access SBC = 250 max calls

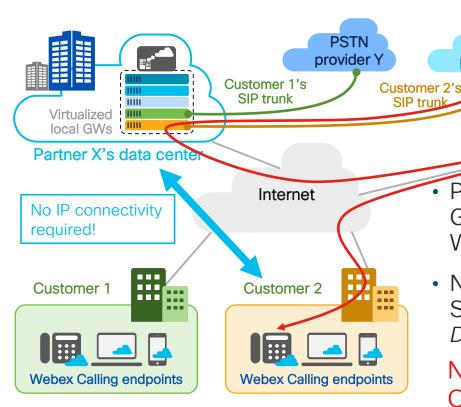


#### Single CUBE platform with two LGWs 2 TLS Connections to Access SBC = 500 max calls Trunk Route voice class uri 200 sip Inbound dial-peer 200201 Trunk voice class tenant 200 I GW1 SIP trunks p bind interface Gig 1 service and voice class sip-profiles 200 Fach I GW with its own TI S connection. destination dpg 200 were previo = 250 max calls per connection configuration W Sear webex Internet Access **SBC** Name voice class uri 300 sip 500 max calls per CUBE Trunk 1 Inbound dial-peer 300301 voice class tenant 300 I GW2 Trunk 2 bind interface Gig 2 voice class sip-profiles 300 destination dpg 300

# Partner hosted Local Gateway (Multi-tenant)

**PSTN** 

provider Z



 Partner hosts and manages customer's Local Gateway (e.g., vCUBE) - connected OTT to Webex Calling

Cisco Webex

Calling

 Not recommended if on-premises PBX or SBC is present (requires VPN between Partner DC and customer network)

Note: Registration-based LGW Concurrent Call Limits per Trunk and per TLS connection Apply

## Single vCUBE instance with two LGWs - Total <u>500</u> calls

#### Trunk1 - LGW1=250 calls

#### dial-peer voice 200201 voip

description In/Out WxC

#### max-conn 250

destination-pattern BAD.BAD session protocol sipv2 session target sip-server destination dpg 100 incoming uri request 200 voice-class sip tenant 200

#### voice class tenant 200

bind control source-interface GigabitEthernet1
bind media source-interface GigabitEthernet1

listen-port secure 5062

tls-profile 2

voice class tls-profile 2
 trustpoint CUBE-TLS

cisco Life!

#### Trunk 2 - LGW2=250 calls

### dial-peer voice 300301 voip description In/Out WxC

#### max-conn 250

destination-pattern BAD.BAD session protocol sipv2 session target sip-server destination dpg 300 incoming uri request 300 voice-class sip tenant 300

#### voice class tenant 300

bind control source-interface GigabitEthernet1 bind media source-interface GigabitEthernet1 listen-port secure 5070

tis emetile 2

tls-profile 3

voice class tls-profile 3
 trustpoint CUBE-TLS



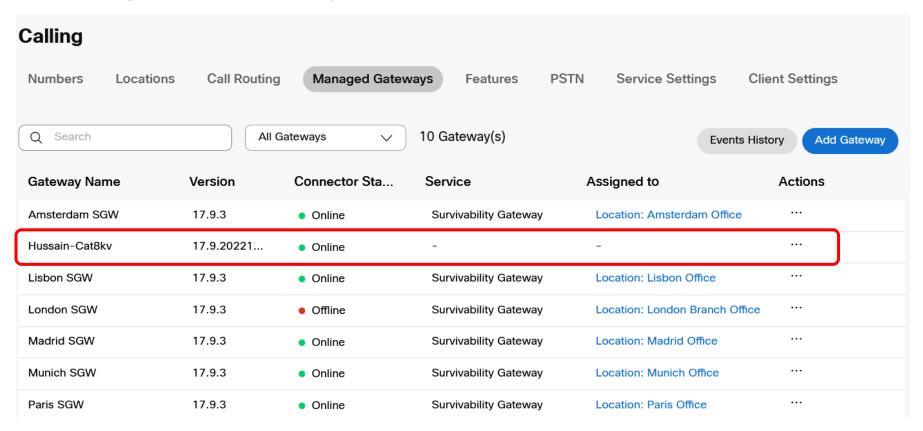
## CUBE Encrypted Audio Call Capacity

Platform  ¹CSR1Kv - Based on tests using Cisco UCS ° C240 host with Intel ° Xeon ° 6132 2.60GHz processors running VMware ESXi 6.0.	Audio IP Telephony calls RTP(G711)-RTP(G711)	Encrypted Audio (SHA1_80) calls sRTP(G711)-RTP(G711)	calls CPS	
1100 series (Default DRAM)	500	300	2	
4321 (4 GB)	500	300	1	
4331 (4 GB)	1000	600	3	
4351 (4 GB)	2000	750	4	
4431 (8 GB)	3000	750	4	
4451 (8 GB)	6000	2100 (16.12.2)	11	
<b>4461</b> (8 GB)	10000 (17.2.1r)	9900 (17.6.4)	<mark>30</mark>	
C8200L-1N-4T (4 GB)	1500 (17.5.1)	400 (17.5.1)	3	
C8200-1N-4T (8 GB)	2500 (17.4.1)	650 (17.4.1)	4	
C8300-1N1S-6T (8 GB)	7000 (17.3.2)	1600 (17.3.2)	9	
C8300-2N2S-6T (8 GB)	7500 (17.3.2)	1800 (17.3.2)	10	
C8300-1N1S-4T2X (8 GB)	8000 (17.3.2)	3500 (17.12+)	<mark>15</mark>	
C8300-2N2S-4T2X (16 GB)	10000 (17.3.2)	4300 (17.3.2)	24	
C8000V-S/CSR1Kv - 1 vCPU1 (4 GB)	1000	300	1	
C8000V-M/CSR1Kv - 2 vCPU1 (4 GB)	3000	1000	6	
C8000V-L/CSR1Kv - 4 vCPU <sup>1</sup> (8 GB)	6000	1080	6	

Validate Registrationbased LGW Configuration through Control Hub

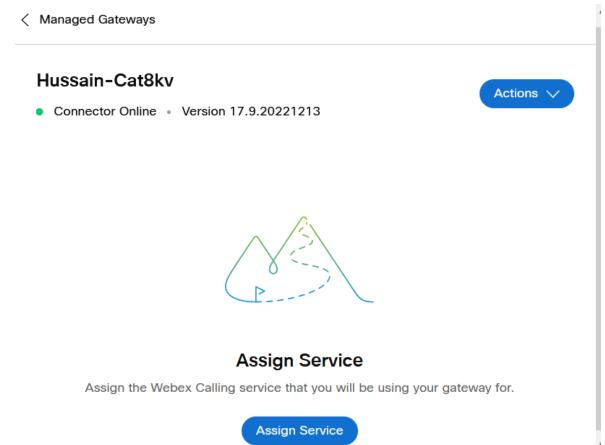


# Managed Gateway now Online





# Assign a Service to the Managed Gateway





# Select a Service Type

#### Assign Service to Hussain-Cat8kv

Select the Webex Calling service that you will be using your gateway for.

Select service type

Cancel

Assign



×

#### Service Type: LGW or SGW

#### Assign Service to Hussain-Cat8kv

Select the Webex Calling service that you will be using your gateway for.





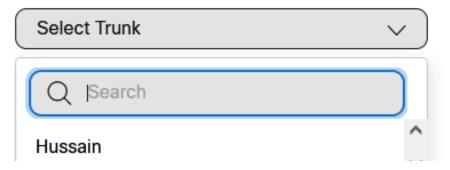
#### For Service Type Local Gateway, specify the Trunk

#### Assign Service to Hussain-Cat8kv

Select the Webex Calling service that you will be using your gateway for.



Select the trunk to assign this gateway to

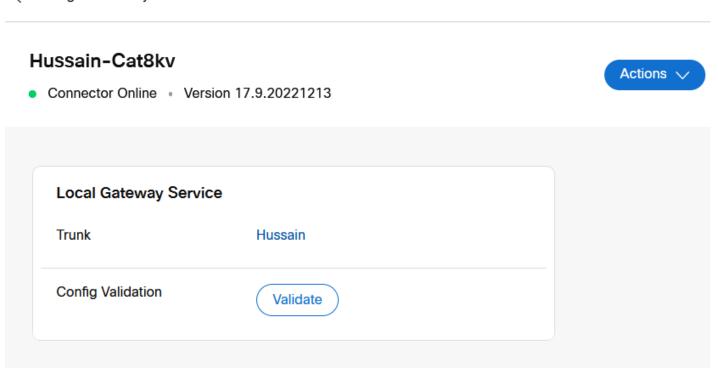






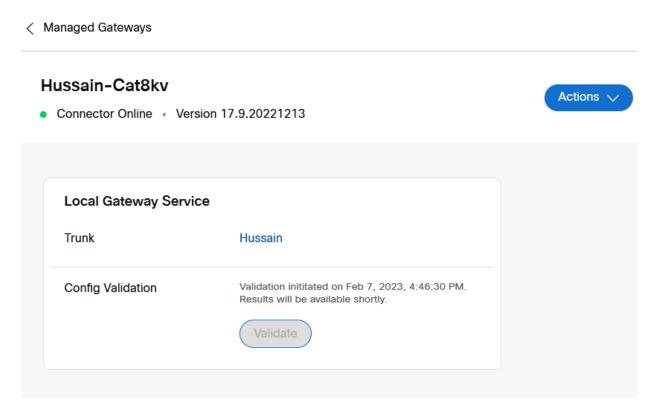
#### Validate Registration-based LGW Configuration

Managed Gateways





#### Validation takes a few minutes

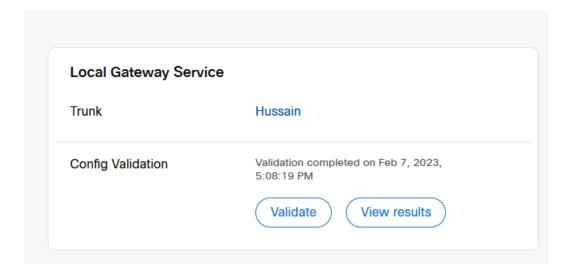


#### View Validation results

< Managed Gateways

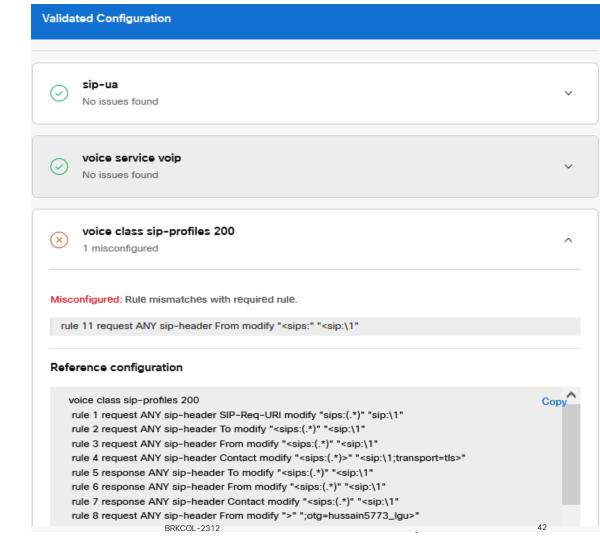
#### Hussain-Cat8kv

Connector Online Version 17.9.20221213:174319



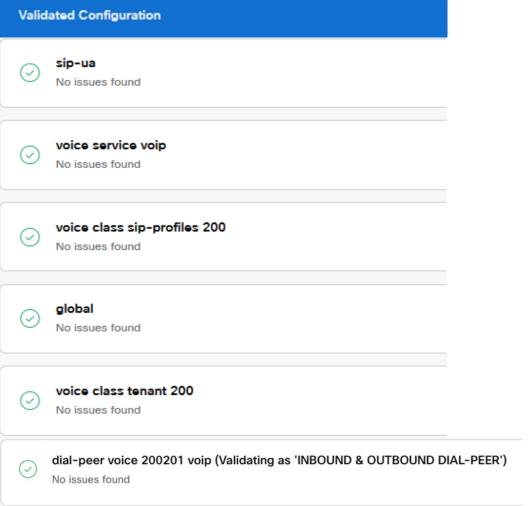


In the Validated Configuration page, verify if there are any misconfigurations





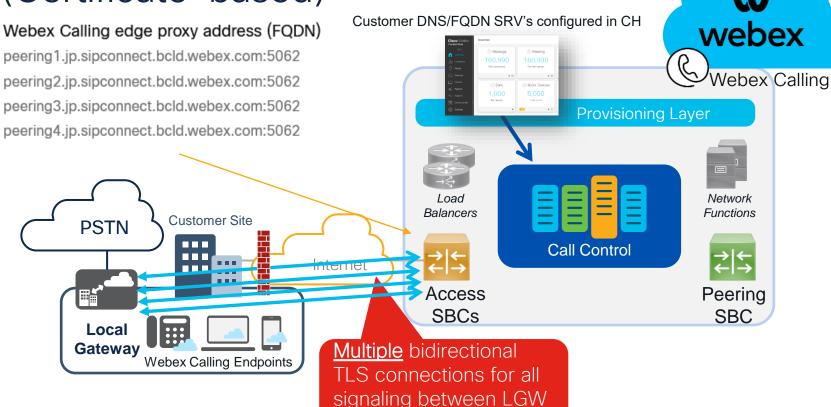
# Fix misconfigurations within the Local Gateway and run validation again



### Introducing Certificatebased Local Gateway



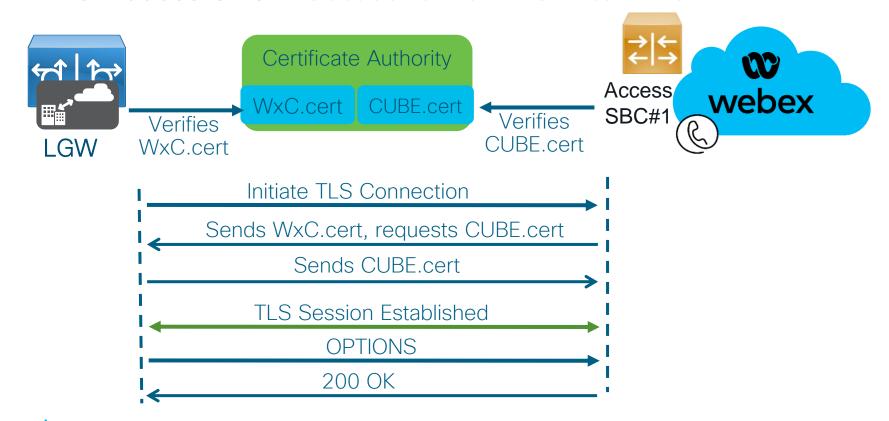
Webex Calling Trunk - Local Gateway (Certificate-based)



and cloud

BRKCOL -2312

### Certificate-based Local Gateway (Trunk Establishment) – 1st WxC Access SBC - Outbound from LGW to WxC



Certificate – based Local Gateway (Trunk Establishment) –

1st WxC Access SBC – Inbound from WxC to LGW

Certificate Authority



Now repeat the process with the 2<sup>nd</sup>, the 3<sup>rd</sup>, and the 4<sup>th</sup> WxC Access SBC

Cisco Unified Border Element

# Configuring a Certificate-based LGW

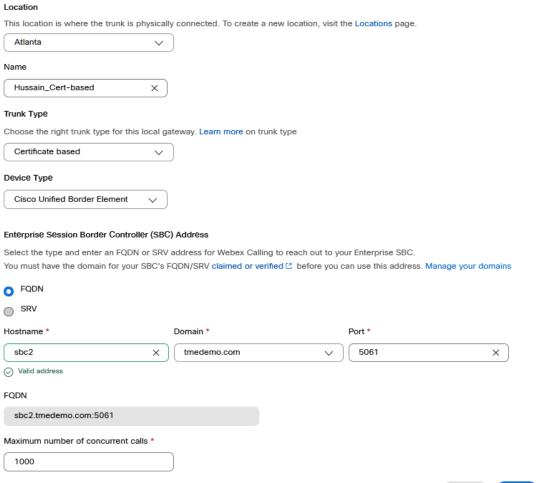


Adding a Certificatebased Trunk in Control Hub



#### Add Trunk

#### Add a Certificatebased Trunk to a Location







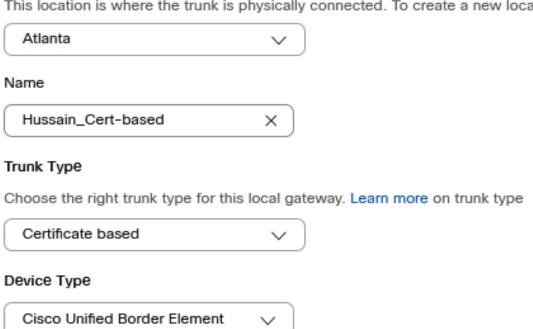


#### Adding a Trunk

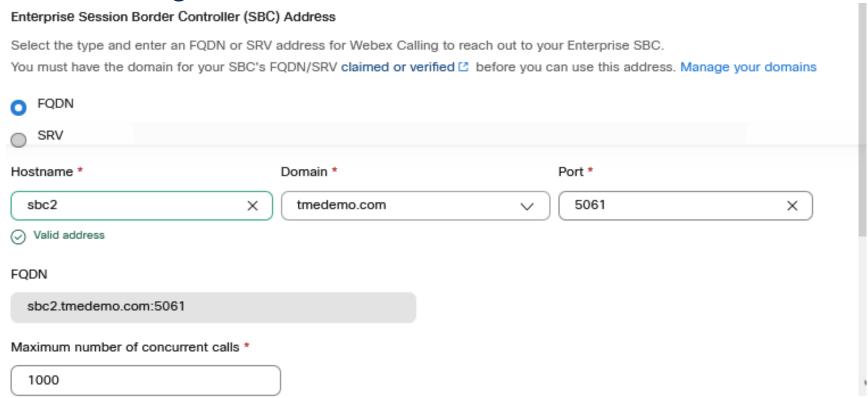
#### Add Trunk

#### Location

This location is where the trunk is physically connected. To create a new location, visit the Locations page.



# Define the LGW hostname and select to resolve the LGW through an FQDN or an SRV





Cancel

Save

# Save the Webex Calling Edge Addresses displayed Add Trunk



#### Hussain\_Cert-based Successfully Created.

Visit Route Group page to add trunk(s) to a route group.

Visit Locations page to configure PSTN connection to individual locations.

Visit Dial Plans page to use this trunk as the routing choice for a dial plan.

#### Trunk Info

Status ①

Unknown

#### Webex Calling edge proxy address (FQDN)

peering1.us.sipconnect.bcld.webex.com:5062 peering2.us.sipconnect.bcld.webex.com:5062 peering3.us.sipconnect.bcld.webex.com:5062 peering4.us.sipconnect.bcld.webex.com:5062

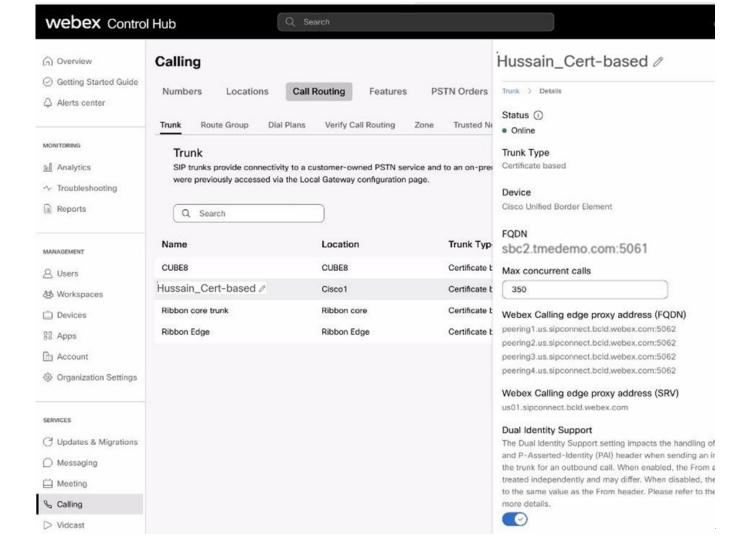
Webex Calling edge proxy address (SRV)

us01.sipconnect.bcld.webex.com





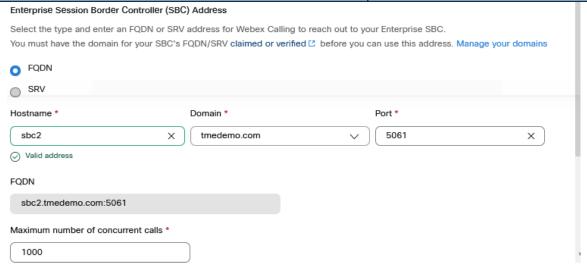
# View your trunk





#### References in this presentation

Top level Domain	tmedemo.com
SBC/CUBE's FQDN (should be publicly reachable)	sbc2.tmedemo.com
Static Public IP associated with the CUBE FQDN	198.135.2.118





# Configuring CUBE as a Certificate-based LGW



# Step by Step CUBE config: Common Global Configuration

Step 1:
Base Platform configuration and Certificates



#### CUBE Reference platform configuration

- Before proceeding with CUBE configuration, ensure baseline platform configuration such as NTPs, ACLs, enable passwords, IP routing, IP Addresses, etc. are configured according to your organization's policies and procedures
- All SIP and media ports on the external interface (Webex Calling facing) of the Local Gateway MUST be accessible to Webex Calling service and vice versa.
- Public IPv4 address(es) must be reachable from the outside and should resolve through a public DNS service
- FQDN for the LGW configured within Control Hub should resolve to this interface IP (Static NAT supported)
- IOS-XE 17.6+ is required.

interface GigabitEthernet 1
 description To Webex Calling - Public IPv4
 ip address 198.135.2.118 255.255.25.0

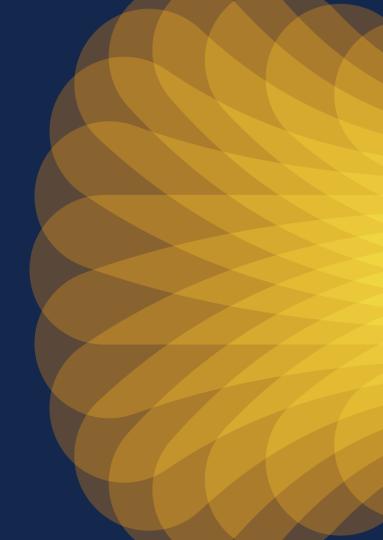
Configure IP Name Server to enable DNS lookup, Domain-name, NTP

```
CUBE#config terminal
CUBE(config)#hostname sbc2
sbc2(config)#ip domain-name tmedemo.com
sbc2(config)#ip name-server 208.67 222.222
sbc2(config)#ntp server 0.us.pool.ntp.org
```

- DNS Servers: ensure the ip name-server is reachable by successfully pinging it. Local Gateway must resolve Webex Calling proxy addresses using this DNS
- Set the same domain name for the platform as defined in Control Hub



### Certificates





#### Trust between Webex Calling and Local Gateway

- A signed certificate is required for a successful authorization and authentication of calls from the trunk. The certificate must meet the following requirements:
  - The certificate MUST be signed by a CA mentioned in What Root <u>Certificate Authorities are Supported for Calls to Cisco Webex Audio</u> and Video Platforms?
  - The trust bundle mentioned in <u>What Root Certificate Authorities are Supported for Calls to Cisco Webex Audio and Video</u>
     <u>Platforms?</u> should be uploaded on to the Local Gateway (CUBE).



# Import Cisco CA bundle for Webex Calling Certificate authentication

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

Reading file from
http://www.cisco.com/security/pki/trs/ios_core.p7b
Loading http://www.cisco.com/security/pki/trs/ios_core.p7b
% PEM files import succeeded.
```



#### Generate an RSA key pair - sbc2-key

crypto key generate rsa general-keys label sbc2-key modulus 4096 exportable

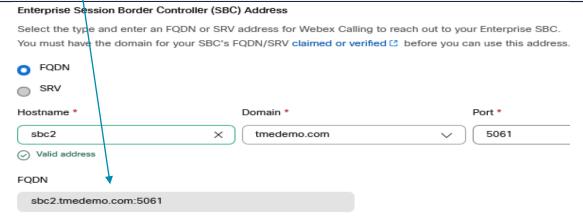
- Create an RSA key matching the certificate length of the root certificate with the above command
- Most CAs require private key size to be at least 2048 bit



# Create a PKI trustpoint to hold the CA-signed CUBE certificate using the RSA key

```
crypto pki trustpoint CUBE_CA_CERT
enrollment terminal pem
serial-number none
subject-name CN=sbc2.tmedemo.com ! (must match platform's DNS hostname through which it is reachable)
subject-alt-name sbc2.tmedemo.com
revocation-check none
rsakeypair sbc2-key ! Created previously
```

- CUBE\_CA\_CERT Trustpoint name can be anything
- Certificates MUST contain the Fully Qualified Domain Name (FQDN) as a common name or subject alternate name in the certificate with the FQDN chosen in the Control Hub



#### Generate a CSR on CUBE

#### crypto pki enroll CUBE\_CA\_CERT

```
% The subject name in the certificate will include: cn=sbc2.tmedemo.com % The subject name in the certificate will include: sbc2.tmedemo.com Display Certificate Request to terminal? [yes/no]: yes
```

- Input certificate details, make sure the LGW FQDN defined in Control Hub is present in the SAN <a href="https://www.sslshopper.com/csr-decoder.html">https://www.sslshopper.com/csr-decoder.html</a>
- Copy and save the CSR

% Start certificate enrollment...

- Send the CSR to your CA, who will send back a certificate for the host and also the root/intermediate CAs
- You may need to add the LGW FQDN (sbc2.tmedemo.com) record to your public DNS before your CA will issue you the cert

#### Paste Certificate Signing Request (CSR)

----BEGIN CERTIFICATE REQUEST-----

MIICpDCCAYwCAQAwPjEaMBgGA1UEAxMRc2JjMi5jdWJlLXRtZS5jt hkiG9w0BCQIWEXNiYzIuY3ViZS10bWUuY29tMIIBIjANBgkqhkiG9v AQ8AMIIBCgKCAQEAjBuVXcBKtrPeAHQM1ips3MxaDYlZT6e9N1hi EtIQPvVnFDjSXS2LTMx9FHnmdpEgYkGOzxVjdd0G+aVcsrG/JqtJeSi yJT86Yre9M5uvswEWiwYy/uq3nz3CDFd5NpyUa3sHYqsdnY5/nAo 2T12i3jMpIMqjoDAnP2izd/zPqJBouRPAkx5LVVGATYm1mjfcggAW KbuoE0Hgaot89mk|xVYKdTHFKZGt1xtQy8QXNMzyiXAe/ElgTbTi5I vCOzcA3ecOWrjrTsbd5hinLq654cyF1c2YVSTQIDAQABoCEwHwYJ} MRIwEDAOBgNVHQ8BAf8EBAMCBaAwDQYJKoZIhvcNAQEFBQAD DTCNQTOpzsCjqI6f5l1z6/DGISwy2Lvm5j9SdTZZ7M7NZndEcFubq c8az2Ss6i0fWP5+JxF1ptbWy1ValsA4fxSgeSHNS2nvLriy9el3F7u8H B1|5hdtqRzanCLR1||gTKRFWqOM/NHqgTWX4LpDmePlq66XAsv+a 2b3kCUGYL324Ys1+9VfuoUeSKUj4lccwNaZmRImCGF0ltgUnCUPk JeuxjTJFdu1MZtXYMfXFCV99axLEgAuGl6Acp6LtpQfvE0rgWgKv+22 Ke9XS3t4KYM=

----END CERTIFICATE REQUEST----

#### **CSR Information:**



Common Name: sbc2.cube-tme.com

# Create a PKI trustpoint to hold the Root Certificate from the Certificate Authority

```
crypto pki trustpoint Root CA CERT
 enrollment terminal
 revocation-check none
crypto pki authenticate Root CA CERT
<paste root CA X.64 based certificate here>
----BEGIN CERTIFICATE----
...! Paste this in Root_CA_CERT
----END CERTIFICATE----
```



# Create a PKI trustpoint to hold the Intermediate Certificate, **if** the root certificate has an intermediate CA

```
crypto pki trustpoint Intermediate CA
 enrollment terminal
 chain-validation continue Root CA CERT
 revocation-check none
crypto pki authenticate Intermediate CA
<paste Intermediate CA X.64 based certificate here>
----BEGIN CERTIFICATE----
...! Paste this in Intermediate_CA
----END CERTIFICATE----
```



### Authenticate and import the CA signed CUBE cert as shown below (Intermediate CA present)

```
! If the root certificate has an intermediate CA, then proceed as
! shown below. Paste in the top-level intermediate cert only that
! can authenticate the host (CUBE) cert
crypto pki authenticate CUBE CA CERT
<paste Intermediate CA X.64 based certificate here>
  --BEGIN CERTIFICATE----
...! Paste this in Intermediate CA
----END CERTIFICATE----
! Import the host(CUBE) certificate as shown below
crypto pki import CUBE CA CERT certificate
<paste CUBE CA X.64 based certificate here>
Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
    -BEGIN CERTIFICATE:
...! Paste this in CUBE CA CERT

    CUBE_CA_CERT - Trustpoint label to associate certificate

    -FND CERTIFICATE-
```

# Authenticate and import the CA signed CUBE cert as shown below (Intermediate CA NOT present)

```
! If the root certificate does not have an intermediate CA, then
! proceed as shown below. Paste in the top-level root cert only
! that can authenticate the host (CUBE) cert
crypto pki authenticate CUBE CA CERT
<paste root CA X.64 based certificate here>
   -BEGIN CERTIFICATE-
...! Paste this in Root CA CERT
----END CERTIFICATE----
! Import the host(CUBE) certificate as shown below
crypto pki import CUBE CA CERT certificate
<paste CUBE CA X.64 based certificate here>
Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
----BEGIN CERTIFICATE-
...! Paste this in CUBE CA CERT
                           CUBE_CA_CERT - Trustpoint label to associate certificate
----END CERTIFICATE-
```

# Specify the default trustpoint and TLS version under SIP-UA

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

• transport tcp tls v1.2 - Default TLS version to be 1.2



# Step by Step CUBE config: Common Global Configuration

Step 2: Trunk Enablement



#### Configure Global CUBE settings

(voice service voip)

```
voice service voip
ip address trusted list
  ipv4 X.X.X.X Y.Y.Y.Y ! Check Webex Calling Port Reference Guide
allow-connections sip to sip
no supplementary-service sip refer
no supplementary-service sip handle-replaces
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
early-offer forced
```



#### Codec Lists

```
voice class codec 100
codec preference 1 opus
codec preference 2 g711ulaw
codec preference 3 g711alaw
```



### Configure STUN to enable ICE-Lite

```
voice class stun-usage 100 stun usage ice lite
```

- Used to enable STUN with ICE-Lite
- Will be applied to all Webex Calling facing dial-peers



### Enable SRTP Crypto and SIP Profiles

```
voice class sip-profiles 100
  rule 10 request ANY sip-header Contact modify "198.135.2.118" "sbc2.tmedemo.com"
  rule 20 response ANY sip-header Contact modify "198.135.2.118" "sbc2.tmedemo.com"
!
voice class srtp-crypto 100
  crypto 1 AES_CM_128_HMAC_SHA1_80
```

- Above SIP Profile applied to all Webex Calling facing dial-peers:
  - 198.135.2.118 is the IP address of the Local Gateway interface facing Webex Calling and sbc2.tmedemo.com is the FQDN of the enterprise SBC (Local Gateway) defined within Control Hub
  - Rules 10 and 20 ensure that the Local Gateway IP address is replaced with the FQDN in the Contact header of SIP request and response messages. This is a requirement for authentication of Certificate-based Local Gateway to be used as a trunk in Webex Calling
- crypto 1 AES\_CM\_128\_HMAC\_SHA1\_80 Used to set the crypto cipher for the Webex Calling trunk.

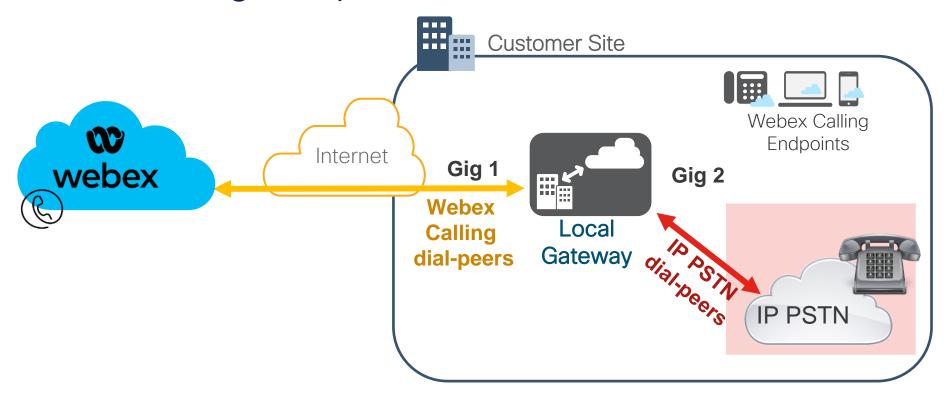


# Step by Step CUBE config:

Step 3: Call Routing



### Call Routing components





#### Outbound Dial-peers to Webex Calling peering proxies

- The following 4 dial peers are used for load balancing
  - 1. Dial-peer voice 201 voip
  - 2. Dial-peer voice 202 voip
  - 3. Dial-peer voice 203 voip
  - 4. Dial-peer voice 204 voip

#### Webex Calling edge proxy address (FQDN)

peering1.jp.sipconnect.bcld.webex.com:5062 peering2.jp.sipconnect.bcld.webex.com:5062 peering3.jp.sipconnect.bcld.webex.com:5062 peering4.jp.sipconnect.bcld.webex.com:5062

IP PSTN Inbound dial-peer 100 invokes voice class dpg 200

```
voice class dpg 200
description Incoming IP PSTN(DP100) to WxC(DP201/202/203/204)
dial-peer 201 preference 1
dial-peer 202 preference 1
dial-peer 203 preference 1
dial-peer 204 preference 1
```

 This dial-peer structure ensures LGW is maintaining multiple active bidirectional connections with Webex Calling edge proxies

#### Outbound Dial-peer 201 - Towards WxC Proxy 1

```
dial-peer voice 201 voip
description Outbound dial-peer towards Webex Calling Proxy 1
destination-pattern BAD.BAD
 session protocol sipv2
 session target dns:peering1.us.sipconnect.bcld.webex.com:5062
 session transport tcp tls
voice-class sip rel1xx disable
voice-class codec 100
voice-class stun-usage 100
voice-class sip profiles 100
voice-class sip srtp-crypto 100
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet 1
voice-class sip bind media source-interface GigabitEthernet 1
dtmf-relay rtp-nte
 srtp
no vad
```



#### Outbound Dial-peer 202 - Towards WxC Proxy 2

```
dial-peer voice 202 voip
description Outbound dial-peer towards Webex Calling Proxy 2
destination-pattern BAD.BAD
 session protocol sipv2
 session target dns:peering2.us.sipconnect.bcld.webex.com:5062
 session transport tcp tls
voice-class sip rel1xx disable
voice-class codec 100
voice-class stun-usage 100
voice-class sip profiles 100
voice-class sip srtp-crypto 100
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet 1
voice-class sip bind media source-interface GigabitEthernet 1
dtmf-relay rtp-nte
 srtp
no vad
```



### Outbound Dial-peer 203 - Towards WxC Proxy 3

```
dial-peer voice 203 voip
description Outbound dial-peer towards Webex Calling Proxy 3
destination-pattern BAD.BAD
 session protocol sipv2
 session target dns:peering3.us.sipconnect.bcld.webex.com:5062
 session transport tcp tls
voice-class sip rel1xx disable
voice-class codec 100
voice-class stun-usage 100
voice-class sip profiles 100
voice-class sip srtp-crypto 100
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet 1
voice-class sip bind media source-interface GigabitEthernet 1
dtmf-relay rtp-nte
 srtp
no vad
```



#### Outbound Dial-peer 204 - Towards WxC Proxy 4

```
dial-peer voice 204 voip
description Outbound dial-peer towards Webex Calling Proxy 4
destination-pattern BAD.BAD
 session protocol sipv2
 session target dns:peering4.us.sipconnect.bcld.webex.com:5062
 session transport tcp tls
voice-class sip rel1xx disable
voice-class codec 100
voice-class stun-usage 100
voice-class sip profiles 100
voice-class sip srtp-crypto 100
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet 1
voice-class sip bind media source-interface GigabitEthernet 1
dtmf-relay rtp-nte
 srtp
no vad
```



## Outbound WxC Dial-peer 2000 - Towards Webex Calling

edge proxy SRV address from Control Hub

```
dial-peer voice 2000 voip
description Outbound dial-peer towards WxC Edge Proxy SRV Address
destination-pattern BAD.BAD
 session protocol sipv2
 session target dns:us01.sipconnect.bcld.webex.com
 session transport tcp tls
voice-class sip rel1xx disable
voice-class codec 100
                                           Webex Calling edge proxy address (SRV)
voice-class stun-usage 100
                                           us01.sipconnect.bcld.webex.com
voice-class sip profiles 100
voice-class sip srtp-crypto 100
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet 1
voice-class sip bind media source-interface GigabitEthernet 1
dtmf-relay rtp-nte
 srtp
                               voice class dpg 200
no vad
                                description Incoming IP PSTN(DP100) to WxC(DP2000)
                                dial-peer 2000 preference 1
```

### Inbound PSTN Call

Webex Calling

SIP Trunk Gig 1

voice class dpg 200

description Incoming IP PSTN(DP100) to WxC(DP201/202/203/204)

dial-peer 201 preference 1 dial-peer 202 preference 1

dial-peer 203 preference 1

dial-peer 204 preference 1

ITSP SIP Trunk

**IP PSTN** 

Gig 2 198.18.133.3

Outbound WxC Dial-Peers #

Inbound PSTN Call

Inbound IP PSTN Dial-Peer

#### dial-peer voice 201 voip

webex

description Outbound dial-peer to Webex Calling Proxy 1 session target dns:peering1.us.sipconnect.bcld.webex.com:5062

#### dial-peer voice 202 voip

description Outbound dial-peer to Webex Calling Proxy 2 session target dns:peering2.us.sipconnect.bcld.webex.com:5062

#### dial-peer voice 203 voip

description Outbound dial-peer to Webex Calling Proxy 3 session target dns:peering3.us.sipconnect.bcld.webex.com:5062

#### dial-peer voice 204 voip

description Outbound dial-peer to Webex Calling Proxy 4 session target dns:peering4.us.sipconnect.bcld.webex.com:5062 voice class uri 100 sip host ipv4:198.18.133.3

dial-peer voice 100 voip description Incoming dial-peer from IP PSTN incoming uri via 100 session protocol sipv2 destination dpg 200 voice-class codec 100 dtmf-relay rtp-nte no vad

### Inbound Dial-peer 200 - From Webex Calling

```
FODN
voice class uri 200 sip
sbc2.tmedemo.com:5061
dial-peer voice 200 voip
description inbound from Webex Calling
session protocol sipv2
session transport tcp tls
                            voice class dpg 100
 incoming uri request 200
                             description Incoming WxC(DP200) to IP PSTN(DP101)
 destination dpg 100
                             dial-peer 101 preference 1
voice-class codec 100
voice-class stun-usage 100
voice-class sip profiles 100
voice-class sip srtp-crypto 100
voice-class sip bind control source-interface GigabitEthernet 1
voice-class sip bind media source-interface GigabitEthernet 1
 dtmf-relay rtp-nte
 srtp
no vad
```

# Outbound PSTN Call

voice class uri 200 sip pattern sbc2.tmedemo.com

dial-peer voice 200 voip description inbound from Webex Calling session protocol sipv2

session transport top tls

destination dpg 100

incoming uri request 200

voice-class codec 100 voice-class stun-usage 100 voice-class sip profiles 100 voice-class sip srtp-crypto 100

voice-class sip bind control source-interface GigabitEthernet 1 voice-class sip bind media source-interface GigabitEthernet 1 dtmf-relay rtp-nte

srtp no vad

Inbound WxC Dial-Peer

Outbound PSTN Call

Outbound IP PSTN Dial-Peer

dial-peer voice 101 voip

destination-pattern BAD.BAD

session protocol sipv2

voice-class codec 100

dtmf-relay rtp-nte

no vad

description Outgoing dial-peer to IP PSTN

session target ipv4:198.18.133.3



Webex Calling

SIP Trunk Gig 1



**ITSP SIP Trunk** 

Gig 2

IP PSTN

198.18.133.3

voice class dpg 100

description Incoming WxC(DP200) to IP PSTN(DP101)

dial-peer 101 preference 1



#### Registration-based Trunk Pros and Cons

#### Pros:

- CUBE can sit on internal network behind a NAT/firewall
  - No need for the customer to expose CUBE's external interface
  - No need for the customer to setup a DMZ
- Easier to deploy: achieves security without a need for certificates
- Recommended method
- Config Validation from the Control Hub

#### Cons:

- Limited scale (single TCP/TLS connection)
  - Scales upto 250 calls (OTT), 500+ (Interconnect)
- Sensitive to network impairments (all calls affected when TCP/TLS connection is lost)

#### Certificate-based Trunk Pros and Cons

#### Pros:

- Higher scale, up to CUBE platform limits (multiple TCP/TLS connections)
- Better resilience (each call is independent)
  - Network drop does not impact new calls as the call could land on the new connection
- Both sides (Webex Calling Access SBC and CUBE) can create connections on demand

#### Cons:

- CUBE must be reachable from the cloud (<u>public IPv4 address</u> on the external interface with inbound FW rules) [CUBE can be behind static NAT]
  - Customer will need to publish an FQDN (IOS-XE 17.6+ - current help.Webex documentation) or SRV (IOS-XE 17.9+) for WxC to reach the LGW
- Requires certificates signed by public CA on each CUBE and a DNS SRV

# 3<sup>rd</sup> Party SBC as a Local Gateway



# Oracle SBC is now Certificate-based only

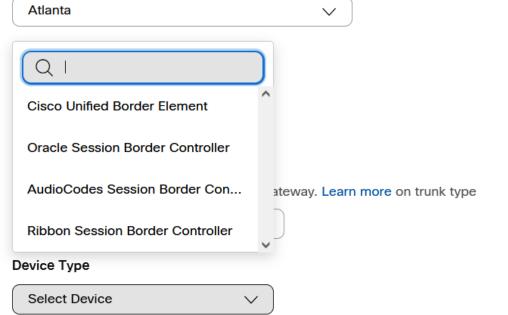
Add Trunk

#### Location

This location is where the trunk is physically connected. To create a new location

Oracle conducted tests with SBC 9.0 software – this on any of the following products:

- AP 1100
- AP 3900
- AP 4600
- AP 6300
- AP 6350
- AP 3950 (Starting from SBC 9.0 version)
- AP 4900 (Starting from SBC 9.0 version)
- VME
- Oracle SBC on Public Cloud



Configuration Guide

Enterprise Session Border Controller (SBC) Address



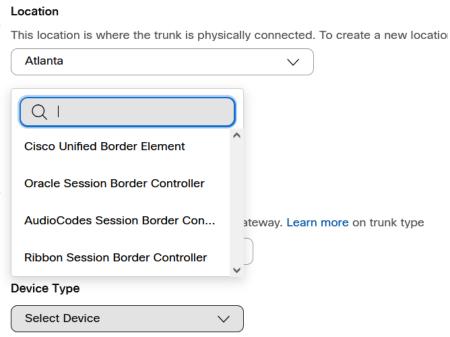
### AudioCodes SBC is now supported as LGW

Certificate-based only

Add Trunk

#### **AudioCodes** Mediant 500 Gateway & E-SBC Mediant 800B/C Gateway & E-SBC Mediant 1000B Gateway & E-SBC Mediant 2600 E-SBC Mediant 4000/B SBC Mediant 9000, 9030, 9080 SBC Mediant Software SBC (VE/SE/CE) 7.40A.250.440 or later

#### Configuration Guide



Enterprise Session Border Controller (SBC) Address

BRKCOL -2312



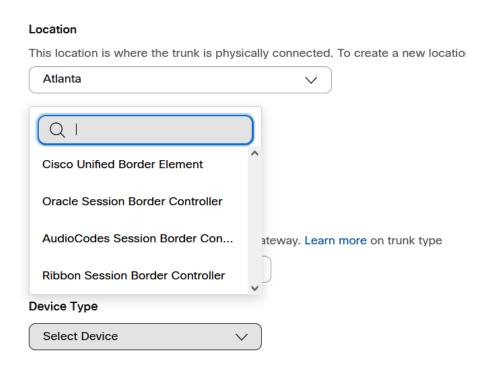
# Ribbon SBC is now supported as LGW Certificate-based only

#### **Add Trunk**

Ribbon Code Version
Platform

SBC 5000 10.1.0
SBC 7000
SBC SWe

**Configuration Guide** 



Enterprise Session Border Controller (SBC) Address



# Resources



#### Resources

For more information take a look at the following resources:

- What's new in Webex Calling: <a href="https://help.webex.com/en-us/article/rdmb0/What's-new-in-webex-Calling">https://help.webex.com/en-us/article/rdmb0/What's-new-in-webex-Calling</a>
- Trunk configuration guide: Webex Calling Trunks
- Configure Local Gateway on Cisco IOS XE for Webex Calling <u>https://help.webex.com/en-us/article/jr1i3r/Configure-Local-Gateway-on-Cisco-IOS-XE-for-Webex-Calling</u>
- https://help.webex.com/en-us/article/n0xb944/Configure-Trunks,-Route-Groups,-and-Dial-Plans-for-Webex-Calling



#### Resources

- Enroll Cisco IOS Managed Gateways to Webex Cloud
- Assign Services to Managed Gateways
- Validate Cisco Local Gateway Configuration through Control Hub
- Webex Integrations: <u>Webex Integrations</u> > Oracle
- Oracle SBC integration with Cisco Webex Calling as 3<sup>rd</sup> party Local Gateway (LGW) <a href="https://www.oracle.com/a/otn/docs/oracle-sbc-integration-with-cisco-webex-calling-v1.0.pdf">https://www.oracle.com/a/otn/docs/oracle-sbc-integration-with-cisco-webex-calling-v1.0.pdf</a>



# Additional sessions on IOS-XE UC (CUBE, Local Gateway, Survivability Gateway)

- BROCOL-2314 CUBE v14 Updates
- Session Room A4 Tuesday 1:45PM 2:45PM



- BRKCOL-2312 High-Capacity Premises-based PSTN Option for Webex Calling
  - Session Room A1 Wednesday 2:30PM 3:30PM
- · Walk-in-Lab: LABCOL-2417 Local Gateway for Webex Calling



- BRKCOL-2993 Enabling Site Survivability for Webex Calling
  - Session Room A9 Thursday 10:30AM 11:30AM
- Walk-in-Lab: LABCOL-2416 Site Survivability for Webex Calling





# Thank you





