



Connecting Cisco Webex Calling to Lumen SIP Trunk via a Certificate-based Local Gateway on Cisco Unified Border Element (CUBE) v14.5 [IOS-XE 17.8.1a]

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Introduction

Service Providers today, such as Lumen are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and centralized IP to TDM POP gateways to provide on-net and off-net services.

Lumen is service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. As an intermediate between Cisco Webex Calling and SIP trunk to Lumen, Cisco Unified Border Element (CUBE) running IOS-XE 17.8.1a can be used as a Certificate-based Local Gateway. The Cisco Unified Border Element provides demarcation, security, interworking and session control services for Cisco Webex Calling connected to Lumen.

This document assumes the reader is knowledgeable with the terminology and configuration of CUBE. The configuration settings specifically required for Webex Calling certificate based- Local gateway are presented. Feature configuration and most importantly the dial plan are customer specific and need individual approach.

- This application note describes how to configure a Certificate-based Local Gateway on CUBE for Cisco Webex Calling for connectivity to Lumen SIP Trunking service. IOS-XE 17.8.1a (CUBE v14.5) was used for this application note.
- Testing was performed in accordance with Webex Calling certificate-based local gateway test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), semi-attended, attended, and blind transfers, call forward and conference.
- The Cisco UBE configuration presented in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Lumen network and Cisco Webex Calling Certificate based Local gateway. The configuration described in this document details the important configuration settings to enable interoperability to be successful and care must be taken by the network administrator deploying Cisco Webex Calling Certificate based local gateway trunk to interoperate to Lumen SIP Trunking network.

Network Topology

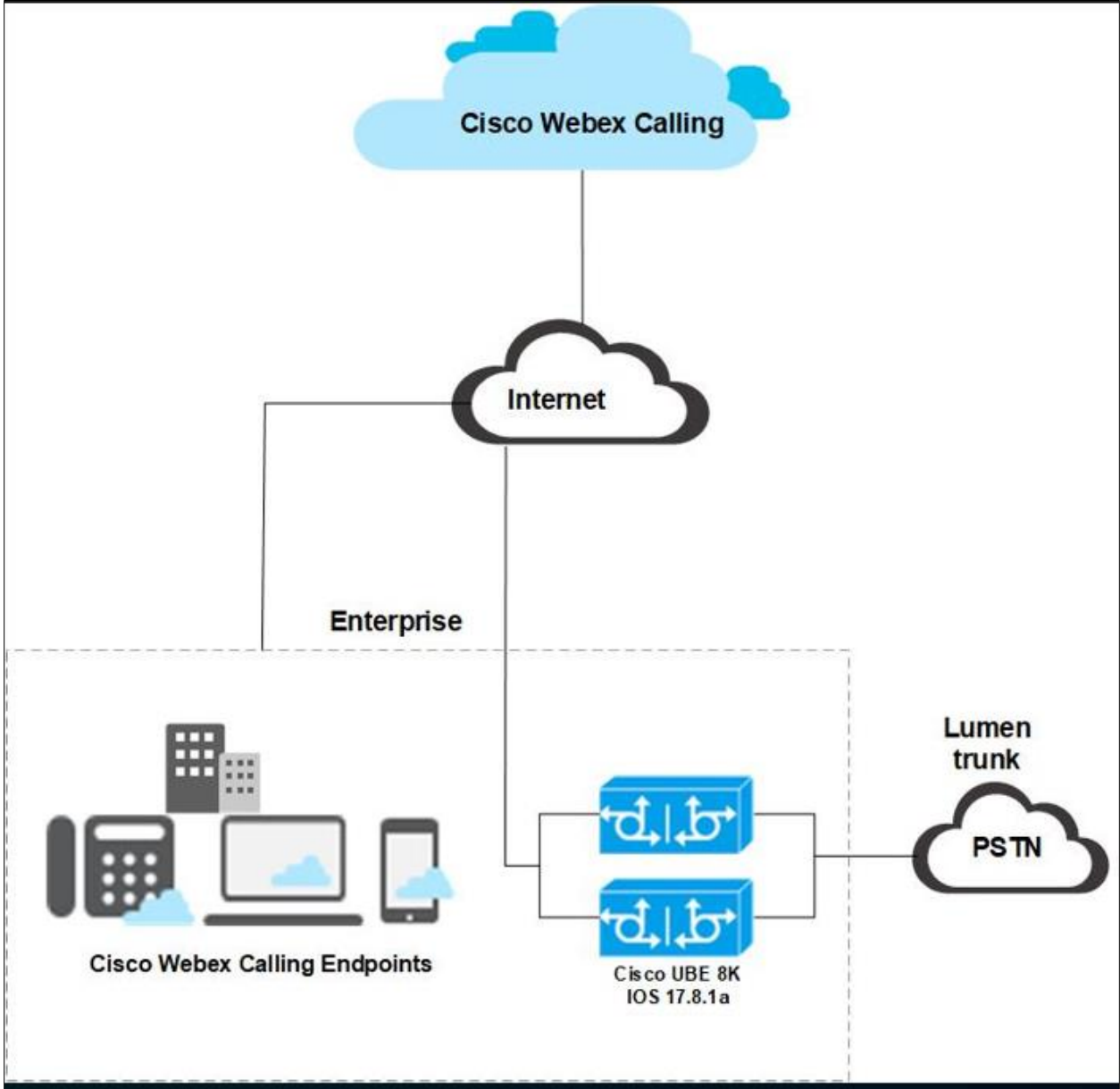


Figure 1: Network Topology with CUBE-HA

Cisco Webex Calling and Cisco UBE Settings:

Setting	Value
Transport from Cisco UBE to Cisco Webex Calling	TLS with SRTP
Transport from Cisco UBE to Lumen	UDP with RTP
Voice Mail Support	YES
Session Refresh	YES
Early Media support	YES

System Components

Hardware Requirements

- CUBE platform 8300-1N1S-6T
- Cisco IP Phones with Multiplatform Firmware
- Cisco ATA 19X

Software Requirements

- CUBE:
 - v14.5 running IOS-XE 17.8.1a
 - Cisco IOS Software [Cupertino], c8000be Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 17.8.1a, RELEASE SOFTWARE (fc3)
- Cisco MPP-Version: sip68xx.11-3-7MPP0001-272.loads
- Cisco ATA 19X-MPP-Version: 11-2-2MPP0101-013

Features

Features Supported

- Incoming and outgoing calls using G711ulaw voice codecs
- Call Conference
- Fax
 - G711 Pass-through
 - T38 Fax
- Voice Mail
- Auto Attendant
- Call hold & Resume(MoH)
- Semi-attended and Attended Call transfer
- Blind Transfer
- Call forward (all, busy and no answer)
- DTMF (RFC2833)
- IP-PBX Calling number privacy

Features Not Supported

- Cisco Webex Calling does not support the following
 - Fax Super G3 speed
 - Tone on Hold

Features Not Tested

- None

Features Not Applicable

- The following are the sections that are not applicable for the test topology
 - Inter-Site Scenarios
 - Codec mid-call re-negotiation
 - IP-PBX Telephone Number Support

Caveats

The following are the observations from CUBE.

- Basic outbound call test cases expected G729 to be offered first but CUBE does not support codec preference list in SRTP to RTP, so calls are established using G711ulaw.
- In long duration calls, Webex does not respond to session refresh Re-INVITEs received from PSTN, resulting in CUBE disconnecting the call. Hence “no session refresh” is configured towards Webex Calling and an “UPDATE” message from Webex Calling was observed for session refresh.
- In Webex Calling hold scenarios, Webex sends two Re-INVITEs on hold. One with send-only and other with send-recv. Webex Calling plays MOH.

Configuration

1 Configuring Cisco Webex Calling

1.1 Add location

Step1:

Login to Cisco Webex Control Hub and navigate to Services

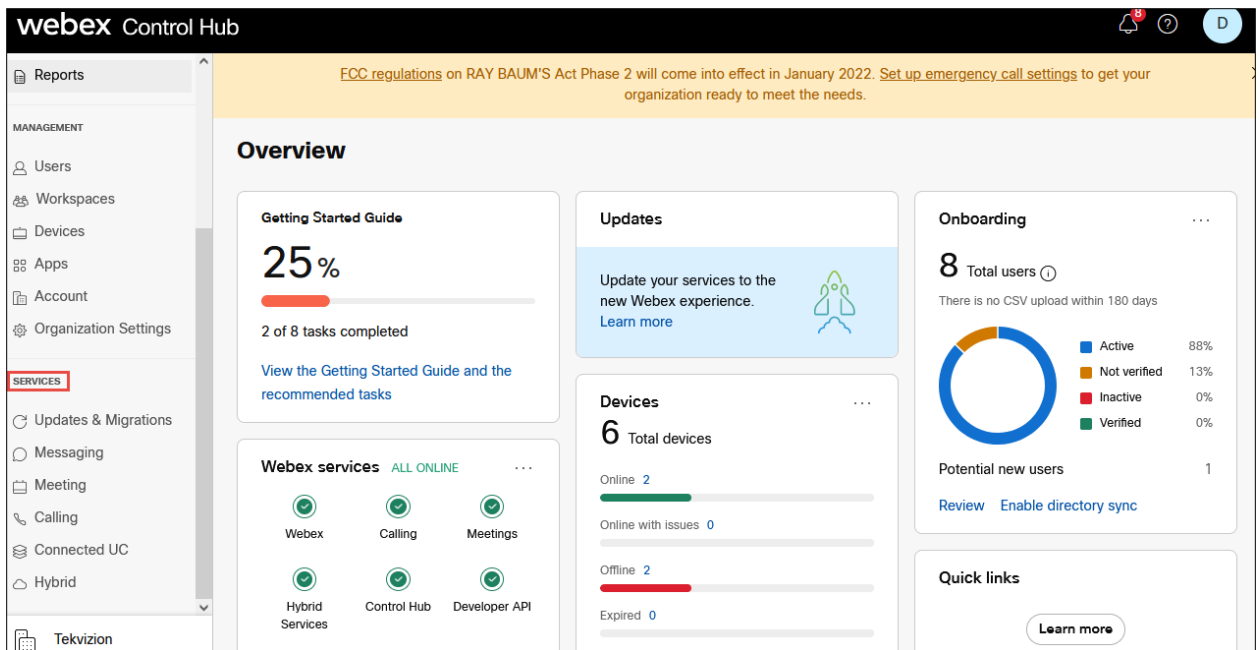


Figure 2: Control Hub Services

Step 2:

Navigate to **Calling** and click on **Locations**

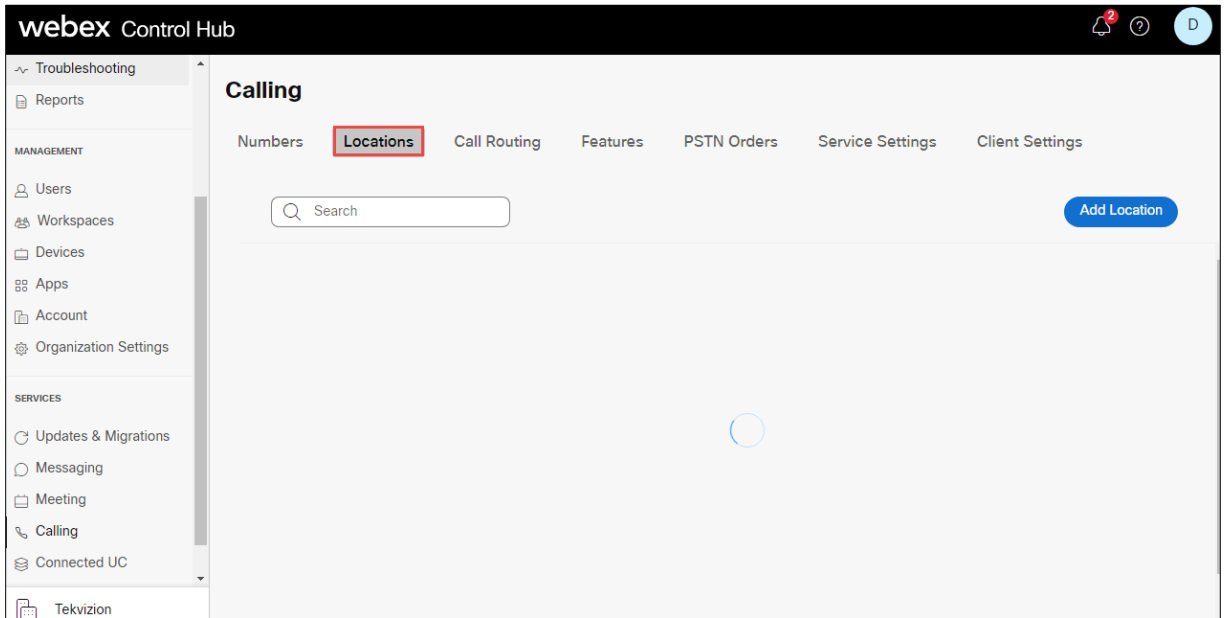


Figure 3: Locations

Step 3:

Click on **Add Location**

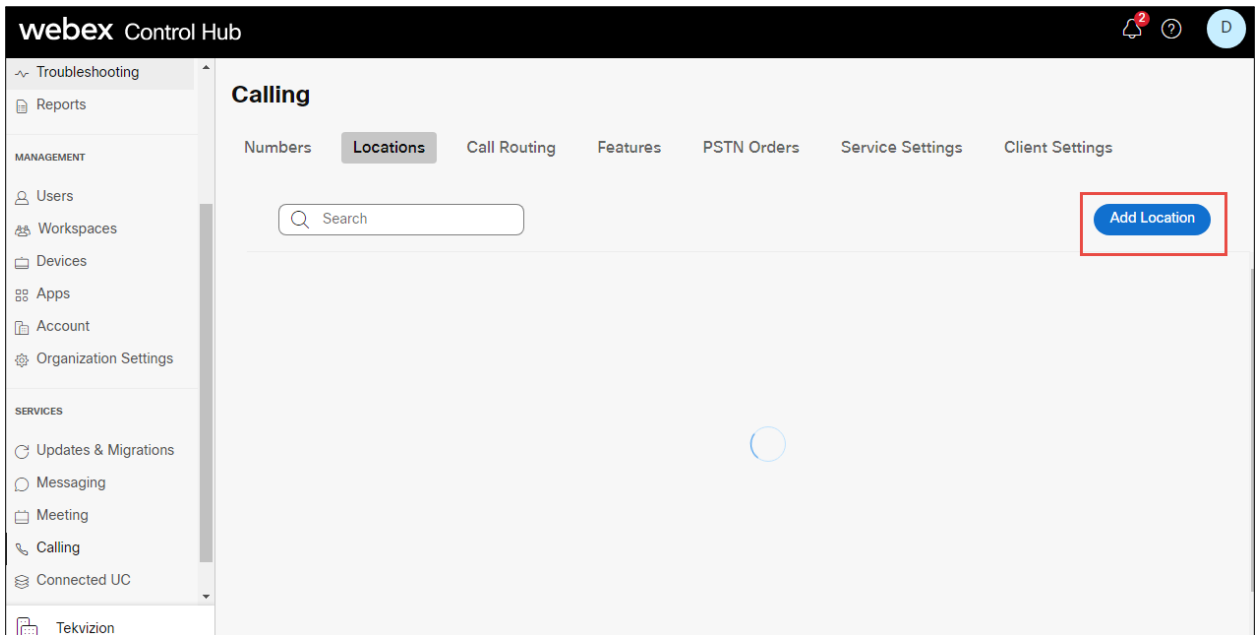


Figure 4: Location creation or selection

Step 4:

Enter **Location** details and click **save**. After adding the location, you will be prompted to add connection type, select No for the connection type. It can be added later.

The screenshot shows a form titled "Add Location" with a red border. The form is divided into two columns. The left column contains: "Location Name" with a text input field containing "Cisco"; "Country/Region" with a dropdown menu showing "United States of America"; "Location Address" with a text input field containing "3701 W Plano Pkwy ste 300" and a secondary field for "Street address line 2 (optional)"; and "City/Town" with a text input field containing "Plano". The right column contains: "Announcement Language" with a dropdown menu showing "English"; "Email Language" with a dropdown menu showing "English - American English"; and "Time zone" with a dropdown menu showing "Select a time zone". Each input field has a small 'x' icon for clearing the text.

Figure 5: Add location details

This screenshot shows the continuation of the "Add Location" form. The left column contains: "City/Town" with a text input field containing "Plano"; "State/Province/Region" with a dropdown menu showing "Texas"; and "Zip/Postal code" with a text input field containing "75075-7840". At the bottom right of the form, there are two buttons: "Cancel" and "Save". The "Save" button is highlighted with a red border.

Figure 6: Add location details Contd.,

Step 5:

Navigate to **Calling** → **Call Routing** → **Add Trunk** and provide the details of Location and name for the SIP Trunk

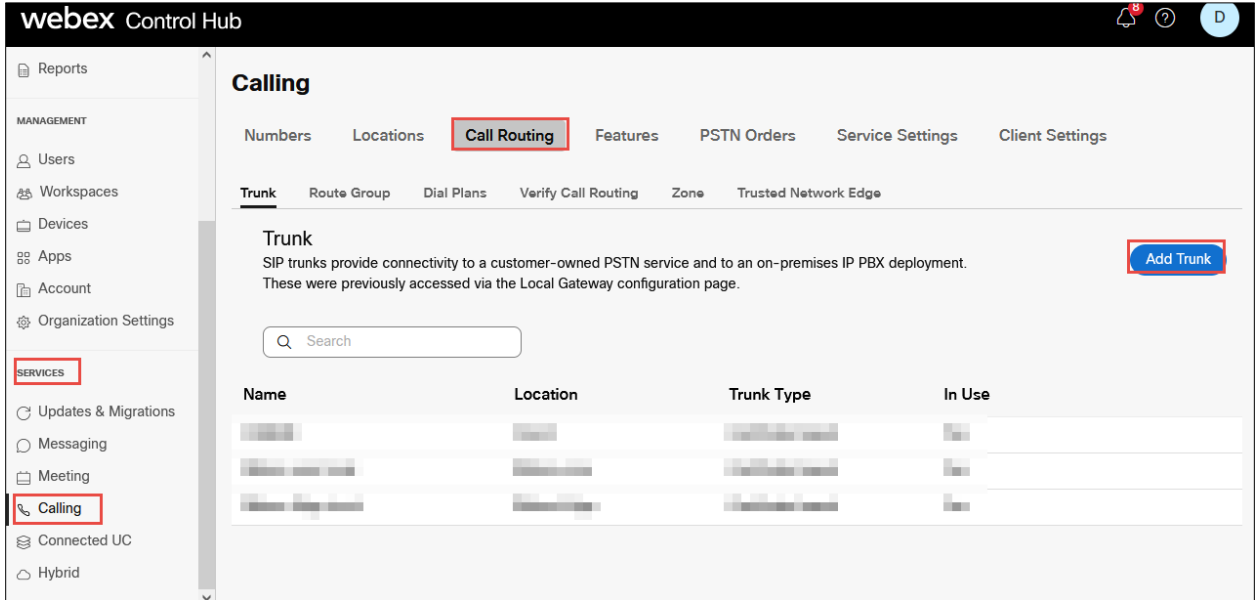


Figure 7: Add Trunk details Contd.,

Add Trunk

Location
This location is where the trunk is physically connected. To create a new location, visit the [Locations](#) page.

Cisco1

Name

CUBE8K|

Trunk Type
Choose the right trunk type for this local gateway. [Learn more](#) on trunk type

Certificate based

Device Type

Cisco Unified Border Element

Enterprise Session Border Controller (SBC) Address

Select the type and enter an FQDN or SRV address for Webex Calling to reach out to your Enterprise SBC. You must also [add and verify](#) your domains before you can use this address. [Manage your domains](#)

FQDN
 SRV

Hostname *	Domain *	Port *
sbc6	tekvizionlabs.com	5061

FQDN

sbc6.tekvizionlabs.com:5061

Maximum number of concurrent calls *

400

Dual Identity Support

The Dual Identity Support setting impacts the handling of the From header and P-Asserted-Identity (PAI) header when sending an initial SIP INVITE to the trunk for an outbound call. When enabled, the From and PAI headers are treated independently and may differ. When disabled, the PAI header is set to the same value as the From header. Please refer to the documentation for more details.

Cancel Save

Figure 8: Add Trunk details Contd.

Add Trunk



CUBE8K 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

Webex Calling edge proxy address (FQDN)

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

Webex Calling edge proxy address (SRV)

us01.sipconnect.bclid.webex.com

Figure 9: Add Trunk details Contd.,

Step 6:

Choose the location and select Manage in PSTN Connection to add Connection type.

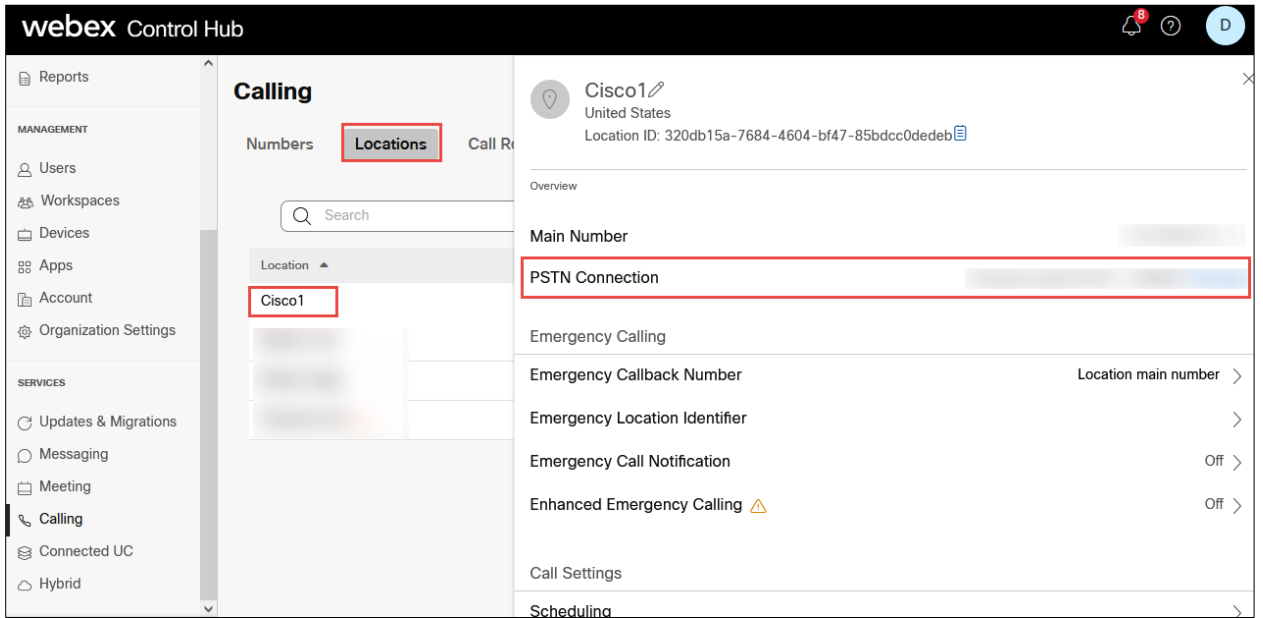


Figure 10: PSTN Connection

Step 7:

Select the **Connection Type** as **Premises-based PSTN** and click on Next

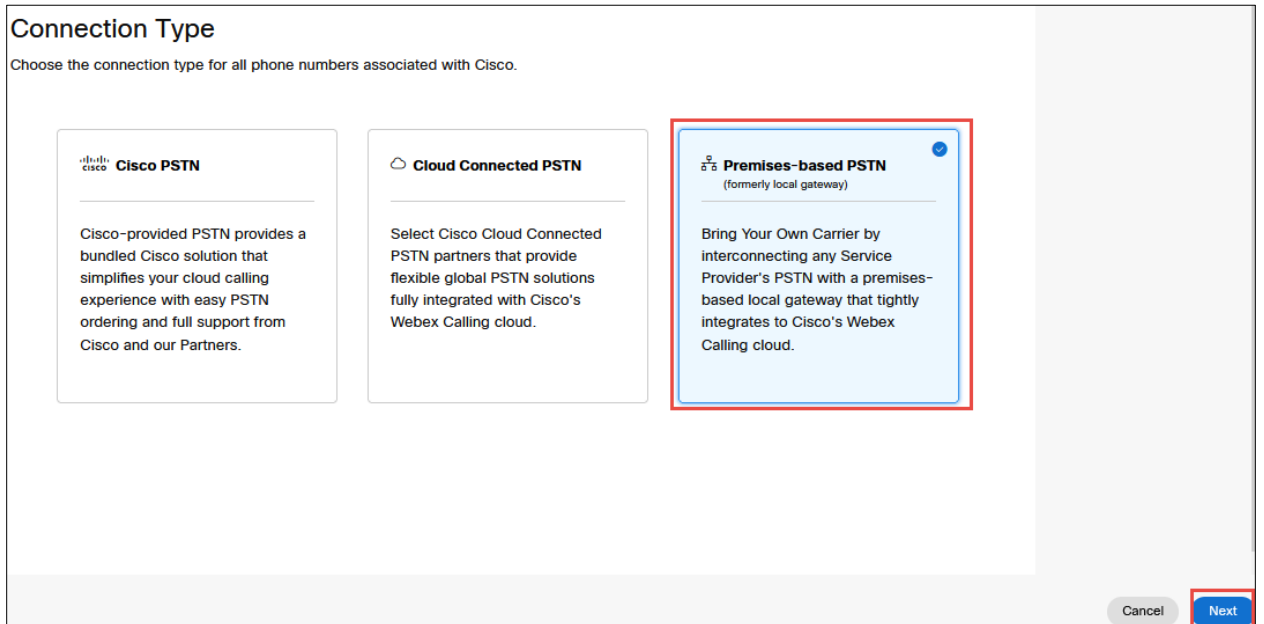


Figure 11: PSTN Connection Contd.,

Step 8:

Select the SIP trunk created earlier and click on Save

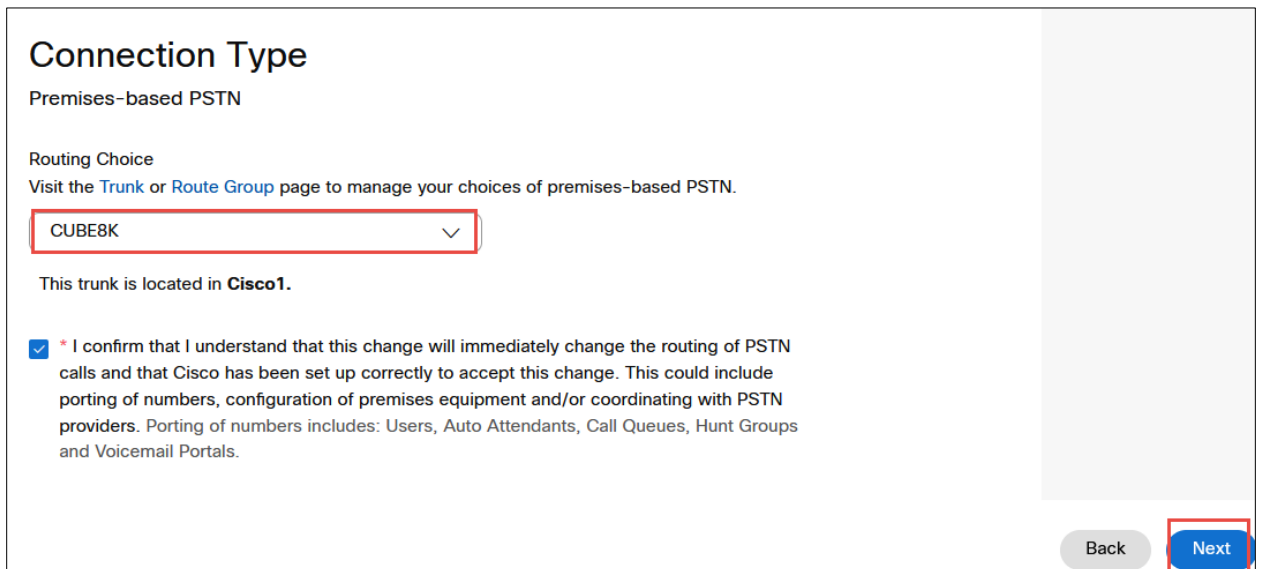


Figure 12: PSTN Connection Contd.,

Step 9:

Select the **Numbers**, Click on **Manage** and choose **Add**. Select the **Location** and **PSTN Connection**

The screenshot displays two parts of a web interface. The top part shows a confirmation message "PSTN connection saved" with a green checkmark. Below it is a summary card for "Premises-based PSTN" with the following details:

- Routing Choice:** CUBE8K
- Type:** Trunk
- Location:** Cisco1

At the bottom right of this card are two buttons: "Done (add numbers later)" and "Add Numbers Now" (highlighted with a red box).

The bottom part of the screenshot is a modal window titled "Add Numbers" with a close button (X) in the top right. It features a progress indicator with three steps: "Select a Location" (active), "Select Numbers", and "Done". The main content area is titled "Choose a Location to Add Numbers" and contains a form with two fields:

- Location:** A dropdown menu with "Cisco1" selected.
- PSTN Connection:** A dropdown menu with "Premises-based PSTN · CUBE8K" selected.

At the bottom right of the modal are "Cancel" and "Next" buttons, with "Next" highlighted by a red box.

Figure 13: Add Numbers

Step 10:

Add the phone numbers provided by service provider and complete the wizard.

Select a Location Select Numbers Done

Enter numbers you want to add

Input your numbers, with area codes, to add them to this location.
Country codes, plus signs, dashes, and parentheses are optional.
Valid examples: 4507832223, (450) 783-2223, 450-783-2223, +1-450-783-2223

Activate Numbers Later ⓘ

✕

(97 17 x (97 18 x

Enter phone numbers separated by commas

2/1000 Phone numbers Clear All

Back Save

Figure 14: Add Numbers Contd.,

Add Numbers

Select a Location Select Numbers Done

☑ Successfully saved numbers

Phone Numbers (1)

(97: 18

Close

Figure 15: Add Numbers Contd.,

1.2 Adding user

Step 1:

In the Cisco Webex Control Hub, select **Users** in the left pane. To add a user, click on **Manage Users** button

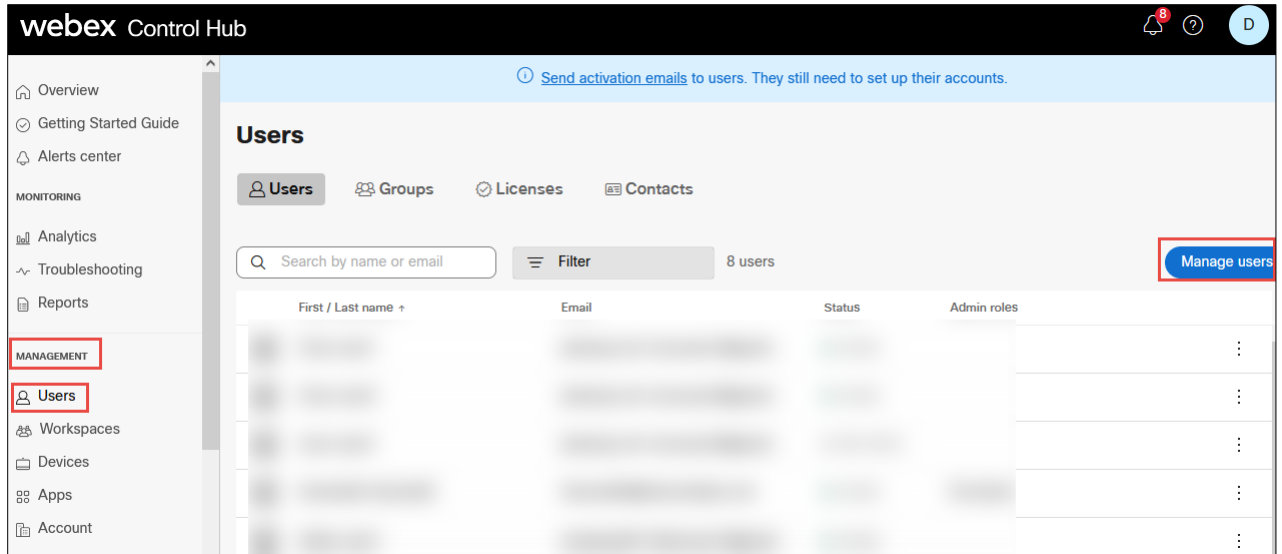


Figure 16: Adding Users

Step 2:

In the Manage Users window, click on **Manually Add or Modify Users** option

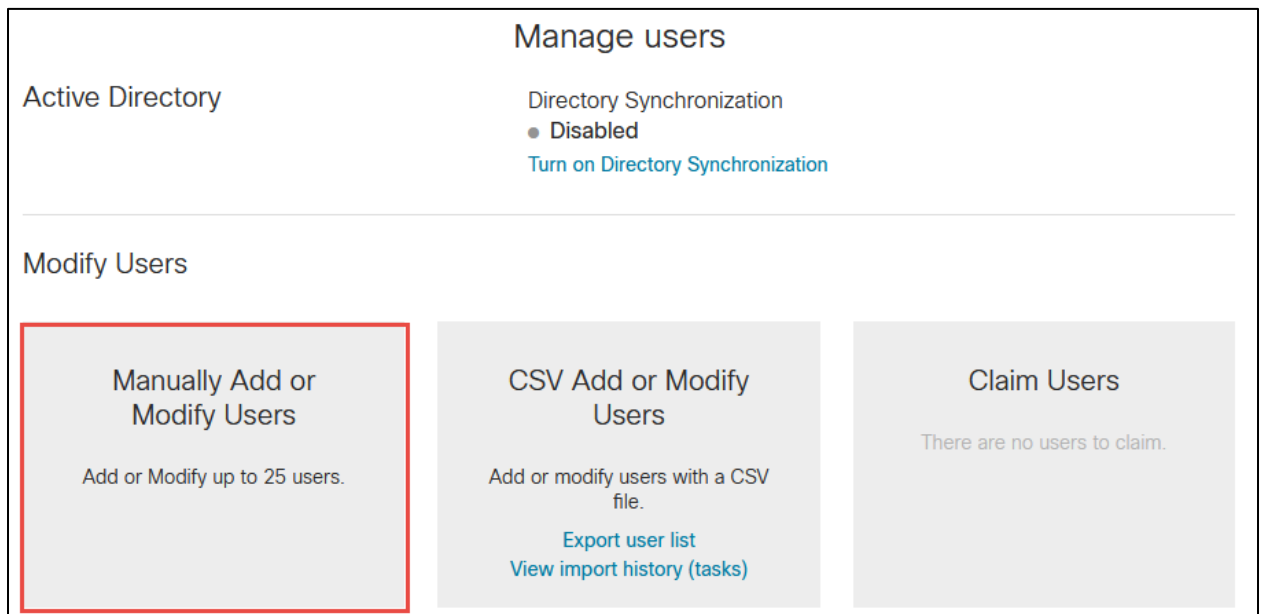


Figure 17: Manually Add or Modify Users

Step 3:

Select either **Email address** or **Names and Email address** and provide the necessary email address. Sample Name and email address provided here is below. Click on **+** symbol to add the user and click on **Next**

Manage users

Manually Add or Modify Users
Enter up to 25 users to modify.

Email address

Names and Email address

Cisco user3 l+ciscouser3@.com +

Back Next

Figure 18: Adding email address and name

Step 4:

Click on **Confirm Adding** button to add the new user and click on **Next**

Manage users

Users to be Added or Modified

Email Address ↑	Name	Status
l+ciscouser3@.com	Cisco user3	New User

Back Next

Figure 19: Confirm Adding

Step 5:

Add Services for the Users. Here select **Webex Calling** under **Calling** section and click **Next**

The screenshot shows the 'Manage users' interface with the following sections:

- Messaging:** Basic Messaging
- Meeting:** Basic Space Meetings
- Calling:** Call on Webex (1:1 call, non-PSTN)

Non-subscription Licenses:

- Register to Unified Communications Manager (UCM)

Licensed Collaboration Services:

- Messaging:** Advanced Messaging
- Meetings:** Advanced Space Meetings, Webex Assistant for Meetings, Webex Meetings Suite
- Calling:** Webex Calling, Professional

Buttons: Back, Next

Figure 20: Add Services for Users

Step 6:

Assign the user to appropriate location and select the phone number and extension. Click on **Finish** button.

The screenshot shows the 'Assign Numbers' interface with the following details:

User	Location	Phone Number	Extension	Calling Plan
Cisco user3 1+ciscouser3@i.c...	Cisco1	+197 18	18	

Buttons: Back, Finish

Figure 21: Assign Numbers

Step 7:

Successful creation of user will be displayed in the Add Users window. Click on Finish button.

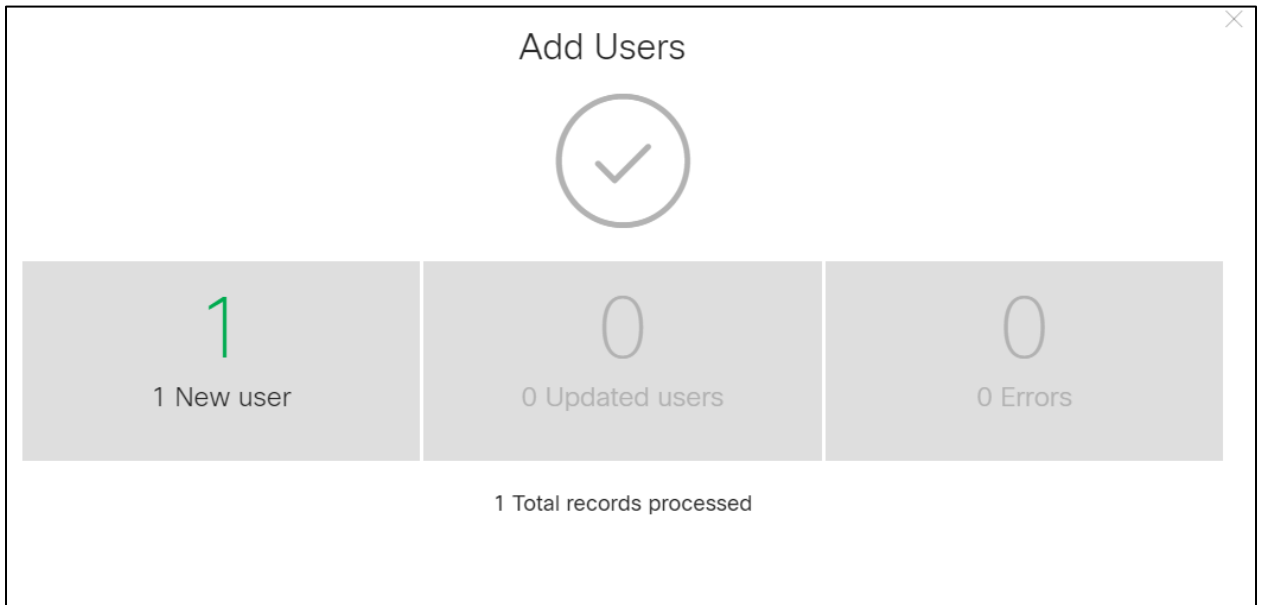


Figure 22: Add User successful

1.3 Adding Devices

Step 1:

To add a device, navigate to **Devices** in Cisco Webex Control Hub. The existing devices will be listed out. Click **Add Device** button.

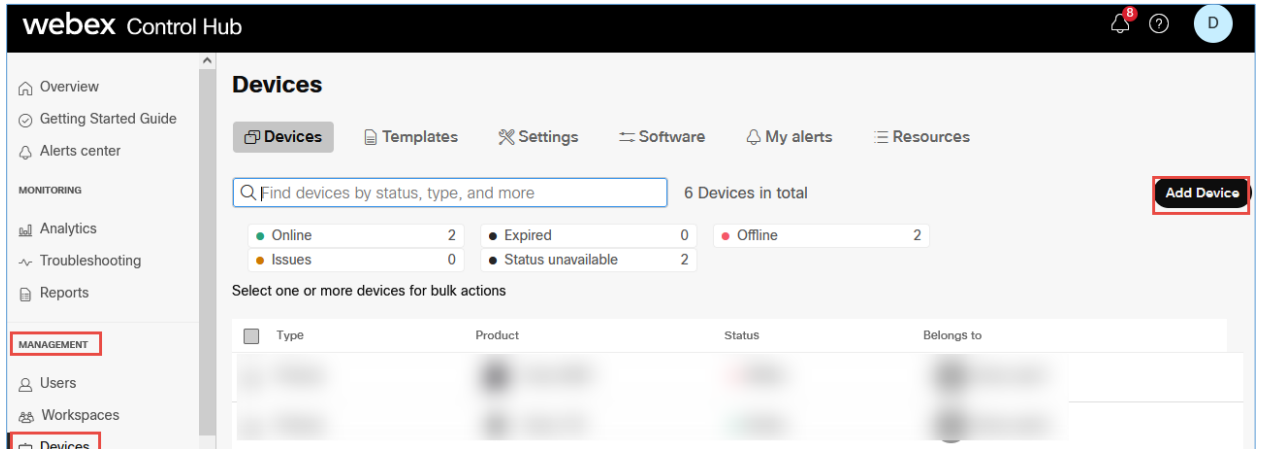


Figure 23: Add Devices Window

Step 2:

In the Add Device window, assign the device to a user or a place. Select **Existing User** and Click on Next.

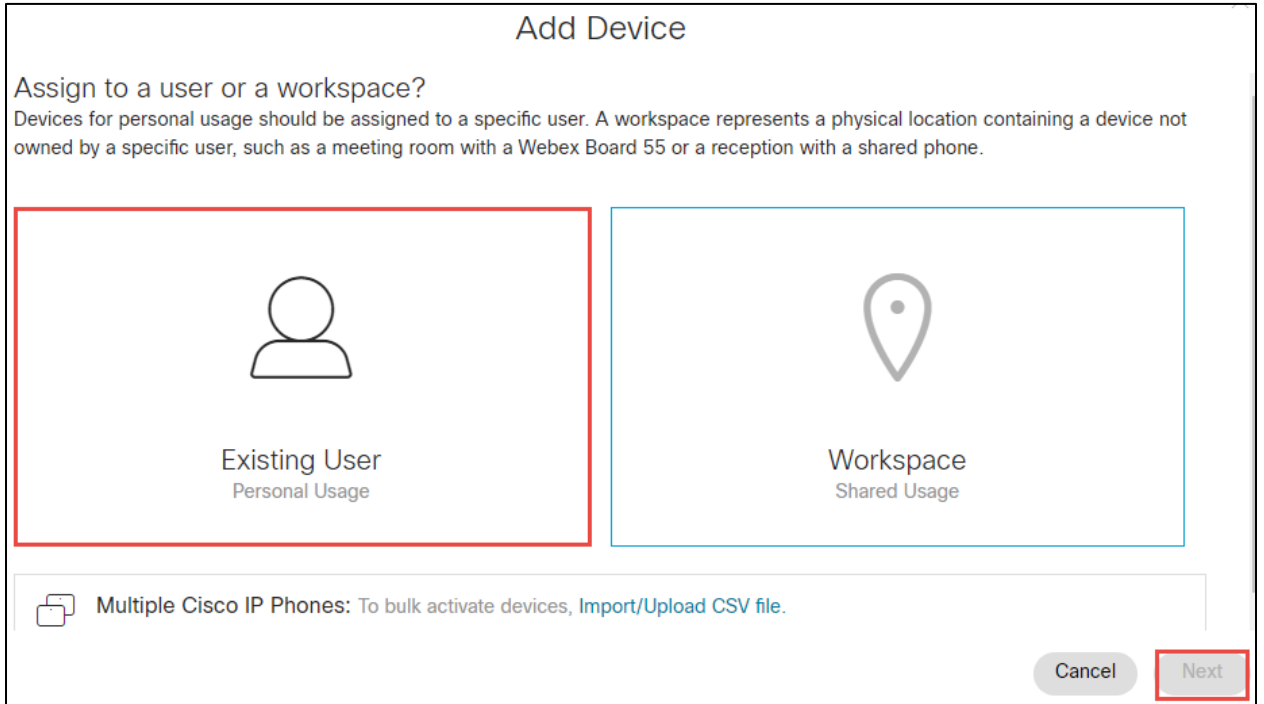
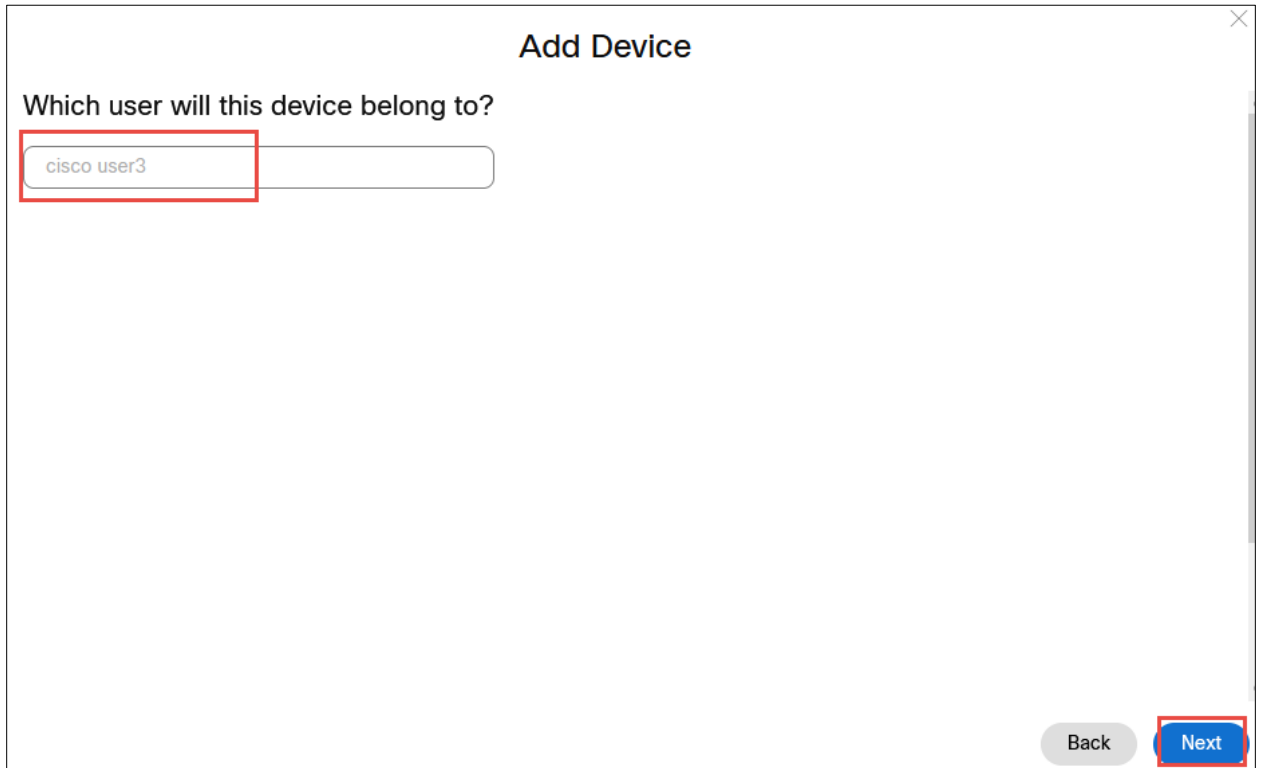


Figure 24: Assign to a user or a place

Step 3:

Select appropriate user from the **search for a user list** and click **Next**



The screenshot shows a dialog box titled "Add Device" with a close button (X) in the top right corner. Below the title is the question "Which user will this device belong to?". A search input field contains the text "cisco user3" and is highlighted with a red border. At the bottom right of the dialog, there are two buttons: a grey "Back" button and a blue "Next" button, with the "Next" button also highlighted by a red border.

Figure 25: User Association with Device

Step 4:

In the Select Device drop down box, select the appropriate **device** and enter the **MAC address**. Click on **Save** button. The device will be added successfully.

Add Device

What kind of device do you want to set up for this user?

Room, Board or Desk series
e.g. Cisco Webex Board, Room, and Desk series, and Webex Share.

Cisco IP Phone
e.g. Cisco 8845, 8865, 8800 and Analog Telephone Adapter ports

Select Device

Cisco 6851

How would you like to setup this device?

By Activation Code

By Mac Address

Enter MAC Address

Enter the MAC address of the IP phone you want to add.

47 | D|

Back Save

Figure 26: Select Device and add MAC address

1.4 Assign main number to a location

Step 1:

Assign number in location as the Main number.

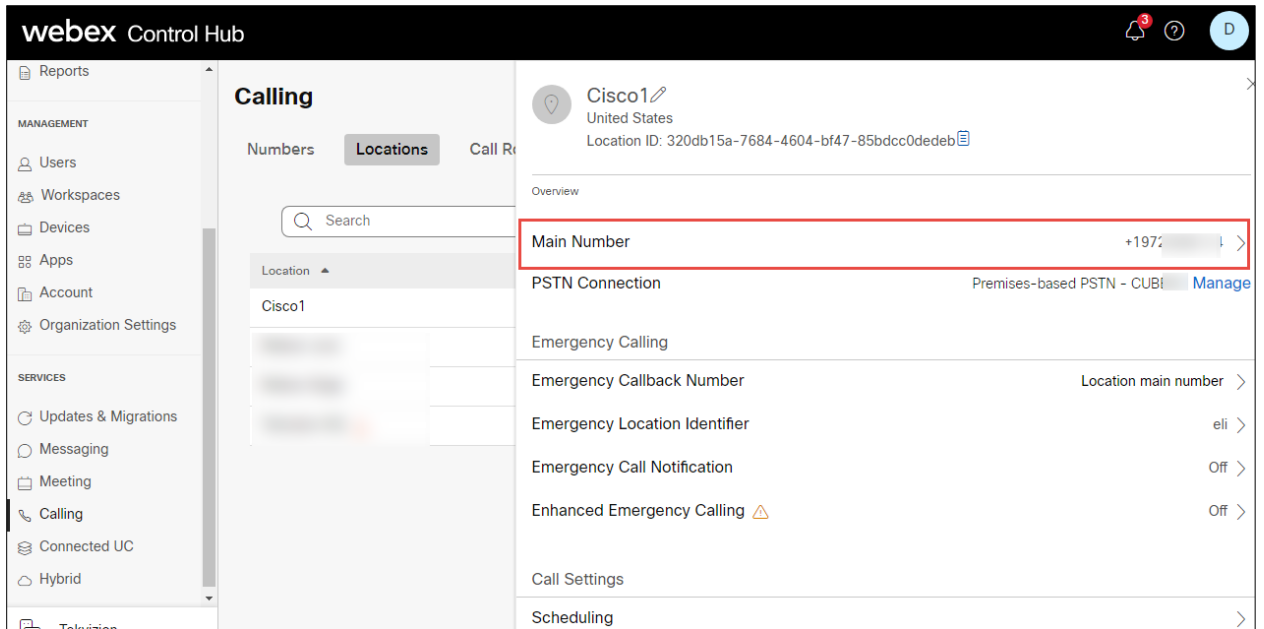


Figure 27: Assign Main number in location

2 Cisco UBE Configuration

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

2.1 IP Networking

```
interface GigabitEthernet0/0/0
description To HA interface
ip address 10.64.5.234 255.255.0.0
negotiation auto
!
interface GigabitEthernet0/0/1
description To 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 Webex Calling
ip address 192.65.79.x 255.255.255.128
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 Same rii id to be used in CUBEs that has same virtual IP
redundancy group 1 ip x.x.x.x exclusive	Enable Redundancy group in physical interface with virtual IP towards PSTN and Webex calling

2.2 IP Routing

2.2.1 To Webex Calling

```
ip route 0.0.0.0 0.0.0.0 192.65.79.129
```

2.2.2 To PSTN

```
ip route 0.0.0.0 0.0.0.0 192.65.79.129
```

2.3 DNS Servers

DNS must be configured to resolve addresses for Webex Calling

```
ip name-server 208.67.222.222 208.67.220.220
```

2.4 NTP Servers

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

```
ntp server 10.10.10.5
```

2.5 Certificates

The following steps describe how to create and install a certificate.

2.5.1 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)
```

2.5.2 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
```

2.5.3 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:
```

2.5.4 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
```

2.5.5 Import signed host certificate

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

```
crypto pki import sbc6 certificate
```

Enter the base 64 encoded CA certificate.

End with a blank line or the word "quit" on a line by itself

2.5.6 Specify the default trust point and TLS version to use

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

2.5.7 Import Cisco CA bundle for Webex calling certificate authentication

Create the CA certificate trust point used to validate Webex Calling SIP Link TLS messages:

```
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.
```


2.5.8 Exporting RSA key and certificate from CUBE 1 for High Availability

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

2.5.9 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 "sbc6"

```
crypto pki import sbc6 pkcs12 ftp://<username>:<password>@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.
```

2.6 Global CUBE settings

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

```
voice service voip
ip address trusted list
  ipv4 139.177.65.53 255.255.255.255
  ipv4 85.119.56.128 255.255.255.192
  ipv4 85.119.57.128 255.255.255.192
  ipv4 135.84.169.0 255.255.255.128
  ipv4 135.84.170.0 255.255.255.128
  ipv4 135.84.171.0 255.255.255.128
  ipv4 135.84.172.0 255.255.255.128
  ipv4 135.84.173.0 255.255.255.128
  ipv4 135.84.174.0 255.255.255.128
  ipv4 139.177.64.0 255.255.255.0
```

```
ipv4 139.177.65.0 255.255.255.0
ipv4 139.177.66.0 255.255.255.0
ipv4 139.177.67.0 255.255.255.0
ipv4 139.177.68.0 255.255.255.0
ipv4 139.177.69.0 255.255.255.0
ipv4 139.177.70.0 255.255.255.0
ipv4 139.177.71.0 255.255.255.0
ipv4 139.177.72.0 255.255.255.0
ipv4 139.177.73.0 255.255.255.0
ipv4 185.115.196.0 255.255.255.128
ipv4 185.115.197.0 255.255.255.128
ipv4 199.19.197.0 255.255.255.0
ipv4 199.19.199.0 255.255.255.0
ipv4 199.59.64.0 255.255.255.128
ipv4 199.59.65.0 255.255.255.128
ipv4 199.59.66.0 255.255.255.128
ipv4 199.59.67.0 255.255.255.128
ipv4 199.59.70.0 255.255.255.128
ipv4 199.59.71.0 255.255.255.128
ipv4 128.177.14.0 255.255.255.128
ipv4 128.177.36.0 255.255.255.192
ipv4 10.64.1.0
address-hiding
mode border-element
media statistics
media bulk-stats
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip refer
no supplementary-service sip handle-replaces
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
trace
sip
  early-offer forced
g729 annexb-all
```

Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
fax protocol	Specifies the fax protocol
no supplementary-service sip refer no supplementary-service sip handle-replaces	Disable forwarding SIP REFER message for call transfers and replace the Dialog-ID in the Replaces header with the peer Dialog-ID
early-offer forced	Forces LGW to send the SDP information in the initial INVITE message
g729 annexb-all	Allows all variants of G729

2.7 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 weightage 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

2.8 Options keepalive to Webex Calling

```
voice class sip-options-keepalive 100
description Keepalive Webex calling
down-interval 5
transport tcp tls
sip-profiles 100
```

2.9 Message Handling Rules

2.9.1 SIP Profiles: Manipulations for outbound messages to Webex Calling

The following sip profile is required to:

Rule 10: Modify Contact header to replace the IP address of the Local Gateway with the FQDN

```
voice class sip-profiles 100
  rule 10 request OPTIONS sip-header Contact modify "<sip.*:"
"<sip:sbc6.tekvizionlabs.com:"
!
voice class sip-profiles 200
  rule 10 request ANY sip-header Contact modify "@.*:" "@sbc6.tekvizionlabs.com:"
  rule 20 response ANY sip-header Contact modify "@.*:" "@sbc6.tekvizionlabs.com:"
```

2.10 SRTP crypto

Used to set the crypto cipher for the Webex Calling

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

2.11 Tenant

2.11.1 Tenant to Webex Calling

```
voice class tenant 200
  no remote-party-id
  connection-reuse
  srtp-crypto 200
  localhost dns:sbc6.tekvizionlabs.com
  session transport tcp tls
  no session refresh
  error-passthru
  bind control source-interface GigabitEthernet0/0/2
  bind media source-interface GigabitEthernet0/0/2
  no pass-thru content custom-sdp
  privacy-policy passthru
```


2.13 Codecs

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

2.14 Dial-peer group

2.14.1 From Webex calling To PSTN

```
voice class dpg 100
  description Incoming WxC(DP200101) to IP PSTN(DP101)
  dial-peer 101 preference 1
```

2.14.2 From PSTN to Webex Calling

```
voice class dpg 200
  description Incoming IP PSTN(DP100) to Webex Calling(DP200201)
  dial-peer 200201 preference 1
  dial-peer 200202 preference 2
  dial-peer 200203 preference 3
  dial-peer 200204 preference 4
```

2.15 Dial peers

2.15.1 Outbound calls to Lumen

```
dial-peer voice 101 voip
description outgoing dial-peer to IP PSTN
translation-profile outgoing 200
destination-pattern BAD.BAD
session protocol sipv2
session target ipv4:10.64.1.x:5060
session transport udp
voice-class codec 100
voice-class sip options-ping 60
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
```

2.15.2 Inbound calls from Lumen

```
voice class uri 100 sip
host 10.80.11.136
!
dial-peer voice 100 voip
description Incoming dial-peer from PSTN
translation-profile incoming 100
session protocol sipv2
destination dpg 200
incoming uri request 100
voice-class codec 100
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
```


2.15.3 Inbound calls from Cisco Webex Calling

```
voice class uri 200 sip
  host sbc6.tekvizionlabs.com
!
dial-peer voice 200101 voip
  description Inbound from Webex Calling
  session protocol sipv2
  session transport tcp tls
  destination dpg 100
  incoming uri request 200
  voice-class codec 100
  voice-class sip tenant 200
  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
```

2.15.4 Outbound calls to Cisco Webex Calling

```
dial-peer voice 200201 voip
  description Outbound Webex Calling 1
  destination-pattern BAD.BAD
  session protocol sipv2
  session target dns:peering1.us.sipconnect.bclld.webex.com:5062
  session transport tcp tls
  voice-class codec 100
  voice-class sip asserted-id pai
  voice-class sip profiles 200
  voice-class sip tenant 200
  voice-class sip options-keepalive profile 100
  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 200202 voip
description Outbound Webex Calling 2
destination-pattern BAD.BAD
session protocol sipv2
session target dns:peering2.us.sipconnect.bclld.webex.com:5062
session transport tcp tls
voice-class codec 100
voice-class sip profiles 200
voice-class sip tenant 200
voice-class sip options-keepalive profile 100
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 200203 voip
description Outbound Webex Calling 3
destination-pattern BAD.BAD
session protocol sipv2
session target dns:peering3.us.sipconnect.bclld.webex.com:5062
session transport tcp tls
voice-class codec 100
voice-class sip profiles 200
voice-class sip tenant 200
voice-class sip options-keepalive profile 100
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 200204 voip
description Outbound Webex Calling 4
destination-pattern BAD.BAD
session protocol sipv2
session target dns:peering4.us.sipconnect.bclld.webex.com:5062
session transport tcp tls
voice-class codec 100
```

```
voice-class sip profiles 200
voice-class sip tenant 200
voice-class sip options-keepalive profile 100
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
!
```

3 Running Configuration

The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously.

3.1 CUBE 1

Building configuration...

```
Current configuration : 11165 bytes
!
version 17.8
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_MTLS_webex
!
boot-start-marker
boot system bootflash:c8000be-universalk9.17.08.01a.SPA.bin
boot-end-marker
!
logging buffered 21474836
no aaa new-model
clock timezone UTC -5 0
clock calendar-valid
!
ip name-server 208.67.222.222 208.67.220.220
ip domain name tekvisionlabs.com
!
login on-success log
!
subscriber templating
!
multilink bundle-name authenticated
!
password encryption aes
!
crypto pki trustpoint SLA-TrustPoint
  enrollment pkcs12
  revocation-check crl
!
crypto pki trustpoint sbc6
  enrollment terminal
  fqdn sbc6.tekvisionlabs.com
  subject-name cn=sbc6.tekvisionlabs.com
```

```
subject-alt-name sbc6.tekvizionlabs.com
revocation-check crl
rsakeypair sbc6
!
crypto pki certificate chain SLA-TrustPoint
certificate ca 01
crypto pki certificate chain sbc6
certificate 00A76F21D0D0E2906D
certificate ca 07
!
crypto pki certificate pool
cabundle nvram:ios_core.p7b
!
voice service voip
ip address trusted list
ipv4 139.177.65.53 255.255.255.255
ipv4 85.119.56.128 255.255.255.192
ipv4 85.119.57.128 255.255.255.192
ipv4 135.84.169.0 255.255.255.128
ipv4 135.84.170.0 255.255.255.128
ipv4 135.84.171.0 255.255.255.128
ipv4 135.84.172.0 255.255.255.128
ipv4 135.84.173.0 255.255.255.128
ipv4 135.84.174.0 255.255.255.128
ipv4 139.177.64.0 255.255.255.0
ipv4 139.177.65.0 255.255.255.0
ipv4 139.177.66.0 255.255.255.0
ipv4 139.177.67.0 255.255.255.0
ipv4 139.177.68.0 255.255.255.0
ipv4 139.177.69.0 255.255.255.0
ipv4 139.177.70.0 255.255.255.0
ipv4 139.177.71.0 255.255.255.0
ipv4 139.177.72.0 255.255.255.0
ipv4 139.177.73.0 255.255.255.0
ipv4 185.115.196.0 255.255.255.128
ipv4 185.115.197.0 255.255.255.128
ipv4 199.19.197.0 255.255.255.0
ipv4 199.19.199.0 255.255.255.0
ipv4 199.59.64.0 255.255.255.128
ipv4 199.59.65.0 255.255.255.128
ipv4 199.59.66.0 255.255.255.128
ipv4 199.59.67.0 255.255.255.128
ipv4 199.59.70.0 255.255.255.128
ipv4 199.59.71.0 255.255.255.128
ipv4 128.177.14.0 255.255.255.128
ipv4 128.177.36.0 255.255.255.192
ipv4 10.64.1.x
address-hiding
mode border-element
media statistics
media bulk-stats
allow-connections sip to sip
```

```

redundancy-group 1
no supplementary-service sip refer
no supplementary-service sip handle-replaces
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback
none
trace
sip
    early-offer forced
    g729 annexb-all
!
!
voice class uri 100 sip
    host 10.80.11.136
!
voice class uri 200 sip
    host sbc6.tekvizionlabs.com
!
voice class codec 100
    codec preference 1 opus
    codec preference 2 g711ulaw
    codec preference 3 g711alaw
!
voice class sip-profiles 100
    rule 10 request OPTIONS sip-header Contact modify "<sip:.*:"
"<sip:sbc6.tekvizionlabs.com:"
!
voice class sip-profiles 200
    rule 10 request ANY sip-header Contact modify "@.*:"
"@sbc6.tekvizionlabs.com:"
!
voice class dpg 100
    description Incoming WxC(DP200101) to IP PSTN(DP101)
    dial-peer 101 preference 1
!
voice class dpg 200
    description Incoming IP PSTN(DP100) to Webex Calling(DP200201)
    dial-peer 200201 preference 1
    dial-peer 200202 preference 2
    dial-peer 200203 preference 3
    dial-peer 200204 preference 4
!
voice class sip-options-keepalive 100
    description Keepalive Webex Calling
    down-interval 5
    transport tcp tls
    sip-profiles 100
!
voice class tenant 200
    no remote-party-id
    connection-reuse
    srtp-crypto 200
    localhost dns:sbc6.tekvizionlabs.com

```

```

    session transport tcp tls
    no session refresh
    error-passthru
    bind control source-interface GigabitEthernet0/0/2
    bind media source-interface GigabitEthernet0/0/2
    no pass-thru content custom-sdp
    privacy-policy passthru
!
voice class tenant 100
    session transport udp
    url sip
    error-passthru
    bind media source-interface GigabitEthernet0/0/1
    bind control source-interface GigabitEthernet0/0/1
    no pass-thru content custom-sdp
    privacy-policy passthru
!
voice class srtp-crypto 200
    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/
    rule 2 /^0\[2-9\].....\)/ /\1/
    rule 4 /^+91\(.*\)/ /01191\1/
!
voice translation-profile 100
    translate calling 100
    translate called 100
!
voice translation-profile 200
    translate calling 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
!
enable secret 9 xxxx
!
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 HA interface
  ip address 10.64.5.234 255.255.0.0
  negotiation auto
!
interface GigabitEthernet0/0/1
  description To 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 Webex Calling
  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
  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 synwait-time 5
ip http server
ip http secure-server
ip http client source-interface GigabitEthernet0/0/2

```



```

ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 172.16.0.0 255.255.0.0 10.64.1.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
dial-peer voice 100 voip
description Incoming dial-peer from PSTN
translation-profile incoming 100
session protocol sipv2
destination dpg 200
incoming uri request 100
voice-class codec 100
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
!
dial-peer voice 200201 voip
description Outbound Webex Calling 1
destination-pattern BAD.BAD
session protocol sipv2
session target dns:peering1.us.sipconnect.bclid.webex.com:5062
session transport tcp tls
voice-class codec 100
voice-class sip asserted-id pai
voice-class sip profiles 200
voice-class sip tenant 200
voice-class sip options-keepalive profile 100
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 200202 voip
description Outbound Webex Calling 2
destination-pattern BAD.BAD
session protocol sipv2
session target dns:peering2.us.sipconnect.bclid.webex.com:5062
session transport tcp tls
voice-class codec 100

```

```

voice-class sip profiles 200
voice-class sip tenant 200
voice-class sip options-keepalive profile 100
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 200203 voip
description Outbound Webex Calling 3
destination-pattern BAD.BAD
session protocol sipv2
session target dns:peering3.us.sipconnect.bclld.webex.com:5062
session transport tcp tls
voice-class codec 100
voice-class sip profiles 200
voice-class sip tenant 200
voice-class sip options-keepalive profile 100
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 200204 voip
description Outbound Webex Calling 4
destination-pattern BAD.BAD
session protocol sipv2
session target dns:peering4.us.sipconnect.bclld.webex.com:5062
session transport tcp tls
voice-class codec 100
voice-class sip profiles 200
voice-class sip tenant 200
voice-class sip options-keepalive profile 100
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 101 voip
description outgoing dial-peer to IP PSTN
translation-profile outgoing 200
destination-pattern BAD.BAD
session protocol sipv2
session target ipv4:10.64.1.72:5060
session transport udp
voice-class codec 100
voice-class sip options-ping 60
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
!
dial-peer voice 200101 voip
description Inbound Webex Calling
session protocol sipv2
session transport tcp tls
destination dpg 100
incoming uri request 200
voice-class codec 100
voice-class sip tenant 200
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
!
!
gateway
timer receive-rtp 1200
!
sip-ua
no remote-party-id
retry invite 2
transport tcp tls v1.2
crypto signaling default trustpoint sbc6
!
line con 0
exec-timeout 5 0
password 7 xxxxx
logging synchronous
login
stopbits 1
line aux 0
line vty 0 4
exec-timeout 60 0
password 7 xxxxx
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
```

3.2 CUBE2

Building configuration...

```
Current configuration : 12534 bytes
!
version 17.8
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
platform hardware throughput crypto 25M
!
hostname CUBE8K
!
boot-start-marker
boot system bootflash:c8000be-universalk9.17.08.01a.SPA.bin
boot-end-marker
!
logging buffered 214748364
no aaa new-model
clock timezone UTC -5 0
clock calendar-valid
!
ip name-server 208.67.222.222 208.67.220.220
ip domain name tekvisionlabs.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 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 pool
  cabundle nvram:ios_core.p7b
!
voice service voip
  ip address trusted list
    ipv4 139.177.65.53 255.255.255.255
    ipv4 85.119.56.128 255.255.255.192
    ipv4 85.119.57.128 255.255.255.192
    ipv4 135.84.169.0 255.255.255.128
    ipv4 135.84.170.0 255.255.255.128
    ipv4 135.84.171.0 255.255.255.128
    ipv4 135.84.172.0 255.255.255.128
    ipv4 135.84.173.0 255.255.255.128
    ipv4 135.84.174.0 255.255.255.128
    ipv4 139.177.64.0 255.255.255.0
    ipv4 139.177.65.0 255.255.255.0
    ipv4 139.177.66.0 255.255.255.0
    ipv4 139.177.67.0 255.255.255.0
    ipv4 139.177.68.0 255.255.255.0
    ipv4 139.177.69.0 255.255.255.0
    ipv4 139.177.70.0 255.255.255.0
    ipv4 139.177.71.0 255.255.255.0
    ipv4 139.177.72.0 255.255.255.0
    ipv4 139.177.73.0 255.255.255.0
    ipv4 185.115.196.0 255.255.255.128
    ipv4 185.115.197.0 255.255.255.128
    ipv4 199.19.197.0 255.255.255.0
    ipv4 199.19.199.0 255.255.255.0
    ipv4 199.59.64.0 255.255.255.128
    ipv4 199.59.65.0 255.255.255.128
    ipv4 199.59.66.0 255.255.255.128
    ipv4 199.59.67.0 255.255.255.128
    ipv4 199.59.70.0 255.255.255.128
    ipv4 199.59.71.0 255.255.255.128
    ipv4 128.177.14.0 255.255.255.128
    ipv4 128.177.36.0 255.255.255.192
    ipv4 10.64.1.x
  address-hiding
  mode border-element
  media statistics
  media bulk-stats
  allow-connections sip to sip
  redundancy-group 1
  no supplementary-service sip refer
```

```

no supplementary-service sip handle-replaces
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback
none
trace
sip
    early-offer forced
    g729 annexb-all
!
voice class uri 100 sip
    host 10.80.11.136
!
voice class uri 200 sip
    host sbc6.tekvizionlabs.com
voice class codec 100
    codec preference 1 opus
    codec preference 2 g711ulaw
    codec preference 3 g711alaw
!
voice class sip-profiles 100
    rule 10 request OPTIONS sip-header Contact modify "<sip:.*:"
"<sip:sbc6.tekvizionlabs.com:"
!
voice class sip-profiles 200
    rule 10 request ANY sip-header Contact modify "@.*:"
"@sbc6.tekvizionlabs.com:"
!
voice class dpg 100
    description Incoming WxC(DP200101) to IP PSTN(DP101)
    dial-peer 101 preference 1
!
voice class dpg 200
    description Incoming IP PSTN(DP100) to Webex Calling(DP200201)
    dial-peer 200201 preference 1
    dial-peer 200202 preference 2
    dial-peer 200203 preference 3
    dial-peer 200204 preference 4
!
voice class sip-options-keepalive 100
    description Keepalive Webex Calling
    down-interval 5
    transport tcp tls
    sip-profiles 100
!
voice class tenant 200
    no remote-party-id
    connection-reuse
    srtp-crypto 200
    localhost dns:sbc6.tekvizionlabs.com
    session transport tcp tls
    no session refresh
    error-passthru
    bind control source-interface GigabitEthernet0/0/2

```

```

bind media source-interface GigabitEthernet0/0/2
no pass-thru content custom-sdp
privacy-policy passthru
!
voice class tenant 100
  session transport udp
  url sip
  error-passthru
  bind media source-interface GigabitEthernet0/0/1
  bind control source-interface GigabitEthernet0/0/1
  no pass-thru content custom-sdp
  privacy-policy passthru
!
voice class srtp-crypto 200
  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/
  rule 2 /^\0\([2-9].....\) / \1/
  rule 4 /^\+91\(.*\) / /01191\1/
!
voice translation-profile 100
  translate calling 100
  translate called 100
!
voice translation-profile 200
  translate calling 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
!
enable secret 9 xxxxxx
!
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
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 POE2 Gig 4/0/19 WAN
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 http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet0/0/2
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 192.65.79.129

```

```

ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 172.16.0.0 255.255.0.0 10.64.1.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 100 voip
  description Incoming dial-peer from PSTN
  translation-profile incoming 100
  session protocol sipv2
  destination dpg 200
  incoming uri request 100
  voice-class codec 100
  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
!
dial-peer voice 200201 voip
  description Outbound Webex Calling 1
  destination-pattern BAD.BAD
  session protocol sipv2
  session target dns:peering1.us.sipconnect.bclld.webex.com:5062
  session transport tcp tls
  voice-class codec 100
  voice-class sip asserted-id pai
  voice-class sip profiles 200
  voice-class sip tenant 200
  voice-class sip options-keepalive profile 100
  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 200202 voip
  description Outbound Webex Calling 2
  destination-pattern BAD.BAD
  session protocol sipv2
  session target dns:peering2.us.sipconnect.bclld.webex.com:5062
  session transport tcp tls
  voice-class codec 100
  voice-class sip profiles 200
  voice-class sip tenant 200
  voice-class sip options-keepalive profile 100

```

```

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 200203 voip
description Outbound Webex Calling 3
destination-pattern BAD.BAD
session protocol sipv2
session target dns:peering3.us.sipconnect.bclid.webex.com:5062
session transport tcp tls
voice-class codec 100
voice-class sip profiles 200
voice-class sip tenant 200
voice-class sip options-keepalive profile 100
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 200204 voip
description Outbound Webex Calling 4
destination-pattern BAD.BAD
session protocol sipv2
session target dns:peering4.us.sipconnect.bclid.webex.com:5062
session transport tcp tls
voice-class codec 100
voice-class sip profiles 200
voice-class sip tenant 200
voice-class sip options-keepalive profile 100
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 101 voip
description outgoing dial-peer to IP PSTN
translation-profile outgoing 200
destination-pattern BAD.BAD
session protocol sipv2
session target ipv4:10.64.1.x:5060
session transport udp
voice-class codec 100
voice-class sip options-ping 60
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

```

```

!
dial-peer voice 200101 voip
description Inbound Webex Calling
session protocol sipv2
session transport tcp tls
destination dpg 100
incoming uri request 200
voice-class codec 100
voice-class sip tenant 200
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
!
gateway
timer receive-rtp 1200
!
sip-ua
no remote-party-id
retry invite 2
transport tcp tls v1.2
crypto signaling default trustpoint sbc6
!
!
line con 0
exec-timeout 5 0
password 7 xxxx
logging synchronous
login
stopbits 1
line aux 0
line vty 0 4
exec-timeout 60 0
password 7 xxxxxx
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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