Application Note

Public

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]

January 31, 2023

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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

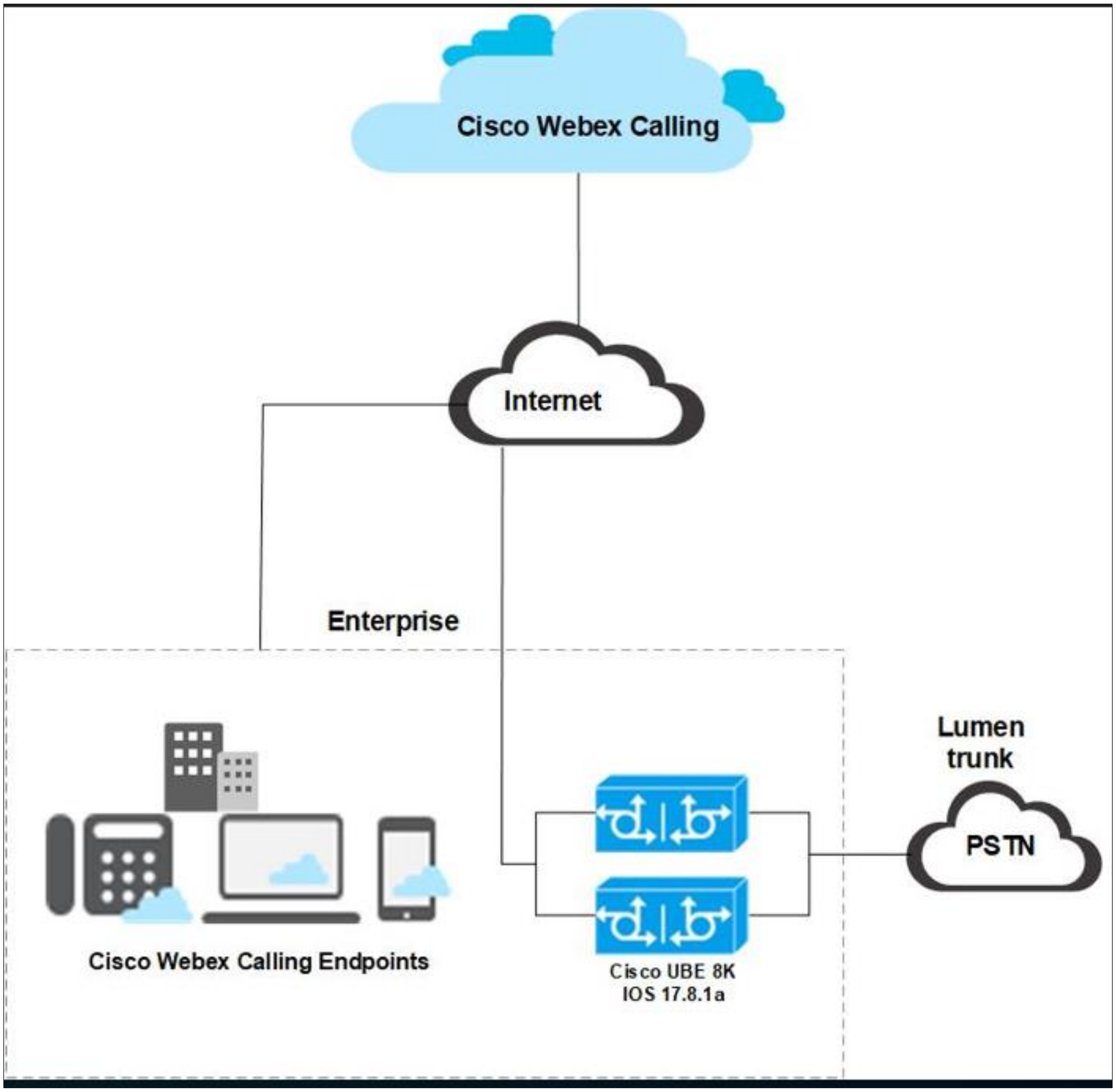


Figure 1: Network Topology with CUBE-HA

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

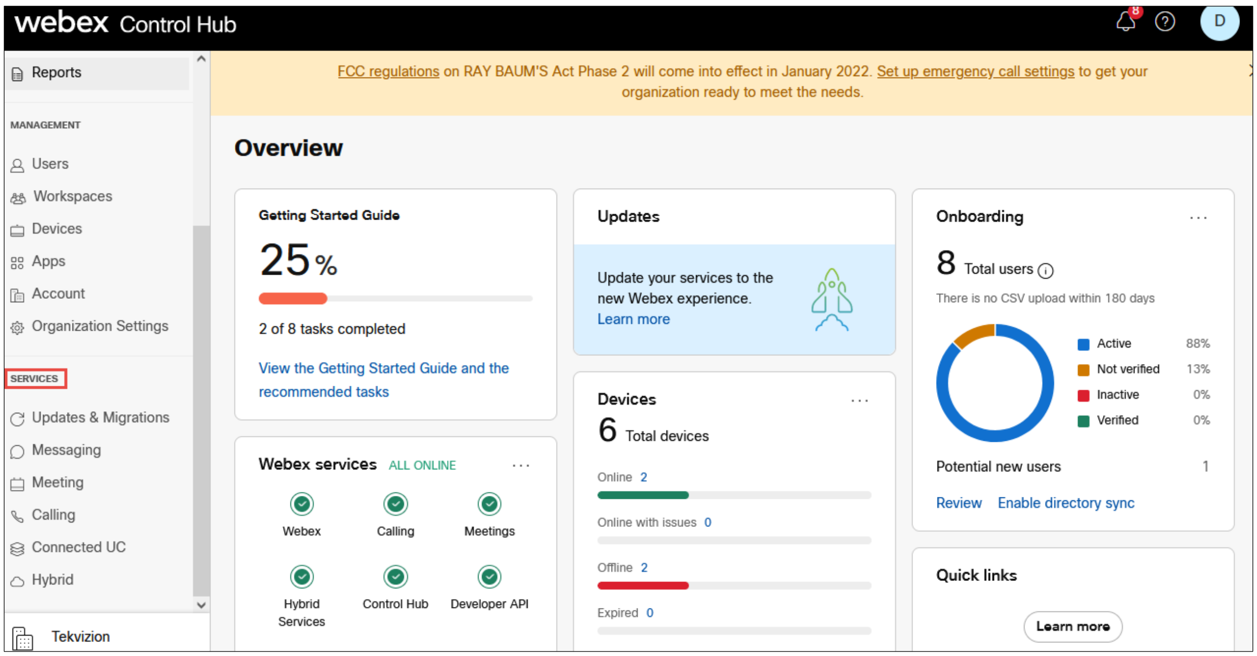


Figure 2: Control Hub Services

Step 2:

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

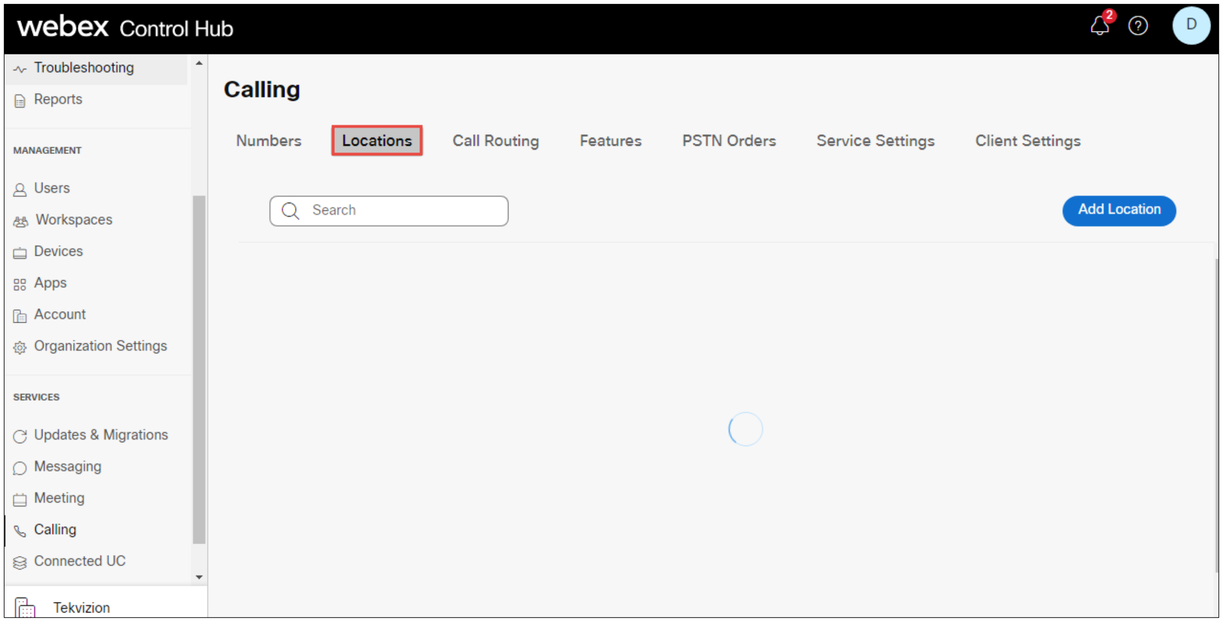


Figure 3: Locations

Step 3:

Click on Add **Location**

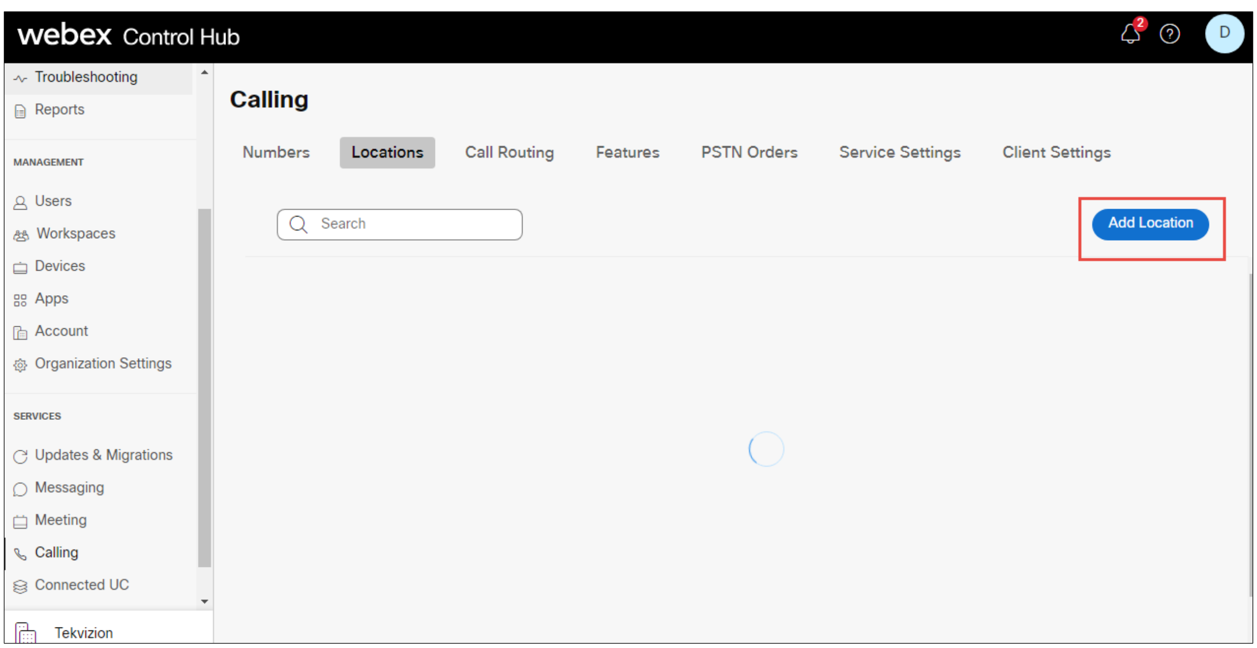


Figure 4: Location creation or selection

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.

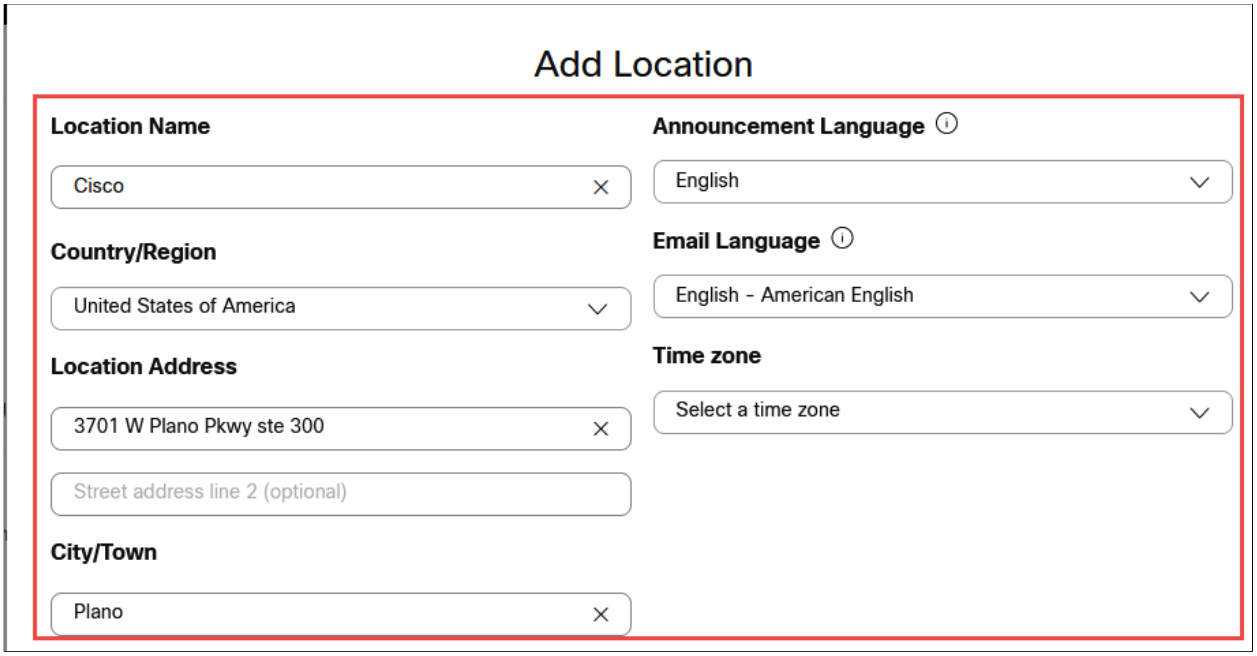


Figure 5: Add location details

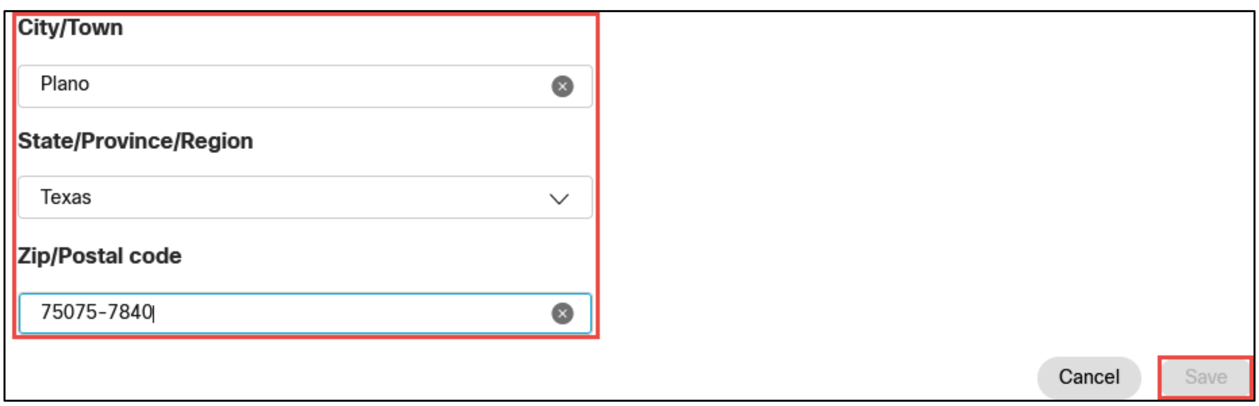


Figure 6: Add location details Contd.,

Step 5:

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

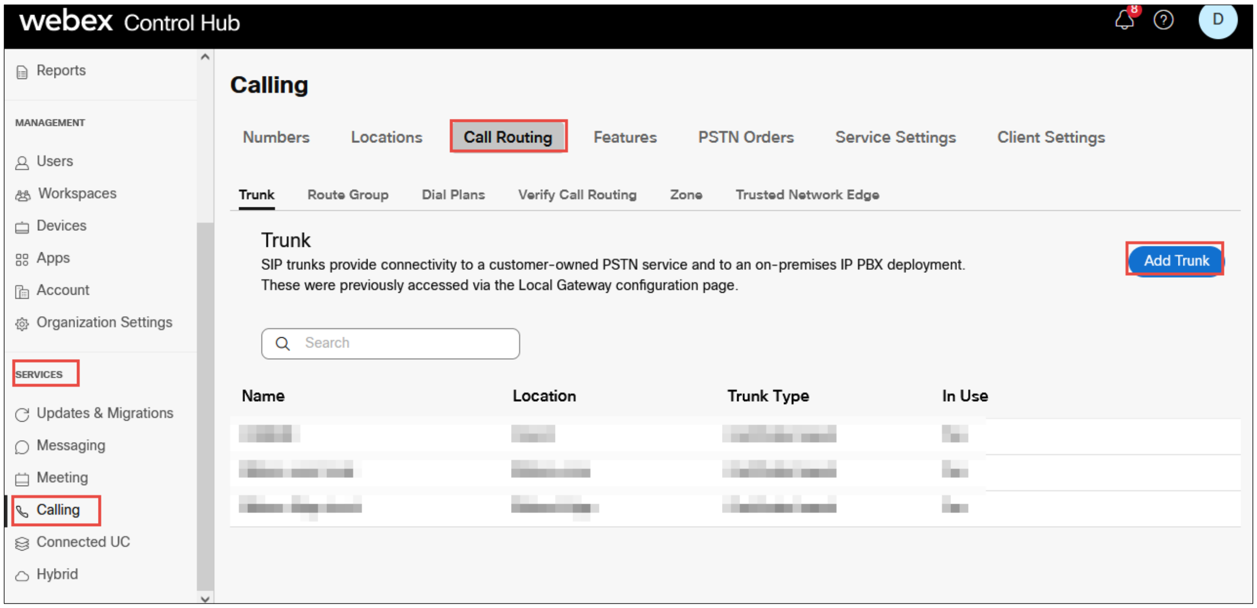
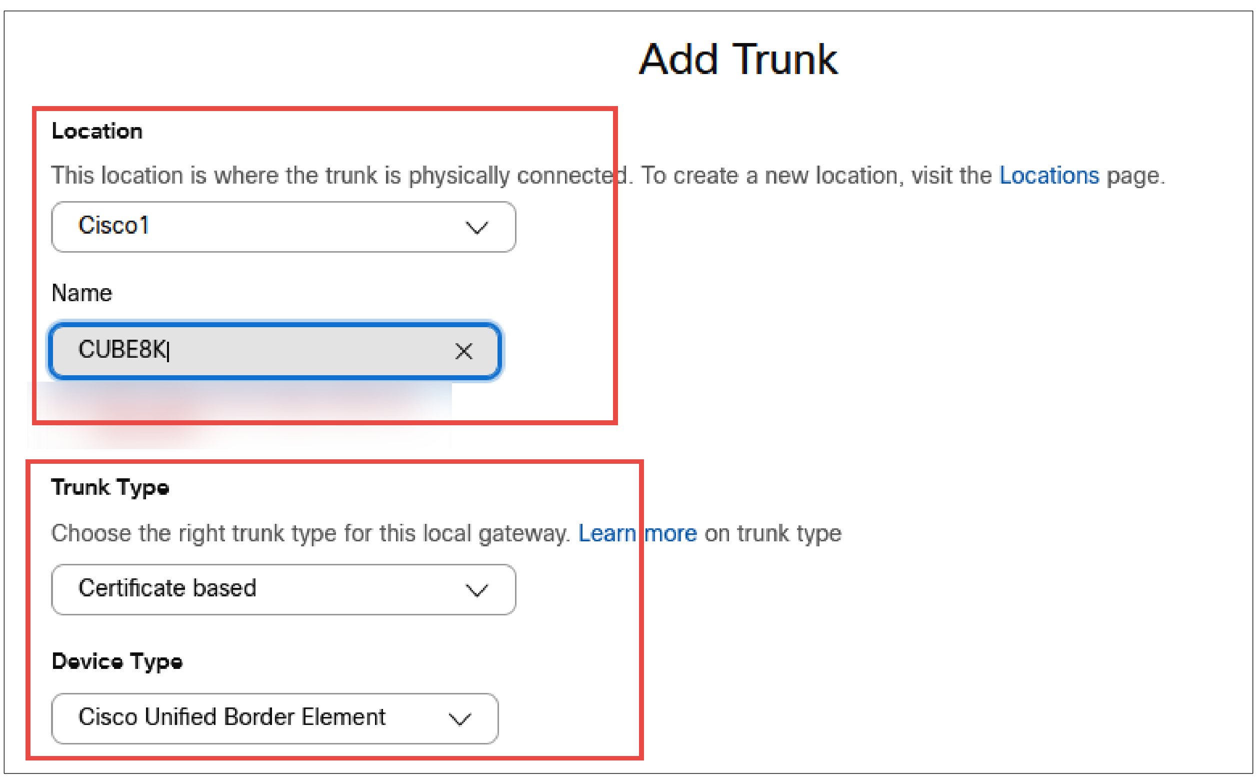


Figure 7: Add Trunk details Contd.,



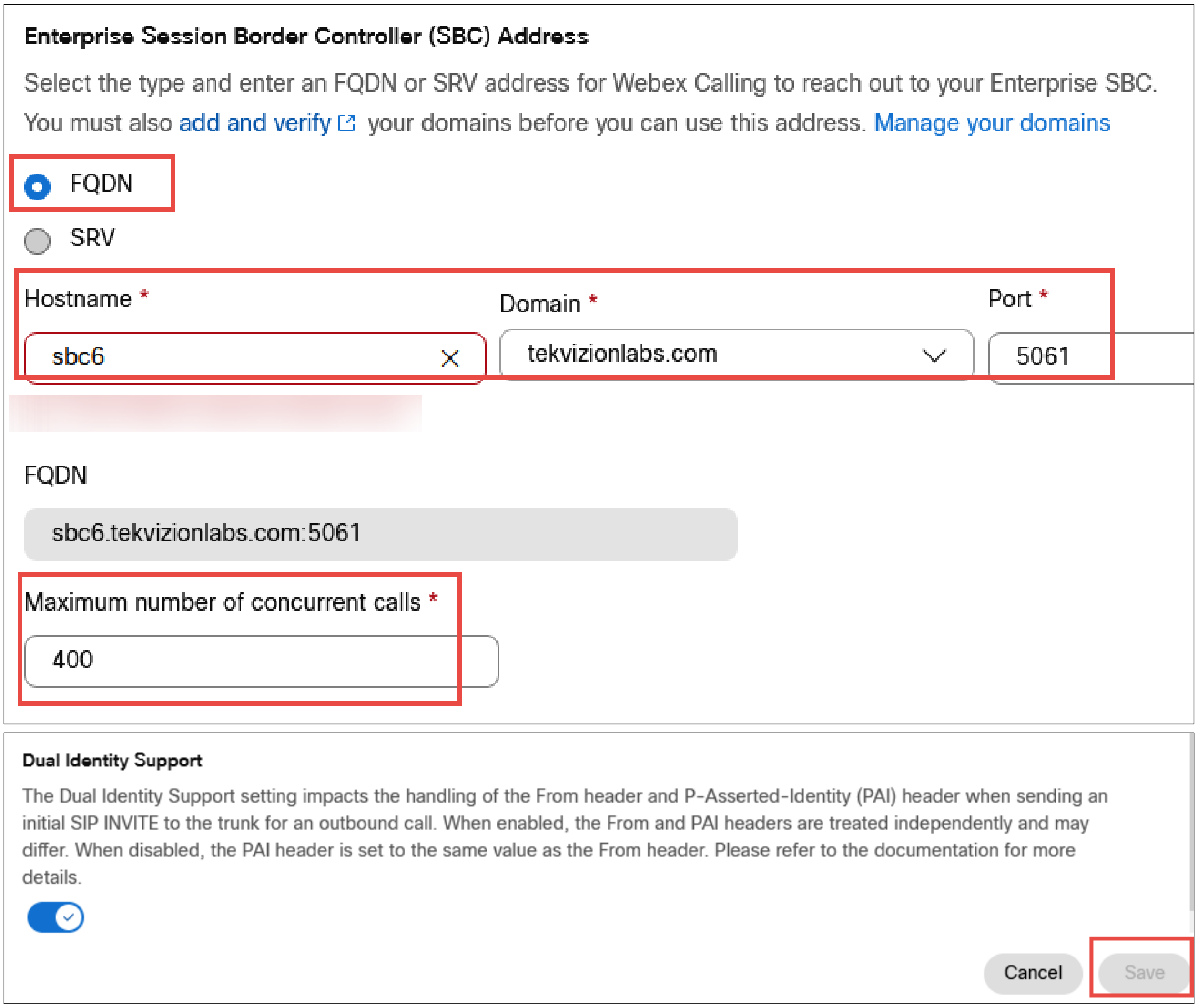


Figure 8: Add Trunk details Contd.

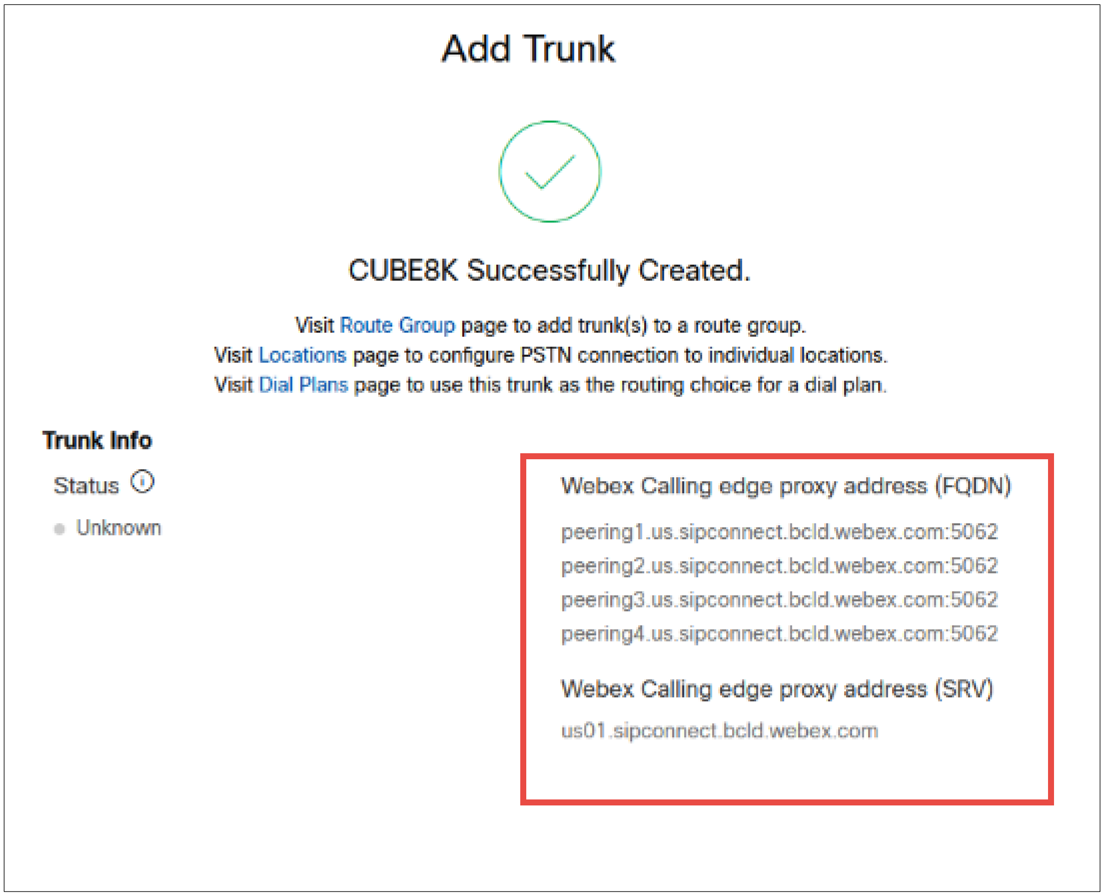


Figure 9: Add Trunk details Contd.,

Step 6:

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

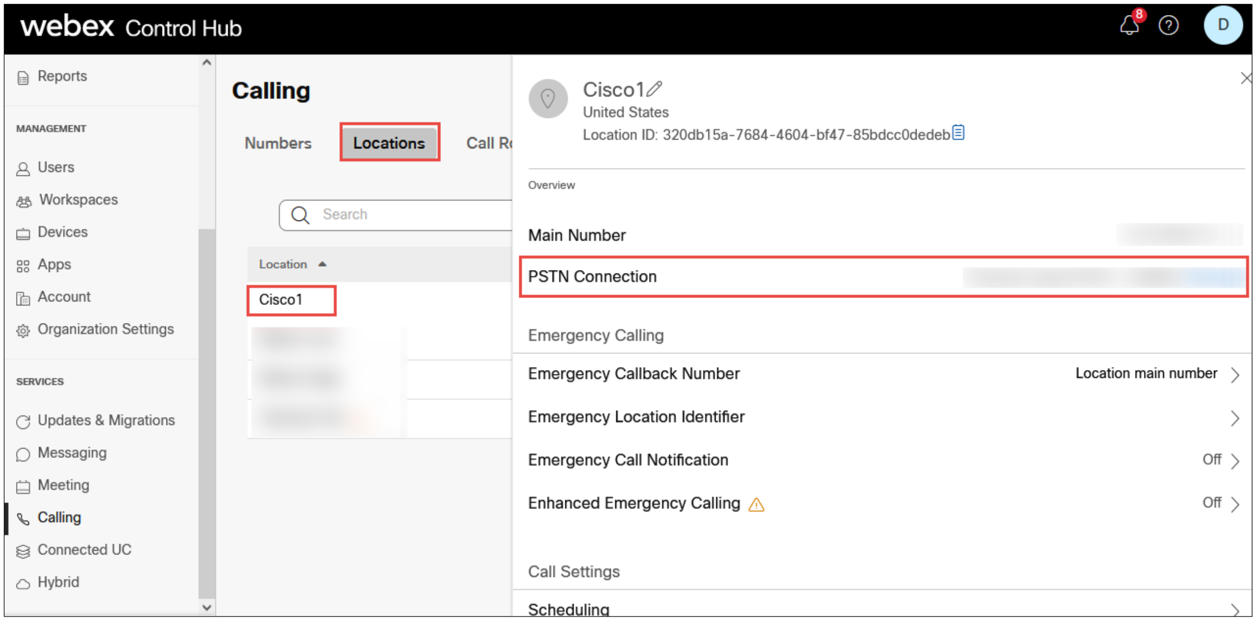


Figure 10: PSTN Connection

Step 7:

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

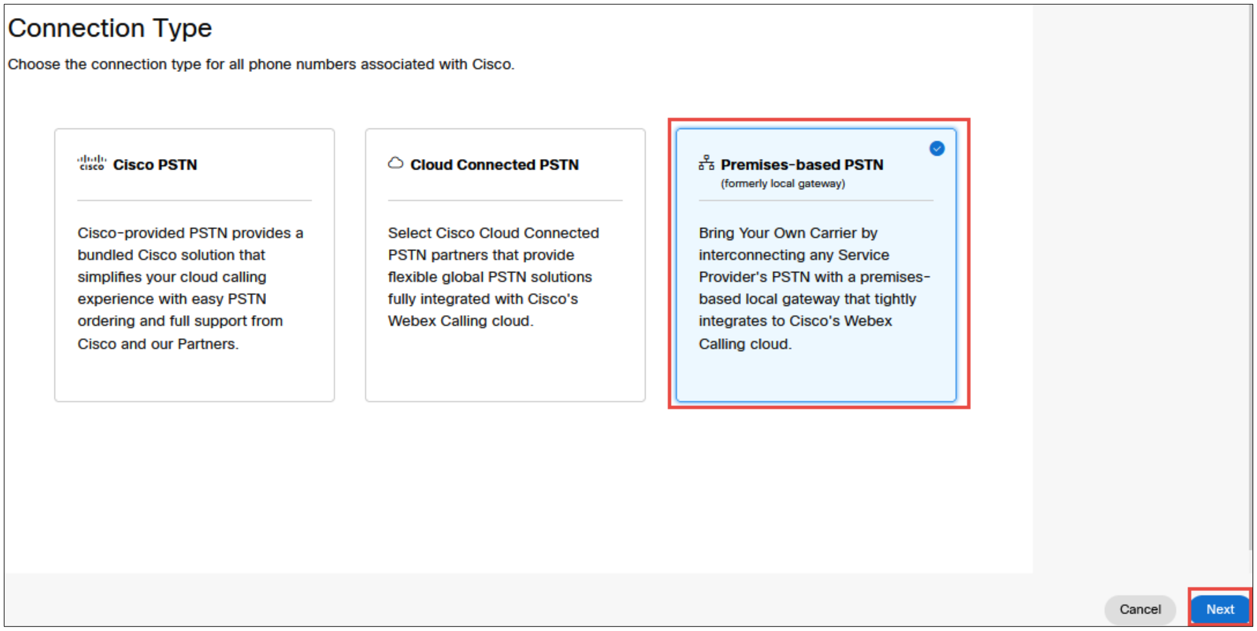


Figure 11: PSTN Connection Contd.,

Step 8:

Select the SIP trunk created earlier and click on Save

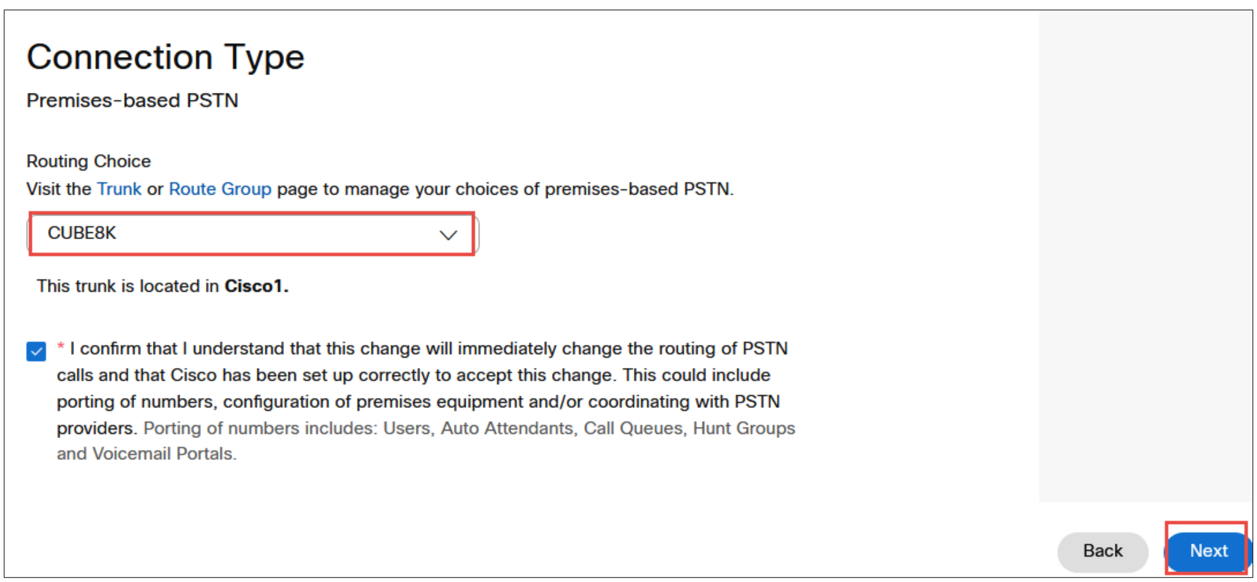


Figure 12: PSTN Connection Contd.,

Step 9:

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

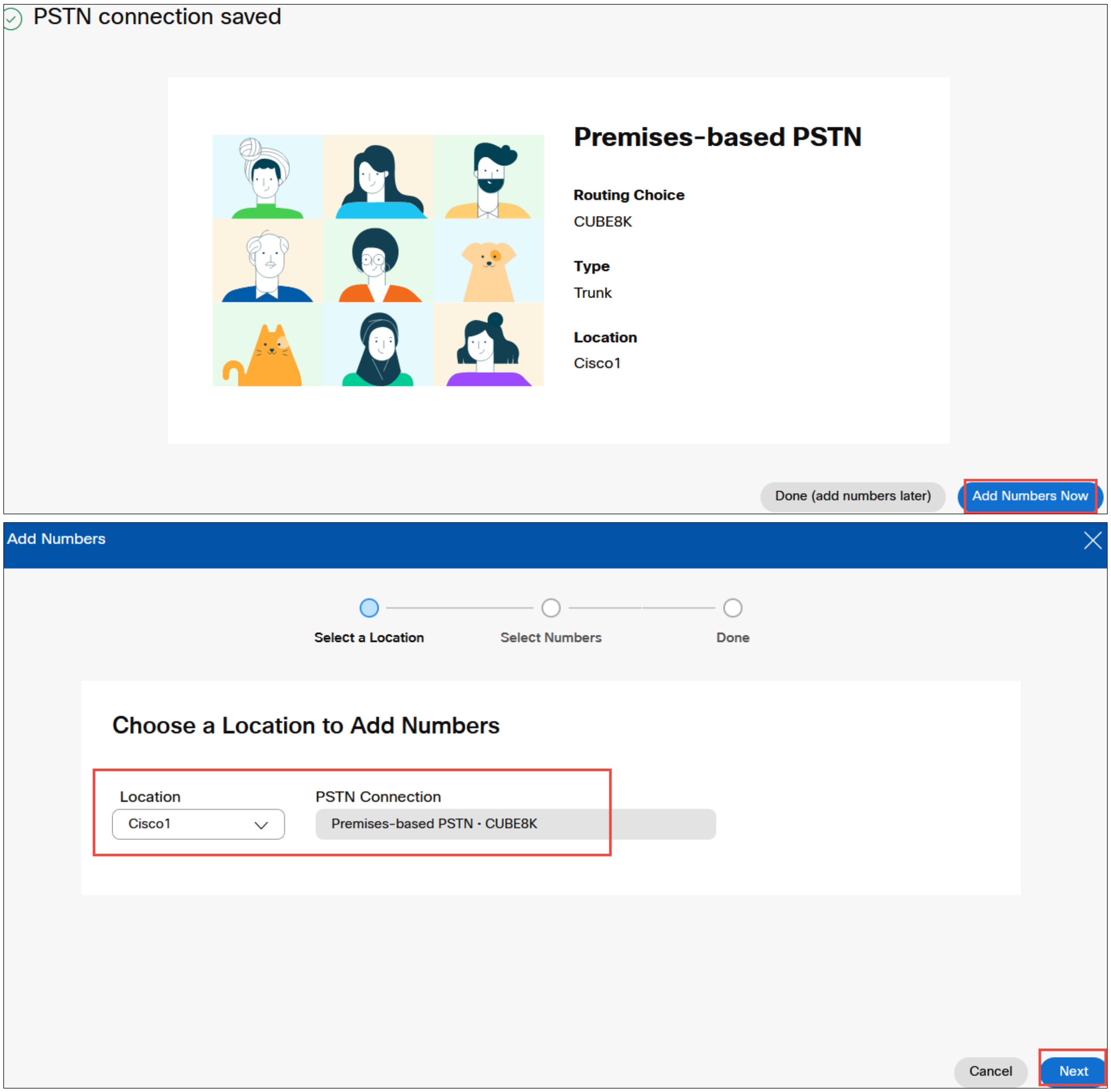


Figure 13: Add Numbers

Step 10:

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

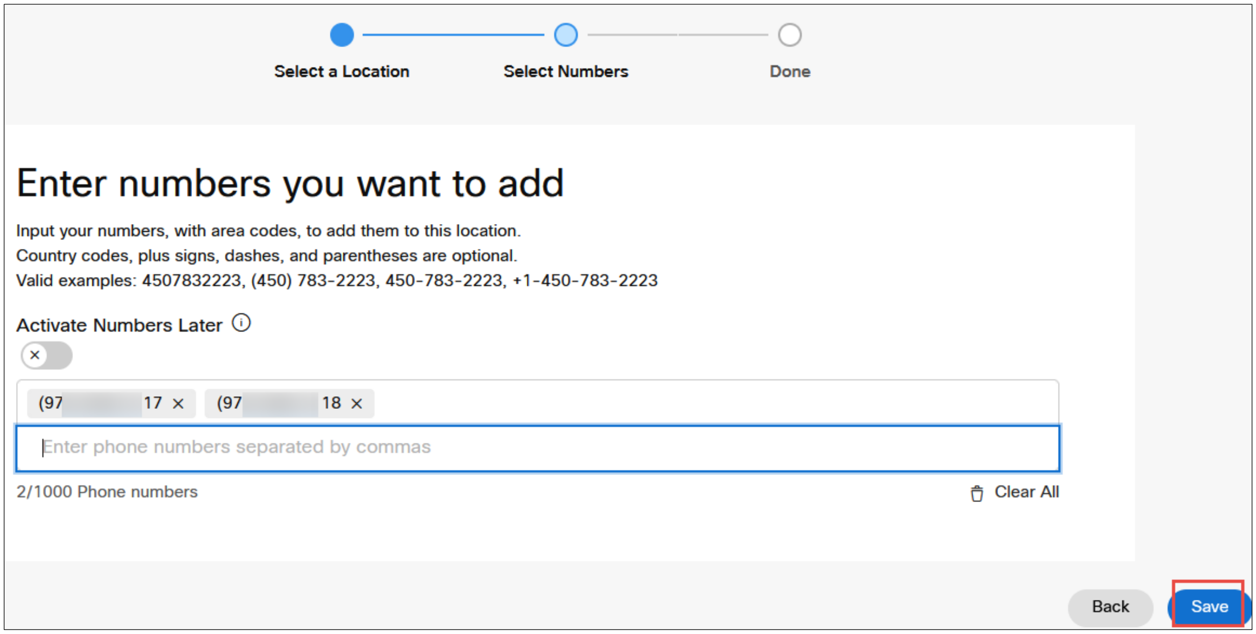


Figure 14: Add Numbers Contd.,

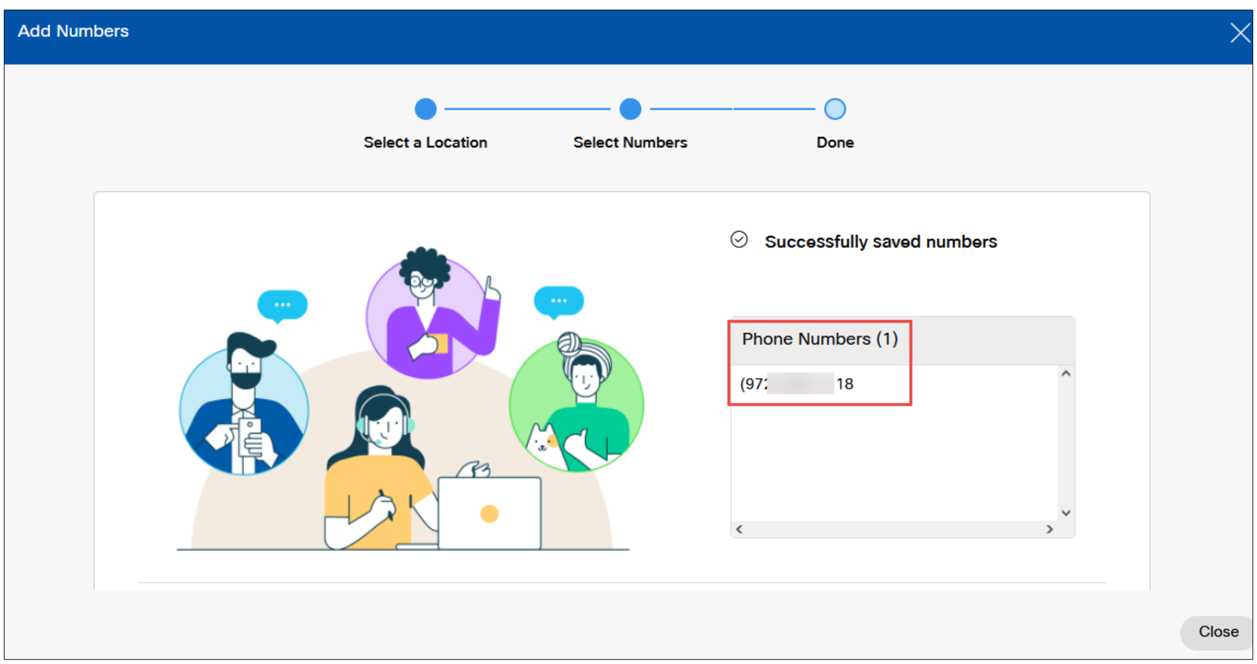


Figure 15: Add Numbers Contd.,

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

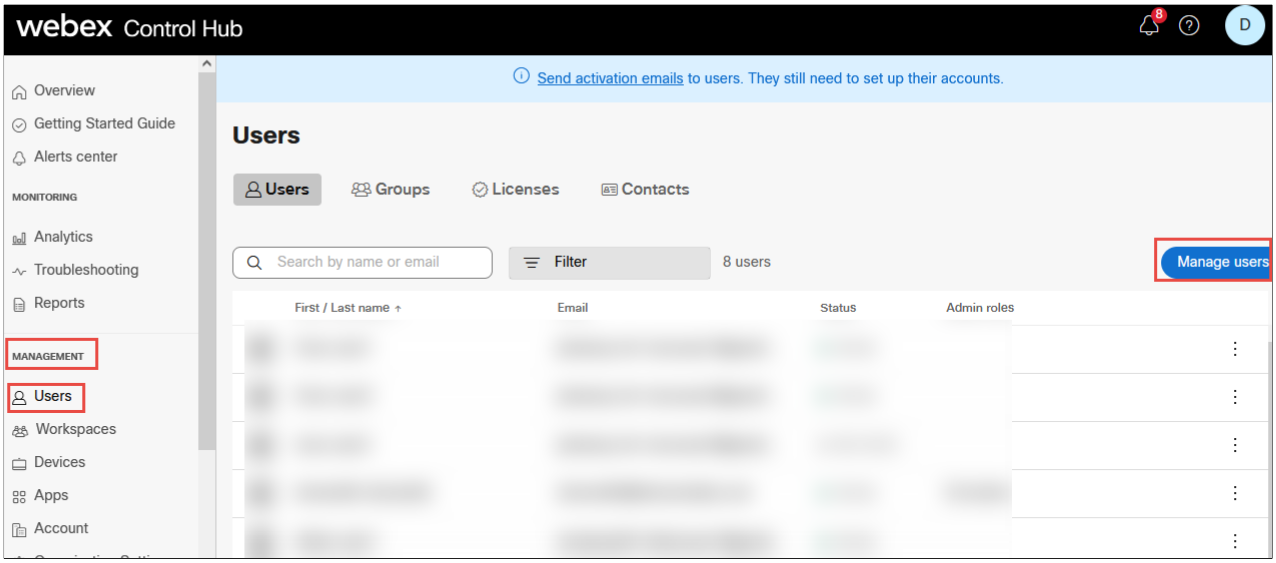


Figure 16: Adding Users

Step 2:

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

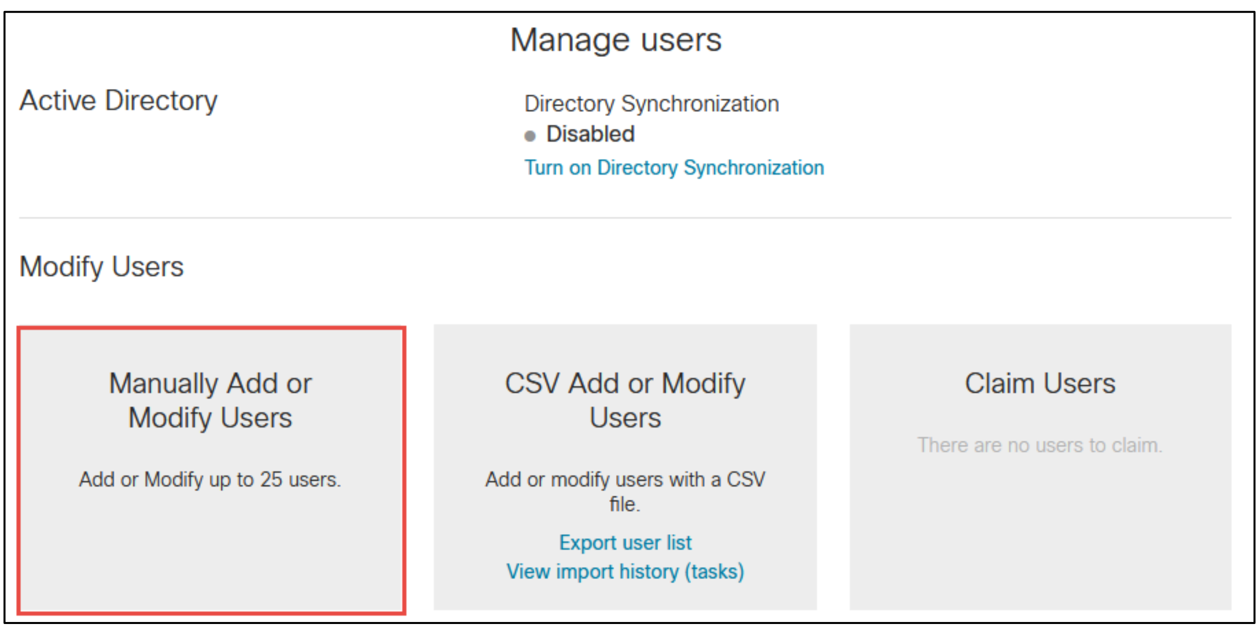


Figure 17: Manually Add or Modify Users

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**

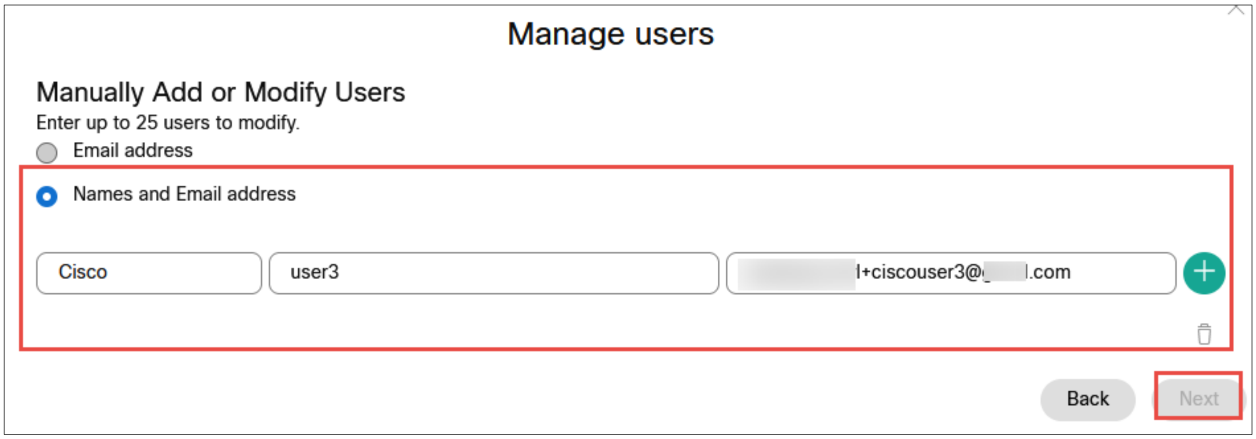


Figure 18: Adding email address and name

Step 4:

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

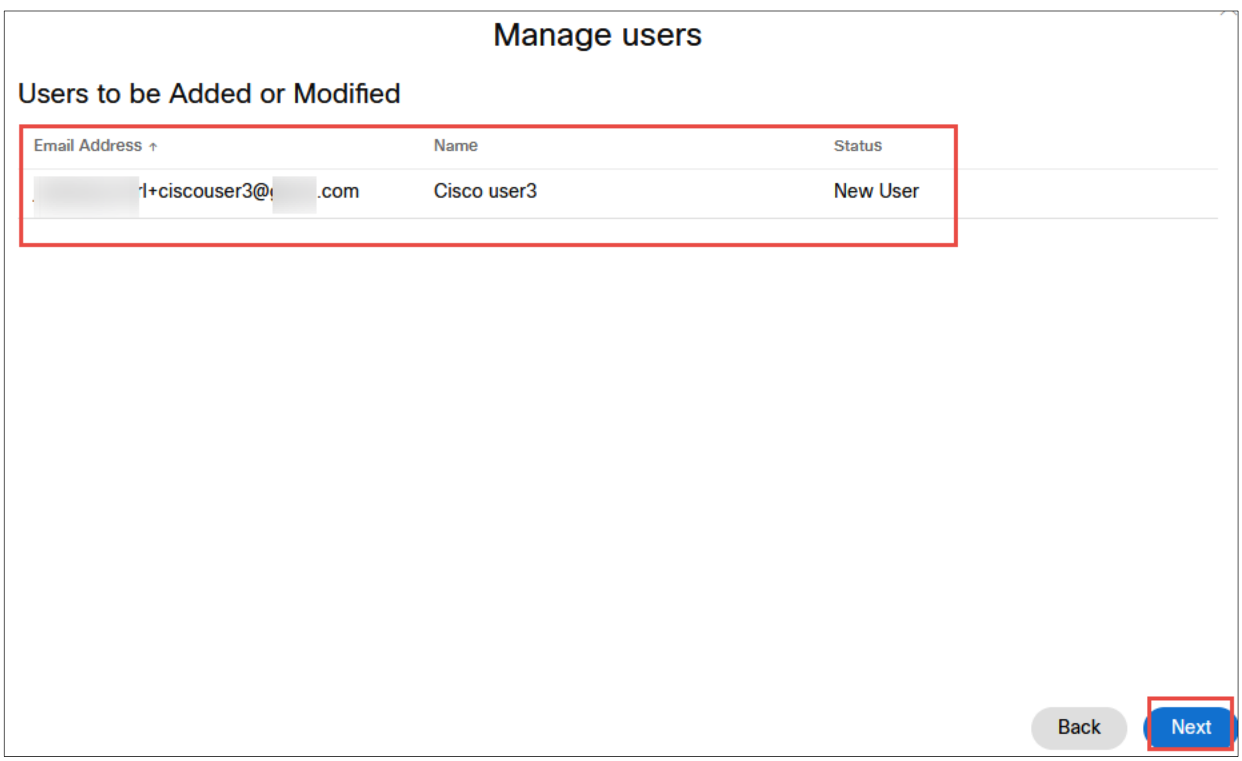


Figure 19: Confirm Adding

Step 5:

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

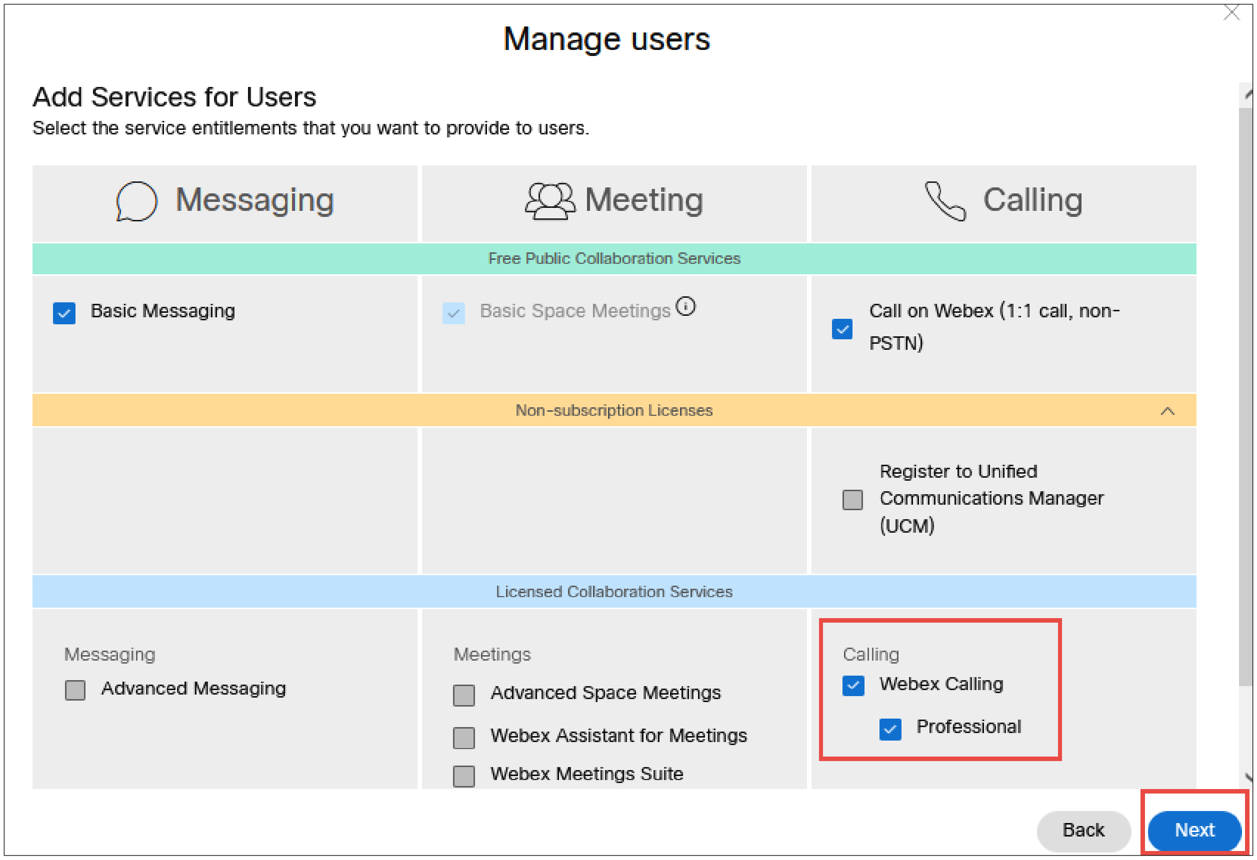


Figure 20: Add Services for Users

Step 6:

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

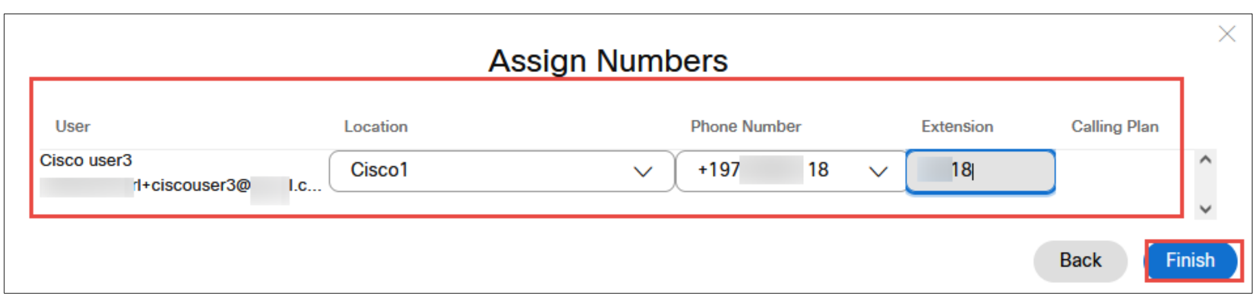


Figure 21: Assign Numbers

Step 7:

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

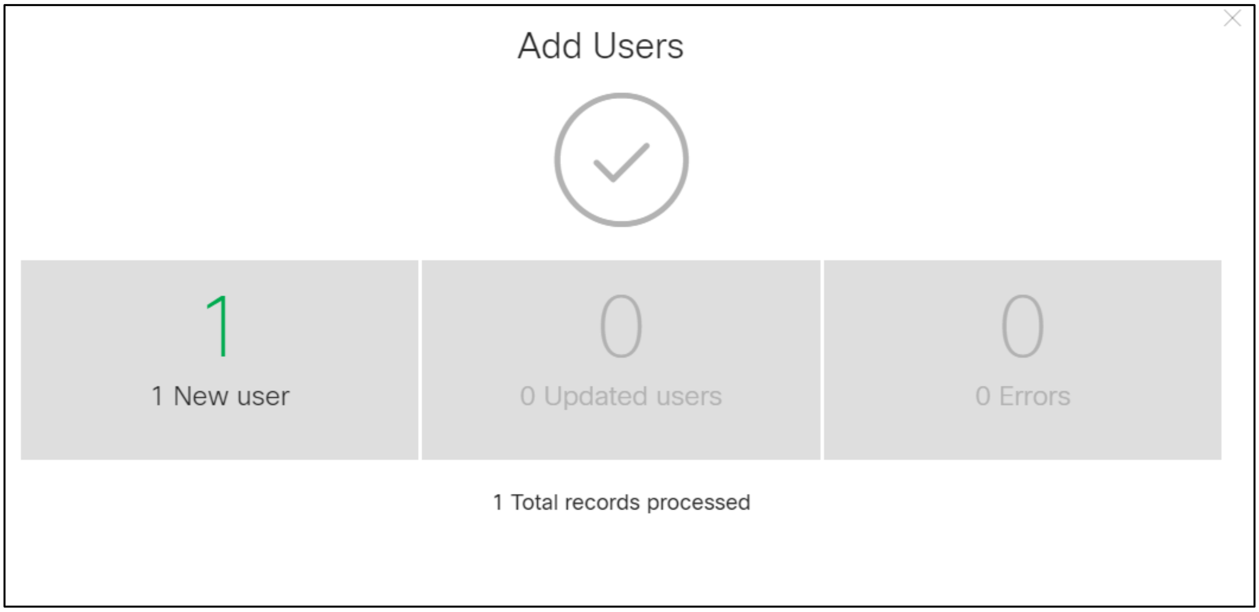


Figure 22: Add User successful

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.

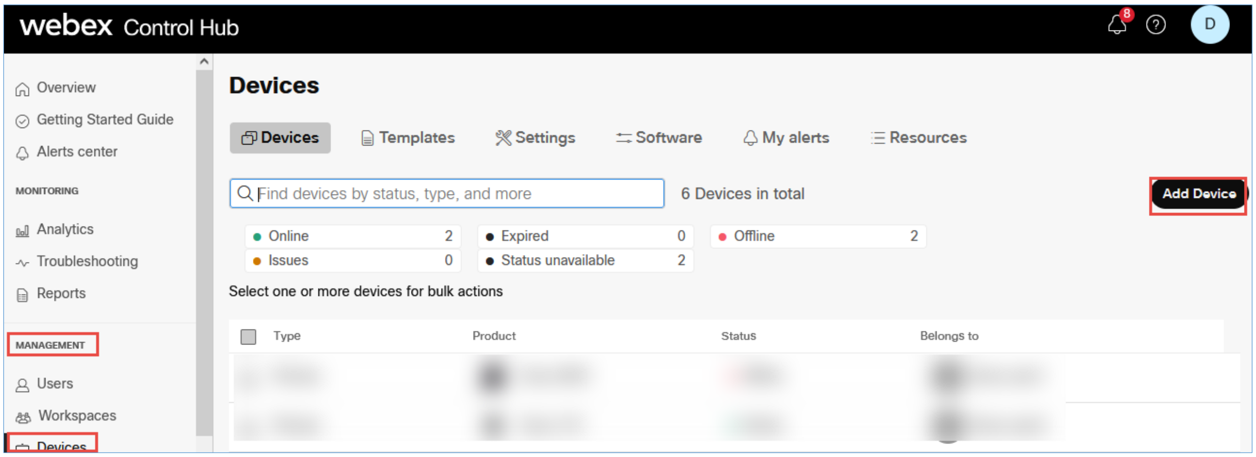


Figure 23: Add Devices Window

Step 2:

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

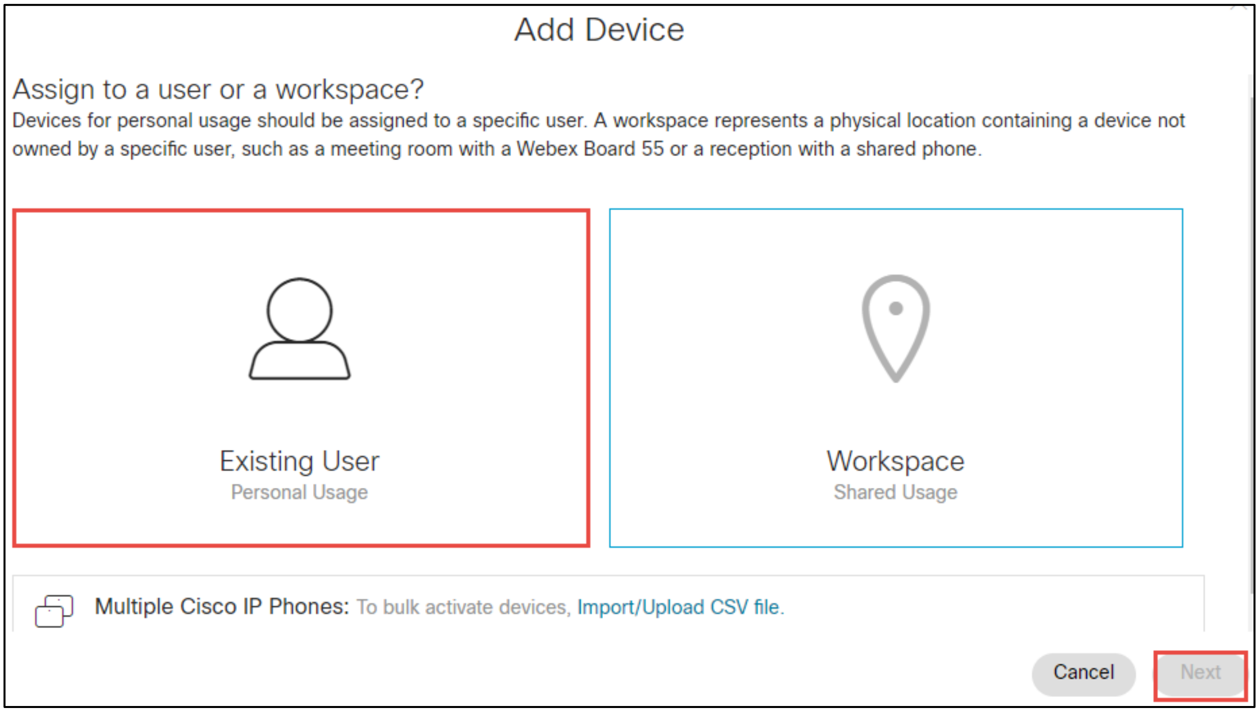


Figure 24: Assign to a user or a place

Step 3:

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

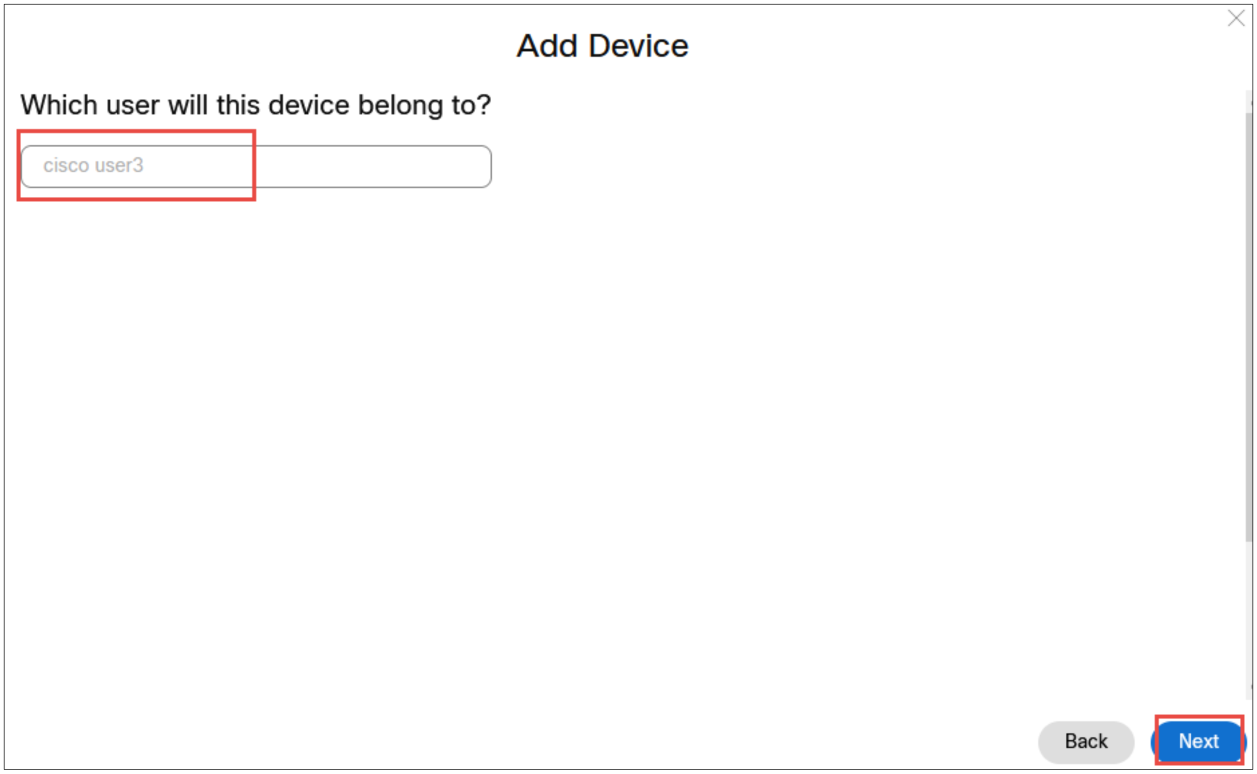


Figure 25: User Association with Device

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.

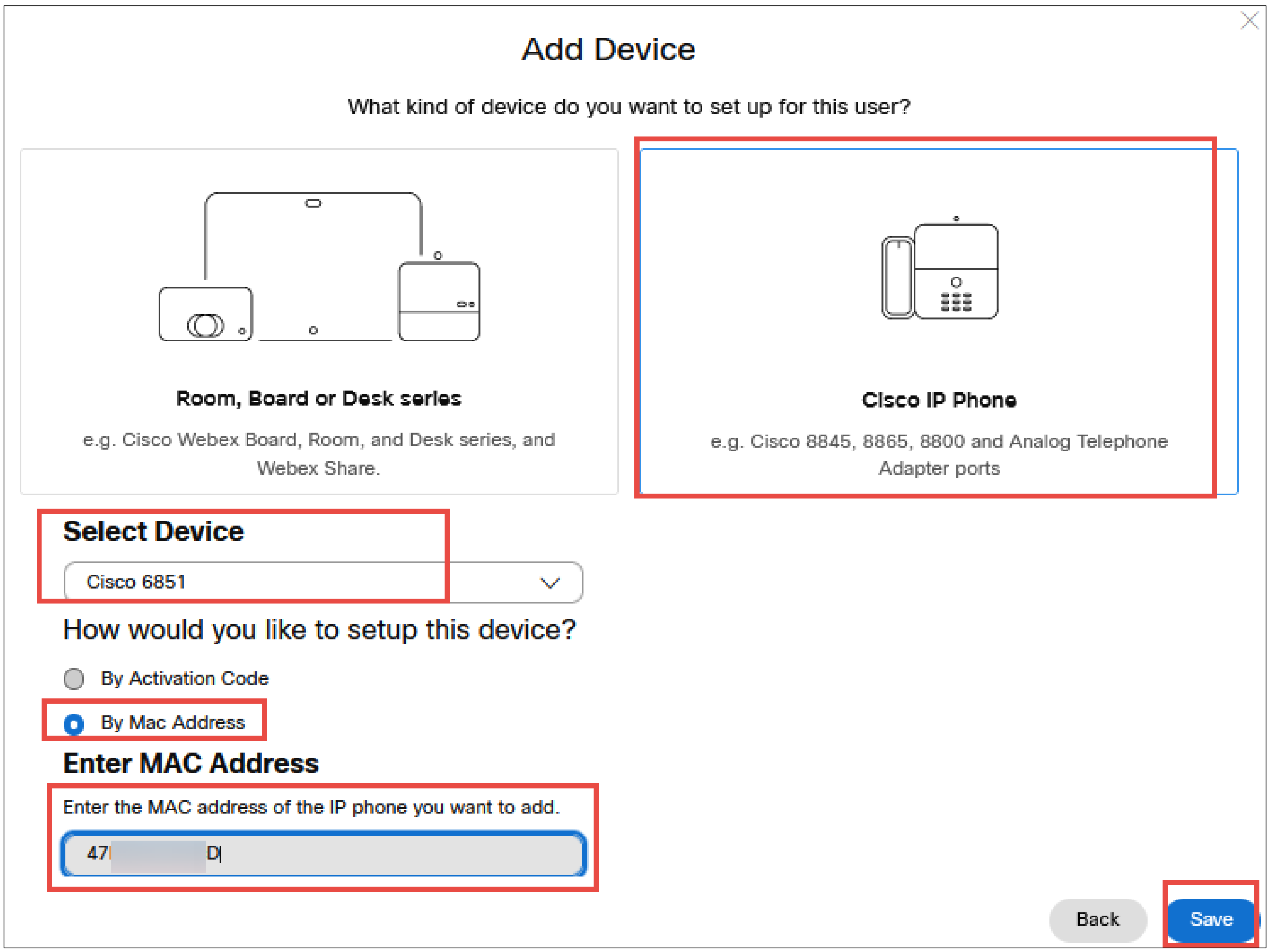


Figure 26: Select Device and add MAC address

1.4 Assign main number to a location

Step 1:

Assign number in location as the Main number.

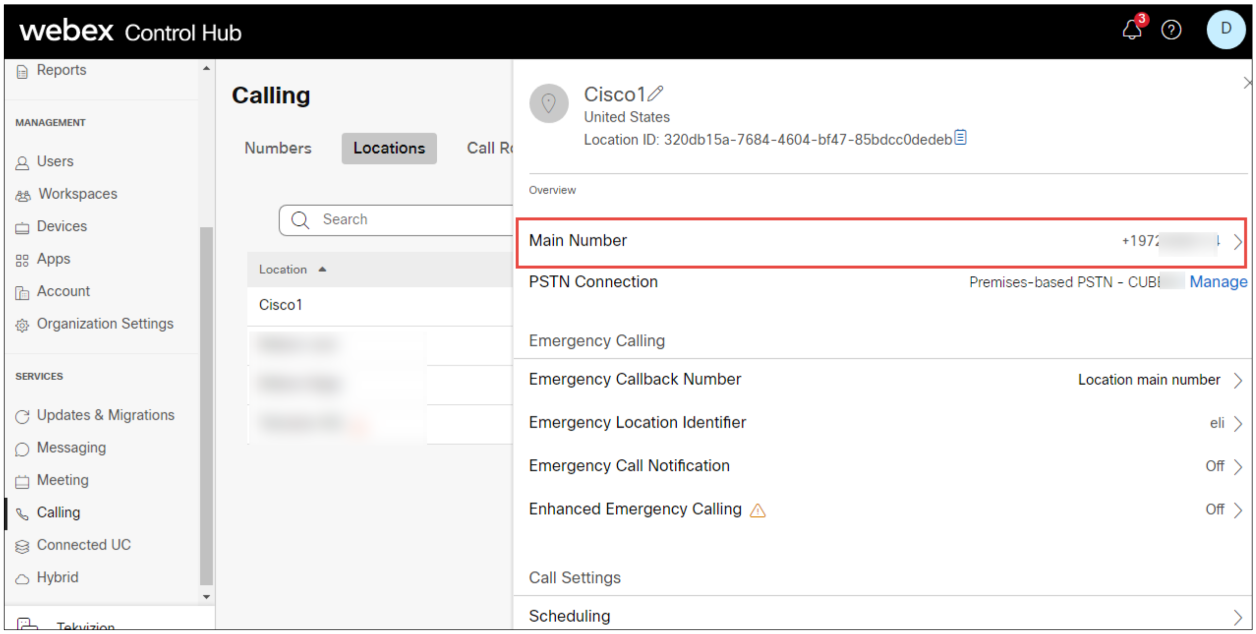


Figure 27: Assign Main number in location

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
```

2.11.2 Tenant to PSTN

```
voice class tenant 100
  session transport udp
  url sip
  error-passthru
  bind control source-interface GigabitEthernet0/0/1
  bind media source-interface GigabitEthernet0/0/1
  no pass-thru content custom-sdp
  privacy-policy passthru
```

2.12 Number translation rules

The following translation rule applies for non +E164 from PSTN/PBX to Webex calling in E164.

2.12.1 To Webex Calling

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

2.12.2 To PSTN

```
voice translation-rule 200
 rule 1 /^\+1\(.*\)/ /\1/
 rule 2 /^\0\([2-9].........\)/ /\1/
 rule 4 /^\+91\(.*\)/ /01191\1/
!
voice translation-profile 200
 translate calling 200
 translate called 200
```

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.bcld.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.bcld.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.bcld.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.bcld.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 tekvizionlabs.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.tekvizionlabs.com
 subject-name cn=sbc6.tekvizionlabs.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.bcld.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.bcld.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.bcld.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.bcld.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 tekvizionlabs.com
!
login on-success log
!
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-2307055185
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-2307055185
 revocation-check none
 rsakeypair TP-self-signed-2307055185
!
crypto pki trustpoint SLA-TrustPoint
 enrollment pkcs12
 revocation-check crl
!
crypto pki trustpoint sbc6
 enrollment pkcs12
 revocation-check crl
 rsakeypair sbc6
!
!
```

```
crypto pki 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.bcld.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.bcld.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.bcld.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.bcld.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 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
```

Important Information

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