Application Note

Public

Connecting Cisco Webex Calling to Lumen SIP Trunk via a Certificate-based Local Gateway on Cisco Unified Border Element (CUBE) v14.6 [IOS-XE 17.9.1a]

January 31, 2023

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Introduction

Customers using Webex Calling the option of connecting to the PSTN using a Cisco Unified Border Element (CUBE) as Certificate-based Local Gateway LGW.

This application note describes a tested CUBE-HA configuration for connecting Webex Calling certificate-based LGW to the PSTN and the PBX using IP Trunking service. The same CUBE platform can be configured to connect with multiple Webex Calling tenants using Certificate-based Local Gateway services. Please refer to the documentation and the content provided at www.cisco.com/go/interoperability for guidance on how to adjust this tested configuration to meet the specific requirements of your trunking service.

This document assumes the reader is knowledgeable with the terminology and configuration of CUBE. The configuration settings specifically required for Webex Calling certificate-based Local Gateway for IOS-XE 17.9.1a or later 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 Cisco Webex Calling Certificate-based LGW on a Catalyst 8300 CUBE platform [IOS-XE 17.9.1] for connectivity to Enterprise IP PBX and PSTN SIP Trunking service.
- 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 CUBE configuration presented in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between PSTN 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 successful interworking with the service provider network.

Network Topology

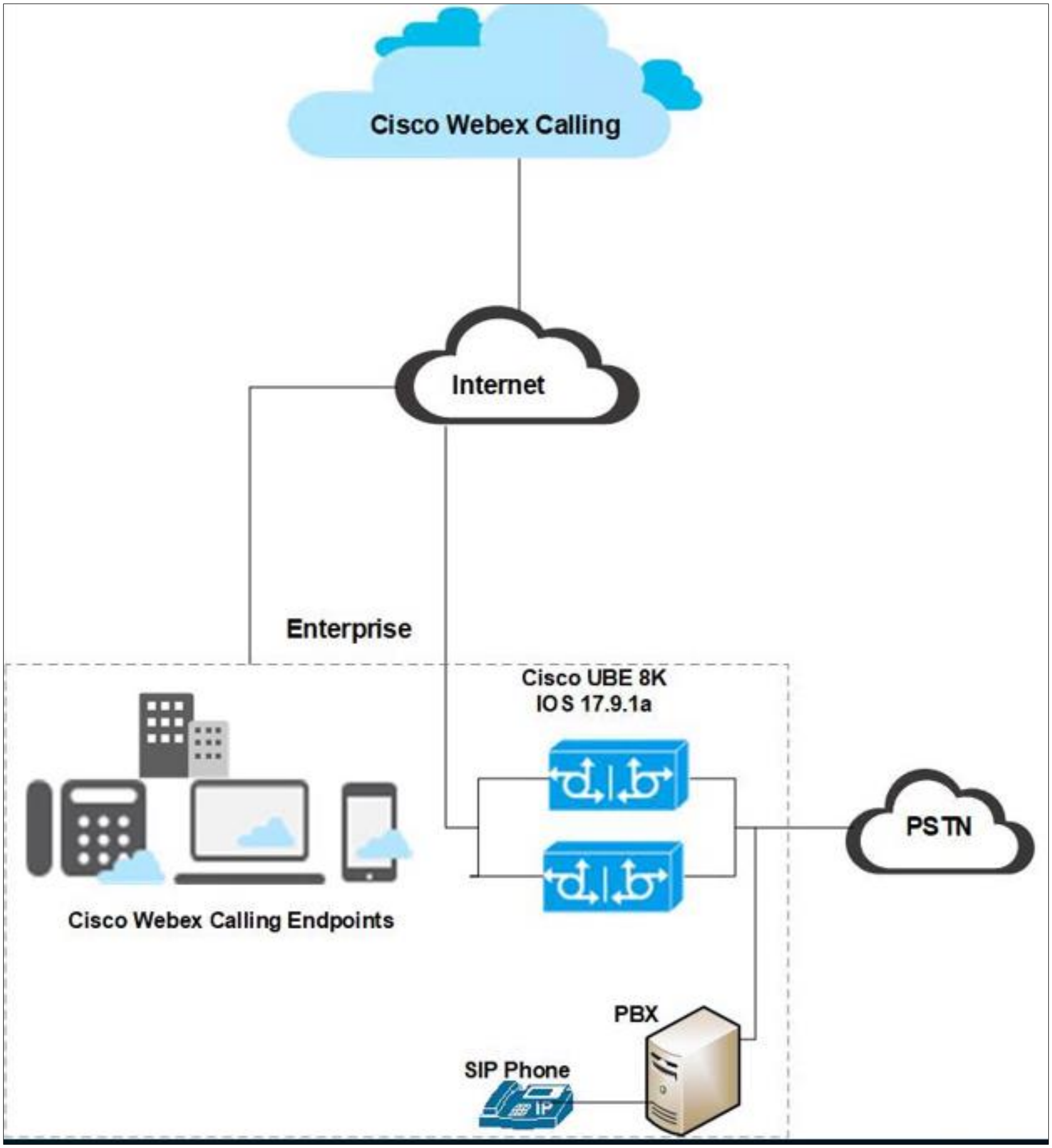


Figure 1: Network Topology

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 PSTN and PBX	UDP with RTP
Session Refresh	YES

System Components

Hardware Requirements

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

Software Requirements

- Cisco UBE:
  - 14.6 running IOS-XE 17.9.1
  - Cisco IOS Software [Cupertino], c8000be Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 17.9.1, RELEASE SOFTWARE (fc3)
- Cisco MPP-Version: sip68xx.11-3-7MPP0001-272.loads
- Cisco MPP-Version: sip8845_65.11-3-7MPP0001-272.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
- Auto Attendant
- Call hold & Resume(MoH)
- Semi-attended and Attended Call transfer
- Blind Transfer
- Call forward all
- DTMF (RFC2833)
- IP-PBX Calling number privacy

Features Not Supported

- None

Features Not Tested

- Scalability
- Multi-tenancy

Features Not Applicable

- None

Caveats

The following are the observations from Cisco UBE.

- In long duration calls, Webex Calling does not respond to a session refresh Re-INVITE received from PSTN, and thus, CUBE disconnects the call. Hence, “no session refresh” is configured towards Webex calling and session refresh from Webex Calling is observed with an “UPDATE” message.
- In Webex Calling call hold scenarios, Webex Calling sends two Re-INVITEs on hold. One with send-only and the other with send-recv. Webex Calling does play MOH.
- Webex Calling does not accept GCM crypto encryption suite.
- In a video call redundancy, audio is preserved on failover, but video is not preserved in an outbound call from a Webex Calling user to an enterprise IP PBX phone. In an inbound call from enterprise IP PBX phone to a Webex Calling user, video is partially preserved in one direction (Webex Calling to IP PBX user) and not in the other direction.
- Webex does not negotiate ICE candidate attributes in an ATA 19X FAX and in video MPP phones.

Configuration

1 Configuring Cisco Webex Calling

1.1 Add location

Step1:

Login to Cisco Webex Control Hub and navigate to Services

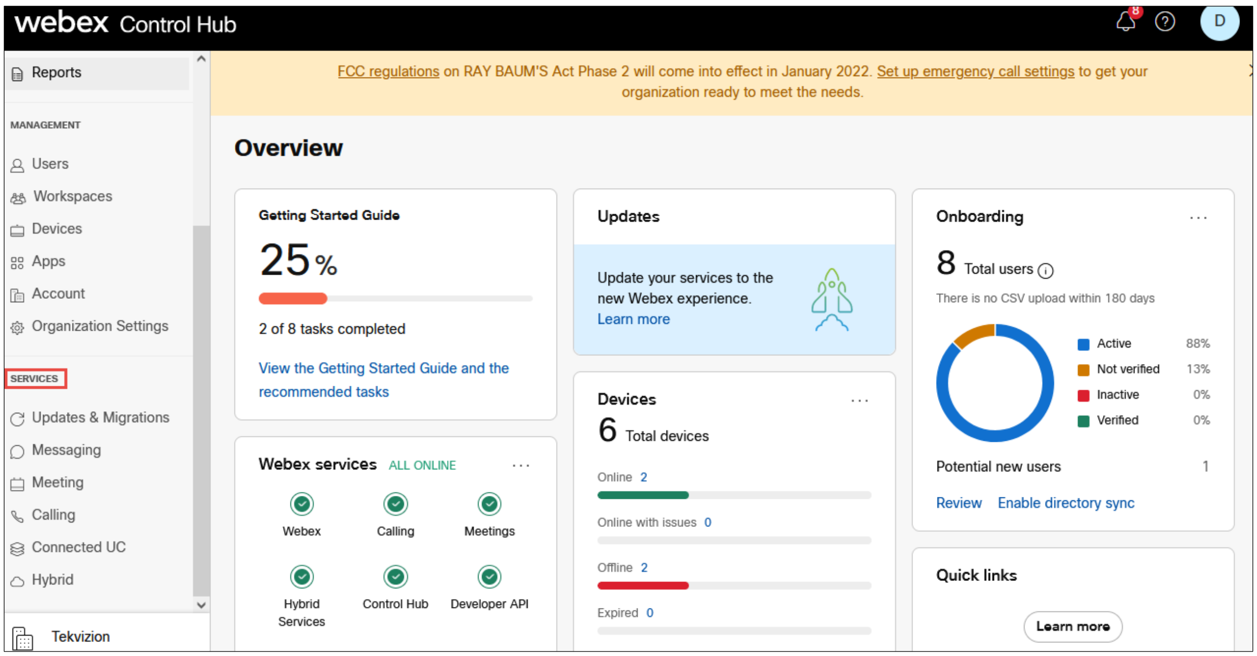


Figure 2: Control Hub Services

Step 2:

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

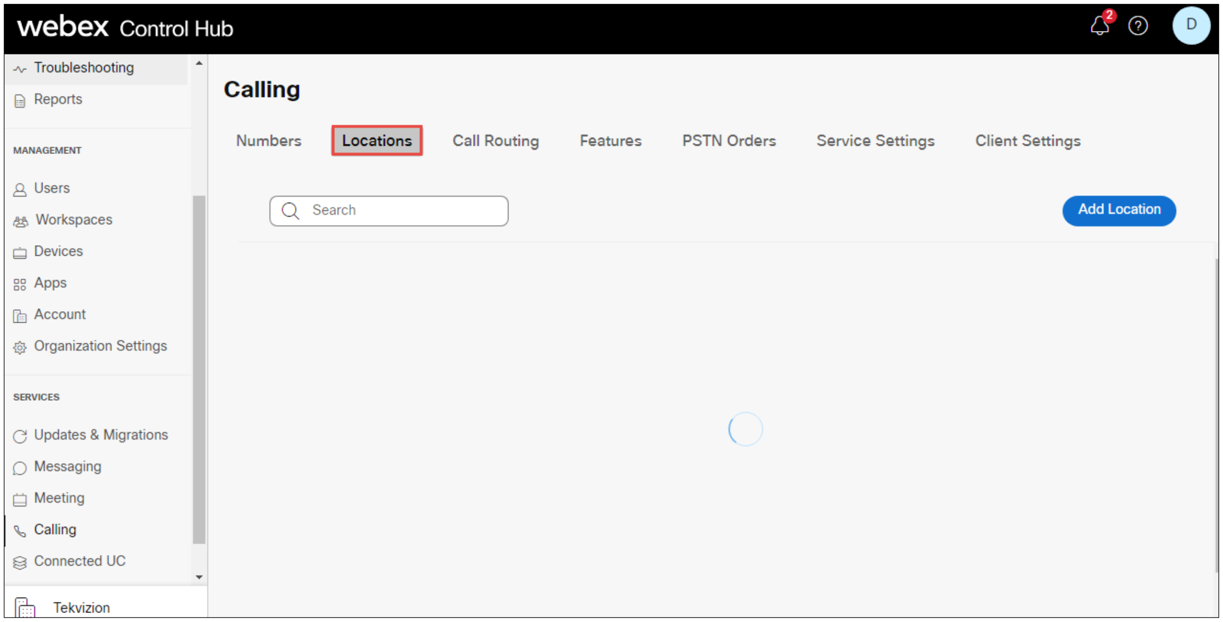


Figure 3: Locations

Step 3:

Click on Add **Location**

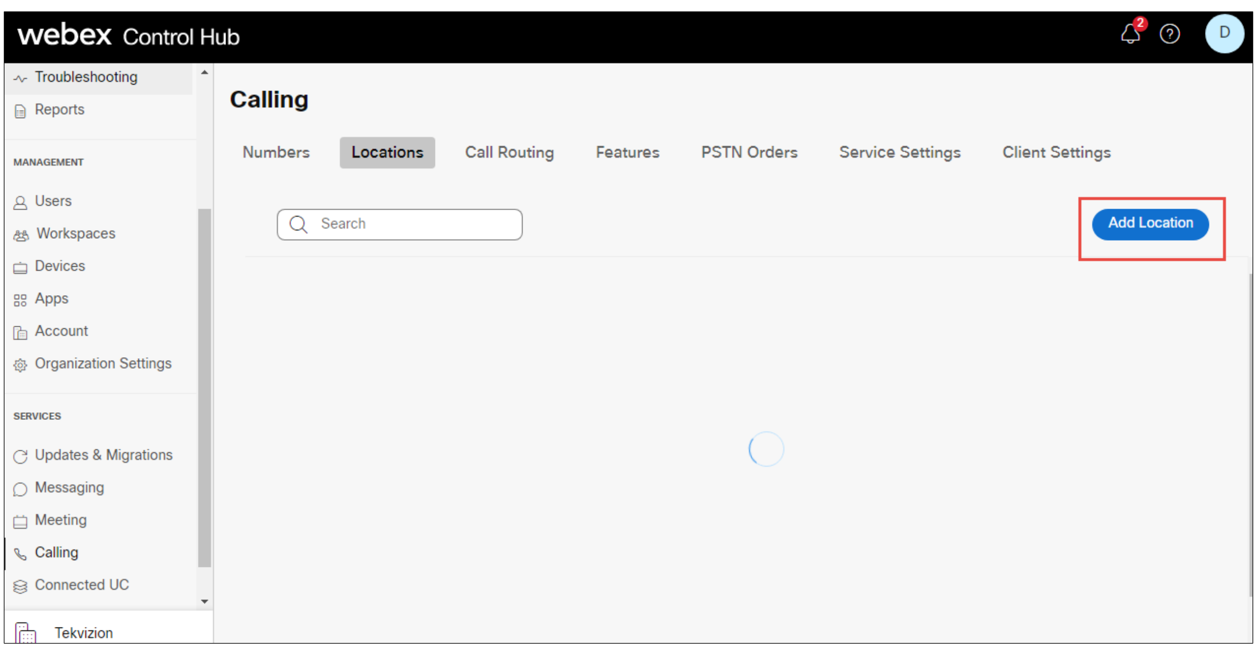


Figure 4: Location creation or selection

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.

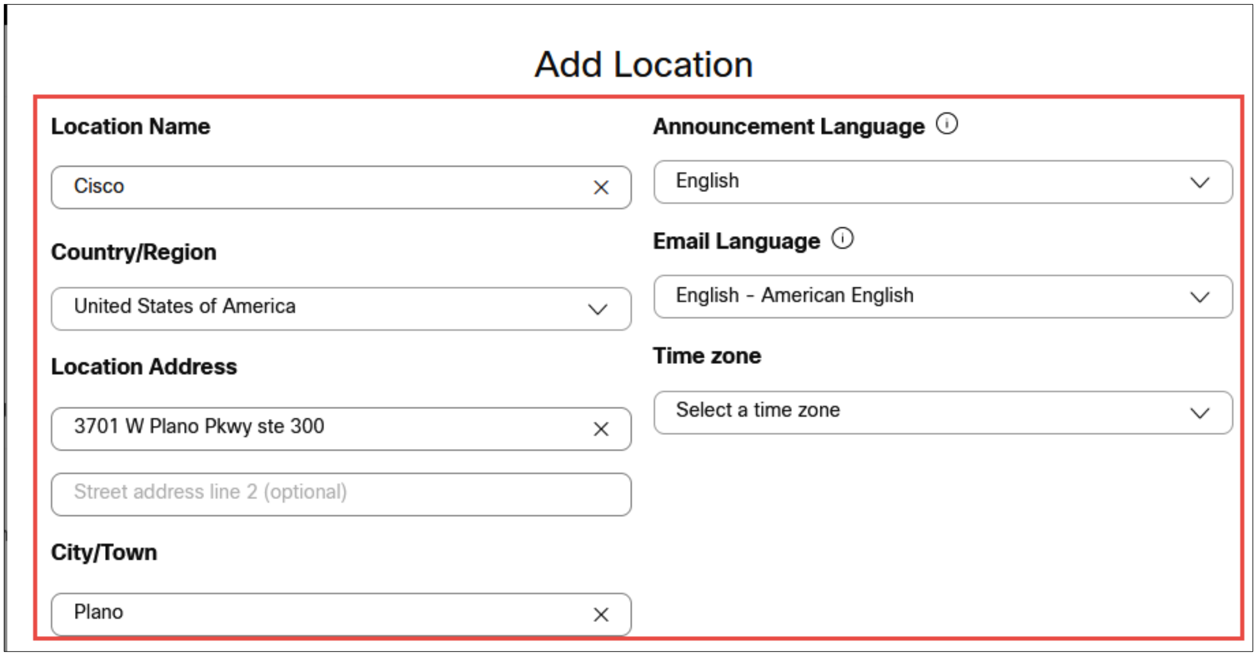


Figure 5: Add location details

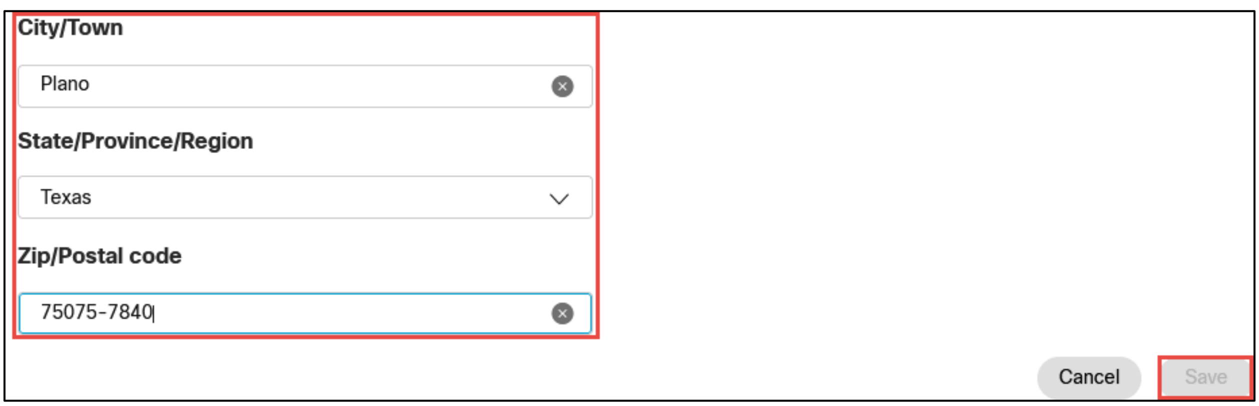


Figure 6: Add location details Contd.,

Step 5:

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

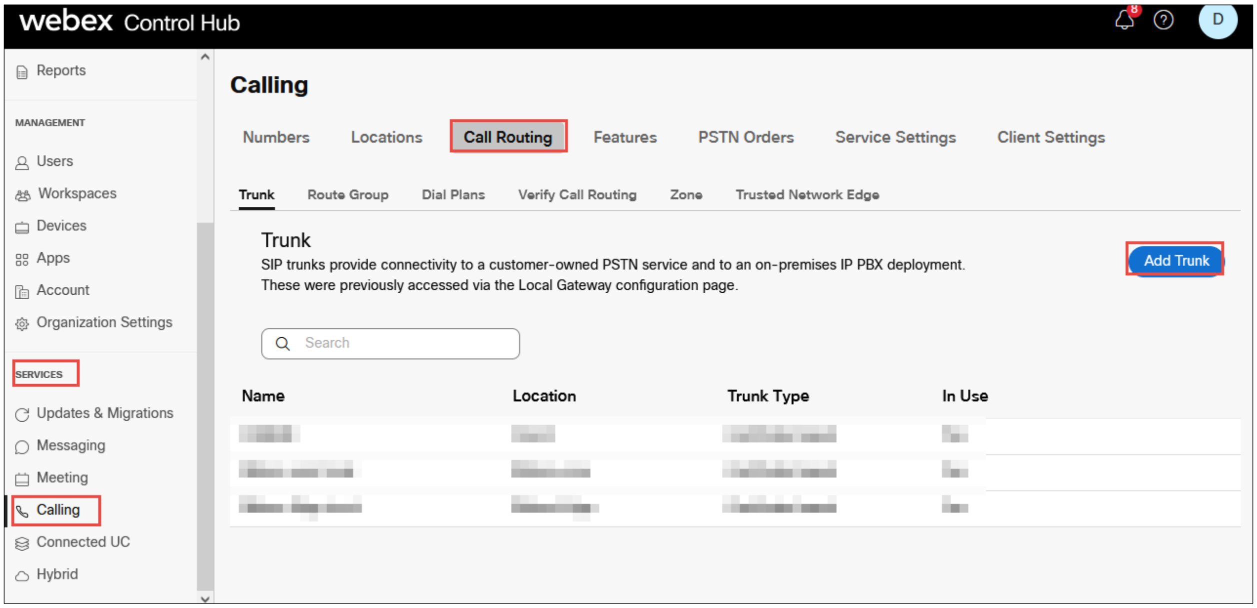
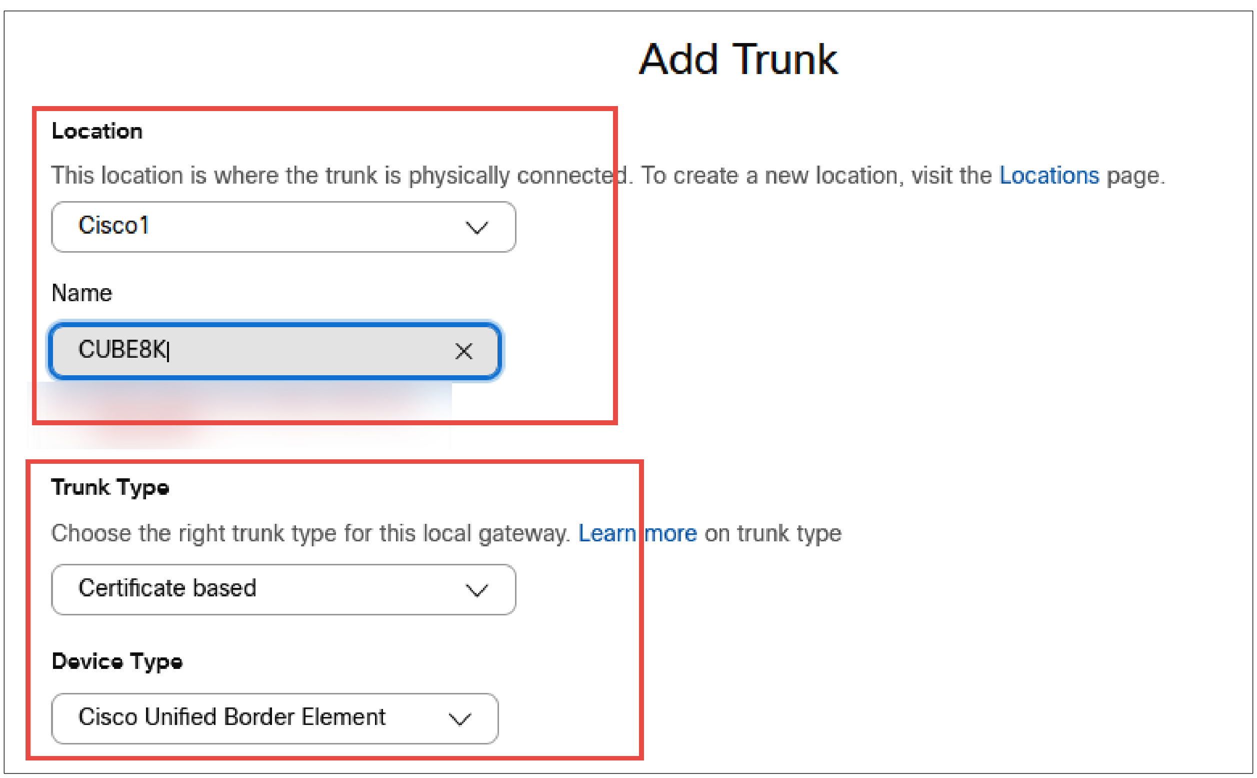


Figure 7: Add Trunk details Contd.,



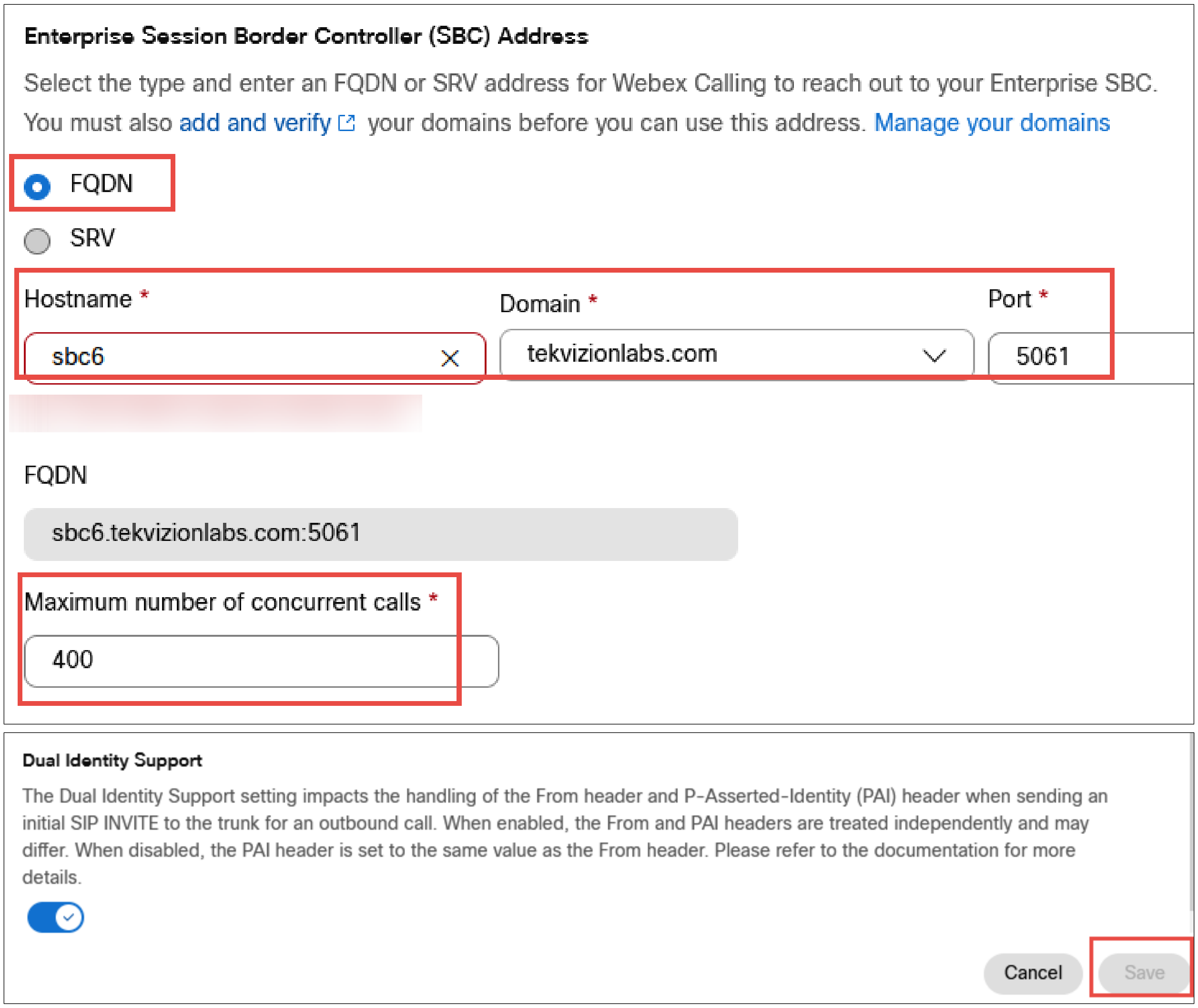


Figure 8: Add Trunk details Contd.

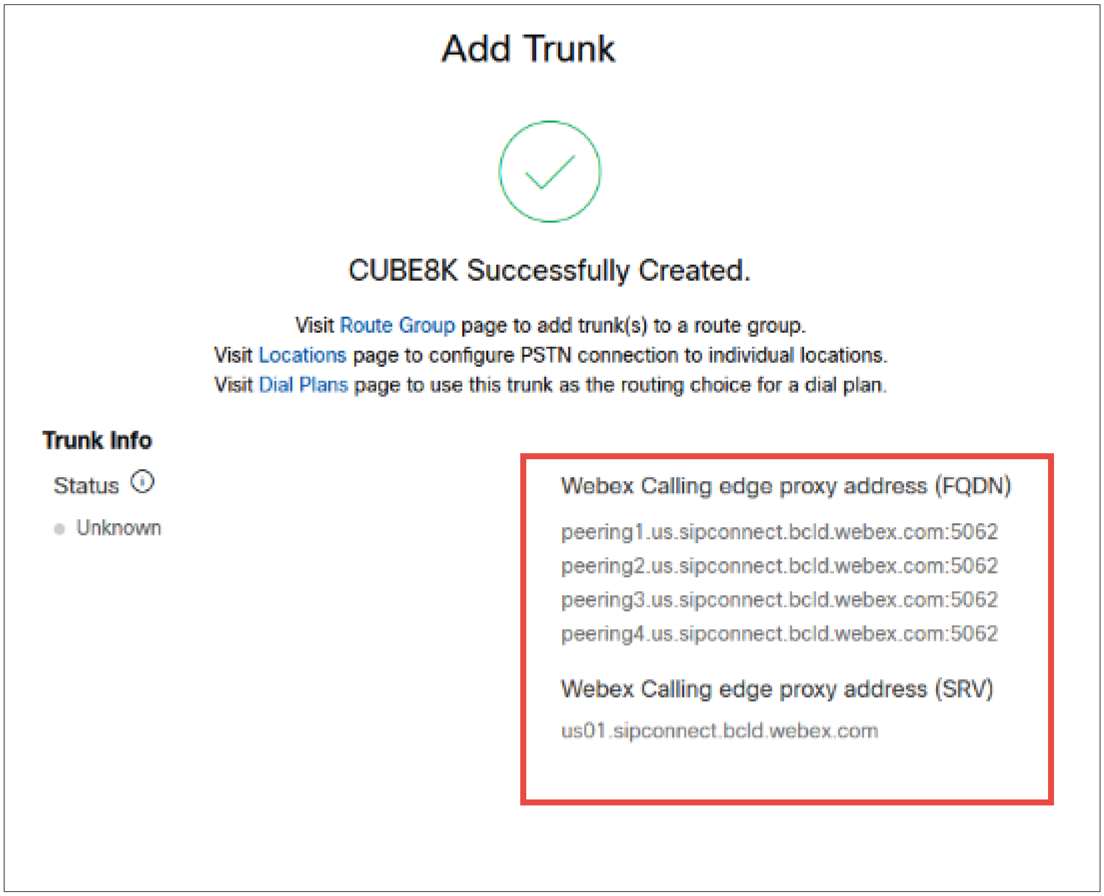


Figure 9: Add Trunk details Contd.,

Step 6:

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

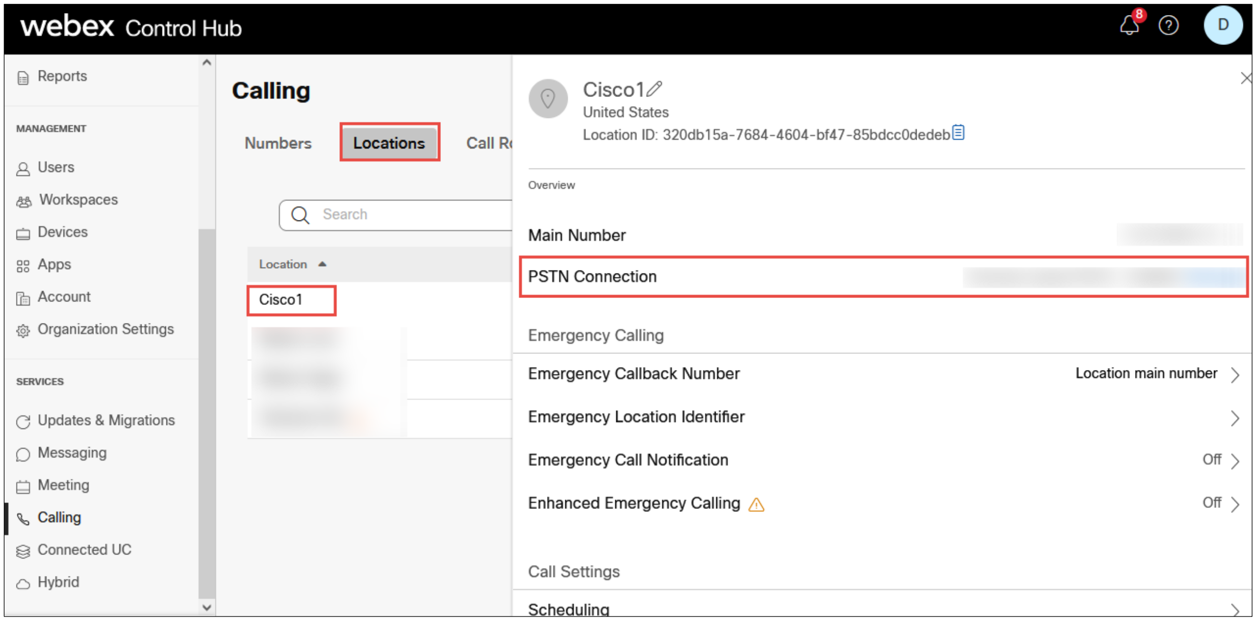


Figure 10: PSTN Connection

Step 7:

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

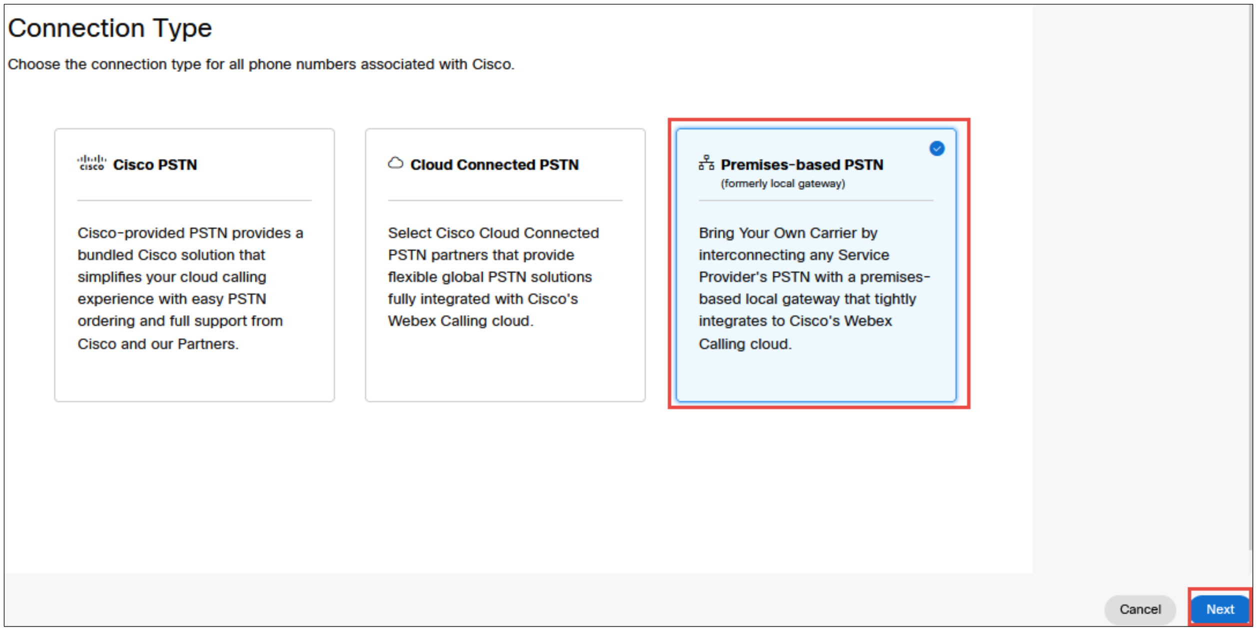


Figure 11: PSTN Connection Contd.,

Step 8:

Select the SIP trunk created earlier and click on Save

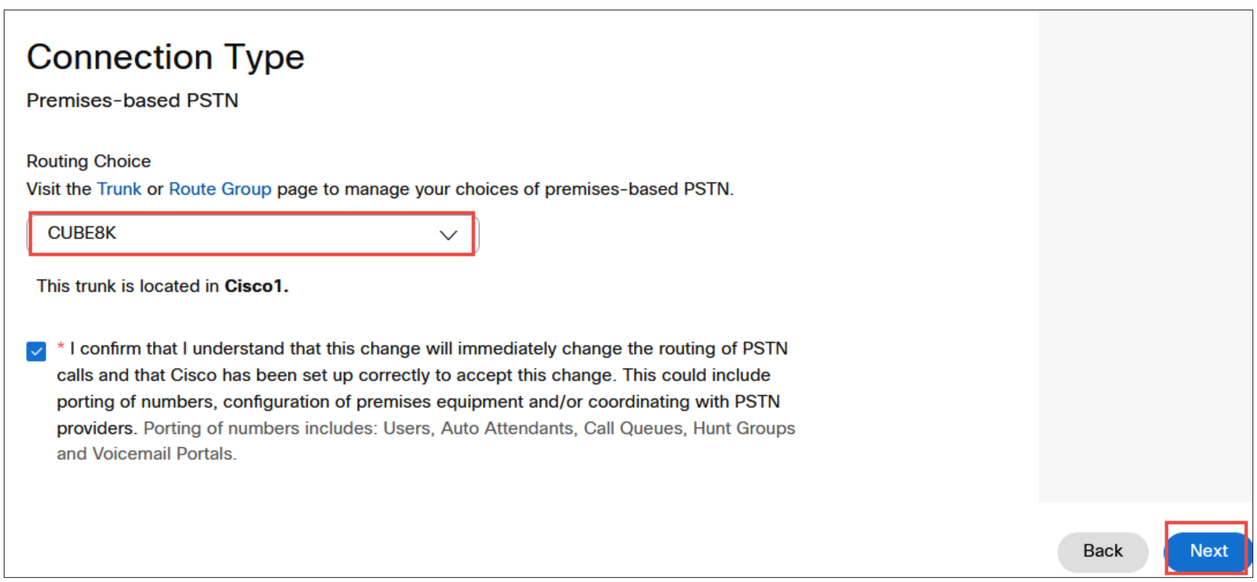


Figure 12: PSTN Connection Contd.,

Step 9:

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

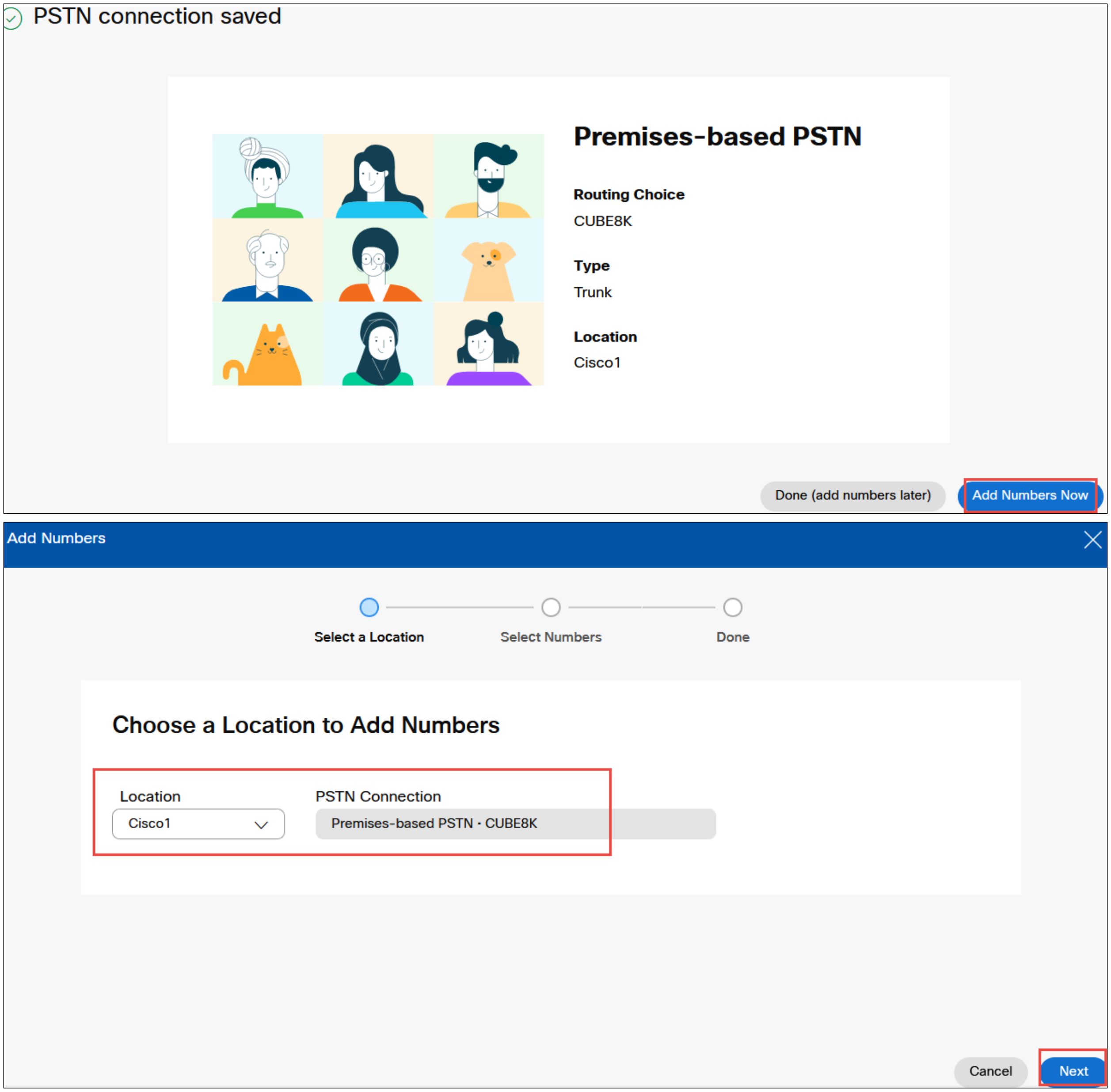


Figure 13: Add Numbers

Step 10:

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

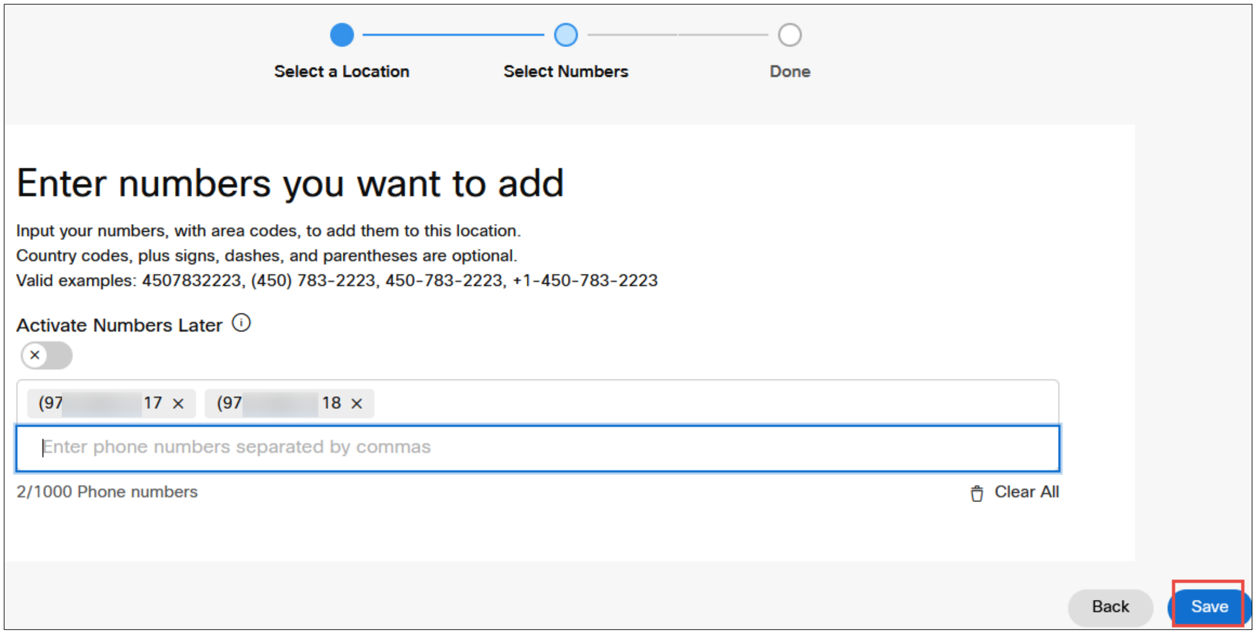


Figure 14: Add Numbers Contd.,

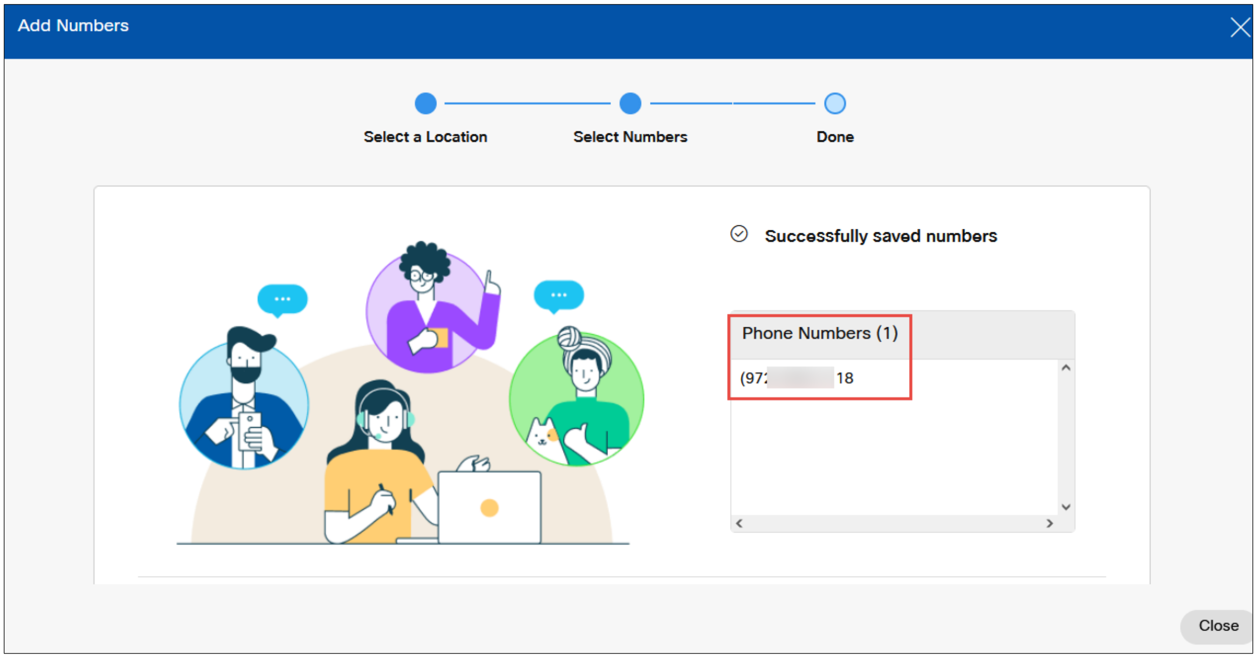


Figure 15: Add Numbers Contd.,

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

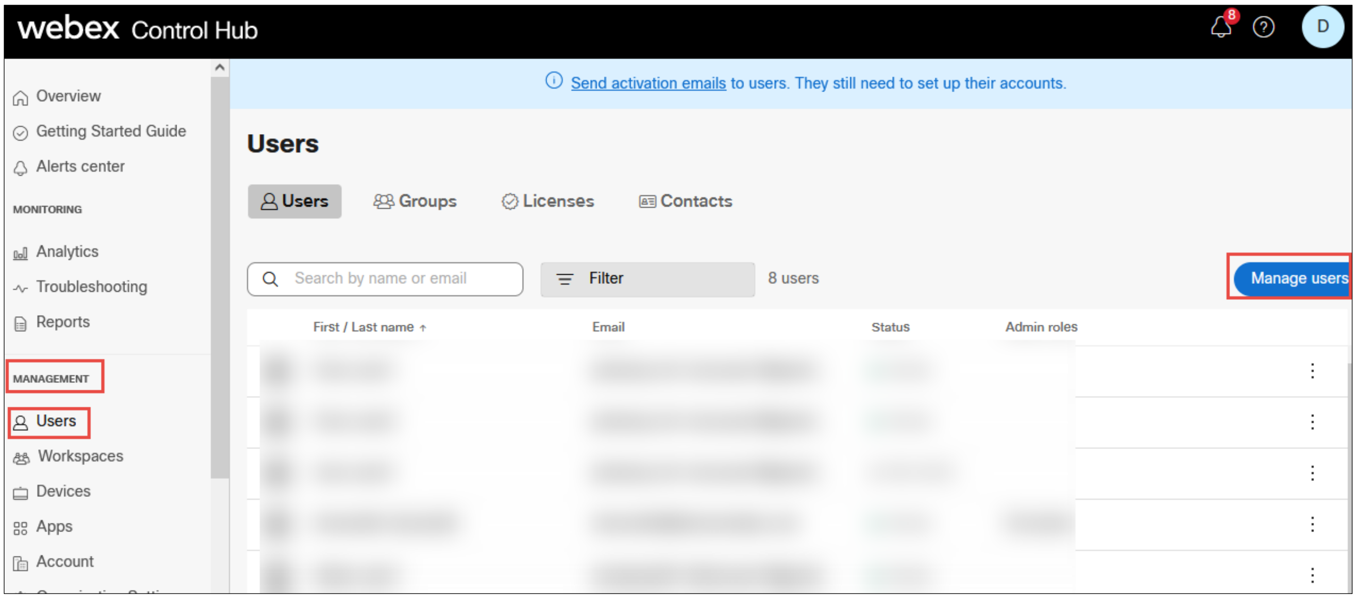


Figure 16: Adding Users

Step 2:

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

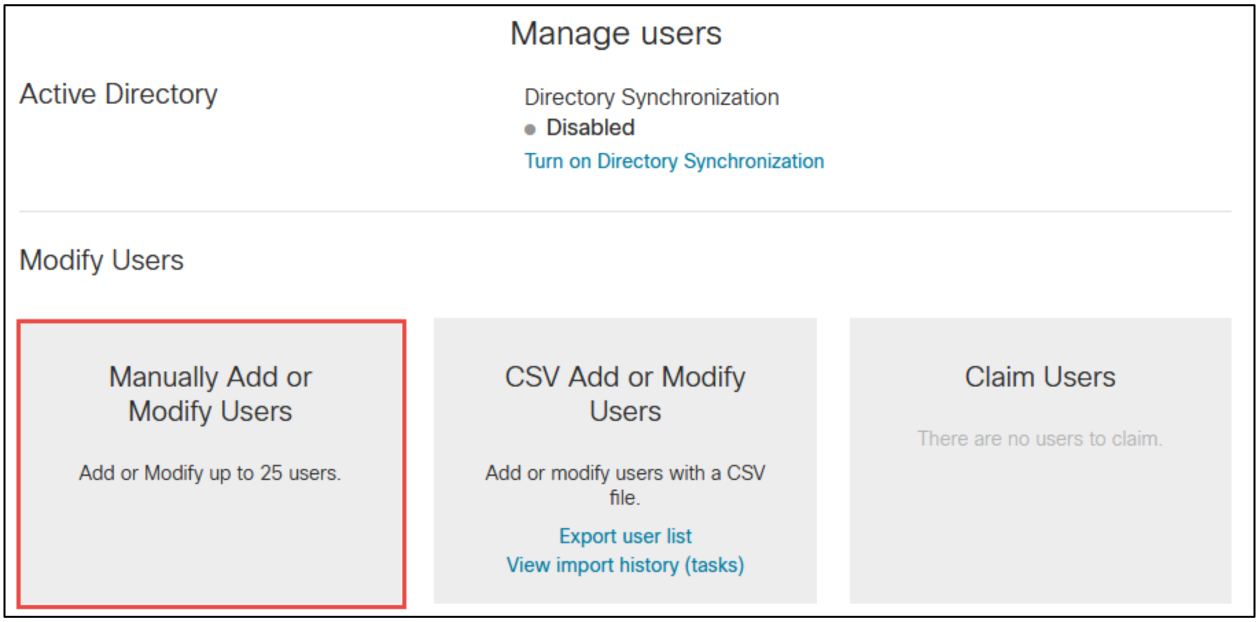


Figure 17: Manually Add or Modify Users

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**

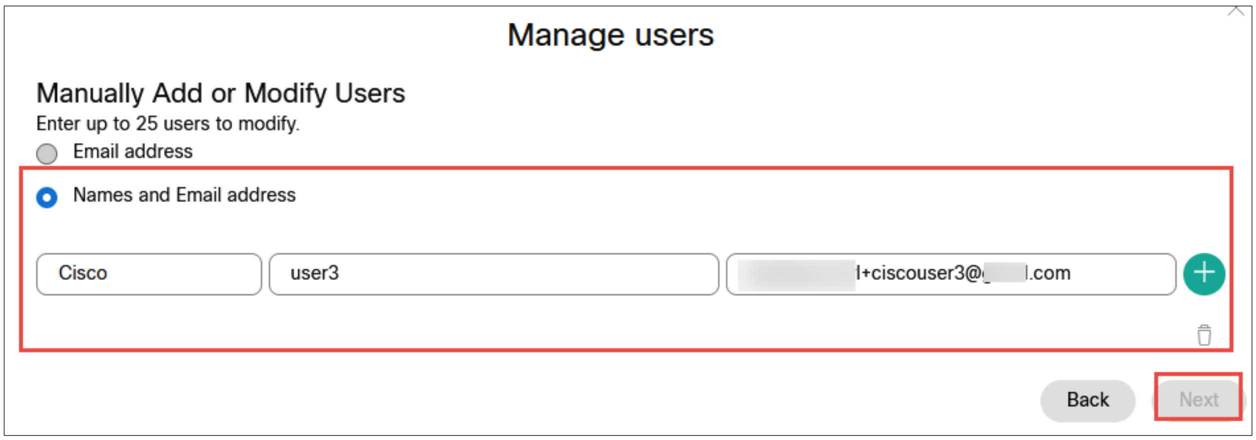


Figure 18: Adding email address and name

Step 4:

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

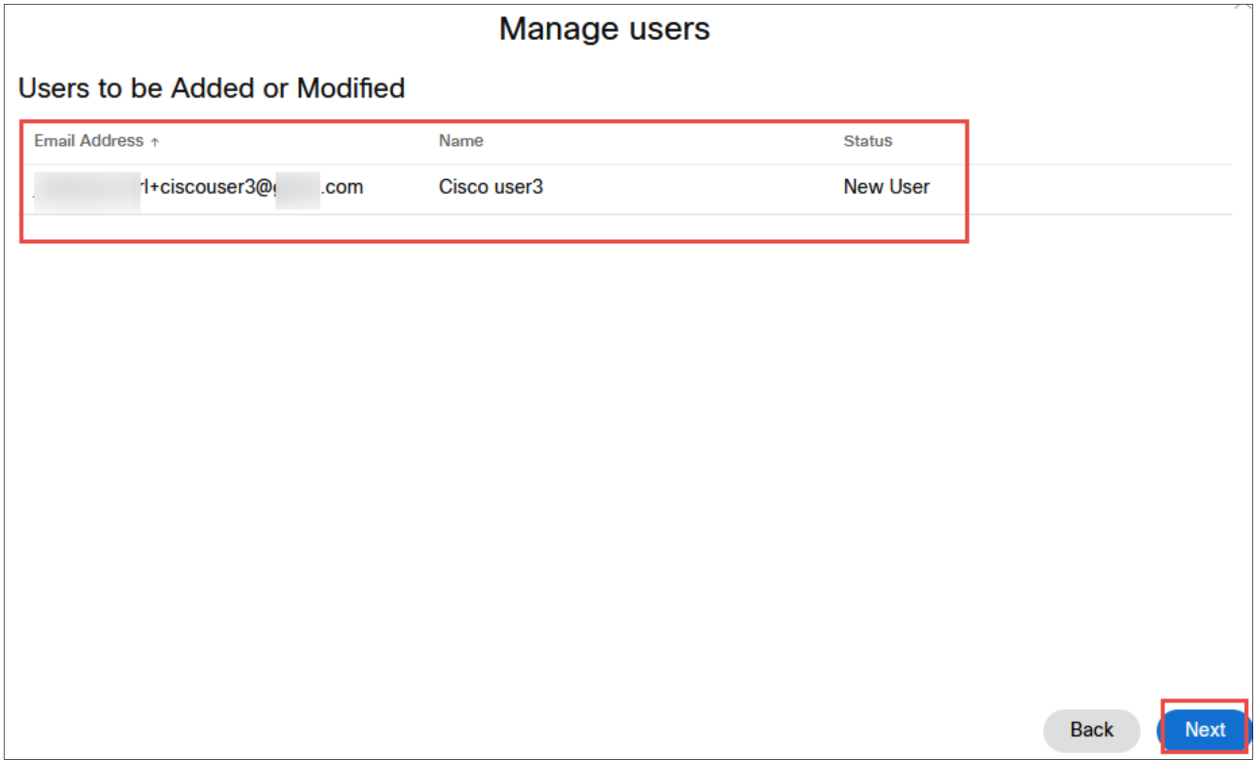


Figure 19: Confirm Adding

Step 5:

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

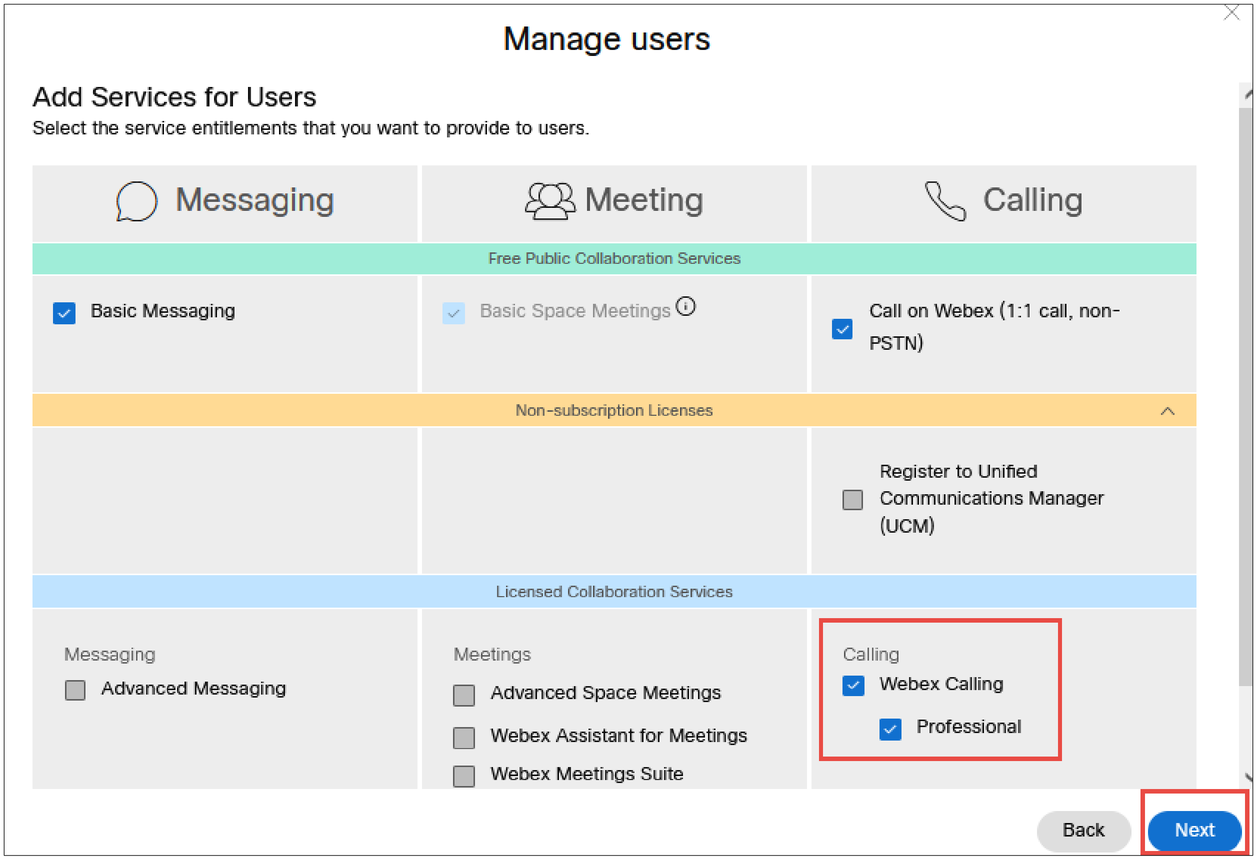


Figure 20: Add Services for Users

Step 6:

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

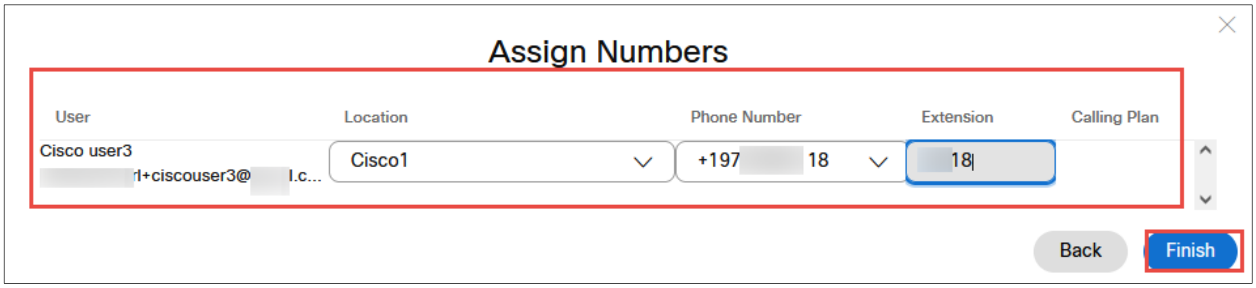


Figure 21: Assign Numbers

Step 7:

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

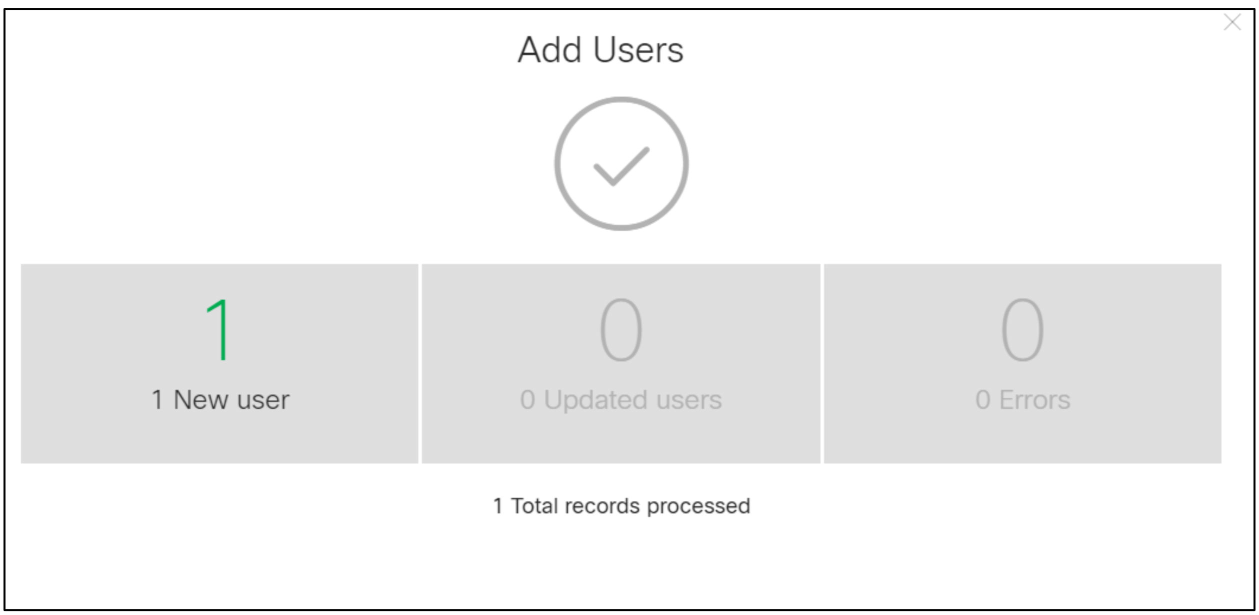


Figure 22: Add User successful

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.

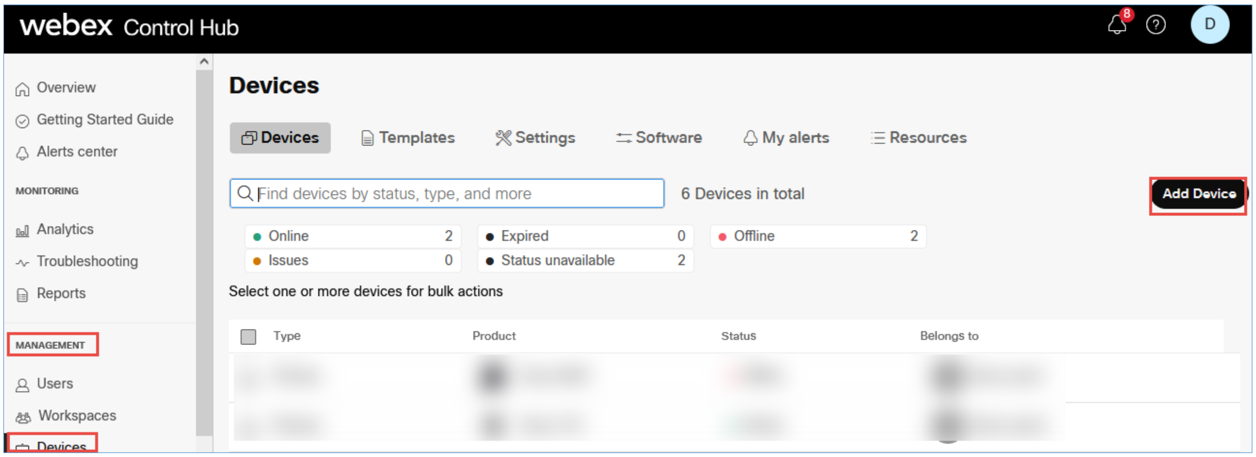


Figure 23: Add Devices Window

Step 2:

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

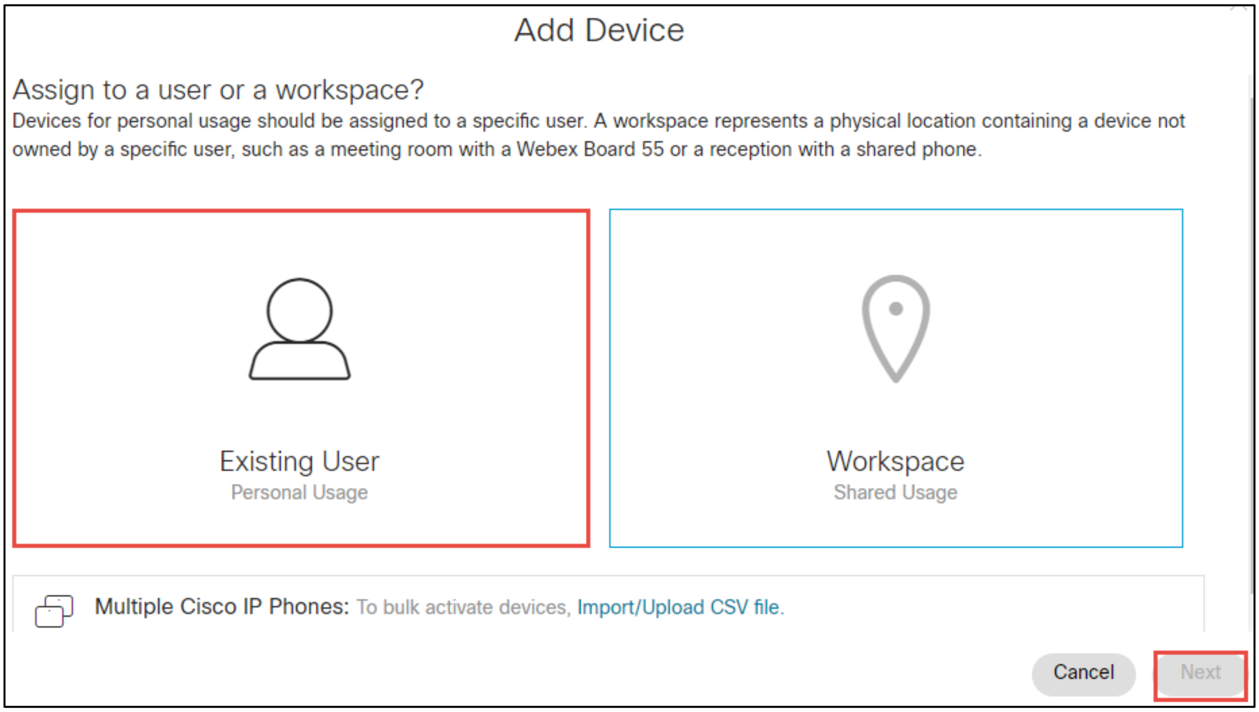


Figure 24: Assign to a user or a place

Step 3:

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

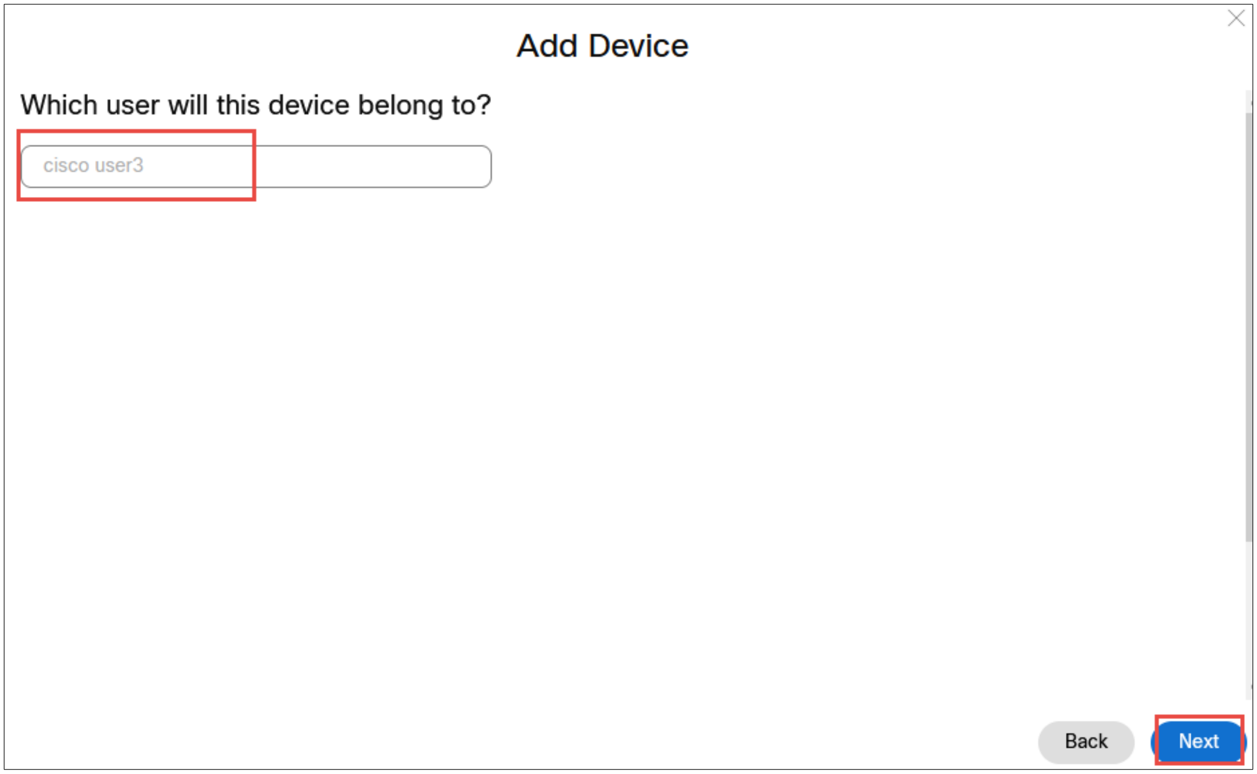


Figure 25: User Association with Device

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.

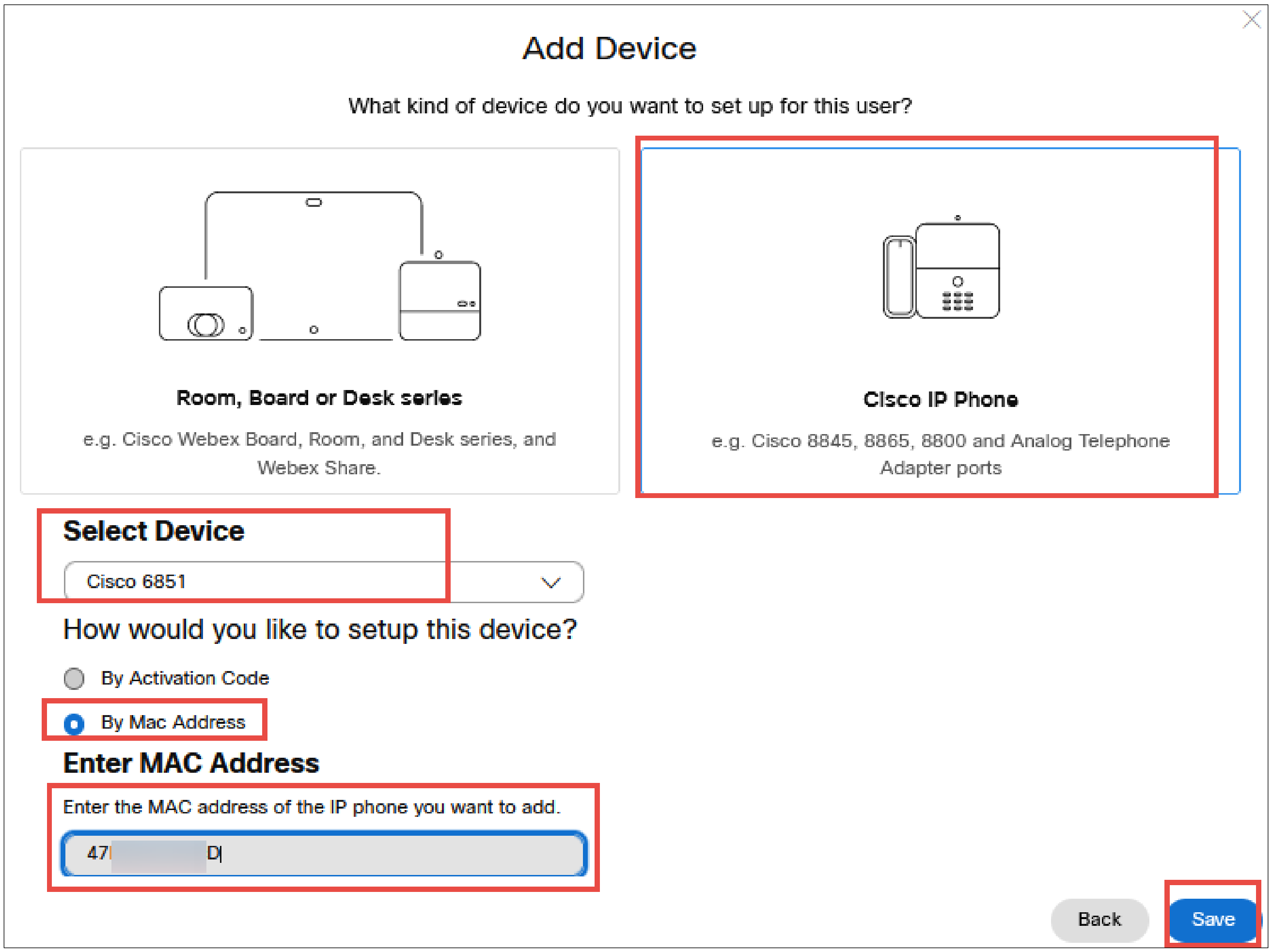


Figure 26: Select Device and add MAC address

1.4 Assign main number in location

Step 1:

Assign number in location as Main number.

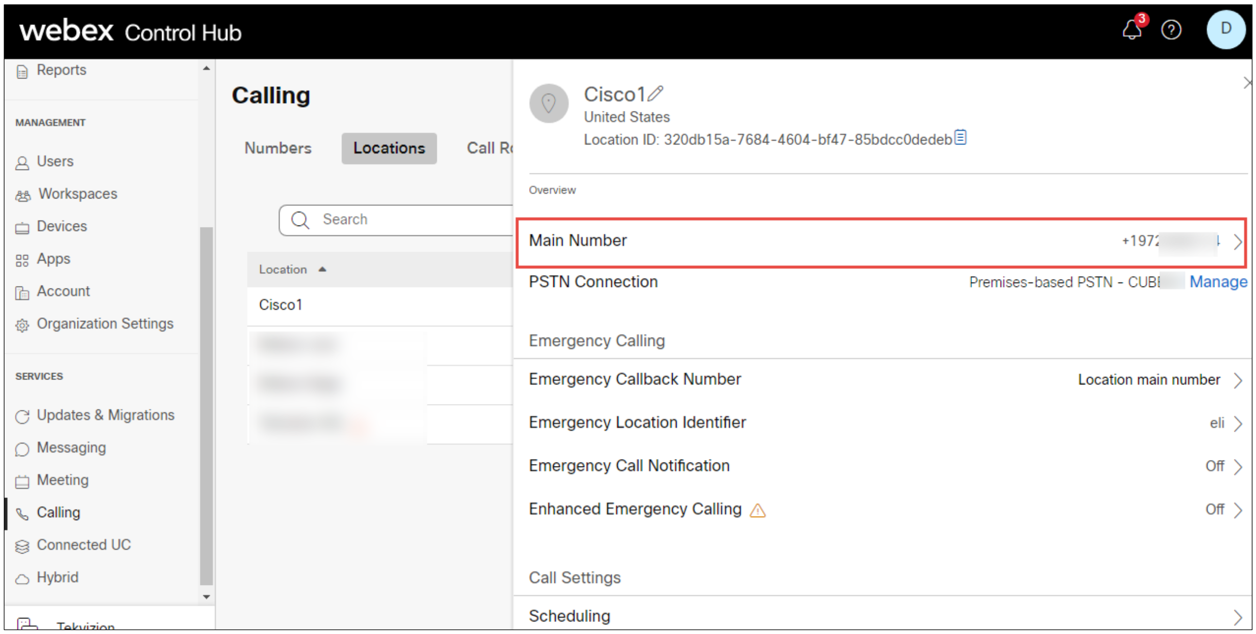


Figure 27: Assign Main number in location

Repeat the same steps in 1.1 in another Webex tenant for Multi tenancy setup.

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 Routing

2.2.1 *To Webex Calling*

```
ip route 0.0.0.0 0.0.0.0 192.65.79.x
```

2.2.2 *To PSTN/PSTN*

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

2.3 DNS and NTP Servers

DNS must be configured to resolve addresses for Webex Calling. Additionally, configure a suitable NTP source to ensure that the correct time is used by the platform

```
ip name-server 208.67.222.222 208.67.220.220
ntp server 10.10.10.5
```

2.4 Certificates

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

2.4.1 *Generate RSA key*

```
crypto key generate rsa general-keys label sbc6 exportable redundancy modulus 4096
The name for the keys will be: sbc6

% The key modulus size is 4096 bits
% Generating 4096 bit RSA keys, keys will be exportable with redundancy...
[OK] (elapsed time was 1 seconds)
```

2.4.2 Create SBC Trustpoint

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

2.4.3 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:
```

2.4.4 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
```

2.4.5 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
```

2.4.6 Specify the TLS version

```
sip-ua
 transport tcp tls v1.2
```

2.4.7 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.
```

2.4.8 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.
```

2.4.9 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.
```

2.5 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
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
listen-port secure  5066
 early-offer forced
 g729 annexb-all
 no call service stop
```

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

2.6 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

2.7 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
```

2.8 STUN ICE-lite

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

2.9 Codecs

2.9.1 To Webex calling/PSTN

```
voice class codec 100
 codec preference 2 g711ulaw
 codec preference 3 g711alaw
 codec preference 4 opus
```

2.9.2 To PBX

```
voice class codec 200
 codec preference 1 opus
```

2.10 Options keepalive to Webex Calling

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

```
voice class sip-profiles 100
 rule 10 request OPTIONS sip-header Contact modify "<sip:.*:" "<sip:sbc6.tekvizionlabs.com:"
!
voice class sip-options-keepalive 100
 description Keepalive Webex calling
 up-interval 5
 transport tcp tls
 sip-profiles 100
```

2.11 Message Handling Rules

2.11.1 SIP Profiles: Manipulations for outbound messages to Webex Calling

The following sip profile is required to:

Rule 10: Modify Contact header with IP to SBC FQDN

```
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:"
```

2.12 Specify the trust point in tls profile

```
voice class tls-profile 100
 description Webexcalling
 trustpoint sbc6
 cn-san validate bidirectional
 cn-san 1 us01.sipconnect.bcld.webex.com
```

Explanation

Command	**Description**
`cn-san validate bidirectional`	Enable CN SAN FQDN validation for bidirectional handshake in certificates
`cn-san 1 us01.sipconnect.bcld.webex.com`	Mention the CN SAN FQDN of Webex Calling to validate

2.13 Tenant

2.13.1 Tenant to Webex Calling

```
voice class tenant 200
  tls-profile 100
  listen-port secure 5061
  no remote-party-id
  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
```

2.13.2 Tenant to PSTN

```
voice class tenant 100
  session transport udp
  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
```

2.13.3 Tenant to PBX

```
voice class tenant 300
  session transport udp
  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
```

2.14 Number translation rules

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

2.14.1 To Webex Calling

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

2.14.2 To PSTN/PBX

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

2.15 Dial peers

2.15.1 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
 incoming uri request 200
 voice-class codec 100
 voice-class stun-usage 100
```

```
 voice-class sip profiles 200
 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
```

2.15.2 Outbound calls to Cisco Webex Calling

```
voice class e164-pattern-map 2002
 description towards Webex tenant1
  e164 +19725980114
!
dial-peer voice 200201 voip
 description Outbound Webex Calling tenant1
 session protocol sipv2
 session target dns:us01.sipconnect.bcld.webex.com
 session transport tcp tls
 destination e164-pattern-map 2002
 voice-class codec 100
 voice-class stun-usage 100
 voice-class sip rel1xx disable
 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
```

2.15.3 Inbound calls from PSTN

```
voice class uri 100 sip
 host 10.64.1.x
!
dial-peer voice 100 voip
 description Incoming dial-peer from PSTN
 translation-profile incoming 100
 session protocol sipv2
 destination dpg 200
 incoming uri from 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
```

2.15.4 Outbound calls to PSTN

```
voice class e164-pattern-map 100
 description towards PSTN
  e164 +1214T
  e164 +1941T
!
dial-peer voice 101 voip
 description outgoing dial-peer to IP PSTN
 translation-profile outgoing 200
 session protocol sipv2
 session target ipv4:10.64.1.x:5060
 session transport udp
 destination e164-pattern-map 100
 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
```

2.15.5 Inbound calls from PBX

```
voice class uri 300 sip
 pattern 10.71.12.11
!
dial-peer voice 200 voip
 description Incoming dial-peer from CUCM
 translation-profile incoming 100
 session protocol sipv2
 incoming uri from 300
 voice-class codec 200
 voice-class sip tenant 300
 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
```

2.15.6 Outbound calls to PBX

```
voice class e164-pattern-map 200
 description towards CUCM
  e164 +1531T
!
dial-peer voice 201 voip
 description outgoing dial-peer to IP CUCM
 translation-profile outgoing 200
 session protocol sipv2
 session target ipv4:10.71.12.11:5060
 session transport udp
 destination e164-pattern-map 200
 voice-class codec 200
 voice-class sip options-ping 60
 voice-class sip tenant 300
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
```

To configure Multi tenancy repeat the configurations towards Webex calling from section 2.10 to 2.15

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 17.9
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-universalk9.17.09.01.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
 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
  listen-port secure  5066
  early-offer forced
  g729 annexb-all
  no call service stop
!
!
voice class uri 100 sip
 host 10.64.1.x
!
voice class uri 200 sip
 host sbc6.tekvizionlabs.com
!
voice class uri 300 sip
 pattern 10.71.12.11
!
voice class codec 100
 codec preference 2 g711ulaw
 codec preference 3 g711alaw
codec preference 4 opus
!
voice class codec 200
 codec preference 1 opus
!
voice class stun-usage 100
 stun usage ice lite
!
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:"
!
voice class e164-pattern-map 100
 description towardsPSTN
  e164 +1214T
  e164 +1941T
!
voice class e164-pattern-map 200
 description towardsCUCM
  e164 +1531T
 !
voice class e164-pattern-map 2002
 description towards Webex tenant1
  e164 +19725980114
```

```
!
voice class sip-options-keepalive 100
 description Keepalive Webex Calling
 up-interval 5
 transport tcp tls
 sip-profiles 100
!
voice class tenant 200
 tls-profile 100
  listen-port secure 5061
  no remote-party-id
  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
  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 tenant 300
  session transport udp
  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
!
voice class srtp-crypto 200
 crypto 1 AES_CM_128_HMAC_SHA1_80
!
voice class tls-profile 100
 description Webexcalling
 trustpoint sbc6
 cn-san validate bidirectional
 cn-san  1 us01.sipconnect.bcld.webex.com
!
voice translation-rule 100
 rule 1 /^\([2-9].........\)/ /+1\1/
!
voice translation-rule 200
 rule 1 /^\+1\(.*\)/ /\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 10.70.0.0 255.255.0.0 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 200101 voip
 description Inbound from Webex Calling
 session protocol sipv2
 session transport tcp tls
 incoming uri request 200
 voice-class codec 100
 voice-class stun-usage 100
 voice-class sip profiles 200
 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
!
dial-peer voice 200201 voip
 description Outbound Webex Calling tenant1
 session protocol sipv2
 session target dns:us01.sipconnect.bcld.webex.com
 session transport tcp tls
 destination e164-pattern-map 2002
 voice-class codec 100
 voice-class stun-usage 100
 voice-class sip rel1xx disable
 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 100 voip
 description Incoming dial-peer from PSTN
 translation-profile incoming 100
 session protocol sipv2
 destination dpg 200
 incoming uri from 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 101 voip
 description outgoing dial-peer to IP PSTN
 translation-profile outgoing 200
 session protocol sipv2
 session target ipv4:10.64.1.x:5060
 session transport udp
 destination e164-pattern-map 100
 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 200 voip
 description Incoming dial-peer from CUCM
 translation-profile incoming 100
```

```
 session protocol sipv2
 incoming uri from 300
 voice-class codec 200
 voice-class sip tenant 300
 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 201 voip
 description outgoing dial-peer to IP CUCM
 translation-profile outgoing 200
 session protocol sipv2
 session target ipv4:10.71.12.11:5060
 session transport udp
 destination e164-pattern-map 200
 voice-class codec 200
 voice-class sip options-ping 60
 voice-class sip tenant 300
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
gateway
 timer receive-rtp 1200
!
sip-ua
 no remote-party-id
transport tcp tls v1.2
!
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 17.9
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-universalk9.17.09.01.SPA.bin
boot-end-marker
!
logging buffered 214748364
no aaa new-model
clock timezone UTC -5 0
clock calendar-valid
!
ip name-server 8.8.8.8
ip domain name tekvizionlabs.com
!
login on-success log
!
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-2307055185
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-2307055185
 revocation-check none
 rsakeypair TP-self-signed-2307055185
!
crypto pki trustpoint SLA-TrustPoint
 enrollment pkcs12
 revocation-check crl
!
crypto pki trustpoint sbc6
 enrollment pkcs12
 revocation-check crl
 rsakeypair sbc6
!
!
```

```
crypto pki 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
 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
 listen-port secure  5066
  early-offer forced
  g729 annexb-all
no call service stop
!
voice class uri 100 sip
 host 10.64.1.x
!
voice class uri 200 sip
 host sbc6.tekvizionlabs.com
!
voice class uri 300 sip
 pattern 10.71.12.11
!
voice class codec 100
 codec preference 2 g711ulaw
 codec preference 3 g711alaw
codec preference 4 opus
!
voice class codec 200
 codec preference 1 opus
!
voice class stun-usage 100
 stun usage ice lite
!
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:"
!
!
voice class sip-options-keepalive 100
 description Keepalive Webex Calling
 up-interval 5
 transport tcp tls
 sip-profiles 100
!
voice class tenant 200
 tls-profile 100
  listen-port secure 5061
  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
  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 tenant 300
  session transport udp
  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
voice class srtp-crypto 200
 crypto 1 AES_CM_128_HMAC_SHA1_80
!
voice class tls-profile 100
 description Webexcalling
 trustpoint sbc6
 cn-san validate bidirectional
 cn-san  1 us01.sipconnect.bcld.webex.com
!
voice translation-rule 100
 rule 1 /^\([2-9].........\)/ /+1\1/
!
voice translation-rule 200
 rule 1 /^\+1\(.*\)/ /\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 10.70.0.0 255.255.0.0 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 200101 voip
 description Inbound from Webex Calling
 session protocol sipv2
 session transport tcp tls
 incoming uri request 200
 voice-class codec 100
 voice-class stun-usage 100
 voice-class sip profiles 200
 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
!
dial-peer voice 200201 voip
 description Outbound Webex Calling tenant1
 session protocol sipv2
 session target dns:us01.sipconnect.bcld.webex.com
 session transport tcp tls
 destination e164-pattern-map 2002
 voice-class codec 100
 voice-class stun-usage 100
 voice-class sip rel1xx disable
 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 100 voip
 description Incoming dial-peer from PSTN
 translation-profile incoming 100
 session protocol sipv2
 destination dpg 200
 incoming uri from 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 101 voip
 description outgoing dial-peer to IP PSTN
 translation-profile outgoing 200
 session protocol sipv2
 session target ipv4:10.64.1.x:5060
 session transport udp
 destination e164-pattern-map 100
 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 200 voip
 description Incoming dial-peer from CUCM
 translation-profile incoming 100
 session protocol sipv2
 incoming uri from 300
 voice-class codec 200
 voice-class sip tenant 300
 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 201 voip
 description outgoing dial-peer to IP CUCM
 translation-profile outgoing 200
 session protocol sipv2
```

```
 session target ipv4:10.71.12.11:5060
 session transport udp
 destination e164-pattern-map 200
 voice-class codec 200
 voice-class sip options-ping 60
 voice-class sip tenant 300
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
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
```

3.3 Show commands

3.3.1 Dial-peer status

```
8K_MTLS_webex#show dial-peer voip keepalive status
 TAG       TENANT    DESTINATION                      OOD-SessID   PRI      WT       STATUS
 200201    200     dns:us01.sipconnect.bcld.webex                                    active
                     sipconnect01ah-us.bcld.webex. 3393         5        25       active
                       ipv4:139.177.64.53:5062
                     sipconnect01ai-us.bcld.webex. 3394         5        25       active
                       ipv4:139.177.64.54:5062
                     sipconnect02ai-us.bcld.webex. 3395         10       25       active
                       ipv4:139.177.65.54:5062
                     sipconnect02ah-us.bcld.webex. 3396         10       25       active
                       ipv4:139.177.65.53:5062
```

Note: Command introduced from 17.9.1a IOS

3.3.2 Dial-peer Summary

```
8K_MTLS_webex#show dial-peer voice summary
dial-peer hunt 0
             AD                                    PRE PASS SESS-SER-GRP\ OUT
TAG   TYPE  MIN OPER PREFIX DEST-PATTERN FER THRU SESS-TARGET   STAT PORT  KEEPALIVE   VRF
100    voip  up up                     0  syst                                         NA
101     voip up up      map:100       0  syst ipv4:10.64.1.x:5060           active      NA
200101  voip up up                     0  syst
200201  voip up up      map:2002      0  syst dns:us01.sipconnect.          active      NA
201     voip up up      map:200       0  syst ipv4:10.71.12.11:506          active      NA
200     voip up up                     0  syst                                         NA
 For server-grp details please execute command:show voice class server-group <tag_id>
To see complete session target for ipv6 use 'sh running-config | section dial-peer <tag>
```

3.3.3 Voice class Keepalive sip Options

```
8K_MTLS_webex#show voice class sip-options-keepalive 100
Voice class sip-options-keepalive: 100           AdminStat: Up
 Description: Keepalive Webex calling
 Transport: tcp tls                Sip Profiles: 100
 Interval(seconds) Up: 5                 Down: 5
```

```
 Retry: 5

  Peer Tag      Server Group    OOD SessID      OOD Stat        IfIndex
  --------      ------------    ----------      --------        -------
  200201                                        Active          16

 OOD SessID: 7479                OOD Stat: Active
  Target: ipv4:139.177.65.54:5062
  Transport: tcp tls             Sip Profiles: 100

 OOD SessID: 7480                OOD Stat: Active
  Target: ipv4:139.177.64.54:5062
  Transport: tcp tls             Sip Profiles: 100

 OOD SessID: 7481                OOD Stat: Active
  Target: ipv4:139.177.65.53:5062
  Transport: tcp tls             Sip Profiles: 100

 OOD SessID: 7482                OOD Stat: Active
  Target: ipv4:139.177.64.53:5062
  Transport: tcp tls             Sip Profiles: 100

------------------------------------------------------
For session target configured as DNS - please execute: show dial-peer voip keepalive status
```

3.3.4 SIP-ua connection details

```
8K_MTLS_webex#Show sip-ua connections tcp tls detail
Total active connections      : 8
No. of send failures          : 36
No. of remote closures        : 2502
No. of conn. failures         : 1880
No. of inactive conn. ageouts : 0
Max. tls send msg queue size of 1, recorded for 139.177.65.54:8934
TLS client handshake failures : 0
TLS server handshake failures : 8

---------Printing Detailed Connection Report---------
Note:
 ** Tuples with no matching socket entry
    - Do 'clear sip <tcp[tls]/udp> conn t ipv4:<addr>:<port>'
      to overcome this error condition
 ++ Tuples with mismatched address/port entry
    - Do 'clear sip <tcp[tls]/udp> conn t ipv4:<addr>:<port> id <connid>'
      to overcome this error condition
 * Connections with SIP OAuth ports

Remote-Agent:139.177.64.53, Connections-Count:2
  Remote-Port Conn-Id Conn-State  WriteQ-Size            Local-Address               TLS-
Version Cipher                         Curve Tenant
  =========== ======= =========== =========== ======================================
=========== ============================== ===== ======
         5062    4450 Established           0 192.65.79.x:62610                        TLSv1.2
ECDHE-RSA-AES256-GCM-SHA384 P-256   200
         8934    4442 Established           0 192.65.79.x:5061                         TLSv1.2
ECDHE-RSA-AES256-GCM-SHA384 P-256   200

Remote-Agent:139.177.65.54, Connections-Count:2
  Remote-Port Conn-Id Conn-State  WriteQ-Size            Local-Address               TLS-
Version Cipher                         Curve Tenant
  =========== ======= =========== =========== ======================================
=========== ============================== ===== ======
         5062    4452 Established           0 192.65.79.x:59714                        TLSv1.2
ECDHE-RSA-AES256-GCM-SHA384 P-256   200
         8934    4449 Established           0 192.65.79.x:5061                         TLSv1.2
ECDHE-RSA-AES256-GCM-SHA384 P-256   200

Remote-Agent:139.177.64.54, Connections-Count:2
  Remote-Port Conn-Id Conn-State  WriteQ-Size            Local-Address               TLS-
Version Cipher                         Curve Tenant
```

```
  =========== ======= =========== =========== ======================================
=========== ============================== ===== ======
         5062    4447 Established           0 192.65.79.x:53007                       TLSv1.2
ECDHE-RSA-AES256-GCM-SHA384 P-256   200
         8934    4435 Established           0 192.65.79.x:5061                        TLSv1.2
ECDHE-RSA-AES256-GCM-SHA384 P-256   200


Remote-Agent:139.177.65.53, Connections-Count:2
  Remote-Port Conn-Id Conn-State  WriteQ-Size            Local-Address               TLS-
Version Cipher                         Curve Tenant
  =========== ======= =========== =========== ======================================
=========== ============================== ===== ======
         5062    4448 Established           0 192.65.79.x:24628                       TLSv1.2
ECDHE-RSA-AES256-GCM-SHA384 P-256   200
         8934    4451 Established           0 192.65.79.x:5061                        TLSv1.2
ECDHE-RSA-AES256-GCM-SHA384 P-256   200



-------------- SIP Transport Layer Listen Sockets ---------------
  Conn-Id             Local-Address                     Tenant
 ==========    ===========================             ========
  0            [0.0.0.0]:5066:                            0
  6            [10.80.11.x]:5066:                         0
  2092         [192.65.79.x]:5066:                        0
  2093         [192.65.79.x]:5061:                       200
```

3.3.5 Show voip trace tenant

INVITE from SBC to Webex

```
8K_MTLS_webex#Show voip trace tenant 200
------------------ Cover Buffer ---------------
Search-key     = +121424259xx:+197259801xx:52161
  Timestamp      = Aug 29 09:48:33.716
  Buffer-Id      = 246
  CallID         = 52161
  Peer-CallID    = 52160
  Correlator     = 73
  Called-Number  = +197259801xx
  Calling-Number = +121424259xx
  SIP CallID     = 949770D5-26B611ED-86669B21-3DD4CEDB@sbc6.tekvizionlabs.com
  SIP Session ID = a17019339fb95e9d86a91cd9bdbe4866
  GUID           = 93F2D0568A1E
  Tenant         = 200
-----------------------------------------------


Sent: SIP TLS message from 192.65.79.x:5061 to 139.177.65.54:5062
INVITE sip:+197259801xx@peering1.us.sipconnect.bcld.webex.com:5062 SIP/2.0
Via: SIP/2.0/TLS 192.65.79.x:5061;branch=z9hG4bK792710C5
From: "214 24259xx" <sip:+121424259xx@sbc6.tekvizionlabs.com>;tag=23D1D36F-7D9
To: <sip:+19725980xxx@peering1.us.sipconnect.bcld.webex.com>
Date: Mon, 29 Aug 2022 09:48:33 GMT
Call-ID: 949770D5-26B611ED-86669B21-3DD4CEDB@sbc6.tekvizionlabs.com
Supported: timer,resource-priority,replaces
Min-SE:  1800
Cisco-Guid: 2482163798-0649466349-2317247510-2368317232
User-Agent: Cisco-SIPGateway/IOS-17.9.1
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO,
REGISTER
CSeq: 101 INVITE
Timestamp: 1661766513
Contact: <sip:+121424259xx@sbc6.tekvizionlabs.com:5061;transport=tls>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 69
P-Asserted-Identity: "214 24259xx" <sip:+121424259xx@sbc6.tekvizionlabs.com>
Session-ID: a17019339fb95e9d86a91cd9bdbe4866;remote=00000000000000000000000000000000
Content-Type: application/sdp
Content-Disposition: session;handling=required
```

```
Content-Length: 580
v=0
o=CiscoSystemsSIP-GW-UserAgent 9363 5690 IN IP4 192.65.79.x
s=SIP Call
c=IN IP4 192.65.79.x
t=0 0
a=ice-lite
m=audio 8500 RTP/SAVP 0 8 101
c=IN IP4 192.65.79.x
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
a=candidate:1 1 UDP 2130706431 192.65.79.1 8500 typ host
a=candidate:1 2 UDP 2130706430 192.65.79.1 8501 typ host
a=rtcp:8501 IN IP4 192.65.79.138
a=ice-ufrag:tee7
a=ice-pwd:jpZBqqlMeCt9kShKy37xkC
```

INVITE from Webex to SBC

```
------------------ Cover Buffer ---------------
Search-key     = +19725980100:+12142425900:54711
  Timestamp      = Aug 29 16:39:36.302
  Buffer-Id      = 249
  CallID         = 54711
  Peer-CallID    = 54712
  Correlator     = 75
  Called-Number  = +12142425900
  Calling-Number = +19725980100
  SIP CallID     = SSE163937446290822-1692560954@139.177.64.54
  SIP Session ID = d7a77ccaeaf653fc92210392efb0f5d2
  GUID           = 00A2A7BE8F30
  Tenant         = 200
-----------------------------------------------
30813: Aug 29 16:39:36.302: //54711/00A2A7BE8F30/CUBE_VT/SIP/Msg/ccsipDisplayMsg:
Received: SIP TLS message from 139.177.64.54:8934 to 192.65.79.1:5061
INVITE sip:+12142425900@sbc6.tekvizionlabs.com:5061;transport=tls;dtg=sbc6.tekvizionlabs.com
SIP/2.0
Via:SIP/2.0/TLS 139.177.64.54:5062;branch=z9hG4bKBroadworksSSE.-192.65.79.1V5061-0-100-
1030372628-1661791177446-
```

```
From:"Cisco user1"<sip:+19725980100@139.177.64.54;user=phone>;tag=1030372628-1661791177446-
To:<sip:+12142425900@91366808.cisco-bcld.com;user=phone>
Call-ID:SSE163937446290822-1692560954@139.177.64.54
CSeq:100 INVITE
Contact:<sip:139.177.64.54:5062;transport=tls>
P-Asserted-Identity:"Cisco user1"<sip:+19725980100@10.21.0.213;user=phone>
Privacy:none
Allow:ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Recv-Info:x-broadworks-client-session-info
X-BroadWorks-Correlation-Info:db169f8d-83a9-4d5c-ab52-3048e65185ba
Accept:application/media_control+xml,application/sdp,multipart/mixed
Supported:
Max-Forwards:69
Session-ID:a2ab704900105000a0004c710c4dce2f;remote=00000000000000000000000000000000
Content-Type:application/sdp
Content-Length:1103

v=0
o=BroadWorks 80901159 1661791177441 IN IP4 135.84.171.105
s=-
c=IN IP4 135.84.171.105
t=0 0
m=audio 32084 RTP/SAVP 99 9 0 8 18 101 108
a=rtpmap:99 opus/48000/2
a=fmtp:99 maxplaybackrate=16000;sprop-
maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:108 telephone-event/48000
a=fmtp:108 0-15
a=ptime:20
a=sendrecv
a=ice-ufrag:TTbS
a=ice-pwd:aKzPENdHwz5C6uFIPqQTOs
a=candidate:1 1 udp 2130706431 192.168.1.113 19608 typ host
a=candidate:1 2 udp 2130706430 192.168.1.113 19609 typ host
a=candidate:3 1 udp 1694494975 122.164.157.102 19608 typ srflx raddr 192.168.1.113 rport 19608
a=candidate:3 2 udp 1694494974 122.164.157.102 19609 typ srflx raddr 192.168.1.113 rport 19609
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
```

```
a=candidate:mse 1 UDP 16777215 135.84.171.105 32084 typ relay
a=candidate:mse 2 UDP 16777214 135.84.171.105 32085 typ relay
```

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