July 2021

Walkthrough Wednesday

Flexible PSTN options for your Webex Calling deployment

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July 2021

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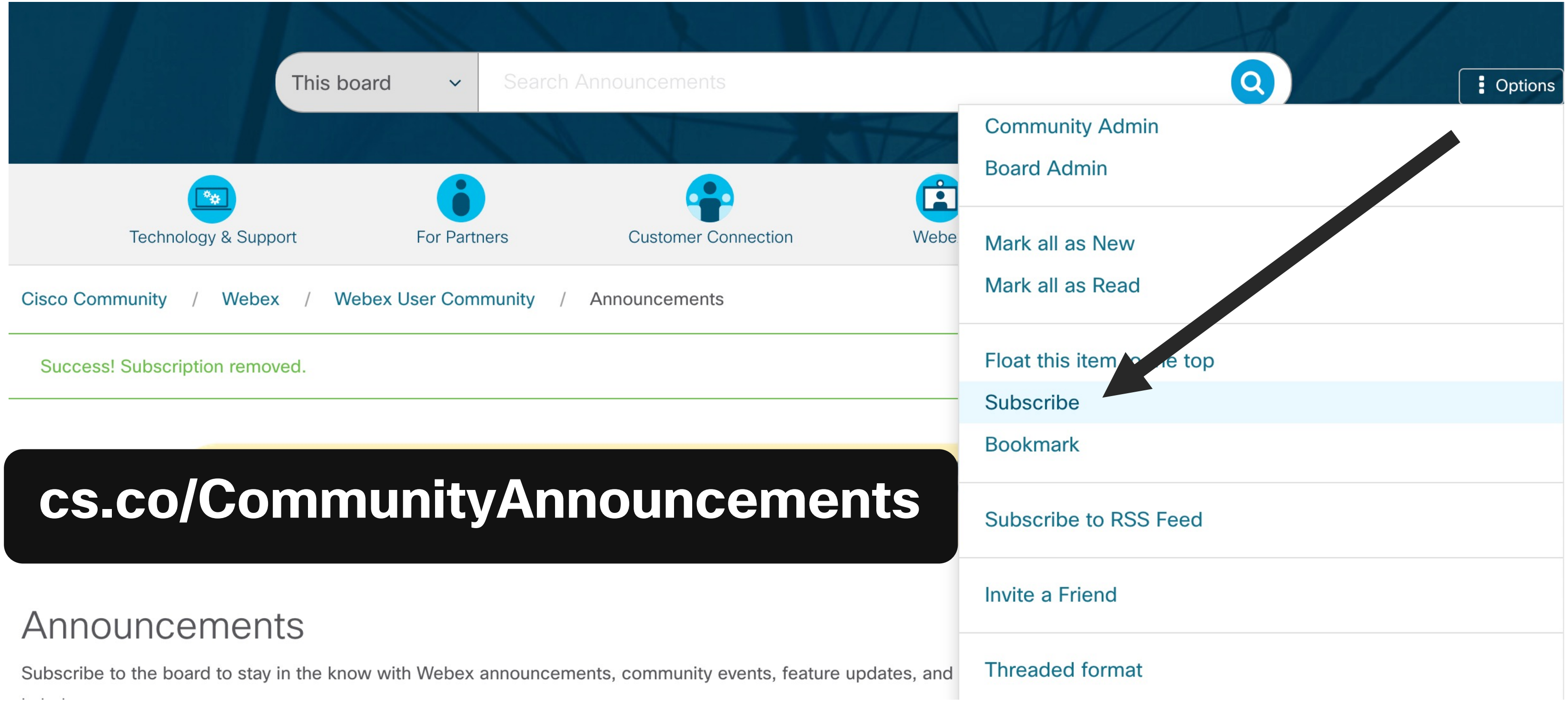
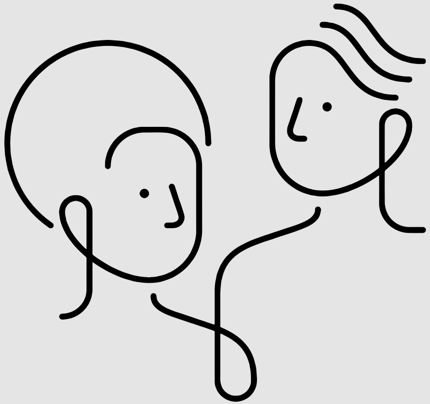


You're invited to continue the conversation in the Webex Community!

- Connect with 100,000+ other Webex users like you
- Ask questions and share knowledge
- Attend live events with esteemed guests
- Submit product ideas that go straight our Product team
- Participate in contests



Webex User Community

This is a place to connect with Webex and other users like you. Ask questions, share feedback, join an event, or help others!

	Posts	Replies	Latest Post
Announcements LATEST POST - Re: Announcing the Webex Ambassador Program!	54	252	07-20-2021 10:44 AM
Webex Meetings and Webex App LATEST POST - Audio Notifications	1824	3618	07-21-2021 05:35 AM
Webex Events App LATEST POST - Re: host role - unmuting an attendees during Q&A session (Events classic)	182	306	07-20-2021 11:04 AM
Webex Training App LATEST POST - Re: Assign Mouse and Keyboard Control	162	294	07-16-2021 02:40 AM
Webex Devices LATEST POST - Reorganize Video Input	141	189	07-20-2021 03:24 PM
Webex Calling LATEST POST - Re: Do Not Disturb/Out of Office Sync	84	134	07-21-2021 06:54 AM

Agenda

- Webex Calling PSTN Options
- Setting up Cisco Calling Plan
- Local Gateway Updates
    - Local gateway sizing
    - Recent Key configuration updates
    - Caller ID handling
    - Working with and creating templates
- Resources

Webex Calling PSTN Options

Integrated and flexible calling plans

Customers can consume the way they want

Local Gateway

Use the customer's local gateway equipment with customers calling solution

Cloud Connected Calling Provider

Buy direct calling solutions from a cloud-connect PSTN/calling provider that is Cisco-approved

Service Provider

(Cisco Cloud Calling + PSTN)

Service Provider offer that bundles their Calling solution with Webex services

Cisco Calling Plans

New

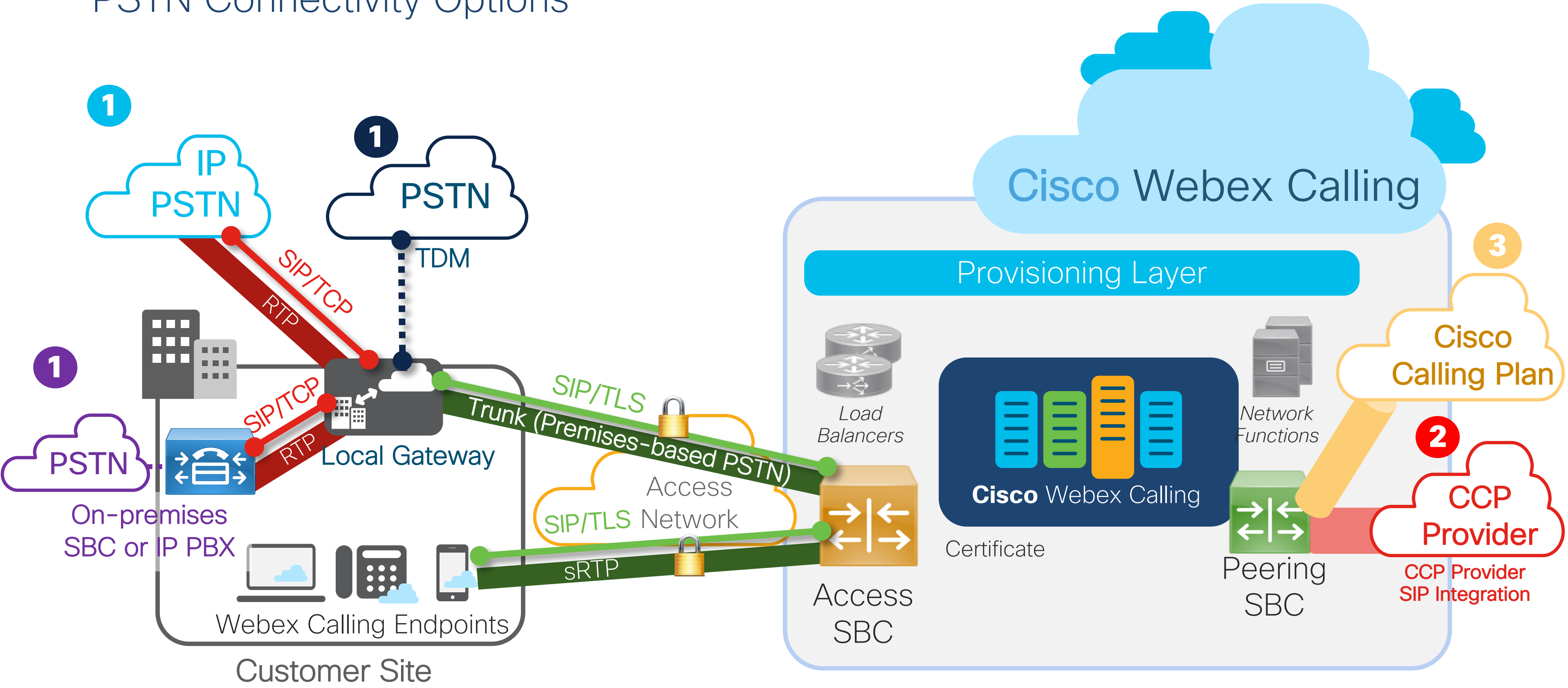
- Intended for VARs ONLY
- Cisco provided Cloud Calling integration into our Webex services
- Easy automated setup & centralized management in Control Hub
- Single order, single bill to customer from partners

Partner Calling Plans Available Today

Cisco Calling Plans

Webex Calling

PSTN Connectivity Options



Setting up Cisco Calling Plan

How Do Customers Order & Provision Cisco Calling?

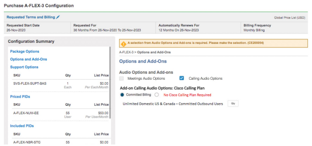
End Customer

Administer

Partners

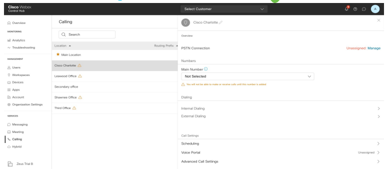
Place an order

Administer



CCW Process

1. Place an order for at least (1) WxC License
2. Provided the option to also purchase Calling Plans
3. Places an order for at least (1) committed Outbound Calling Plan



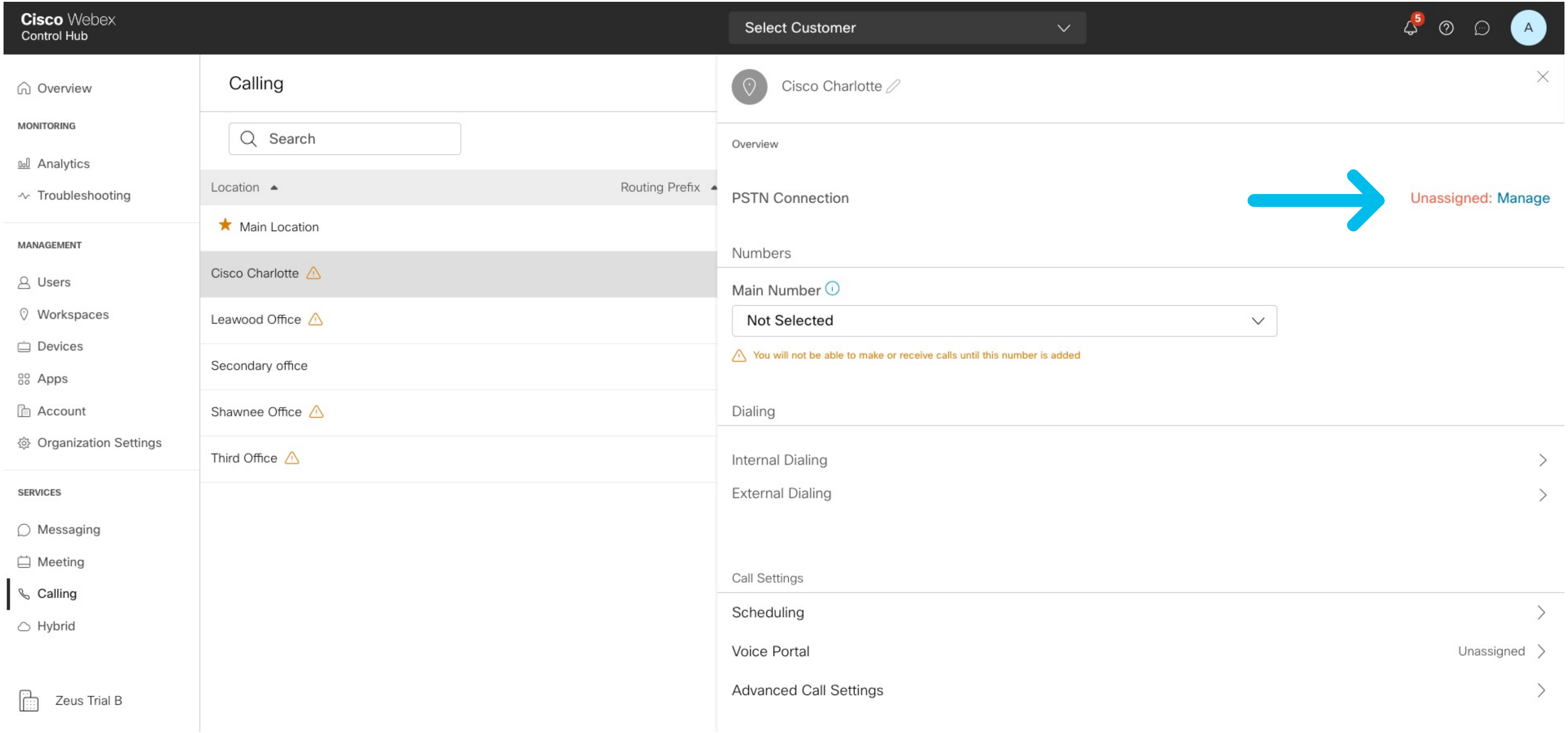
Actions Available in Control Hub

1. Assign any committed Outbound Calling Plans purchased in CCW
2. Order (and assign) uncommitted Outbound Calling Plans
3. Order (and assign) Telephone Numbers
4. Register E911 address

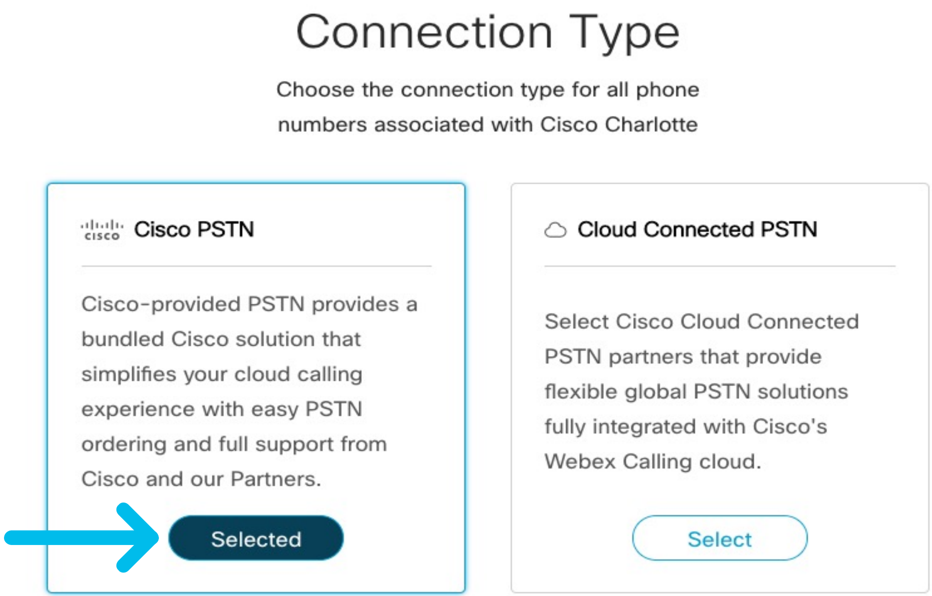
How to Provision (Control Hub)

1. Setting up Cisco PSTN

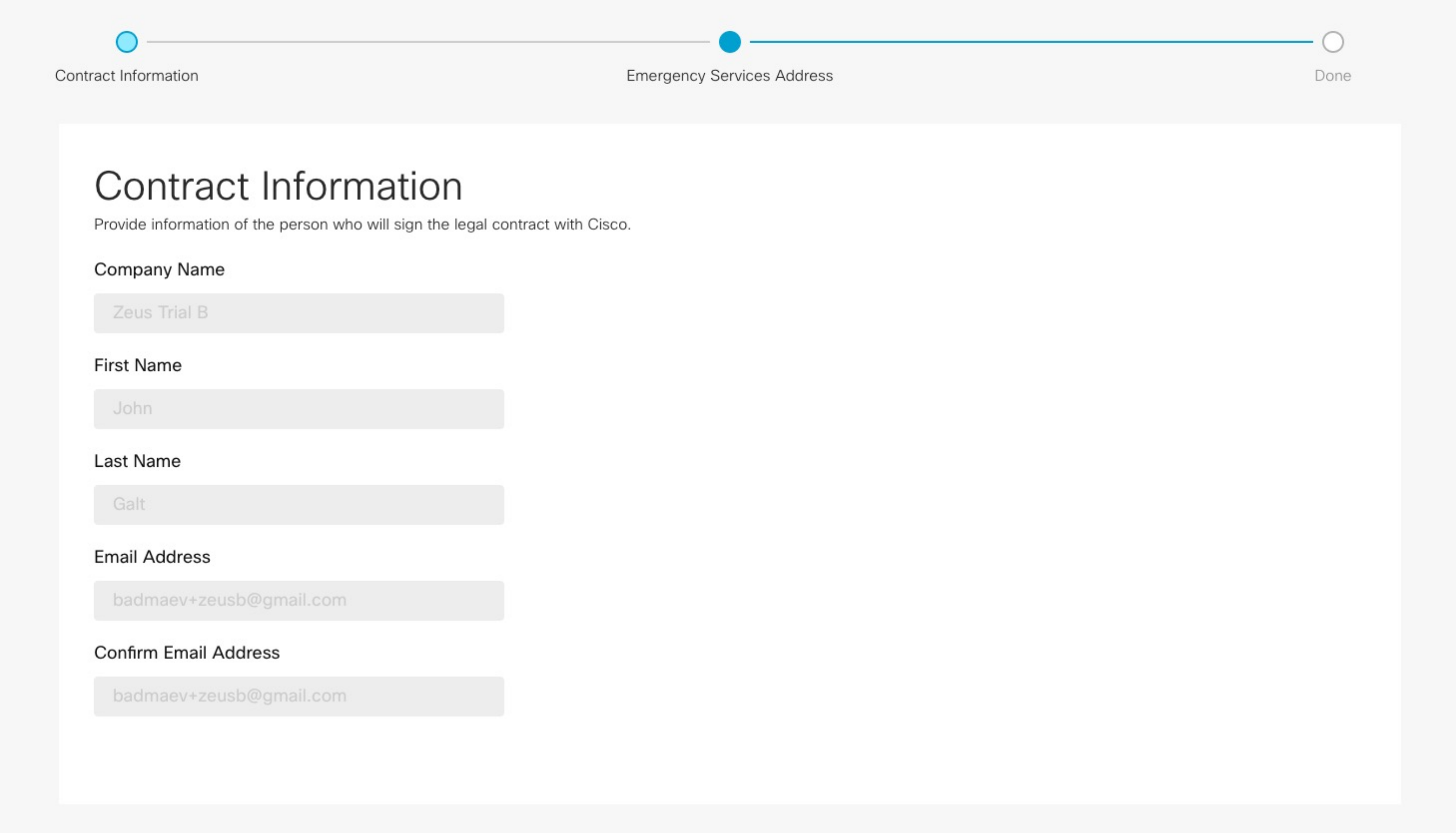
Managing PSTN for a Location



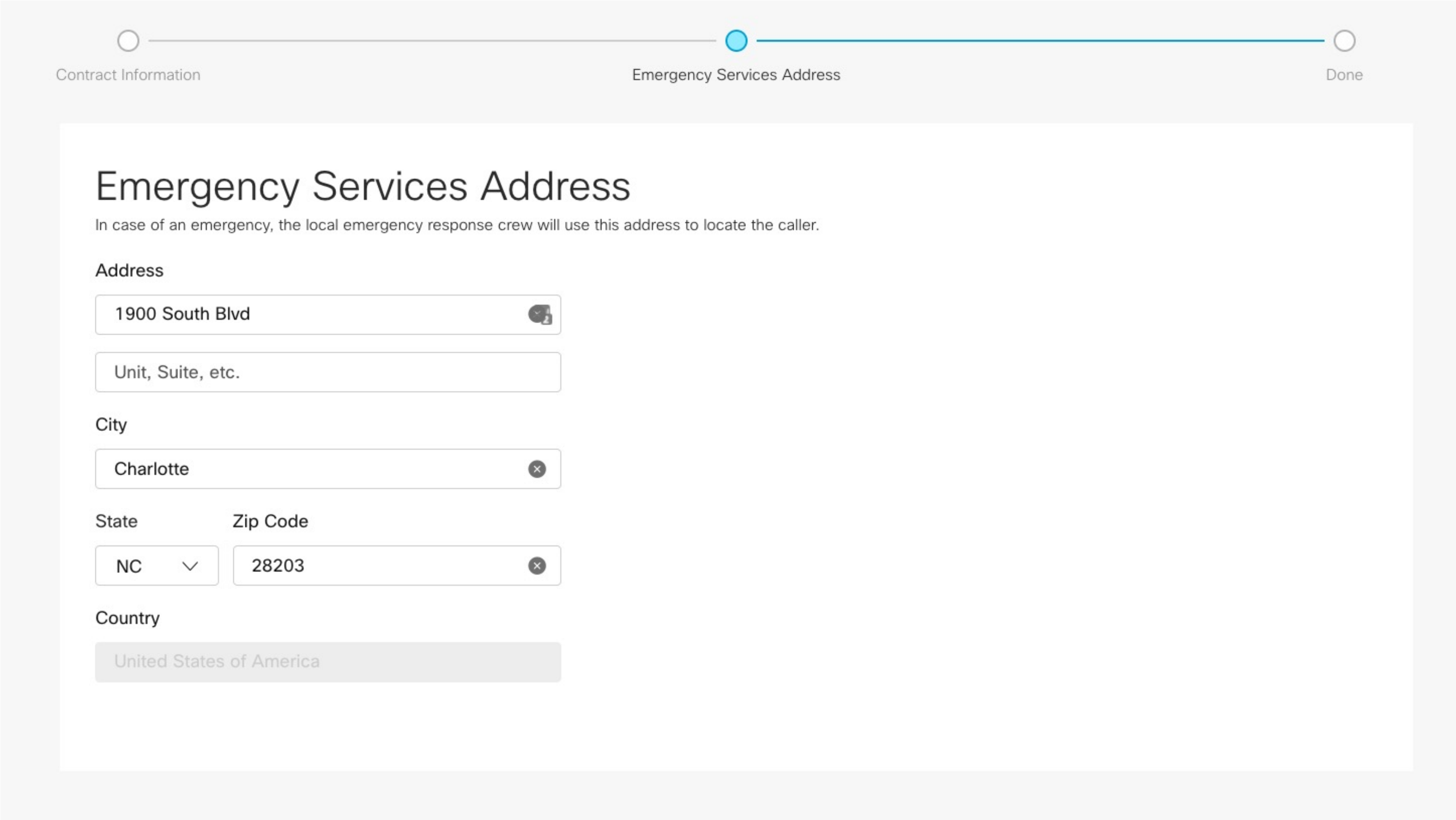
Adding Cisco PSTN for a Location



Registering Contract Info



Registering 911 Address



Registration for Cisco PSTN Complete

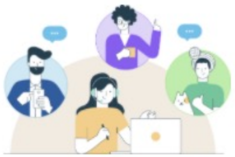
Contract Information — Emergency Services Address — Done

PSTN Connection Saved

Your Information

Contract Information
Zeus Trial B
John Galt
badmaev+zeusb@gmail.com

Emergency Services Address
1900 South Blvd
Charlotte, NC 28203
United States of America



Recommended

Next, add numbers to your trial

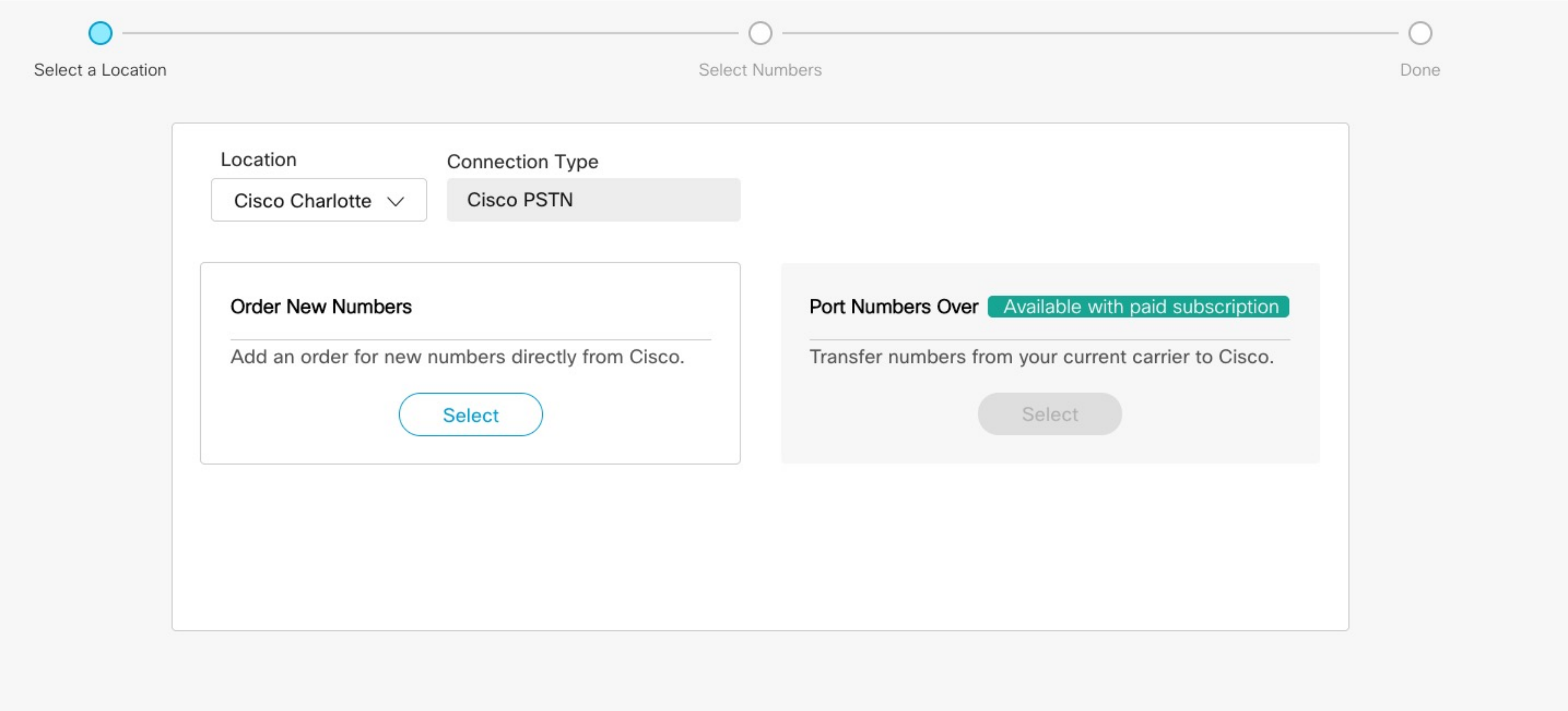
We can Automatically choose some numbers and add them for you.

How many? 1 Get Numbers

I prefer to search and select my own numbers

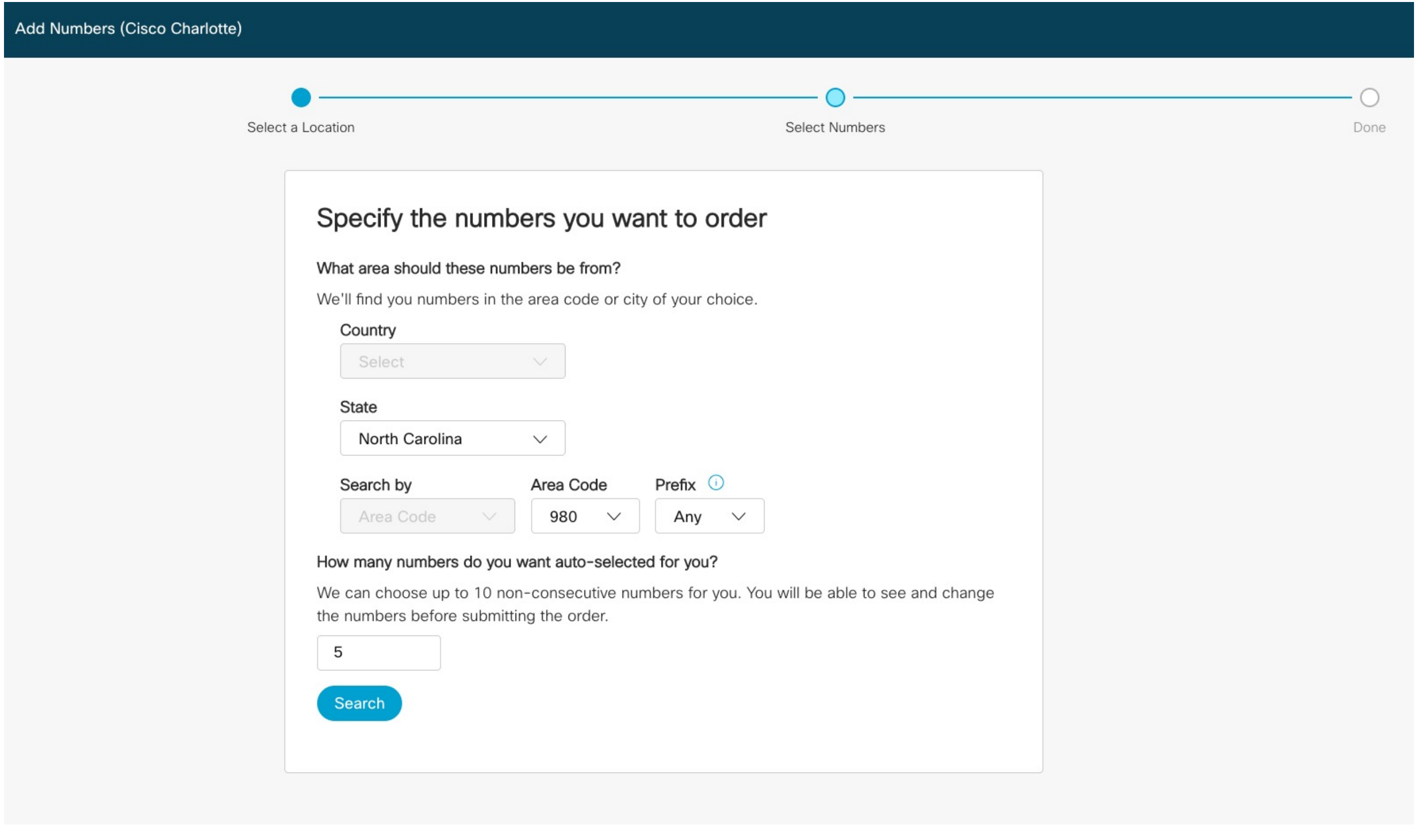
2. Ordering Telephone Numbers (TNs)

Adding TNs

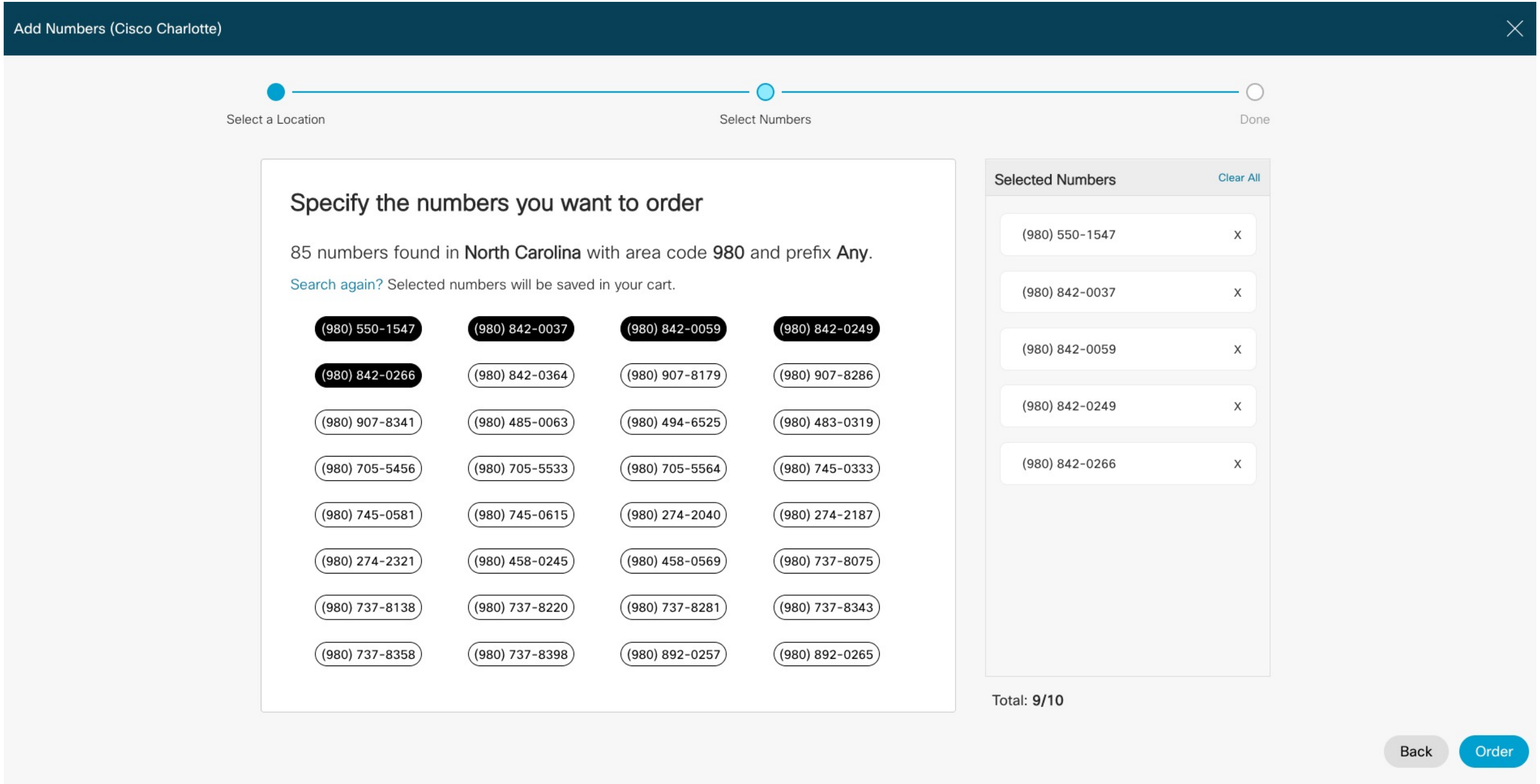


Select a Location
Select Numbers
Done
Location
Cisco Charlotte
Connection Type
Cisco PSTN
Order New Numbers
Add an order for new numbers directly from Cisco.
Select
Port Numbers Over
Available with paid subscription
Transfer numbers from your current carrier to Cisco.
Select

Searching for New TNs



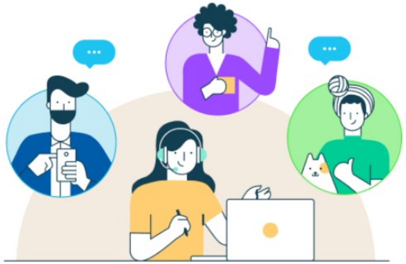
Selecting New TNs



Submitting a TN Order

Add Numbers

Select a Location — Select Numbers — Done



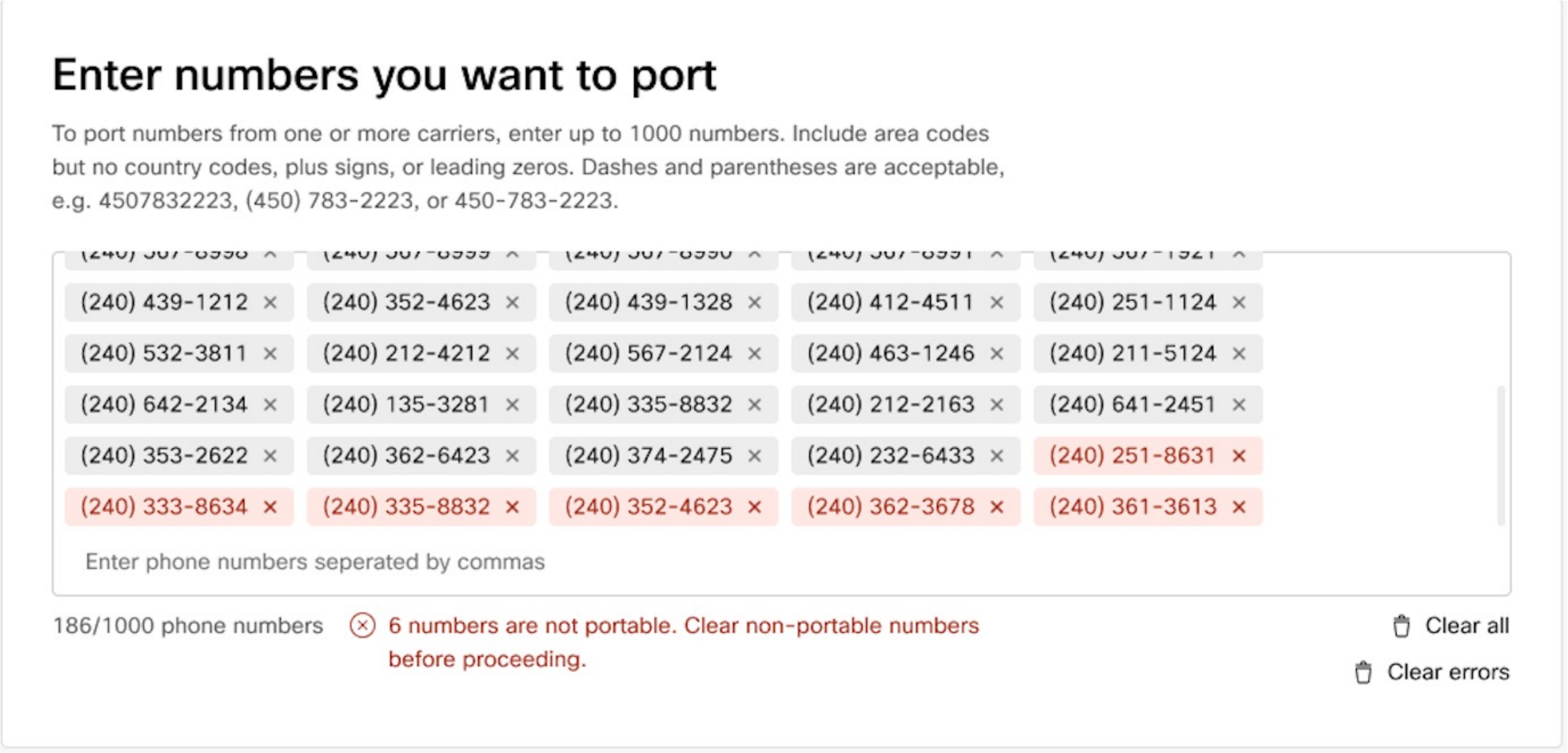
Order Submitted

You can check your order status in PSTN Orders

Phone Numbers (5)
(980) 550-1547
(980) 842-0037
(980) 842-0059
(980) 842-0249
(980) 842-0266

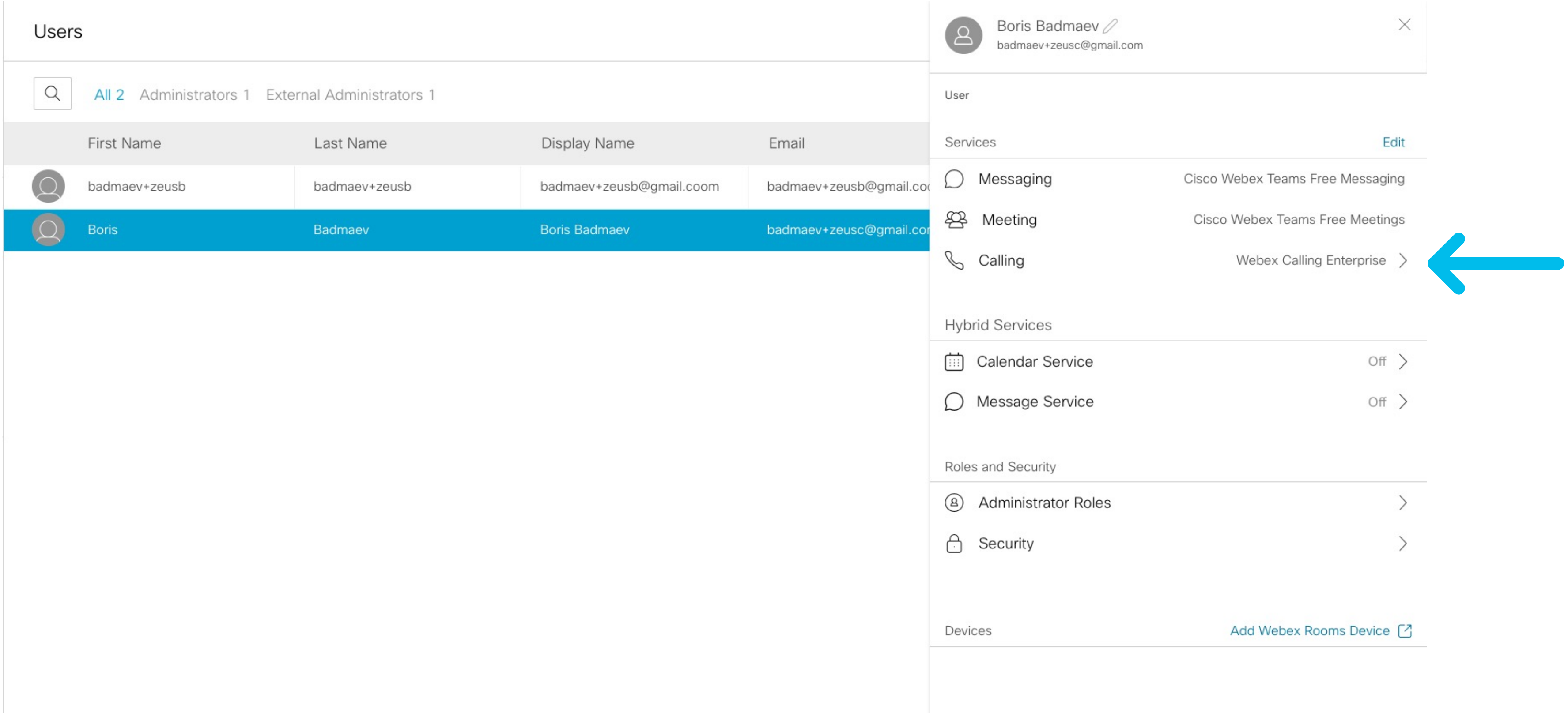
Close | View orders

Porting Existing TNs

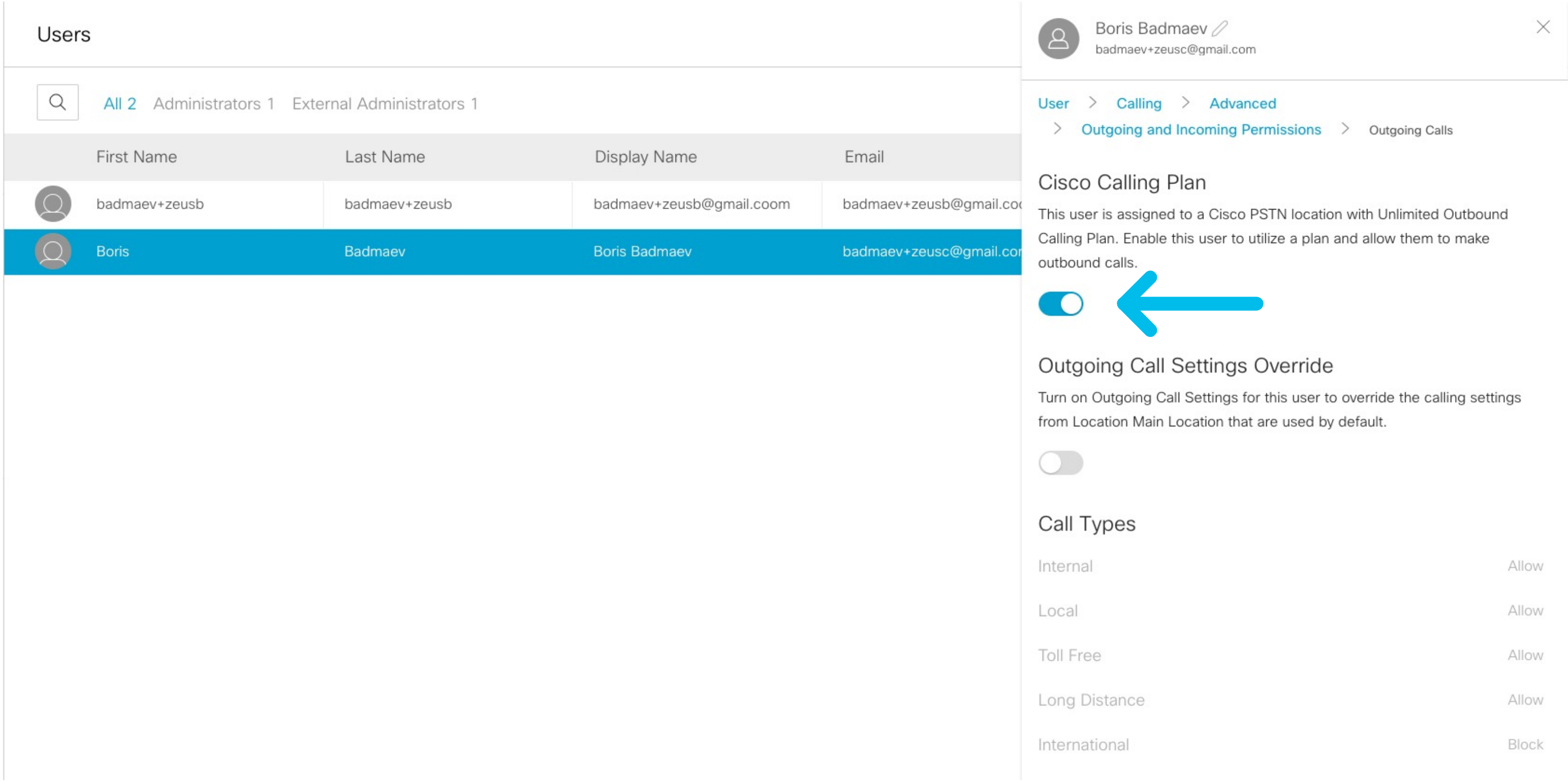


3. Assigning Outbound Calling Plans (OCPs)

Assigning an OCP to a User



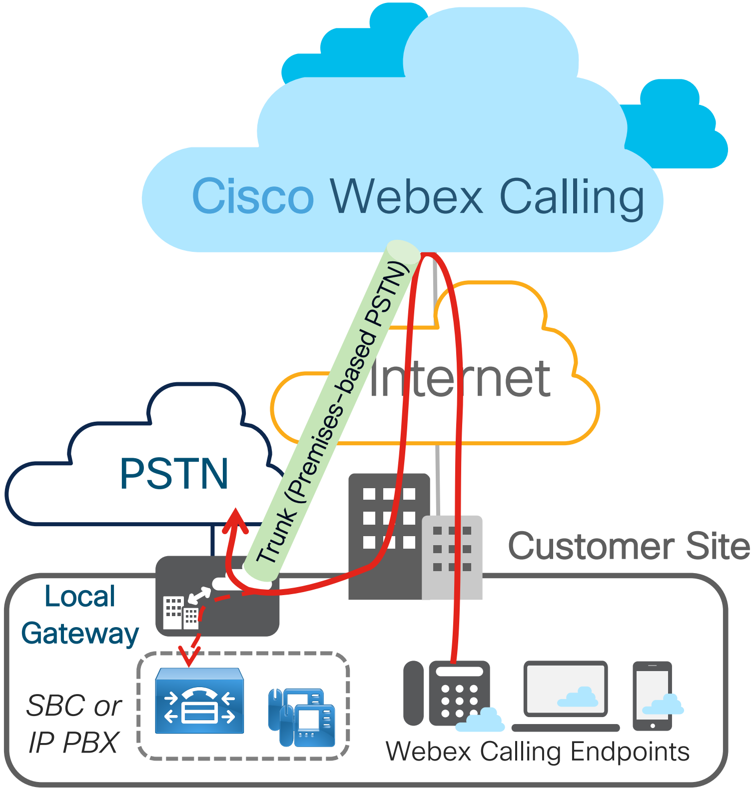
Assigning an OCP to a User



Local Gateway Updates

Webex Calling Trunk - Local Gateway

(Premises-based PSTN) Deployment

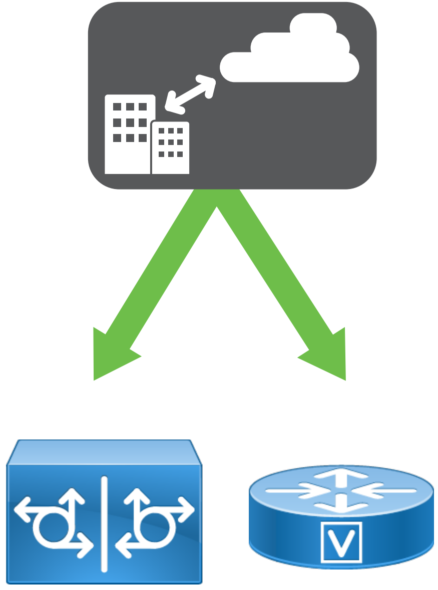


- Provides connectivity to a customer-owned premises-based PSTN service
- May also provide connectivity to an on-premises IP PBX or dedicated SBC/PSTN GW
- Enables on-prem to Webex Calling transition
- **Endpoint registration is NOT proxied through Local Gateway, unlike CUBE Lineside.** Endpoints directly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

Local Gateway

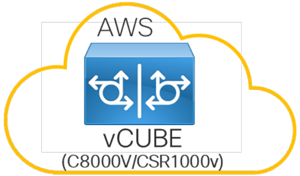
Platform Support

Local Gateway (LGW)

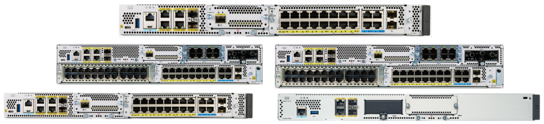


CUBE   IOS-XE GW

- **Cisco CUBE** (for IP-based connectivity) or
  Cisco IOS Gateway (for TDM-based connectivity)
- Hardware and software requirements:
  - ISR 4321, 4331, 4351, 4431, 4451, 4461 (IOS XE 17.3.3)
  - vCUBE in AWS



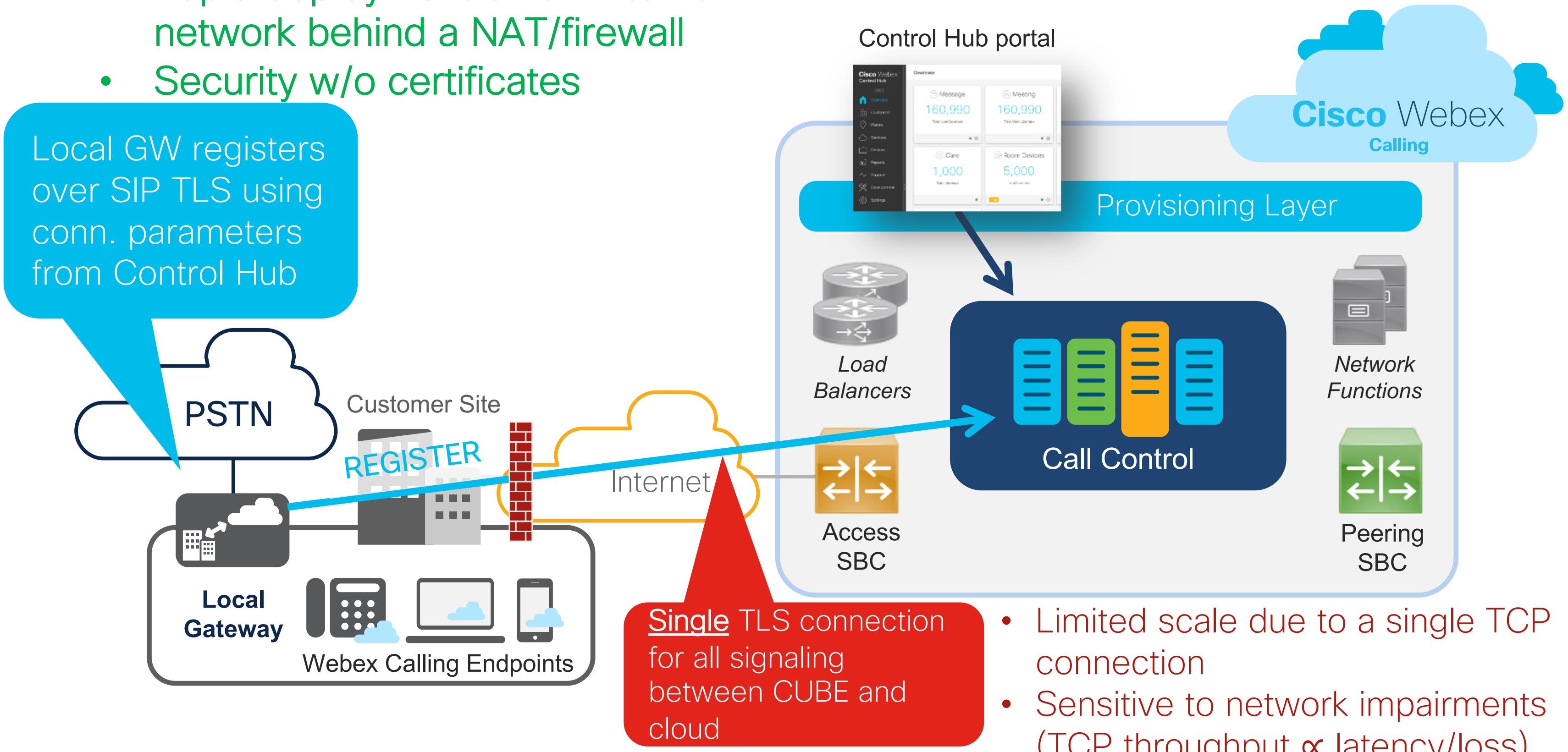
  - Catalyst 8200/8300 series



  - CSR 1000v (vCUBE) (16.12.5 or later) –
    - CSR 1000v licenses are not included in Webex Calling Flex and need to be purchased separately
    - Estimate 200 kbps total data throughput for every audio call
  - ISR 1100 (IOS-XE 16.12.5 or later)

Registering Trunk regardless of SBC

- Rapid deployment on an internal network behind a NAT/firewall
- Security w/o certificates



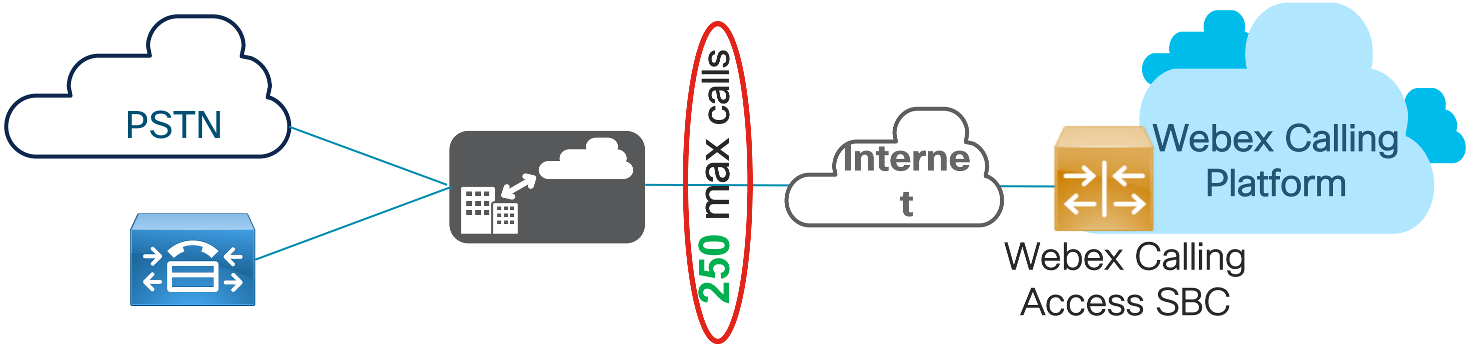
- Limited scale due to a single TCP connection
- Sensitive to network impairments (TCP throughput ∝ latency/loss)

Update IP Trust based on latest subnets

```
LocalGateway#configure terminal
LocalGateway(config)#voice service voip
 ip address trusted list
  ipv4 A.B.C.D X.X.X.Y  !← Always check the Port Reference guide for latest IP list
 media statistics
 media bulk-stats
 allow-connections sip to sip
 no supplementary-service sip refer
 no supplementary-service sip handle-replaces
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 stun
  stun flowdata agent-id 1 boot-count 4
  stun flowdata shared-secret 0 Password123$
 sip
  g729 annexb-all
  early-offer forced
```

Webex Calling Trunk – Local gateway Concurrent Call Limits

- Regardless of LGW platform, premises trunks between LGW and Webex Calling cannot exceed 250 concurrent calls when connected over the Internet (OTT).
  - This assumes a maximum of 100ms one-way latency with no more than10ms jitter, less than 0.5% packet loss
  - Poor network conditions between Local Gateway and Webex Calling access SBC can limit the performance of the signaling connection leading to an even lower concurrent calls limit.
- Multiple LGWs with Trunk and Route groups can be deployed for higher scale:
  - Premises → cloud calls: load balancing supported today *(e.g., CUCM route groups)*
  - Cloud → premises calls: Webex Calling Trunk and Route Groups



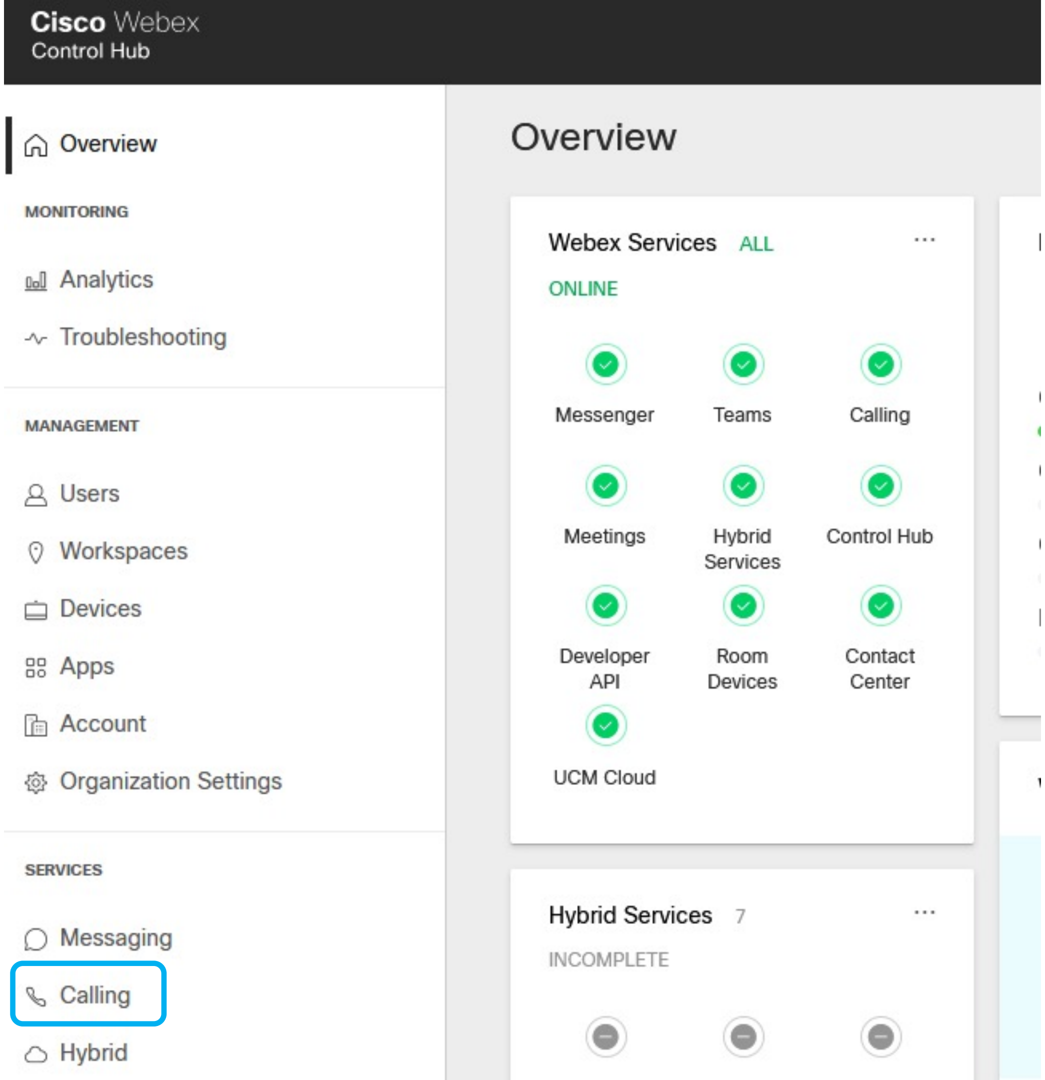
Note: *Contact your Cisco account team if you need more than 250 concurrent calls **per** LGW*

Key Local Gateway Configuration Updates Required

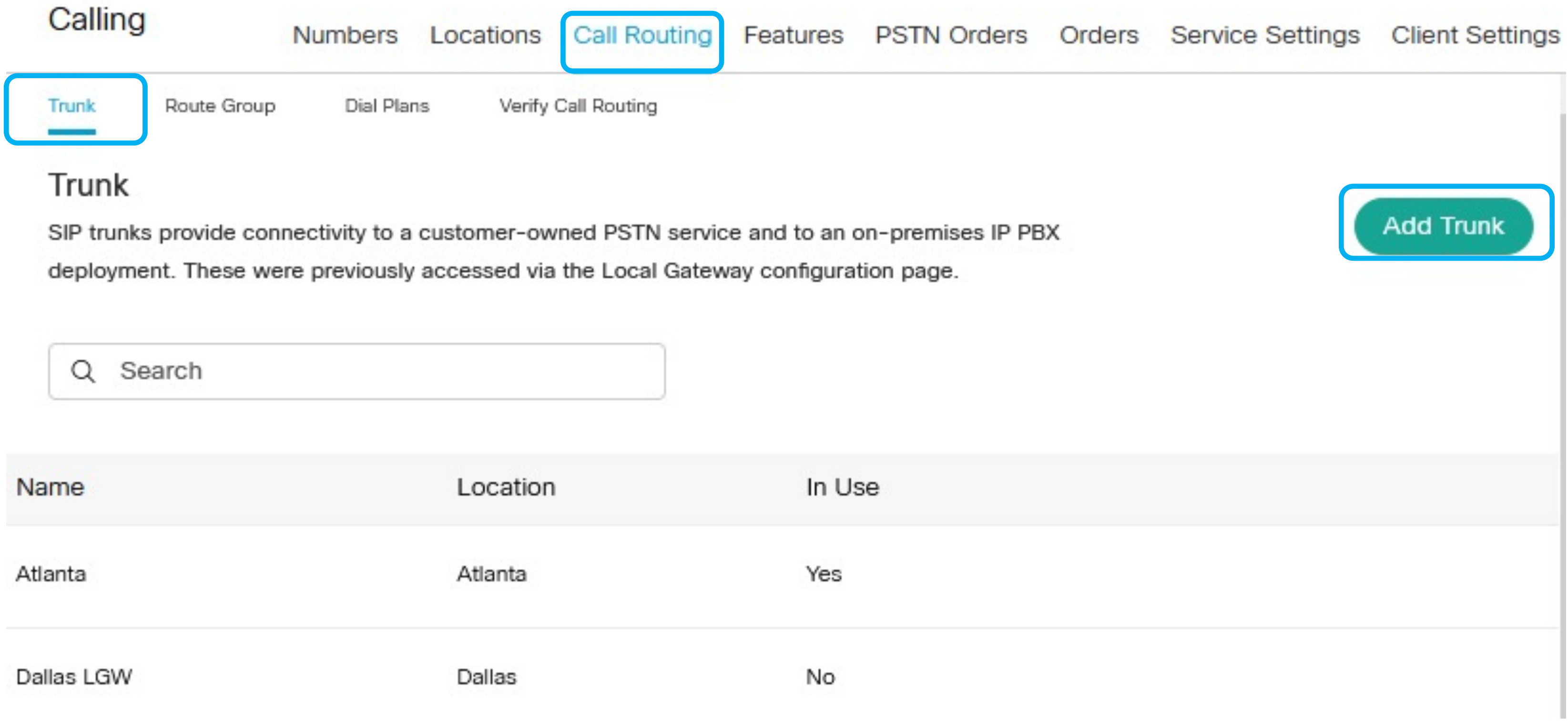
Update

Onboarding Process
Webex Calling Trunk

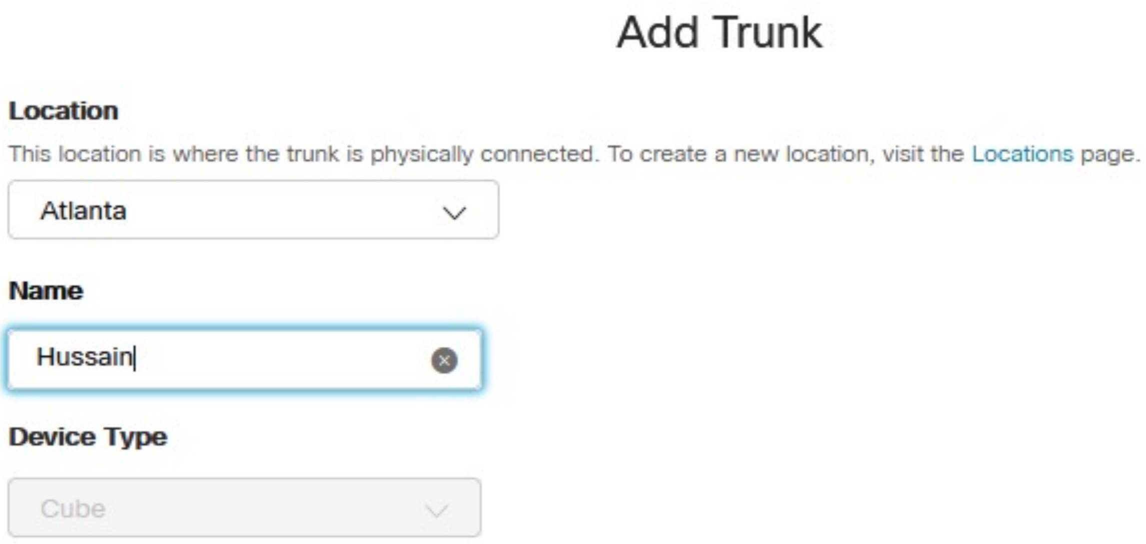
1a. Log in to customer portal and navigate to Services – Click Calling



1b. Navigate to Trunk within Call Routing and select Add Trunk



1c. Add a new Trunk for the desired Location



- Trunk name is limited to 24 characters

1g. Save the Trunk parameters to build the LGW CLI

Parameters on this display required for building LGW CLI

Add Trunk

Hussain Successfully Created.

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

Visit Locations page to configure PSTN connection to individual locations.

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

Trunk Info

Status

● unknown

Trunk Group OTG/DTG
hussain2572_lgu

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

Registrar Domain
40462196.cisco-bcld.com

Line/Port
Hussain6346_LGU@40462196.cisco-bcld.com

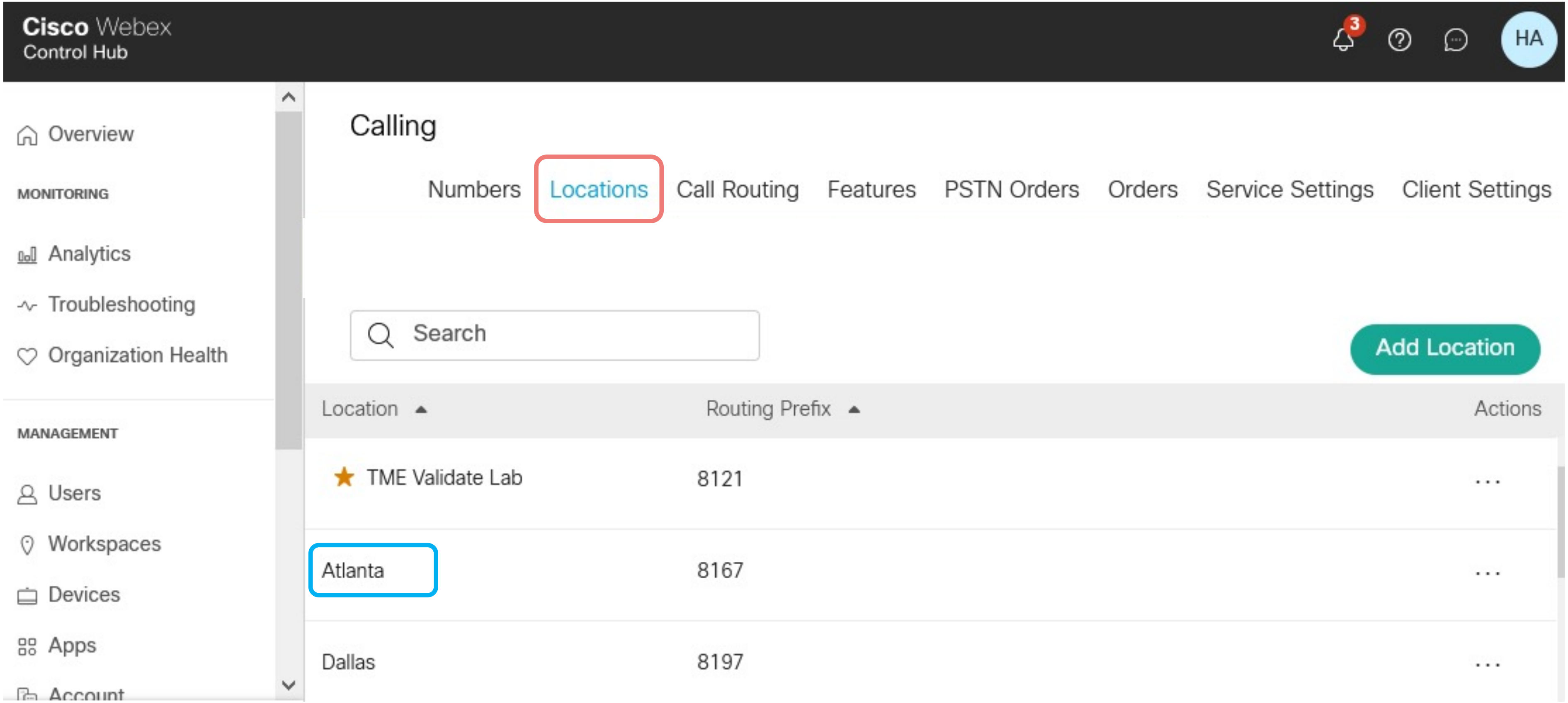
Authentication Information

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

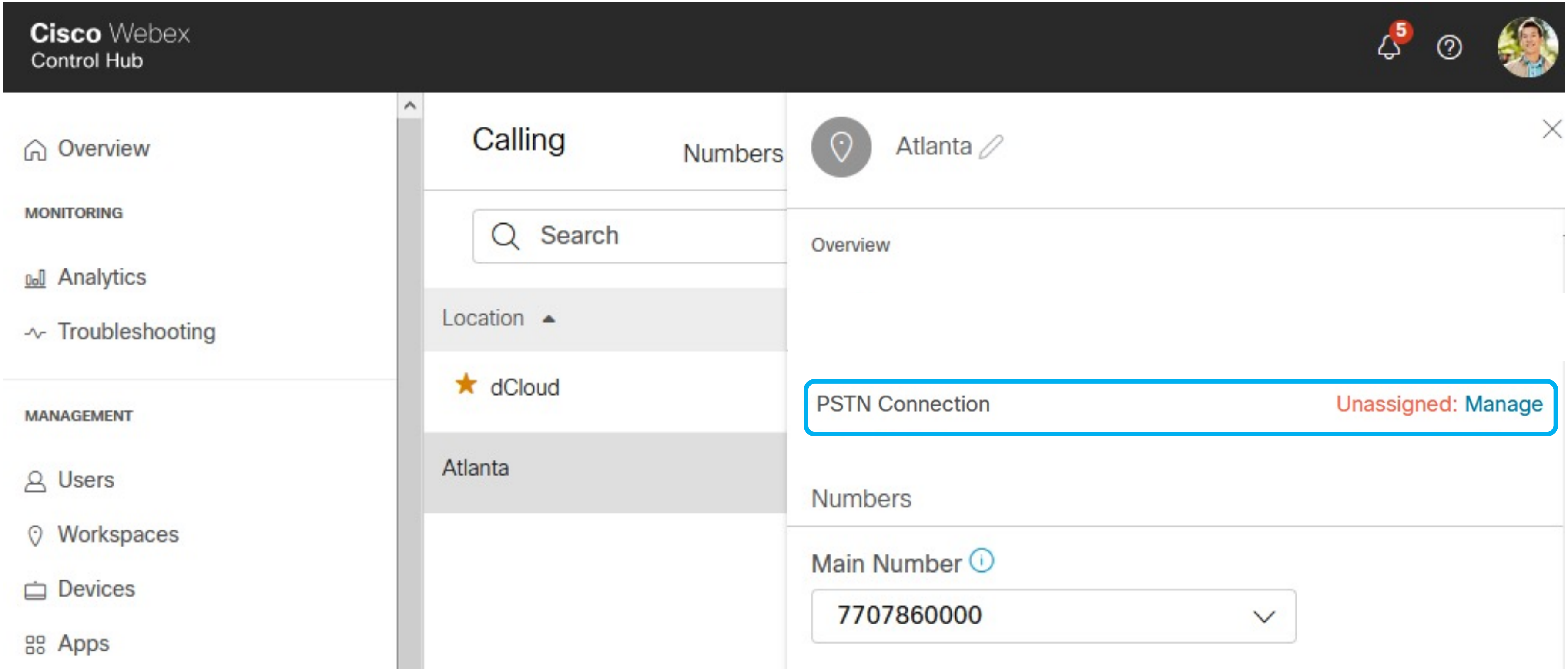
Username: Hussain2572_LGU

Password: meX7]~)VmF

1f. Navigate to Locations under Calling and select the desired location



1g. Click on Unassigned under PSTN Connection



1h. Select Premises-based PSTN
(formerly local gateway)

Connection Type

Choose the connection type for all phone numbers associated with Smyrna.

Cisco PSTN

Cisco-provided PSTN provides a bundled Cisco solution that simplifies your cloud calling experience with easy PSTN ordering and full support from Cisco and our Partners.

Unavailable; talk to your partner.

Cloud Connected PSTN

Select Cisco Cloud Connected PSTN partners that provide flexible global PSTN solutions fully integrated with Cisco's Webex Calling cloud.

Select

Premises-based PSTN
(formerly local gateway)

Bring Your Own Carrier by interconnecting any Service Provider's PSTN with a premises-based local gateway that tightly integrates to Cisco's Webex Calling cloud.

Selected

1i. Select the Trunk, verify the Control Hub Location, and click Save

Connection Type

Premises-based PSTN

Routing Choice

Visit the Trunk or Route Group page to manage your choices of premises-based PSTN.

Hussain

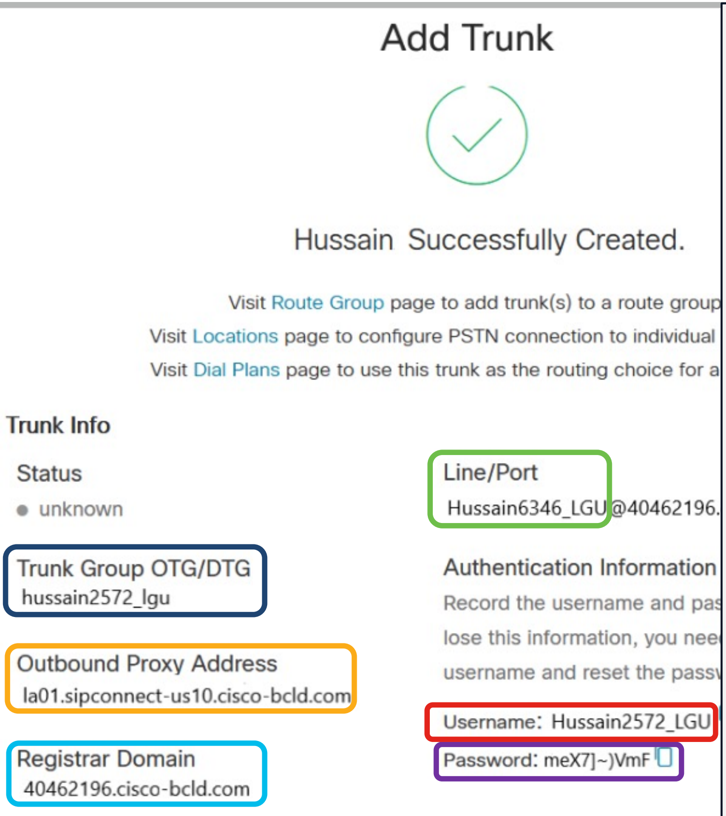
This trunk is located in **Atlanta**.

* I confirm that I understand that this change will immediately change the routing of PSTN calls and that Smyrna has been set up correctly to accept this change. This could include porting of numbers, configuration of premises equipment and/or coordinating with PSTN providers. Porting of numbers includes: Users,

Back Save

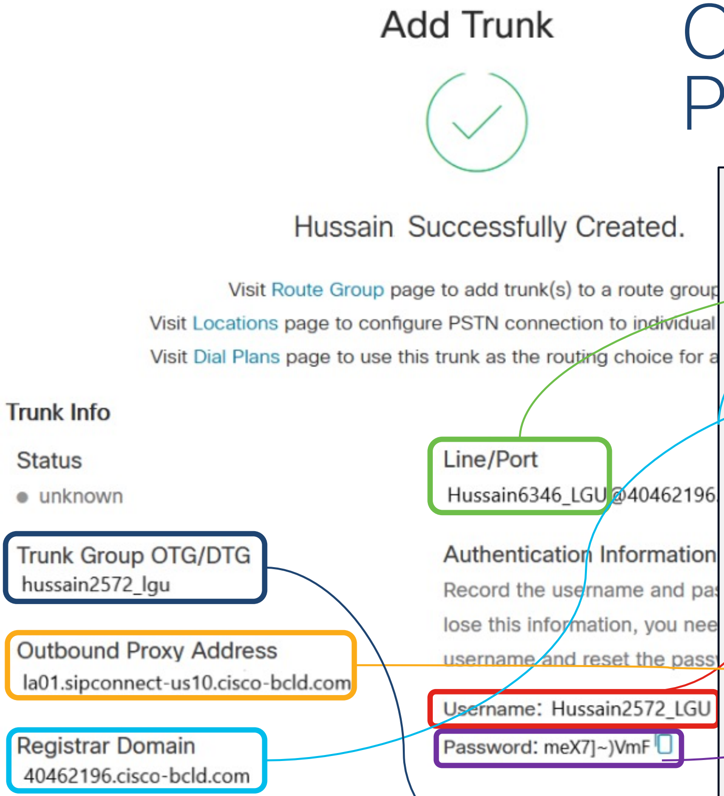
Updating Outbound Proxy

Control Hub Trunk Info Connection Parameters → LGW CLI Config




```
voice class tenant 200
  registrar dns:40462196.cisco-bcld.com scheme sips expires 240 refresh-ratio 50 tcp tls
  credentials number Hussain6346_LGU username Hussain2572_LGU password 0 meX7]~)VmF realm
BroadWorks
  authentication username Hussain2572_LGU password 0 meX7]~)VmF realm BroadWorks
  authentication username Hussain2572_LGU password 0 meX7]~)VmF realm 40462196.cisco-bcld.com
  sip-server dns:40462196.cisco-bcld.com
  connection-reuse
  srtp-crypto 200
  session transport tcp tls
  url sips
  error-passthru
  bind control source-interface GigabitEthernet0/0/1
  bind media source-interface GigabitEthernet0/0/1
  no pass-thru content custom-sdp
  sip-profiles 200
  outbound-proxy dns:la01.sipconnect-us10.cisco-bcld.com
…
voice class sip-profiles 200
 rule 1 request ANY sip-header SIP-Req-URI modify "sips:" "sip:"
 rule 10 request ANY sip-header To modify "<sips:" "<sip:"
 rule 11 request ANY sip-header From modify "<sips:" "<sip:"
 rule 12 request ANY sip-header Contact modify "<sips:(.*)>" "<sip:\1;transport=tls>"
 rule 13 response ANY sip-header To modify "<sips:" "<sip:"
 rule 14 response ANY sip-header From modify "<sips:" "<sip:"
 rule 15 response ANY sip-header Contact modify "<sips:" "<sip:"
 rule 16 request ANY sip-header From modify ">" ";otg=hussain2572_lgu>"
 rule 17 request ANY sip-header P-Asserted-Identity modify "<sips:" "<sip:"
```

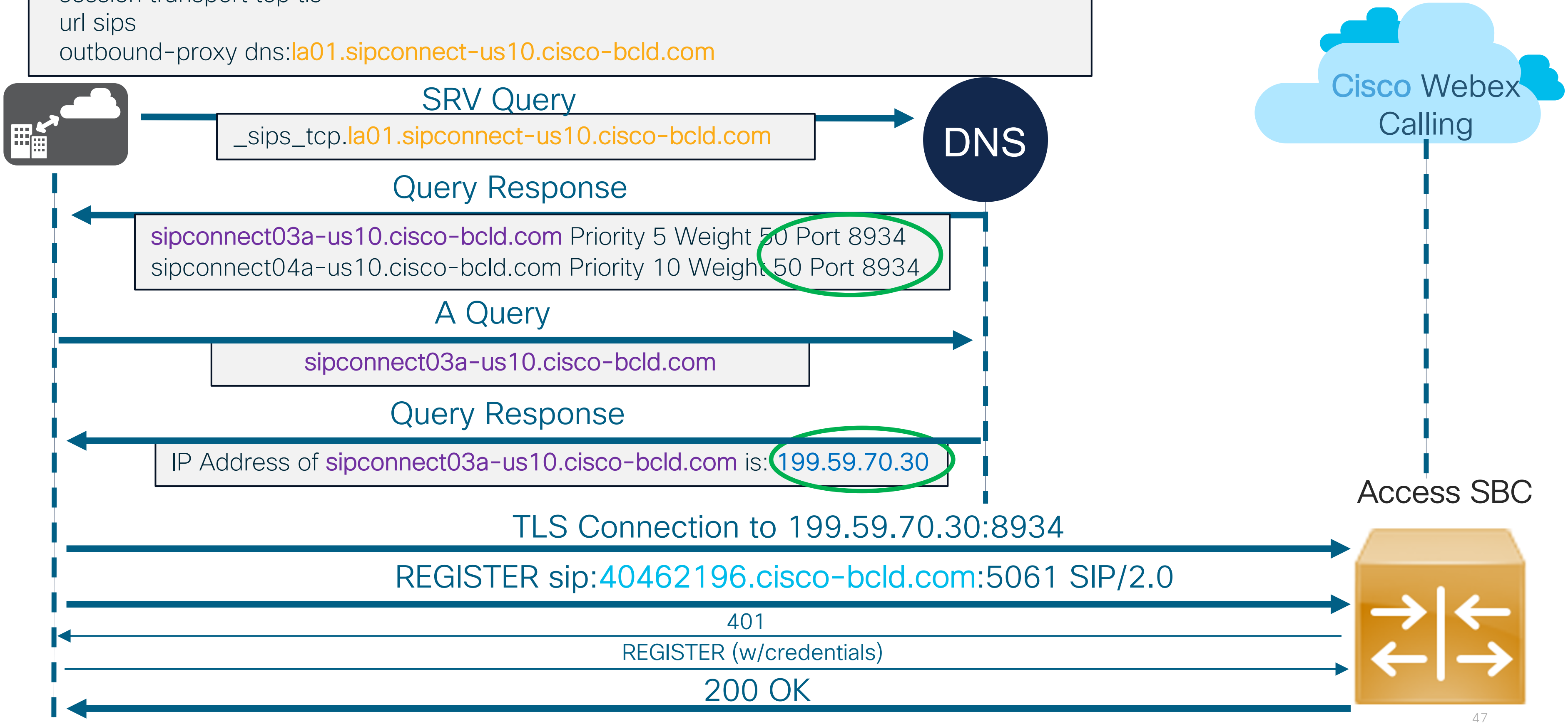
Control Hub Trunk Info Connection Parameters → LGW CLI Config




```
voice class tenant 200
  registrar dns:40462196.cisco-bcld.com scheme sips expires 240 refresh-ratio 50 tcp tls
  credentials number Hussain6346_LGU username Hussain2572_LGU password 0 meX7]~)VmF realm
BroadWorks
  authentication username Hussain2572_LGU password 0 meX7]~)VmF realm BroadWorks
  authentication username Hussain2572_LGU password 0 meX7]~)VmF realm 40462196.cisco-bcld.com
  sip-server dns:40462196.cisco-bcld.com
  connection-reuse
  srtp-crypto 200
  session transport tcp tls
  url sips
  error-passthru
  bind control source-interface GigabitEthernet0/0/1
  bind media source-interface GigabitEthernet0/0/1
  no pass-thru content custom-sdp
  sip-profiles 200
  outbound-proxy dns:la01.sipconnect-us10.cisco-bcld.com
…
voice class sip-profiles 200
 rule 1 request ANY sip-header SIP-Req-URI modify "sips:" "sip:"
 rule 10 request ANY sip-header To modify "<sips:" "<sip:"
 rule 11 request ANY sip-header From modify "<sips:" "<sip:"
 rule 12 request ANY sip-header Contact modify "<sips:(.*)>" "<sip:\1;transport=tls>"
 rule 13 response ANY sip-header To modify "<sips:" "<sip:"
 rule 14 response ANY sip-header From modify "<sips:" "<sip:"
 rule 15 response ANY sip-header Contact modify "<sips:" "<sip:"
 rule 16 request ANY sip-header From modify ">" ";otg=hussain2572_lgu>"
 rule 17 request ANY sip-header P-Asserted-Identity modify "<sips:" "<sip:"
```

Establishing Secure Connectivity b/w LGW and Webex Calling

```
voice class tenant 200
  registrar dns:40462196.cisco-bcld.com scheme sips expires 240 refresh-ratio 50 tcp tls
  session transport tcp tls
  url sips
  outbound-proxy dns:la01.sipconnect-us10.cisco-bcld.com
```



Step by Step outbound proxy upgrade process

(Reload not required)

1. Update IP Trust list based on [Webex Calling Port Reference guide]
2. Update any applicable / matching firewall rules based on above IP ranges
3. Get the new outbound proxy from control hub
4. In `voice class tenant 200` issue `no registrar` and `no outbound-proxy`

```
voice class tenant 200
  no registrar !-> sends a REGISTER to Access SBC with Expires:0
  no outbound-proxy
```

5. Update with the new `outbound-proxy` within `voice class tenant 200` and add the `registrar` back

```
voice class tenant 200
  outbound-proxy dns:<new outbound proxy fqdn>
  registrar dns:<same registrar fqdn>
```

6. Save the local gateway configuration using the `write` command
7. Validate the registration for OTG is successful with `show sip-ua register status`

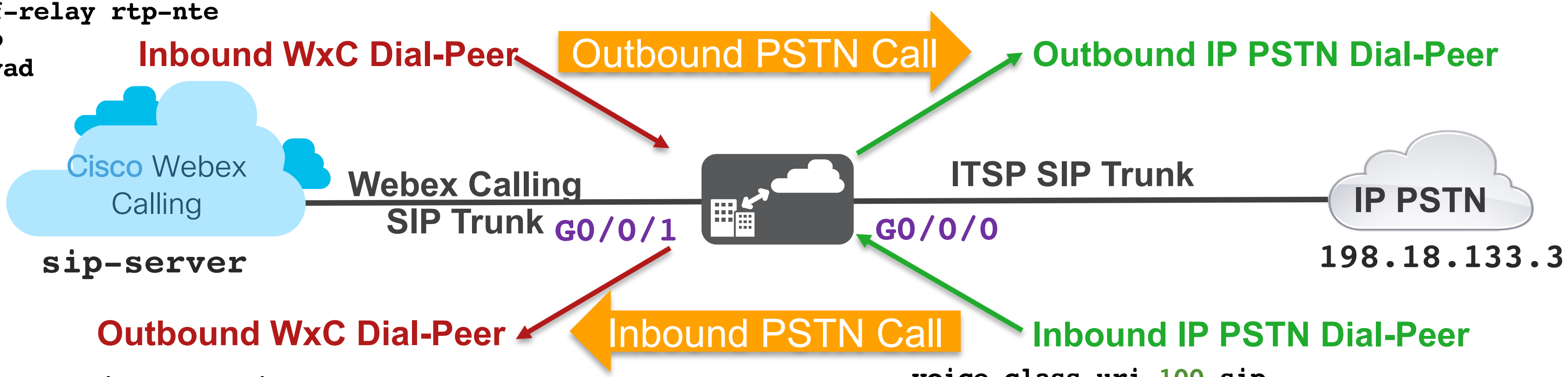
Update

Single Dial-peer facing Webex Calling for Inbound/Outbound Calls

Existing Dial-peer structure

```
voice class uri 200 sip
  pattern dtg=hussain3847.lgu

dial-peer voice 200 voip
  description Incoming dial-peer from Webex Calling
  incoming uri request 200
  destination dpg 100
  voice-class codec 99
  voice-class stun-usage 200
  voice-class sip tenant 200
  dtmf-relay rtp-nte
  srtp
  no vad
```

```
dial-peer voice 101 voip
  description Outgoing dial-peer to IP PSTN
  destination-pattern BAD.BAD
  session protocol sipv2
  session target ipv4:198.18.133.3
  voice-class codec 99
  voice-class sip tenant 100
  dtmf-relay rtp-nte
  no vad
```

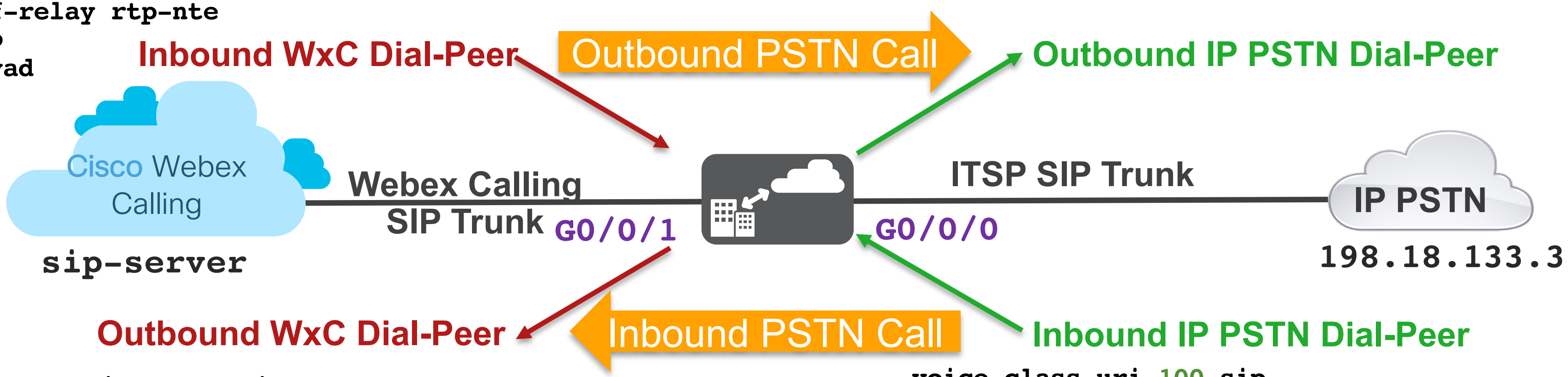



```
dial-peer voice 201 voip
  description Outgoing dial-peer to Webex Calling
  destination-pattern BAD.BAD
  session target sip-server
  voice-class codec 99
  voice-class stun-usage 200
  no voice-class sip localhost
  voice-class sip tenant 200
  dtmf-relay rtp-nte
  srtp
  no vad
```

```
voice class uri 100 sip
  host ipv4:198.18.133.3

dial-peer voice 100 voip
  description Incoming dial-peer from IP PSTN
  incoming uri via 100
  session protocol sipv2
  destination dpg 200
  voice-class codec 99
  voice-class sip tenant 300
  dtmf-relay rtp-nte
  no vad
```

```
voice class uri 200 sip
  pattern dtg=hussain3847.lgu

dial-peer voice 200 voip
  description Incoming dial-peer from Webex Calling
  incoming uri request 200
  destination dpg 100
  voice-class codec 99
  voice-class stun-usage 200
  voice-class sip tenant 200
  dtmf-relay rtp-nte
  srtp
  no vad
```

```
dial-peer voice 101 voip
 description Outgoing dial-peer to IP PSTN
 destination-pattern BAD.BAD
 session protocol sipv2
 session target ipv4:198.18.133.3
 voice-class codec 99
 voice-class sip tenant 100
 dtmf-relay rtp-nte
 no vad
```



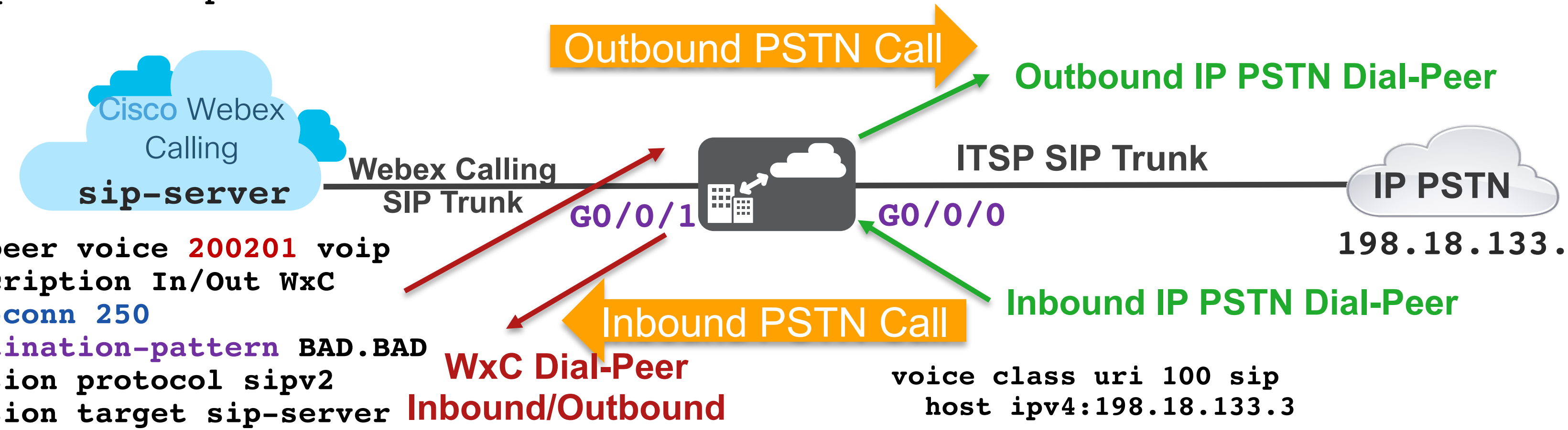

```
dial-peer voice 201 voip
  description Outgoing dial-peer to Webex Calling
  destination-pattern BAD.BAD
  session target sip-server
  voice-class codec 99
  voice-class stun-usage 200
  no voice-class sip localhost
  voice-class sip tenant 200
  dtmf-relay rtp-nte
  srtp
  no vad
```

```
voice class uri 100 sip
  host ipv4:198.18.133.3

dial-peer voice 100 voip
  description Incoming dial-peer from IP PSTN
  incoming uri via 100
  session protocol sipv2
  destination dpg 200
  voice-class codec 99
  voice-class sip tenant 300
  dtmf-relay rtp-nte
  no vad
```

New Dial-peer structure

```
voice class uri 200 sip
  pattern dtg=hussain3847.lgu

voice class dpg 100
 description Incoming WxC(DP200201) to IP PSTN(DP101)
 dial-peer 101 preference 1

voice class dpg 200
 description Incoming IP PSTN(DP100) to WxC(DP200201)
 dial-peer 200201 preference 1
```

```
dial-peer voice 101 voip
 description Outgoing dial-peer to IP PSTN
 destination-pattern BAD.BAD
 session protocol sipv2
 session target ipv4:198.18.133.3
 voice-class codec 99
 voice-class sip tenant 100
 dtmf-relay rtp-nte
 no vad
```




```
dial-peer voice 200201 voip
  description In/Out WxC
  max-conn 250
  destination-pattern BAD.BAD
  session protocol sipv2
  session target sip-server
  destination dpg 100
  incoming uri request 200
  voice-class codec 99
  voice-class stun-usage 200
  no voice-class sip localhost
  voice-class sip tenant 200
  dtmf-relay rtp-nte
  srtp
  no vad
```

```
voice class uri 100 sip
  host ipv4:198.18.133.3

dial-peer voice 100 voip
  description Incoming dial-peer from IP PSTN
  incoming uri via 100
  session protocol sipv2
  destination dpg 200
  voice-class codec 99
  voice-class sip tenant 300
  dtmf-relay rtp-nte
  no vad
```

Single vCUBE instance with two LGWs – Total 500 calls

Trunk1 – LGW1=250 calls

```
dial-peer voice 200201 voip
  description In/Out WxC
  max-conn 250
  destination-pattern BAD.BAD
  session protocol sipv2
  session target sip-server
  destination dpg 100
  incoming uri request 200
  voice-class sip tenant 200

voice class tenant 200
  bind control source-interface GigabitEthernet0/0/1
  bind media source-interface GigabitEthernet0/0/1
```

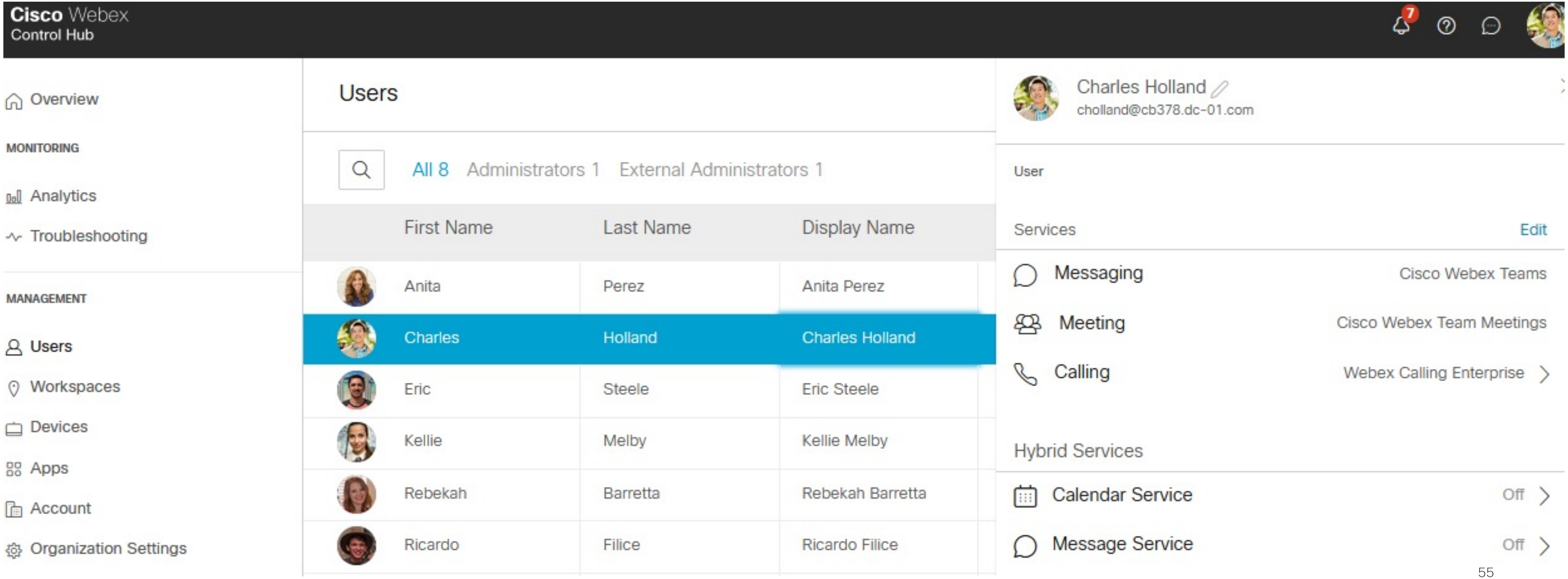
Trunk 2 – LGW2=250 calls

```
dial-peer voice 300301 voip
  description In/Out WxC
  max-conn 250
  destination-pattern BAD.BAD
  session protocol sipv2
  session target sip-server
  destination dpg 300
  incoming uri request 300
  voice-class sip tenant 300

voice class tenant 300
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
```

Caller ID handling

Caller ID settings

- A user's caller ID setting can be modified by administrators from within the Control Hub (determines user's display information for outgoing calls)
- Select **Users** from MANAGEMENT and click on a user and then **Calling**

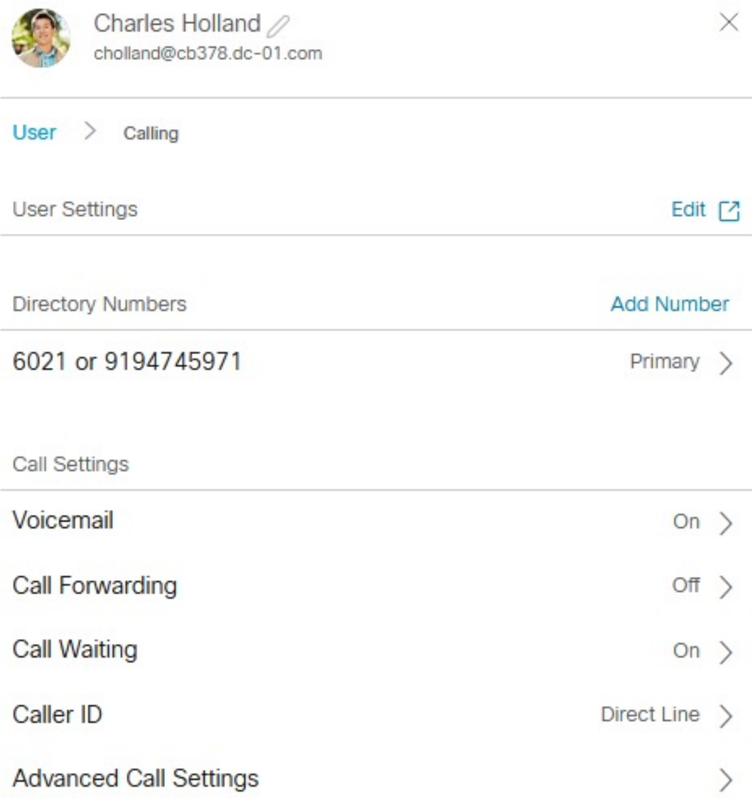


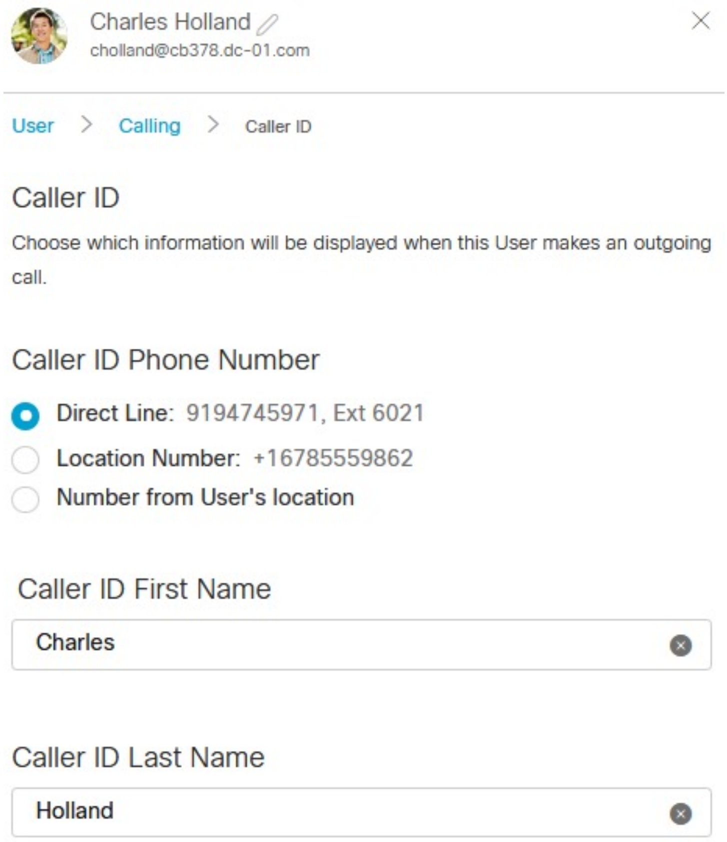
Caller ID settings

- Select **Caller ID** from within **Calling**
- Select the Caller ID option you want to display for a user's outgoing calls
  - **Direct Line –** User's Phone Number / Extension
  - **Location Number** – The main number for the Location.
  - **Number from the user's location** – If you select this option, choose an assigned number from the user's location to appear when the user is making outgoing calls.
    - Helpful when you want all the users from the same department to display the same outgoing number. E.g., Customer Service
- User's Caller ID first and last name can also be modified

Caller ID settings

- Select **Caller ID** from within **Calling**





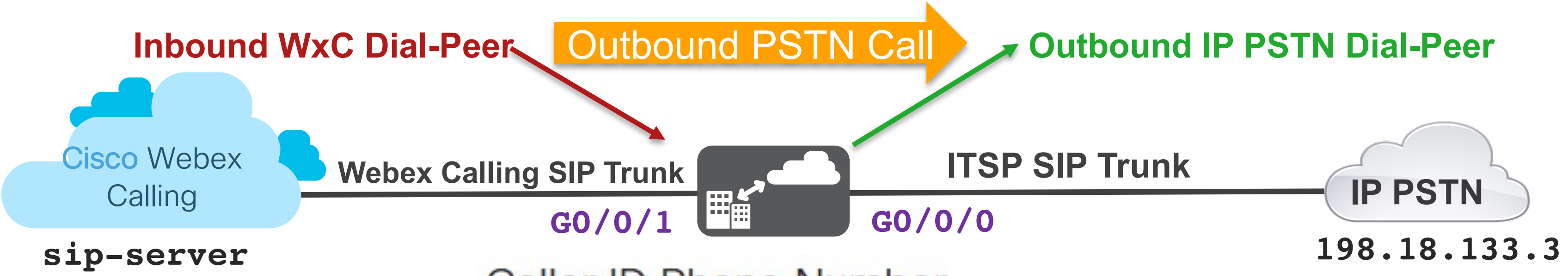
Caller ID and Local Gateway

- Webex Calling supports PAI, which must be configured on LGW and RPID has to be disabled on LGW as it is on by default
- LGW also has to be configured to transparently pass across privacy header values from incoming (Webex Calling) to the outgoing leg (ITSP/IP PBX)
- Above options configured under voice class tenant 200 which is applied to Webex calling facing dial-peer (dial-peer 200201)

```
voice class tenant 200
  no remote-party-id
  asserted-id pai
  privacy-policy passthru

dial-peer voice 200201 voip
  description Inbound/Outbound Webex Calling
  voice-class sip tenant 200
```

Outbound LGW call – Location Number (Main Number)

```
Received:
INVITE sip:+16784695555@198.18.1.226:5061;transport=tls;dtg=hussain3847_lgu SIP/2.0
Via:SIP/2.0/TLS 139.177.64.10:8934;branch=z9hG4bKBroadworksSSE.-64.100.12.6V26076-0-100-1980643282-1607401962594-
From:"Charles Holland"<sip:+16785559862@139.177.64.10;user=phone>;tag=1980643282-1607401962594-
P-Asserted-Identity:"Charles Holland"<sip:+19194745971@10.21.0.214;user=phone>
```



Caller ID Phone Number

- Direct Line: 9194745971 Ext 6021
- Location Number: +16785559862
- Number from User's location

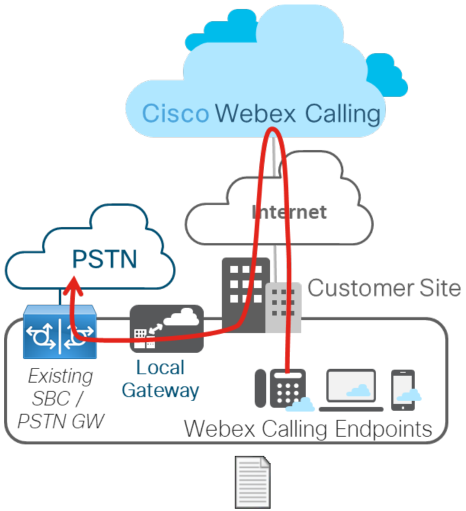
```
Sent:
INVITE sip:+16784695555@198.18.133.3:5060 SIP/2.0
Via: SIP/2.0/UDP 198.18.133.226:5060;branch=z9hG4bK2B7841DC4
From: "Charles Holland" <sip:+19194745971@198.18.133.226>;tag=65C7279C-23C7
P-Asserted-Identity: "Charles Holland" <sip:+19194745971@198.18.133.226>
```

Troubleshooting and Templates

Common LGW/CUBE Commands and Debugs

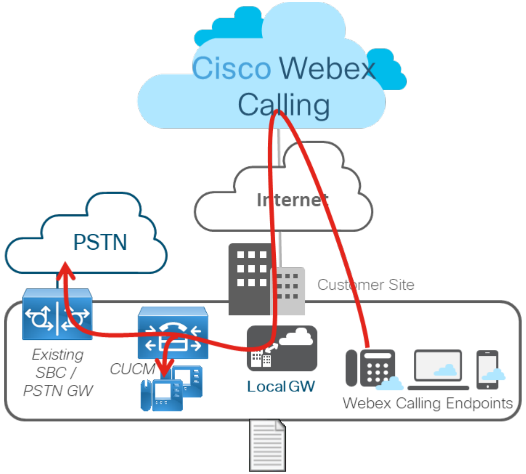
Command/Debug	Explanation
`show sip-ua register status`	display the registration status of a tenant
`show call active voice brief/compact`	display parameters of active calls
`show run`	View running CUBE configuration
`show log`	View CUBE debug logs, logged to a buffer
`clear log`	Clear CUBE debug logs
`debug ccsip non-call`	Enable non-call context trace (REGISTER, OPTIONS), origination from LGW/CUBE
`debug ccsip message`	Enable SIP messaging traces
`debug ccsip error`	Enable SIP error debugging trace
`debug ccsip transport`	Enable SIP Transport layer debugging traces
`debug voip ccapi inout`	Enable end to end IOS-XE VoIP call control (dial-peers) debugs

Local Gateway configuration templates



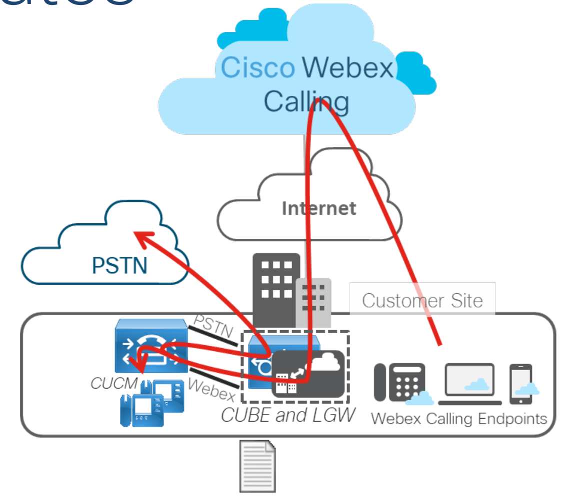
LGW Config Template - IP PSTN based - IOS-XE 16.12.4.txt

```
! LOCAL GATEWAY CONFIGURATION TEMPLATE
!-------------------------------------
! HUSSAIN SHOAIB ALI
! TECHNICAL MARKETING ENGINEER
! https://cisco.box.com/WebexCalling
! ASK-CUBE@EXTERNAL.CISCO.COM
! UPDATED : NOVEMBER 2020
!-------------------------------------

!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! This template is for building a LGW (on a vCUBE) deployment not involving CUCM!
! Verify LGW interfaces for dial-peer bind statements based on your network     !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! XXXXX needs to replaced with the correct parameters from the Control Hub      !
! Refer to the complete Local Gateway Slide deck                                !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! This configuration template can be used for both Customer site  !
! or partner hosted LGW deployments                               !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

!!!!!!!!!!!!!!!!!!!!!!!!! IOS-XE 16.12.4 !!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

!%%%%%%% BEGIN

configure terminal

!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! SELECT A MASTER PASSWORD FOR YOUR PLATFORM AND DO NOT USE  !
! Password123 AS SHOWN BELOW                                 !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

key config-key password-encrypt Password123
password encryption aes
!
crypto pki trustpoint dummyTp
revocation-check crl
exit
sip-ua
crypto signaling default trustpoint dummyTp cn-san-validate server
transport tcp tls v1.2
tcp-retry 1000
end


configure terminal
crypto pki trustpool import clean url http://www.cisco.com/security/pki/trs/ios_core.p7b
end
show crypto pki trustpool | include DigiCert


configure terminal
```



LGW Config Template - PSTN with UCM - IOS-XE 16.12.4.txt

```
! LOCAL GATEWAY CONFIGURATION TEMPLATE
!-------------------------------------
! HUSSAIN SHOAIB ALI
! TECHNICAL MARKETING ENGINEER
! https://cisco.box.com/WebexCalling
! ASK-CUBE@EXTERNAL.CISCO.COM
! UPDATED : NOVEMBER 2020
!-------------------------------------

!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! This template is for building a LGW (on a vCUBE) deployment with CUCM where the CUCM !
! has its own dedicated PSTN GW -----------------------------------------------------!!
! Verify LGW interfaces for dial-peer bind statements based on your network ----------!!
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! XXXXX needs to replaced with the correct parameters from the Control Hub      !
! Refer to the complete Local Gateway Slide deck                                !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

!!!!!!!!!!!!!!!!!!!!!!!!! IOS-XE 16.12.4 !!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

!%%%%%%% BEGIN

configure terminal

!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! SELECT A MASTER PASSWORD FOR YOUR PLATFORM AND DO NOT USE  !
! Password123 AS SHOWN BELOW                                 !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

key config-key password-encrypt Password123
password encryption aes
!
crypto pki trustpoint dummyTp
revocation-check crl
exit
sip-ua
crypto signaling default trustpoint dummyTp cn-san-validate server
transport tcp tls v1.2
tcp-retry 1000
end


configure terminal
crypto pki trustpool import clean url http://www.cisco.com/security/pki/trs/ios_core.p7b
end
show crypto pki trustpool | include DigiCert


configure terminal
voice service voip
  ip address trusted list
  !! Verify updated trust list from the Webex Calling Config Guide!!
  ipv4 85.119.56.128 255.255.255.192
```



LGW Config Template - Co-located PSTN CUBE and UCM - IOS-XE 16.12.4.txt

```
! LOCAL GATEWAY CONFIGURATION TEMPLATE
!-------------------------------------
! HUSSAIN SHOAIB ALI
! TECHNICAL MARKETING ENGINEER
! https://cisco.box.com/WebexCalling
! ASK-CUBE@EXTERNAL.CISCO.COM
! UPDATED : NOVEMBER 2020
!-------------------------------------

!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! This template is for building a LGW (on a vCUBE) deployment with CUCM where the PSTN !
! CUBE and Local Gateway are co-located on the same platform -------------------------!!
! Verify LGW interfaces for dial-peer bind statements based on your network ----------!!
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! XXXXX needs to replaced with the correct parameters from the Control Hub      !
! Refer to the complete Local Gateway Slide deck                                !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

!!!!!!!!!!!!!!!!!!!!!!!!! IOS-XE 16.12.4 !!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

!%%%%%%% BEGIN

configure terminal

!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! SELECT A MASTER PASSWORD FOR YOUR PLATFORM AND DO NOT USE  !
! Password123 AS SHOWN BELOW                                 !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

key config-key password-encrypt Password123
password encryption aes
!
crypto pki trustpoint dummyTp
revocation-check crl
exit
sip-ua
crypto signaling default trustpoint dummyTp cn-san-validate server
transport tcp tls v1.2
tcp-retry 1000
end


configure terminal
crypto pki trustpool import clean url http://www.cisco.com/security/pki/trs/ios_core.p7b
end
show crypto pki trustpool | include DigiCert


configure terminal
voice service voip
  ip address trusted list
  !! Verify updated trust list from the Webex Calling Config Guide!!
  ipv4 85.119.56.128 255.255.255.192
```

```
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! This template is for building a LGW (on a vCUBE) deployment not involving CUCM!
! Verify LGW interfaces for dial-peer bind statements based on your network     !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! XXXXX needs to replaced with the correct parameters from the Control Hub      !
! Refer to the complete Local Gateway Slide deck                                !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! This configuration template can be used for both Customer site  !
! or partner hosted LGW deployments                               !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

!!!!!!!!!!!!!!!!!!!!!!!!!! IOS-XE 16.12.4 !!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

!%%%%%%% BEGIN

configure terminal

!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! SELECT A MASTER PASSWORD FOR YOUR PLATFORM AND DO NOT USE  !
! Password123 AS SHOWN BELOW                                 !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

key config-key password-encrypt Password123
password encryption aes
!
crypto pki trustpoint dummyTp
revocation-check crl
exit
sip-ua
crypto signaling default trustpoint dummyTp cn-san-validate server
transport tcp tls v1.2
tcp-retry 1000
end


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

```
configure terminal
voice service voip
  ip address trusted list
  !! Verify updated trust list from the Webex Calling Config Guide!!
  ipv4 85.119.56.128 255.255.255.192
  ipv4 85.119.57.128 255.255.255.192
  ipv4 185.115.196.0 255.255.255.128
  ipv4 185.115.197.0 255.255.255.128
  ipv4 128.177.14.0 255.255.255.128
  ipv4 128.177.36.0 255.255.255.192
  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 135.84.172.0 255.255.255.192
  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 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 199.19.197.0 255.255.255.0
  ipv4 199.19.199.0 255.255.255.0
  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
    exit
   allow-connections sip to sip
  media statistics
  media bulk-stats
  no supplementary-service sip refer
  no supplementary-service sip handle-replaces
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  stun
    stun flowdata agent-id 1 boot-count 4
```

```
  stun
    stun flowdata agent-id 1 boot-count 4
    stun flowdata shared-secret 0 Password123$
  sip
    g729 annexb-all
    early-offer forced
    end


!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! XXXXX needs to replaced with the correct parameters from the Control Hub      !
! Refer to the complete Local Gateway Slide deck                                !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

configure terminal
voice class sip-profiles 200
  rule 9 request ANY sip-header SIP-Req-URI modify "sips:(.*)" "sip:\1"
  rule 10 request ANY sip-header To modify "<sips:(.*)" "<sip:\1"
  rule 11 request ANY sip-header From modify "<sips:" "<sip:\1"
  rule 12 request ANY sip-header Contact modify "<sips:(.*)>" "<sip:\1;transport=tls>"
  rule 13 response ANY sip-header To modify "<sips:(.*)" "<sip:\1"
  rule 14 response ANY sip-header From modify "<sips:(.*)" "<sip:\1"
  rule 15 response ANY sip-header Contact modify "<sips:(.*)" "<sip:\1"
  rule 20 request ANY sip-header From modify ">" ";otg=XXXXXX>"
  rule 30 request ANY sip-header P-Asserted-Identity modify "sips:(.*)" "sip:\1"


voice class codec 99
  codec preference 1 g711ulaw
  codec preference 2 g711alaw
  exit

voice class srtp-crypto 200
  crypto 1 AES_CM_128_HMAC_SHA1_80
  exit

voice class stun-usage 200
  stun usage firewall-traversal flowdata
  exit
```

```
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! XXXXX needs to replaced with the correct parameters from the Control Hub      !
! Refer to the complete Local Gateway Slide deck                                !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

voice class tenant 200
  registrar dns:XXXXXX scheme sips expires 240 refresh-ratio 50 tcp tls
  credentials number XXXXXX username XXXXXX password 0 XXXXXX realm BroadWorks
  authentication username XXXXXX password 0 XXXXXX realm BroadWorks

  authentication username XXXXXX password 0 XXXXXX realm XXXXXX
  no remote-party-id
  sip-server dns:XXXXXX
  connection-reuse
  srtp-crypto 200
  session transport tcp tls
  url sips
  error-passthru
  asserted-id pai
  bind control source-interface GigabitEthernet1
  bind media source-interface GigabitEthernet1
  no pass-thru content custom-sdp
  sip-profiles 200
  outbound-proxy dns:XXXXXX
  privacy-policy passthru


voice class tenant 100
  session transport udp
  url sip
  error-passthru
  bind control source-interface GigabitEthernet2
  bind media source-interface GigabitEthernet2
  no pass-thru content custom-sdp

voice class tenant 300
  bind control source-interface GigabitEthernet2
  bind media source-interface GigabitEthernet2
  no pass-thru content custom-sdp
```

```
!%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
!! REPLACE A.B.C.D with ITSP's IP Address %
!%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
voice class uri 100 sip
 host ipv4:A.B.C.D

!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
! XXXXX needs to replaced with the correct parameters from the Control Hub      !
! Refer to the complete Local Gateway Slide deck                                !
!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!

voice class uri 200 sip
 pattern dtg=XXXXXX.lgu


dial-peer voice 101 voip
 description Outgoing dial-peer to IP PSTN
 destination-pattern BAD.BAD
 session protocol sipv2
 !%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
 !! REPLACE A.B.C.D with ITSP's IP Address %
 !%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
 session target ipv4:A.B.C.D
 voice-class codec 99
 voice-class sip tenant 100
 dtmf-relay rtp-nte
 no vad


dial-peer voice 200201 voip
 description Inbound/Outbound Webex Calling
 max-conn 150
 destination-pattern BAD.BAD
 session protocol sipv2
 session target sip-server
 incoming uri request 200
 voice-class codec 99
 voice-class stun-usage 200
 no voice-class sip localhost
 voice-class sip tenant 200
 dtmf-relay rtp-nte
 srtp
 no vad
```

```
voice class dpg 100
 description Incoming WxC(DP200201) to IP PSTN(DP101)
 dial-peer 101 preference 1


voice class dpg 200
 description Incoming IP PSTN(DP100) to WxC(DP200201)
 dial-peer 200201 preference 1


dial-peer voice 100 voip
 description Incoming dial-peer from IP PSTN
 session protocol sipv2
 destination dpg 200
 incoming uri via 100
 voice-class codec 99
 voice-class sip tenant 300
 dtmf-relay rtp-nte
 no vad

dial-peer voice 200201 voip
 destination dpg 100


end


copy run start

!!%%%%% END-TEMPLATE
```

Resources

Resources

- CUBE Box – https://cisco.box.com/CUBE-Enterprise (request access via email)
- Webex Calling – https://cisco.box.com/WebexCalling (request access via email)
  - Email ASK-CUBE@EXTERNAL.CISCO.COM with your Box.com account id (email) for access to the Box.com links above. Free Box.com account is fine as well
- Webex Calling Deployment Guide
- Local Gateway Configuration Guide
- Collaboration Transitions
- Webex Calling PA
- Dcloud Labs
  - Cisco Webex Calling v3
  - Transitioning from Unified CM to Webex Calling

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