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The bridge to possible

Local Gateway Design and Deployment for Webex Calling

Hussain Ali, CCIE# 38068 (Voice, Collaboration)
Technical Marketing Engineer
https://www.linkedin.com/in/hussaincube

BRKCOL-2048

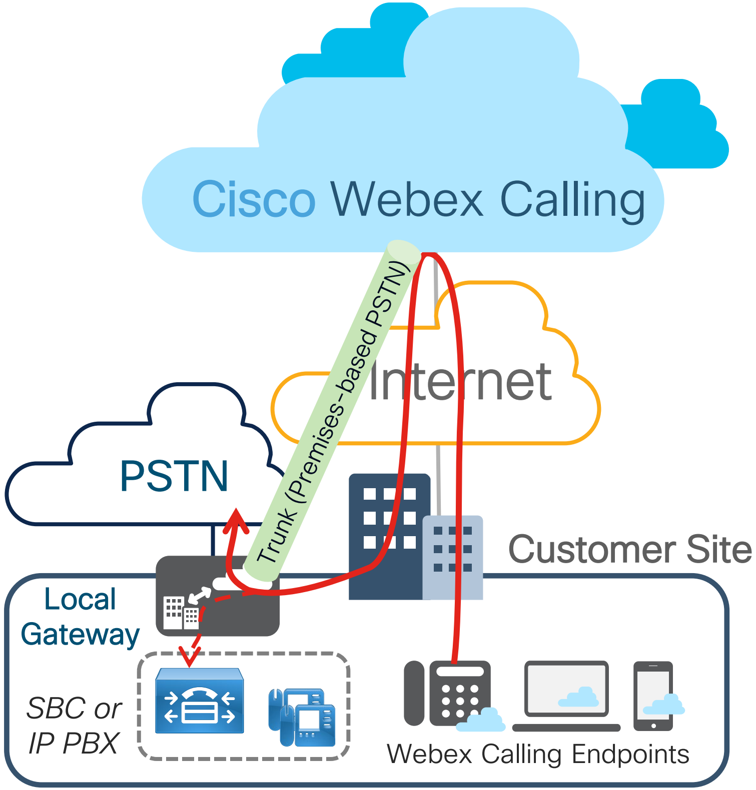
Agenda

- Local Gateway overview and sizing
- Key configuration updates required
- Caller ID handling
- Templates, Troubleshooting, and Resources

Local Gateway
Overview and Sizing

Webex Calling Trunk - Local Gateway

(Premises-based PSTN) Deployment

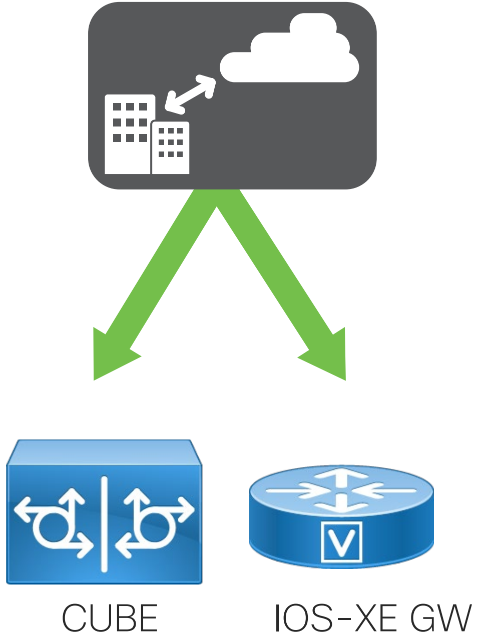


- Provides connectivity to a customer-owned premises-based PSTN service
- May also provide connectivity to an on-premises IP PBX or dedicated SBC/PSTN GW
- Enables on-prem to Webex Calling transition
- **Endpoint registration is NOT proxied through Local Gateway, unlike CUBE Lineside.** Endpoints directly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

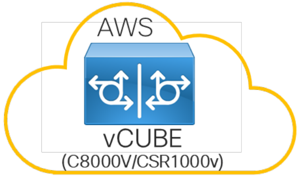
Local Gateway

Platform Support

Local Gateway (LGW)



- **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 17.3.2)
  - vCUBE in AWS



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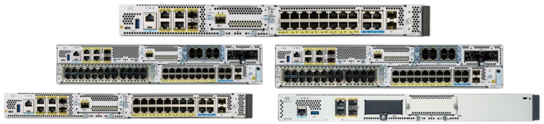
  - Catalyst 8200/8300 series



  - CSR 1000v (vCUBE) (IOS XE 16.9(6) and 16.12.4 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 16.12.4 or later)

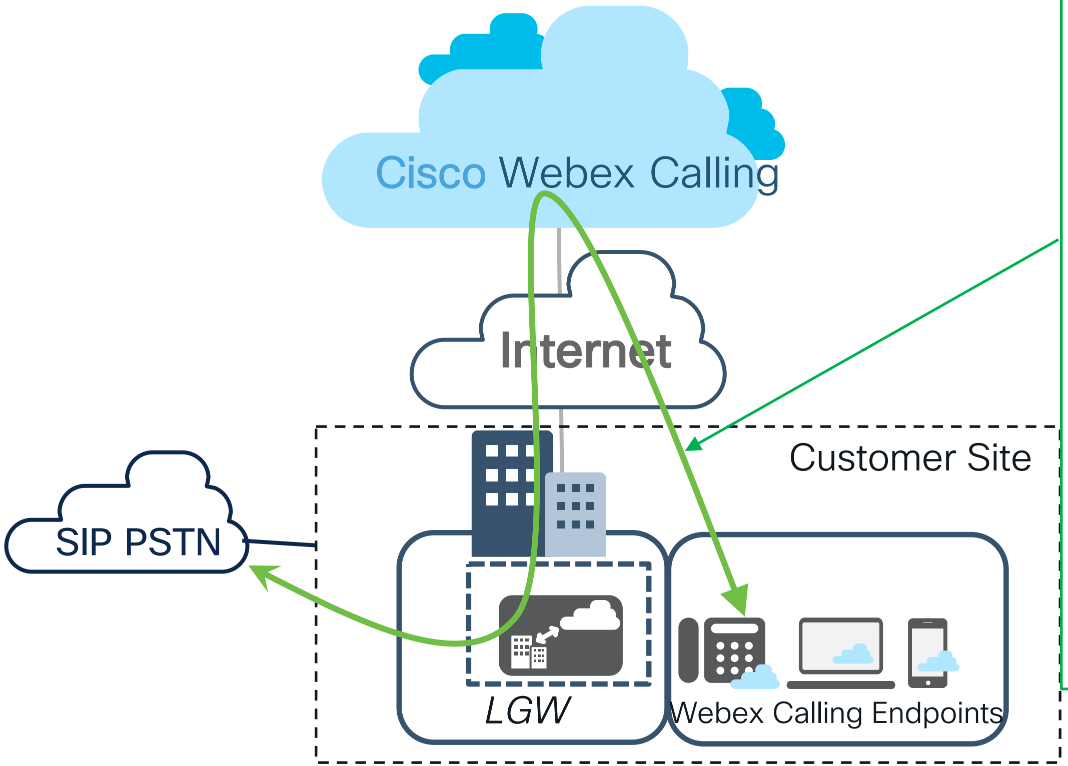
CUBE Software Release Mapping

CUBE Version	Initial IOS-XE Release for this CUBE version and Release date		Subsequent IOS-XE Release for this CUBE version
12.0.0	16.6.1	July 2017	16.6.2 – 16.6.8
12.0.0	16.7.1	Nov 2017	16.7.2 – 16.7.3
12.1.0	16.8.1	March 2018	16.8.2 – 16.8.3
12.1.0	16.9.1	July 2018	16.9.2 – 16.9.5 – 16.9.6*
12.5.0	16.10.1a	Nov 2018	16.10.2 – 16.10.3
12.6.0	16.11.1a	March 2019	16.11.1b
12.7.0	16.12.1c	July 2019	16.12.1a – 16.12.4 – 16.12.5
12.7.1	17.1.1	Nov 2019	–
12.8.0	17.2.1r	March 2020	17.2.2
14.0	17.3.1a	July 2020	–
14.1	17.3.2*	Oct 2020	–
14.2	17.4.1a	Nov 2020	–
14.3	17.5.1	March 2021	–
TBD	17.6.1	July 2021	

Smart Licensing Mandatory

Local Gateway – CUBE Trunk Licenses

Standalone Local Gateway w/SIP PSTN termination



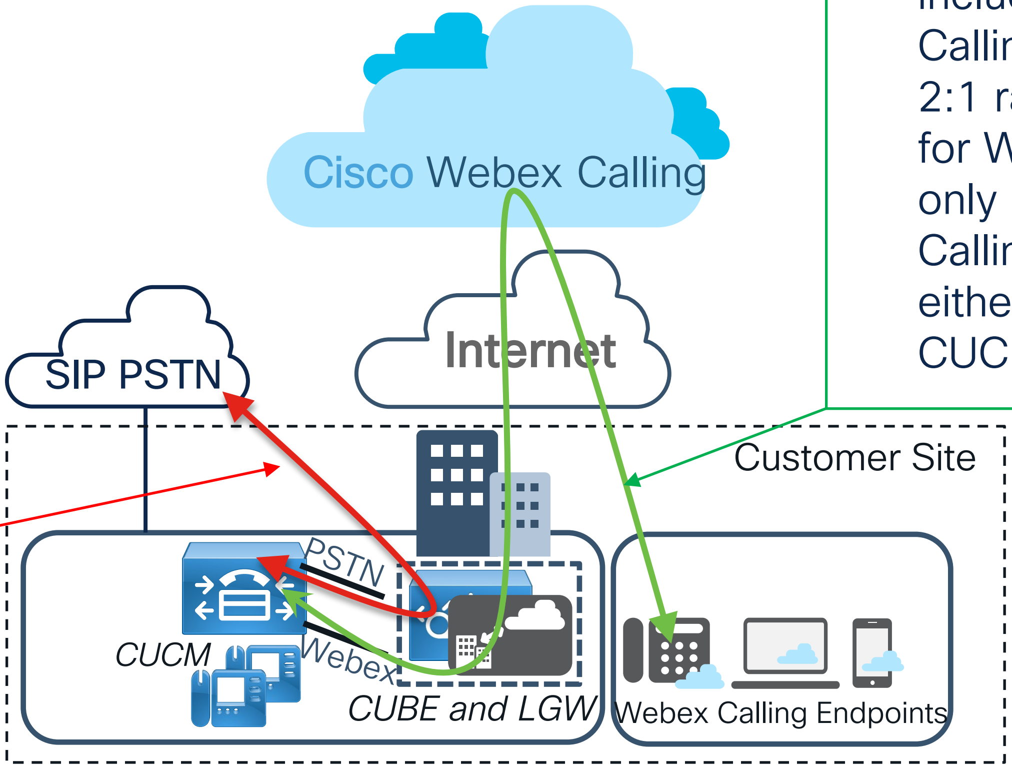
- CUBE Trunk licenses (STD or RED) are included in Webex Calling Flex License at 2:1 ratio (users:trunk) for Webex Calling calls (1 leg to Webex Calling and the other to either the ITSP or CUCM)
- A case needs to be raised to request CUBE licenses.
- Case not required for **A-FLEX-3** customers as **A-FLEX-LGW-CUBE** is automatically deposited into their accounts

https://www.cisco.com/c/en/us/products/collateral/unified-communications/unified-border-element/guide-c07-742037.html#LocalGatewayforCiscoWebexCallingdeploymentoptions

Local Gateway – CUBE Trunk Licenses

Co-located PSTN CUBE and LGW

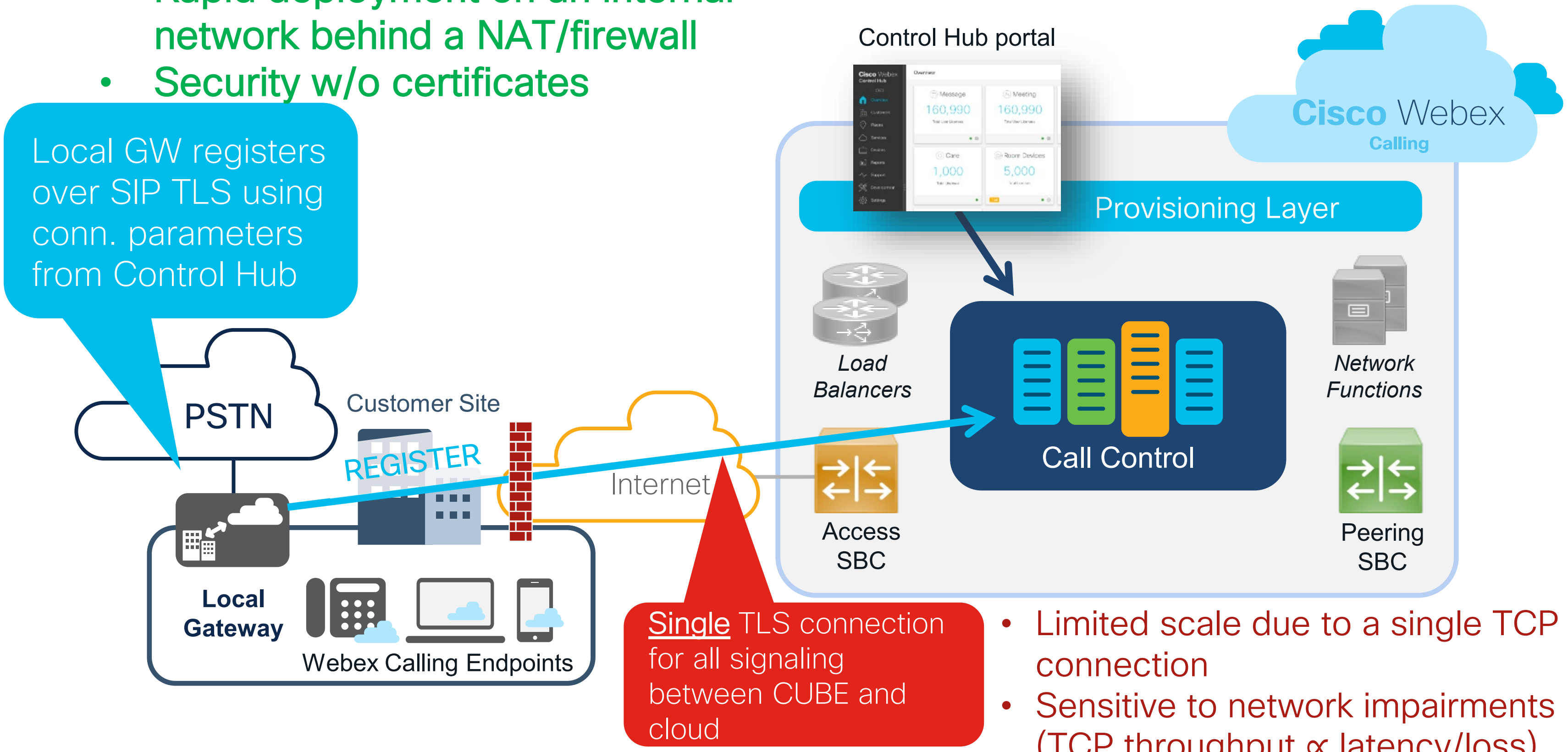
CUBE Trunk licenses (STD or RED) for this leg are **not** included in Webex Calling Flex



- CUBE Trunk licenses (STD or RED) are included in Webex Calling Flex License at 2:1 ratio (users:trunk) for Webex Calling calls only (1 leg to Webex Calling and the other to either the ITSP or CUCM)

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)

Local Gateway

Firewall and NAT traversal – IP Addresses and Ports

Updated IP list for Webex Calling Access SBC

<table>
<tr><th>Purpose</th><th>Source IP</th><th>Source ports</th><th>Protocol</th><th>Global Destination IP</th><th>Destination ports</th></tr>
<tr><td>SIP signaling</td><td>LGW Cisco Webex Calling facing interface</td><td>8000-65535</td><td>TLS TCP</td><td rowspan="2">85.119.56.128/26
85.119.57.128/26
135.84.169.0/25
135.84.170.0/25
135.84.171.0/25
135.84.172.0/25
135.84.173.0/25
135.84.174.0/25
185.115.196.0/25
185.115.197.0/25
199.19.197.0/24
199.19.199.0/24
199.59.64.0/25
199.59.65.0/25
199.59.66.0/25
199.59.67.0/25
199.59.70.0/25
199.59.71.0/25
128.177.14.0/25
128.177.36.0/26
139.177.64.0/24
139.177.65.0/24
139.177.66.0/24
139.177.67.0/24
139.177.68.0/24
139.177.69.0/24
139.177.70.0/24
139.177.71.0/24
139.177.72.0/24
139.177.73.0/24</td><td>8934</td></tr>
<tr><td>sRTP media</td><td>LGW Cisco Webex Calling facing interface</td><td>8000-48000</td><td>UDP</td><td>19560-65535</td></tr>
</table>

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 150 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

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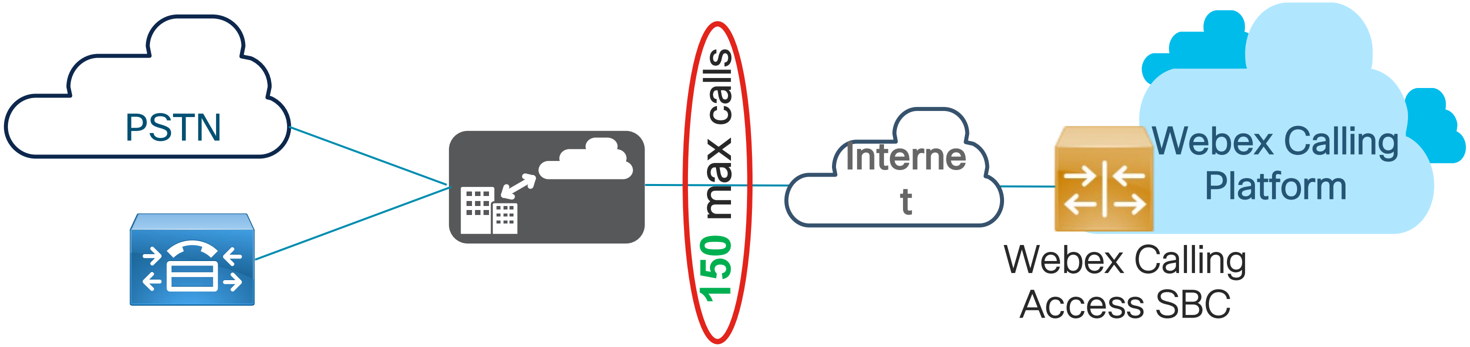


Note: *Contact your Cisco account team if you need more than 150 concurrent calls **per** LGW*

Agenda

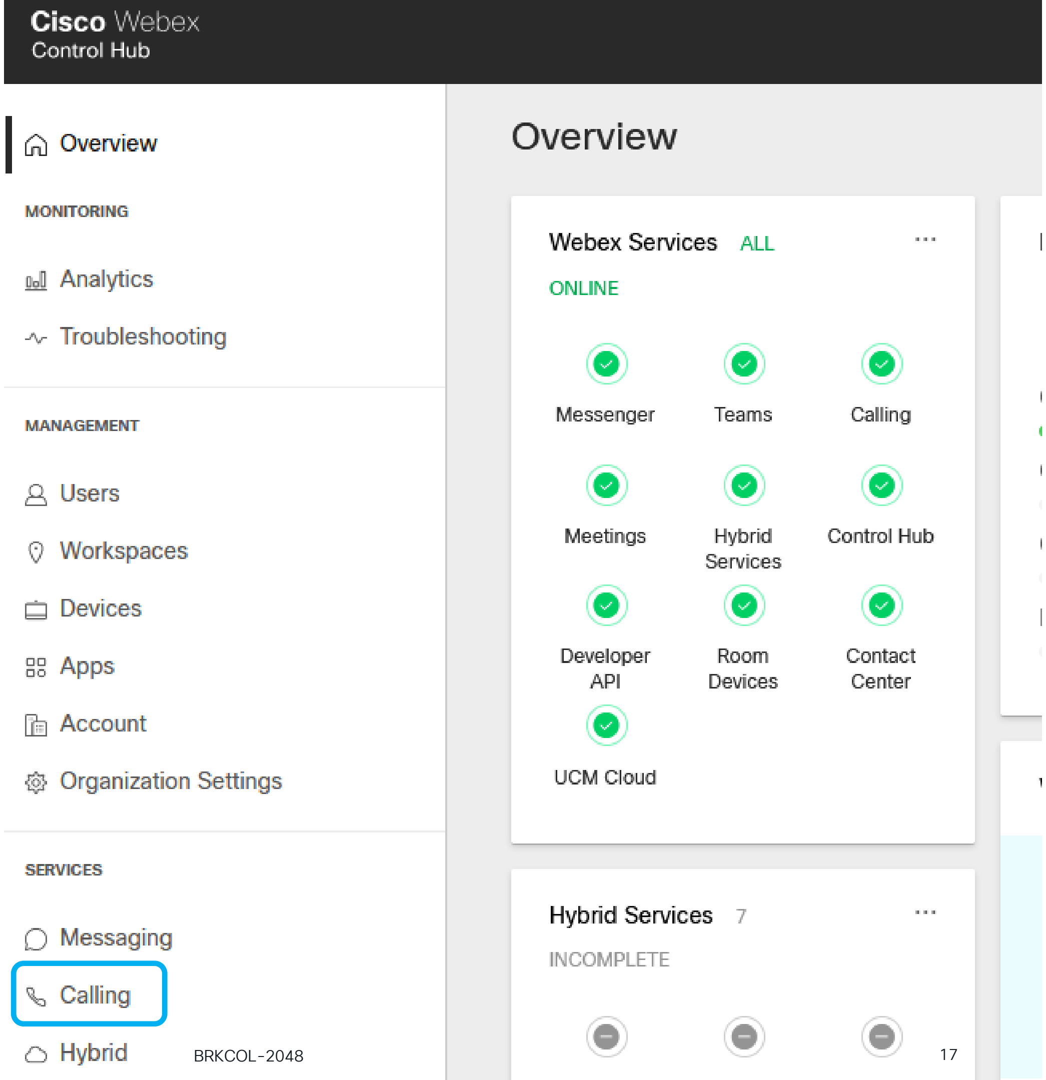
- Local Gateway overview and sizing
- Key configuration updates required
- Caller ID handling
- Templates, Troubleshooting, and Resources

Key Configuration Updates Required

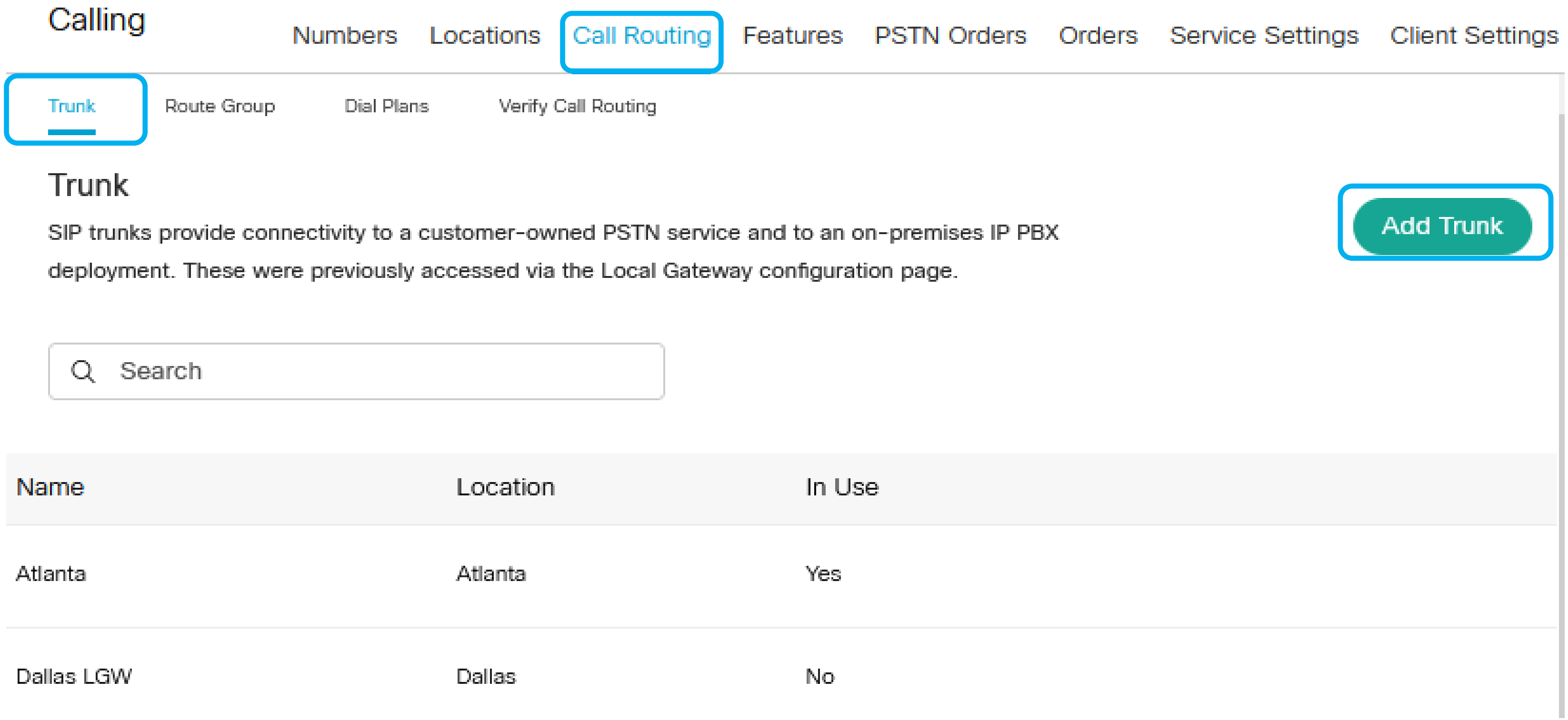
Update

Onboarding Process Webex Calling Trunk

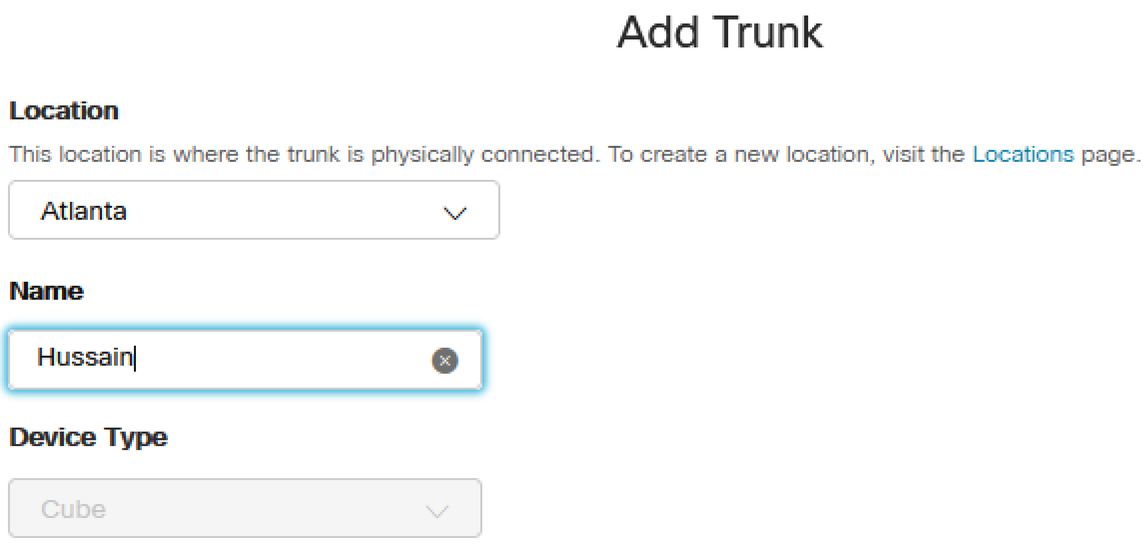
1a. Log in to customer portal and navigate to Services – Click Calling



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

Line/Port

Hussain6346_LGU@40462196.cisco-bcld.com

Trunk Group OTG/DTG

hussain2572_lgu

Authentication Information

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

Outbound Proxy Address

la01.sipconnect-us10.cisco-bcld.com

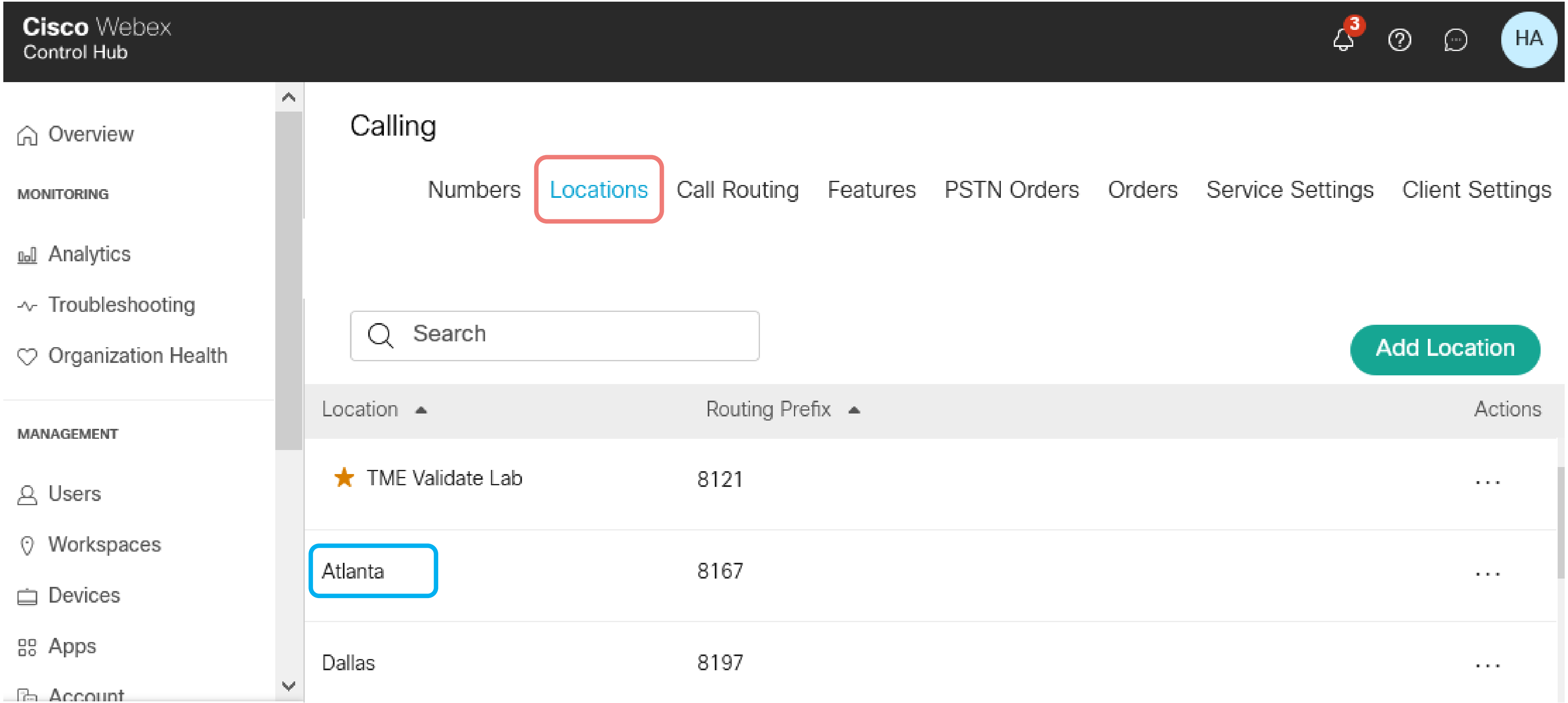
Username: Hussain2572_LGU

Password: meX7]~)VmF

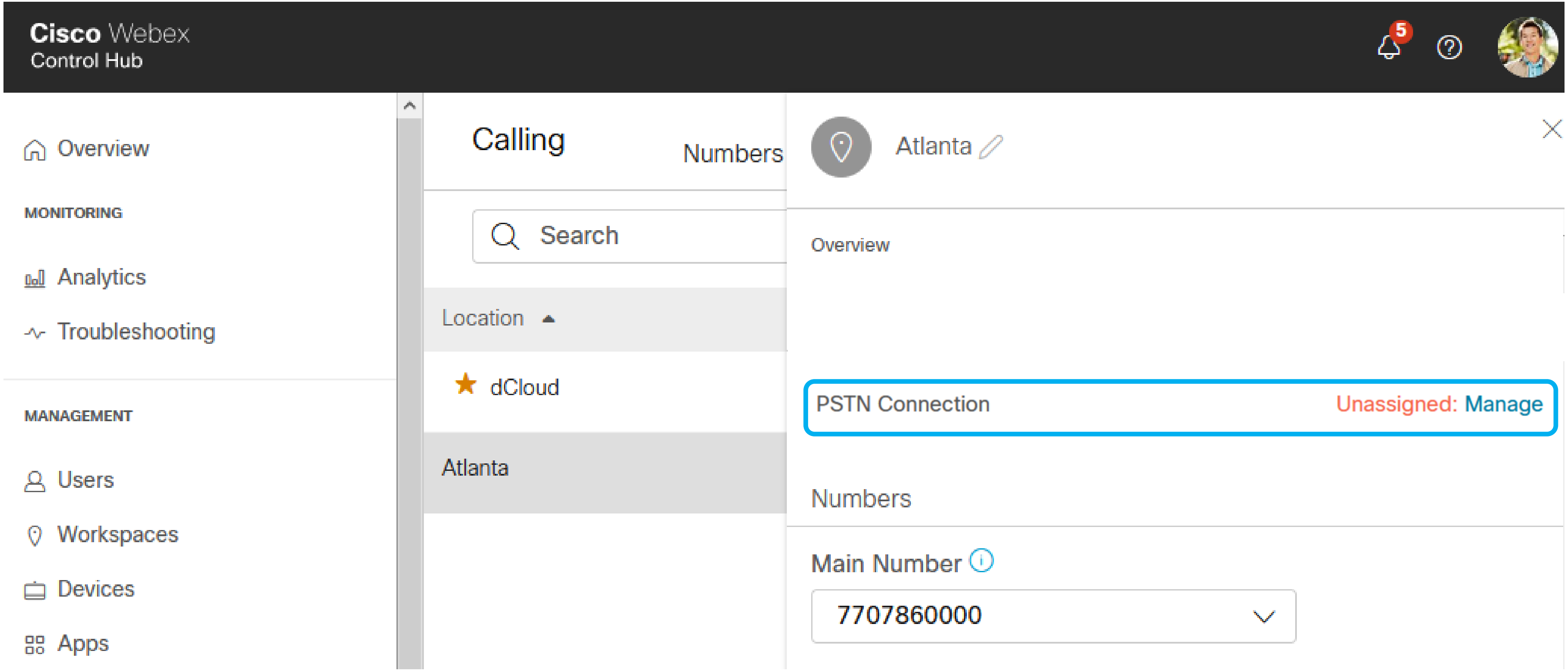
Registrar Domain

40462196.cisco-bcld.com

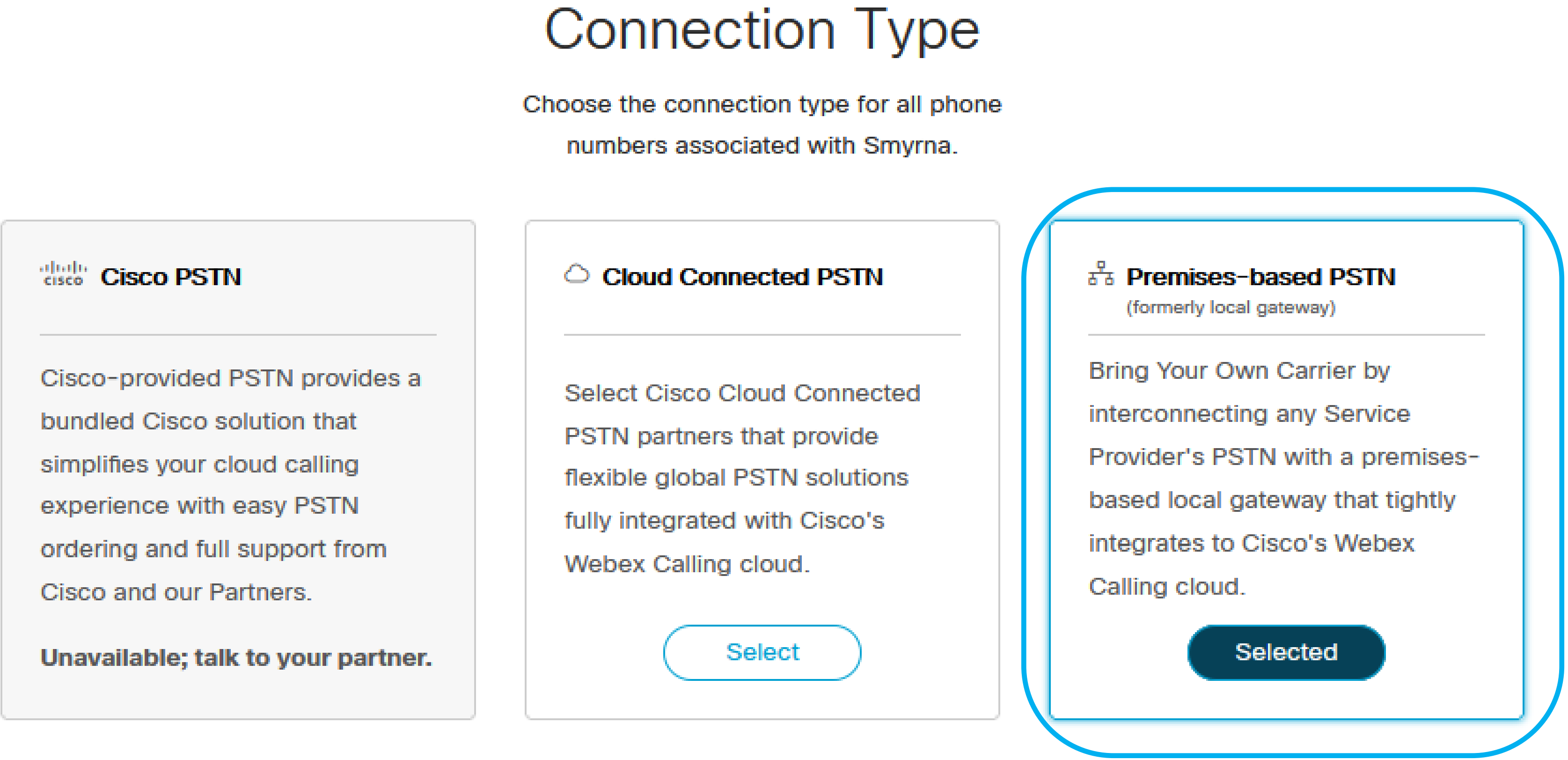
1f. Navigate to Locations under Calling and select the desired location



1g. Click on Unassigned under PSTN Connection



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



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

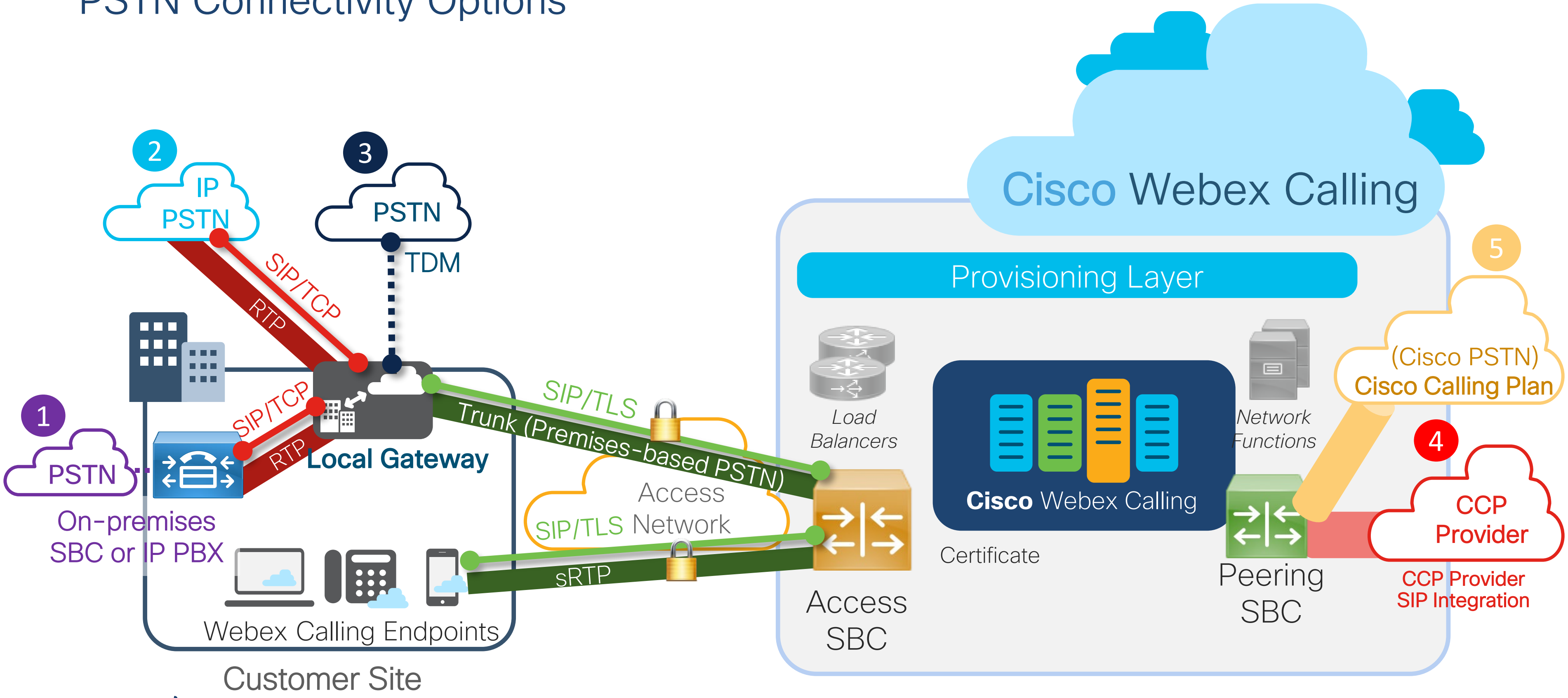
This trunk is located in **Atlanta**.

* 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

Webex Calling

PSTN Connectivity Options



Updating Outbound Proxy

Control Hub Trunk Info Connection Parameters → LGW CLI Config

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

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

Trunk Info

Status

● unknown

Line/Port

Hussain6346_LGU@40462196.

Trunk Group OTG/DTG

hussain2572_lgu

Authentication Information

Record the username and pas

lose this information, you nee

username and reset the passw

Outbound Proxy Address

la01.sipconnect-us10.cisco-bcld.com

Username: Hussain2572_LGU

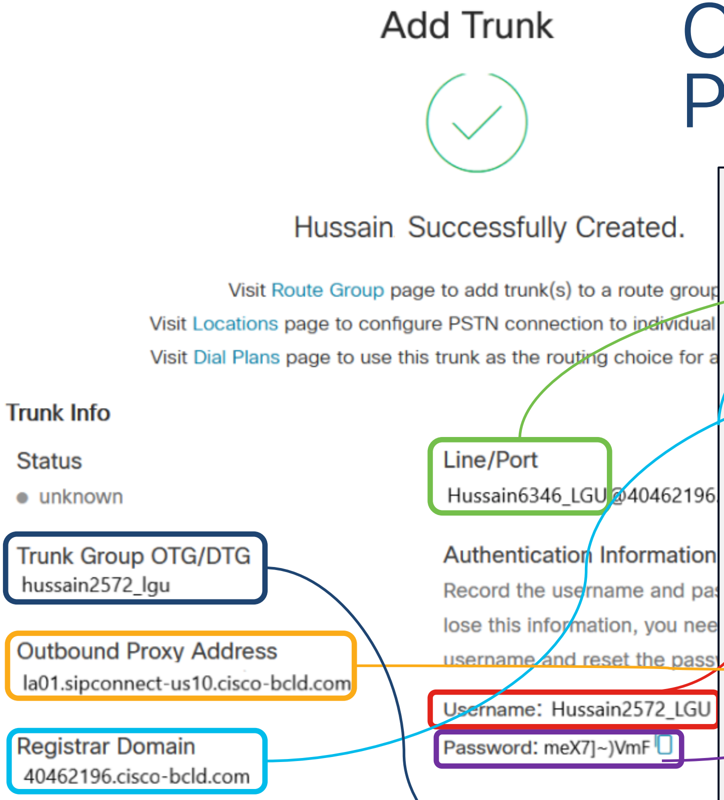
Password: meX7]~)VmF

Registrar Domain

40462196.cisco-bcld.com

```
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:"
```

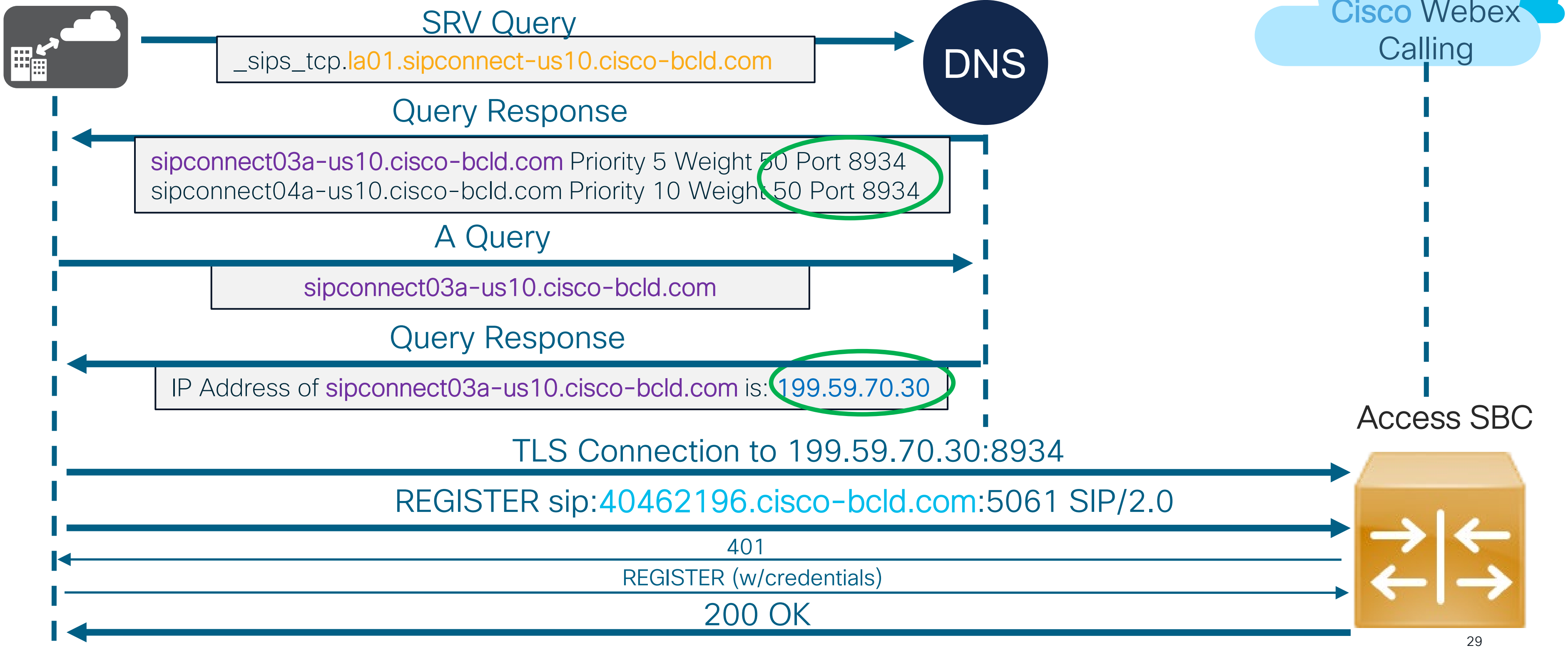
Control Hub Trunk Info Connection Parameters → LGW CLI Config




```
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 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 outbound-proxy`

```
voice class tenant 200
  no outbound-proxy
```

5. Update with the new `outbound-proxy` within `voice class tenant 200`

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

6. Save the local gateway configuration using the `write` command, and reload
7. Validate the registration for OTG is successful with `show sip-ua register status` (may take a few minutes)

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`

Coming Soon

Utilizing Media Path Optimization

Configure STUN ICE-Lite to utilize Media Path Optimization

```
voice class stun-usage 200
 stun usage ice lite
```

BRKCOL-2046

- stun usage ice lite is required for LGW call flows utilizing media path optimization.
- IOS-XE 16.12.5 required
- voice class stun-usage 200 already applied to Webex Calling facing dial-peer
- To utilize media path Optimization, the LGW's Webex Calling facing interface must have a direct network path to and from the Webex Calling endpoints.
- If there is no direct network path between the endpoints and the LGW's Webex Calling facing interface, or if the endpoints are in a different location, then that LGW interface must have a public IP address to benefit from media path optimization
- Ensure the LGW configuration has the latest Outbound Proxy from the Trunk Info page within the Control Hub

Update

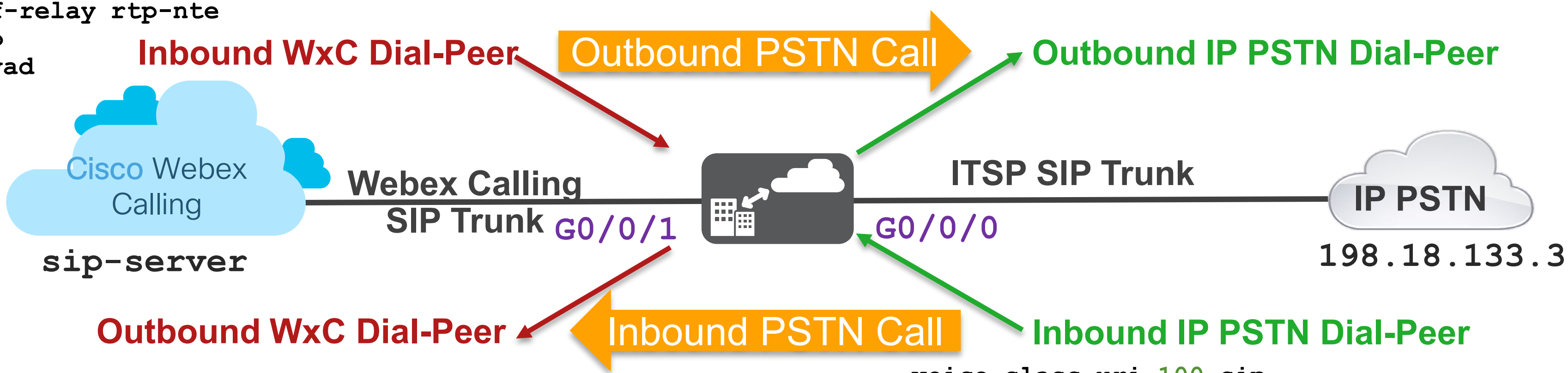
Single Dial-peer facing Webex Calling

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
```

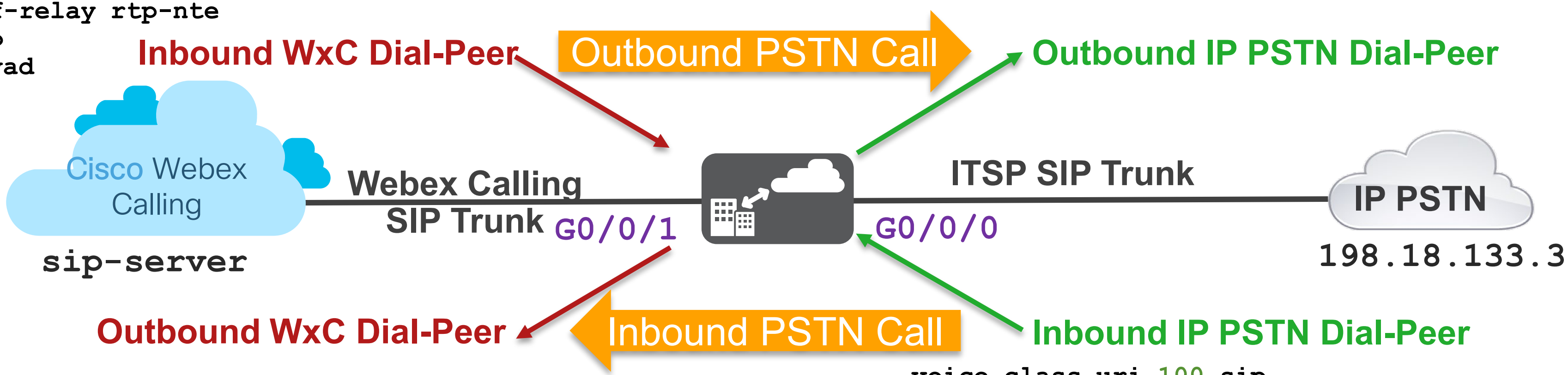



```
dial-peer voice 201 voip
  description Outgoing dial-peer to Webex Calling
  destination-pattern BAD.BAD
  session target sip-server
  voice-class codec 99
  voice-class stun-usage 200
  no voice-class sip localhost
  voice-class sip tenant 200
  dtmf-relay rtp-nte
  srtp
  no vad
```

```
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
```



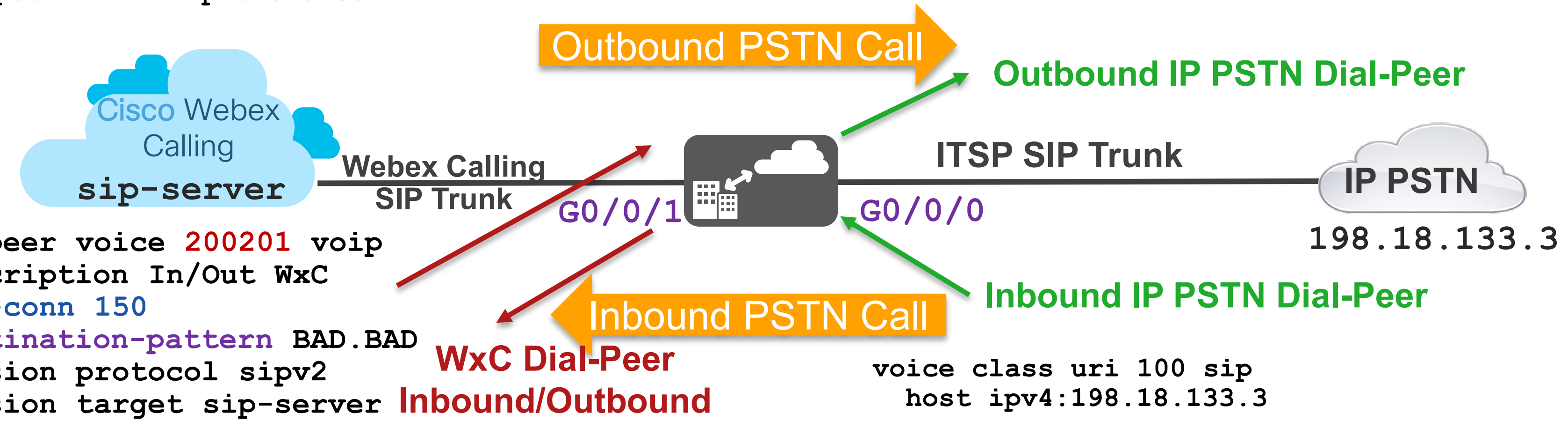

```
dial-peer voice 201 voip
  description Outgoing dial-peer to Webex Calling
  destination-pattern BAD.BAD
  session target sip-server
  voice-class codec 99
  voice-class stun-usage 200
  no voice-class sip localhost
  voice-class sip tenant 200
  dtmf-relay rtp-nte
  srtp
  no vad
```

```
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 150
  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 dial-peer steps for existing LGW deployments

1. Update Dial-peer Group (DPG) 100's description

```
voice class dpg 100
 description Incoming WxC(DP200201) to IP PSTN(DP101)
```

2. Create dial-peer 200201 to serve as a single Webex Calling facing dial-peer

```
dial-peer voice 200201 voip
  description In/Out WxC
  max-conn 150
  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
```

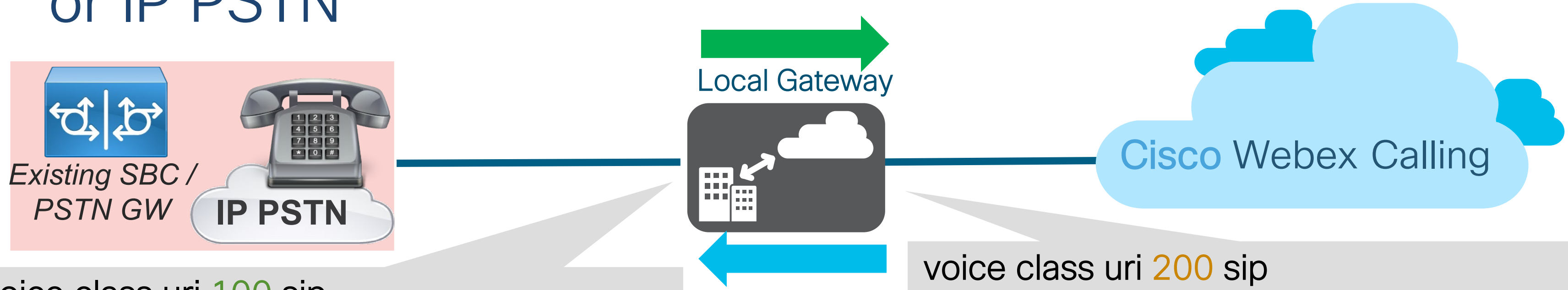
3. Update Dial-peer Group (DPG) 200's description, remove dial-peer 201 and add dial-peer 200201 as preference 1

```
voice class dpg 200
 description Incoming IP PSTN(DP100) to WxC(DP200201)
 no dial-peer 201 preference 1
 dial-peer 200201 preference 1
```

4. Delete dial-peers 200 and 201

```
no dial-peer voice 200
no dial-peer voice 201
```

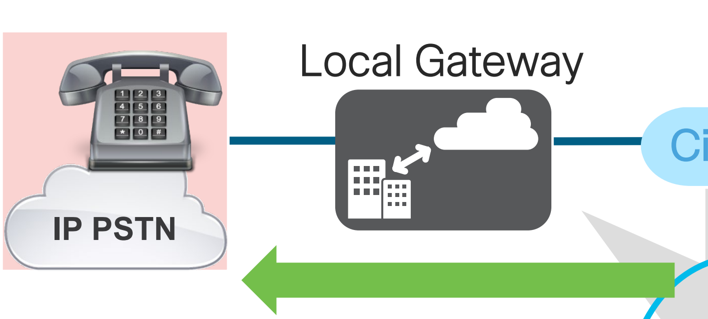

Local Gateway call routing to dedicated PSTN GW/SBC or IP PSTN




```
voice class uri 100 sip
  host <pstn ip address>
  ! Or existing SBC / PSTN GW

dial-peer voice 100 voip
  description Incoming dial-peer from IP PSTN
  incoming uri via 100
  destination dpg 200

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 target ipv4: <pstn ip address>
```

```
voice class uri 200 sip
  pattern dtg=hussain2572.lgu
  ! pattern uniquely identifies a Local gateway site within an
  ! Enterprise Trunk Group OTG/DTG from Control Hub

dial-peer voice 200201 voip
  description Inbound/Outbound Webex Calling
  incoming uri request 200
  destination dpg 100

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

dial-peer voice 200201 voip
  description Inbound/Outbound Webex Calling
  destination-pattern BAD.BAD
  session target sip-server
```

Local Gateway call routing based on Trunk Group ID




```
voice class uri 200 sip
 pattern dtg=hussain2572.lgu
 ! pattern uniquely identifies a Local gateway site
 ! within an Enterprise, Trunk Group OTG/DTG from
 ! Control Hub

dial-peer voice 200201 voip
 description Webex Calling
 incoming uri request 200
 destination dpg 100
```

Note: LGW today doesn't support underscore "_" in the match pattern. As a workaround we use dot "." (match any) to match the underscore "_"

INVITE Received by Local Gateway from Webex Calling

Received:
INVITE sip:+16785551234@198.18.1.226:5061;transport=tls;dtg=hussain2572_lgu SIP/2.0
Via: SIP/2.0/TLS 199.59.70.30:8934;branch=z9hG4bK2hokad30fg14d0358060.1

TCP Retry count, Timers, and Voice class codec

- Set tcp-retry count to 1000 (5 msec multiples = 5 seconds) (default 1 second)

- [IOS-XE 17.3.2+] Set timers connection establish tls *<wait-timer in sec>*. Range is between 5 and 20 seconds. Default is 20 sec. (**LGW takes 20 secs to detect the TLS connection failure before it attempts to establish a connection to the next available Webex Calling Access SBC. This CLI allows the admin to change the value to accommodate network conditions**)

sipconnect03a-us10.cisco-bcld.com Priority 5 Weight 50 Port 8934
sipconnect04a-us10.cisco-bcld.com Priority 10 Weight 50 Port 8934

```
LocalGateway(config)#sip-ua
LocalGateway(config-sip-ua)# tcp-retry 1000
LocalGateway(config-sip-ua)# timers connection establish tls <5-20>
```

- Update voice class codec to include only the codecs required

```
LocalGateway(config)# voice class codec 99
LocalGateway(config-class)# codec preference 1 g711ulaw
LocalGateway(config-class)# codec preference 2 g711alaw
```

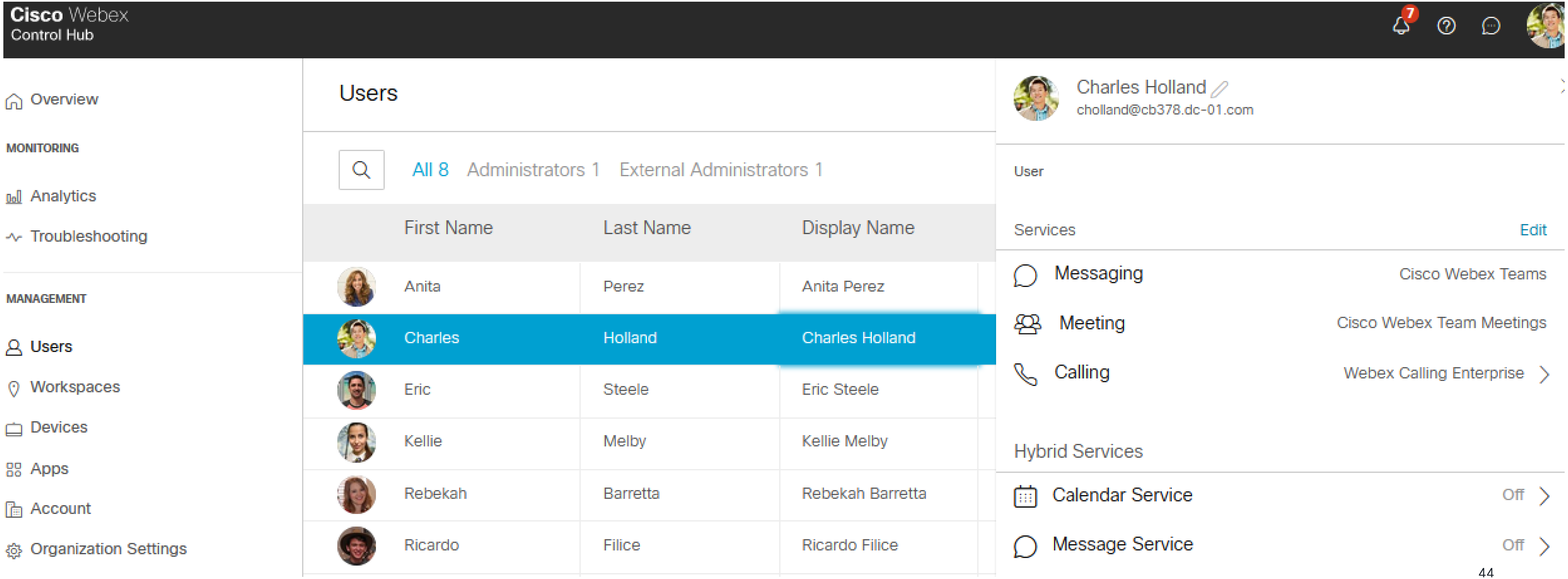
Agenda

- Local Gateway overview and sizing
- Key configuration updates required
- Caller ID handling
- Templates, Troubleshooting, and Resources

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**

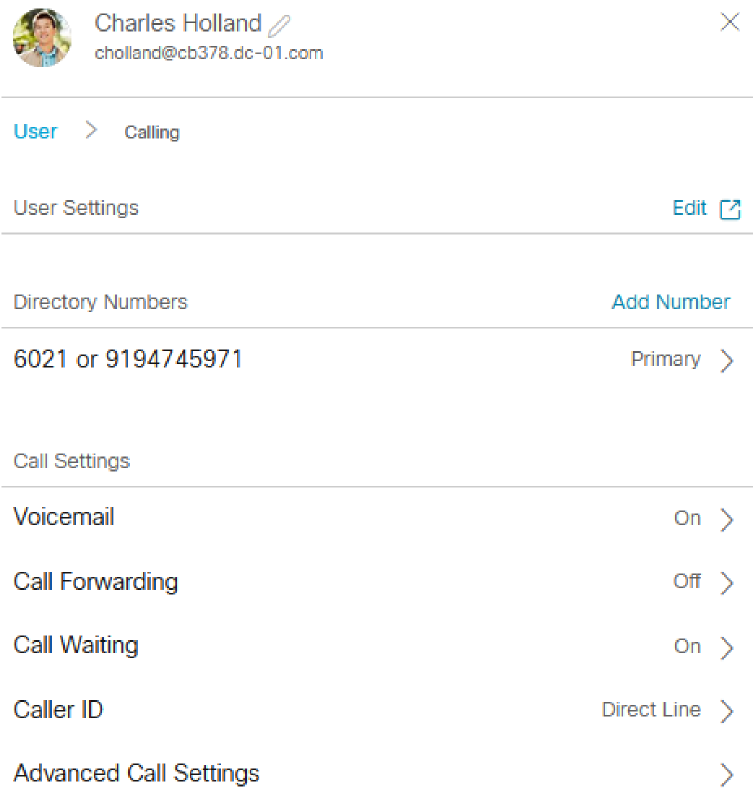


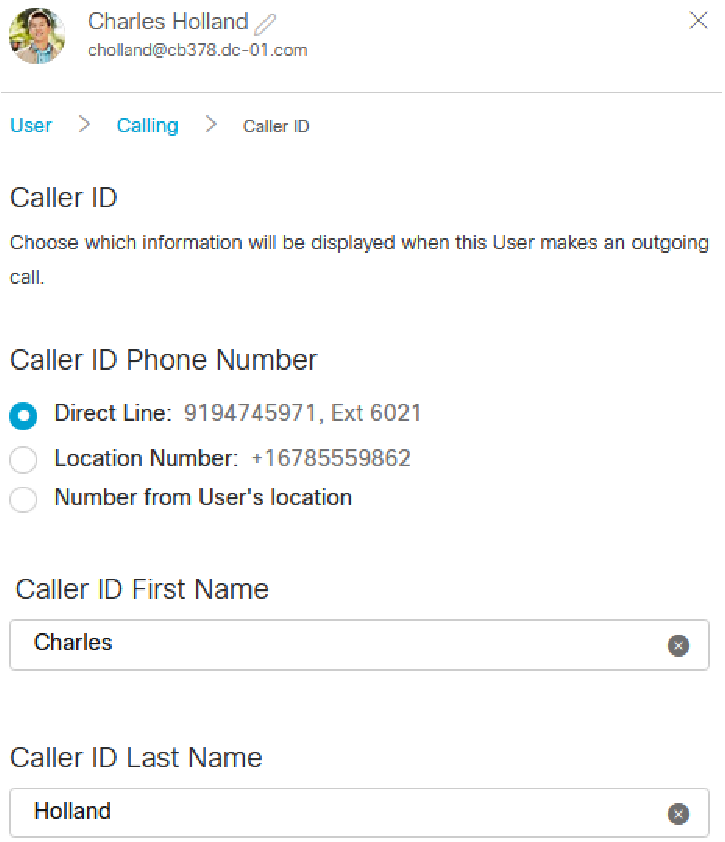
Caller ID settings

- 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
  - **Note:** Special characters are supported for the user's first and last name, but they are not supported in the Caller ID names. Unsupported special characters are removed when the Caller ID first and last names are generated.

Caller ID settings

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





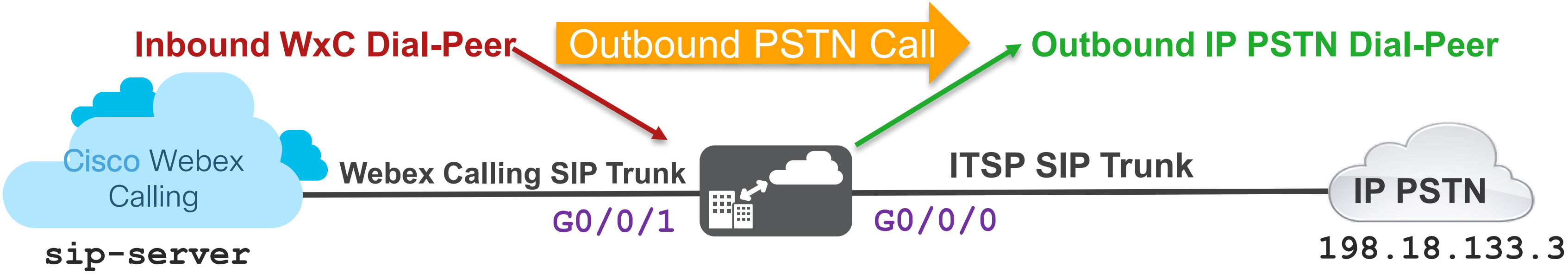
Caller ID and Local Gateway

- 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 – Direct Line

```
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:+19194745971@139.177.64.10;user=phone>;tag=1980643282-1607401962594-
P-Asserted-Identity:"Charles Holland"<sip:+19194745971@10.21.0.214;user=phone>
```



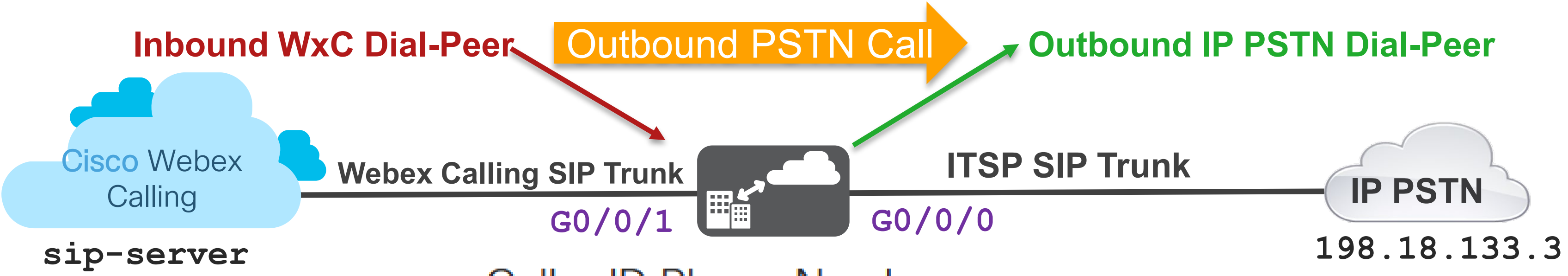
Caller ID Phone Number

- 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>
```

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>
```



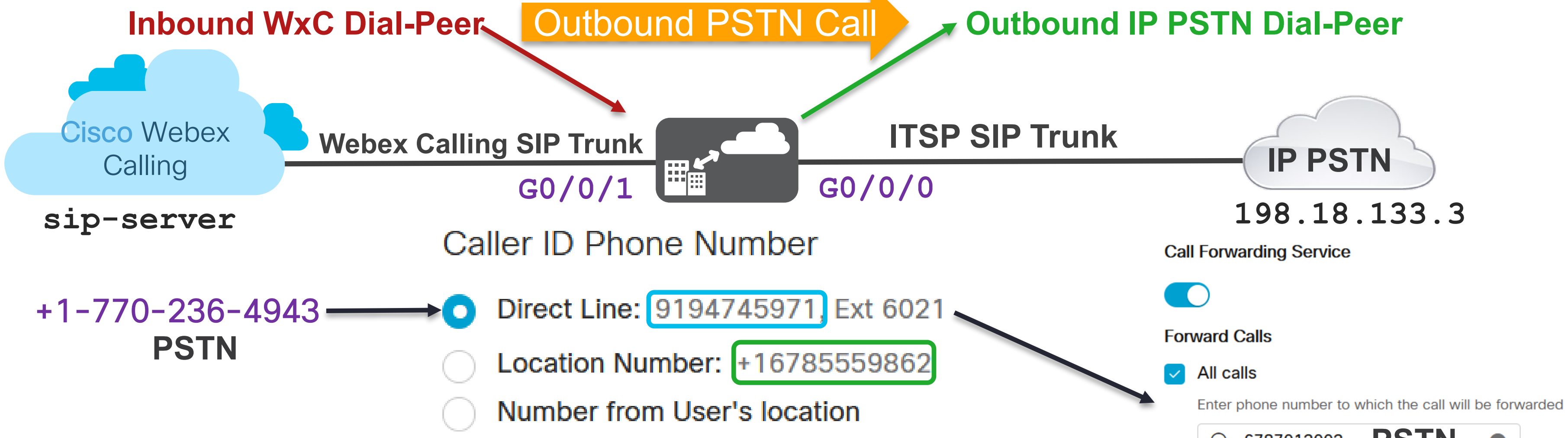
Caller ID Phone Number

- ○ 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>
```

Outbound LGW forwarded (PSTN to MPP to PSTN) call – Direct Line

```
Received:
INVITE sip:+16787013003@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.-64.100.12.6V26076-0-100-1666408278-1607559617941-
From:<sip:+17702364943@139.177.64.10;user=phone>;tag=1666408278-1607559617941-
P-Asserted-Identity:"LG Pilot User"<sip:+16785559862@10.21.0.214;user=phone>
Diversion:"Charles Holland"<sip:+19194745971@10.21.0.214;user=phone>;reason=unconditional;privacy=off;counter=1
```




```
Sent:
INVITE sip:+16787013003@198.18.133.3:5060 SIP/2.0
Via: SIP/2.0/UDP 198.18.133.226:5060;branch=z9hG4bK2C26D17F
From: "LG Pilot User" <sip:+16785559862@198.18.133.226>;tag=6F2CC8C3-4B5
P-Asserted-Identity: "LG Pilot User" <sip:+16785559862@198.18.133.226>
Diversion: "Charles Holland"<sip:+19194745971@10.21.0.214>;privacy=off;reason=unconditional;counter=1
```

SIP Profile to send the original From number

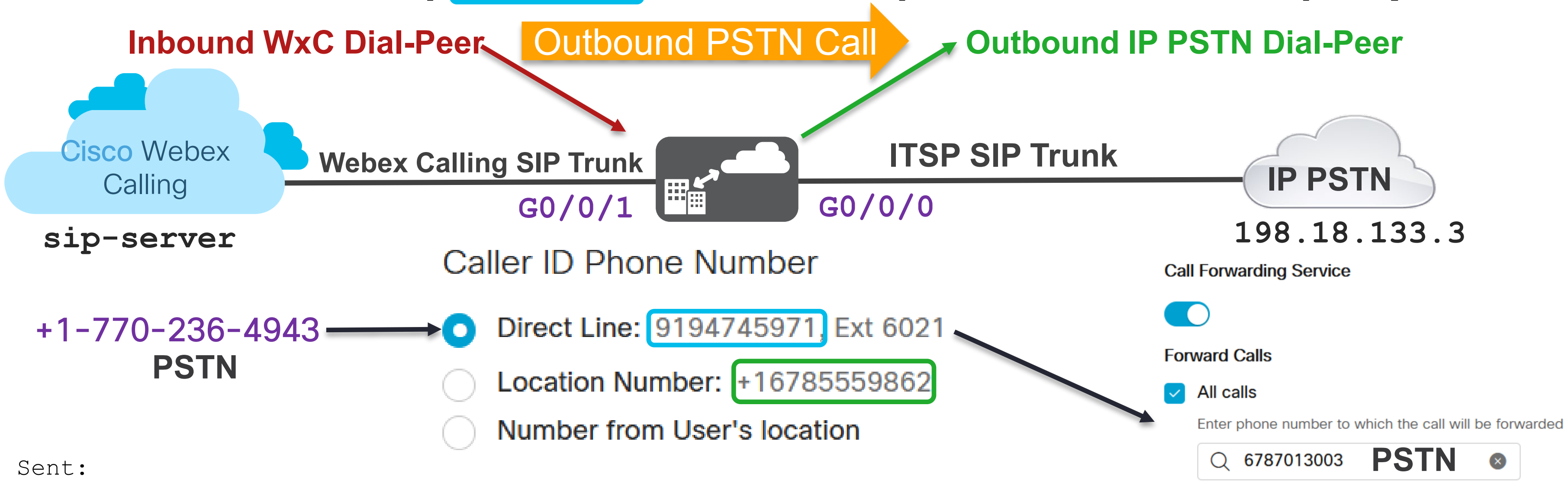
Voice class sip-copylist 200 - Create a copy list (tag 200) and apply it to the inbound dial-peer facing Webex Calling (dial-peer 200201 from documentation). In this way we copy the original FROM in the incoming INVITE from Webex Calling and use it to re-write the FROM in the outgoing PSTN INVITE.

```
voice class sip-copylist 200
 sip-header From

dial-peer voice 200201 voip
 description Inbound/Outbound Webex Calling
 voice-class sip copy-list 200

voice class sip-profiles 100
 rule 1 request INVITE peer-header sip From copy "sip:(.*)@" u03
 rule 2 request INVITE sip-header From modify "sip:(.*)@" sip:\u03@

dial-peer voice 101 voip
 description Outgoing dial-peer to IP PSTN
 voice-class sip profiles 100
```


Outbound LGW forwarded (PSTN to MPP to PSTN) call – Direct Line

With SIP Profile

```
Received:
INVITE sip:+16787013003@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.-64.100.12.6V26076-0-100-1666408278-
1607559617941-
From:<sip:+17702364943@139.177.64.10;user=phone>;tag=1666408278-1607559617941-
P-Asserted-Identity:"LG Pilot User"<sip:+16785559862@10.21.0.214;user=phone>
Diversion:"Charles Holland"<sip:+19194745971@10.21.0.214;user=phone>;reason=unconditional;privacy=off;counter=1
```




```
Sent:
INVITE sip:+16787013003@198.18.133.3:5060 SIP/2.0
Via: SIP/2.0/UDP 198.18.133.226:5060;branch=z9hG4bK2C26D17F
From: "LG Pilot User" <sip:+17702364943@198.18.133.226>;tag=6F2CC8C3-4B5
P-Asserted-Identity: "LG Pilot User" <sip:+16785559862@198.18.133.226>
Diversion: "Charles Holland"<sip:+19194745971@10.21.0.214>;privacy=off;reason=unconditional;counter=1
```

Agenda

- Local Gateway overview and sizing
- Key configuration updates required
- Caller ID handling
- Templates, Troubleshooting, and Resources

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

How to obtain a Packet capture in LGW/CUBE

- `show ip interface brief | exclude una`
- `monitor capture {CAPTURE_NAME:-TAC} interface {INTERFACE_NAME:-GigabitEthernet1} both`
- `monitor capture TAC match ipv4 protocol tcp any any`
- `show monitor capture`
- `monitor capture TAC start`
- `monitor capture TAC stop`
- `monitor capture TAC clear`
- `monitor capture TAC export {FTP_SERVER}`
- `no monitor capture TAC`
- Tutorial - https://community.cisco.com/t5/switching/ios-xe-monitor-capture-issue/td-p/2855298

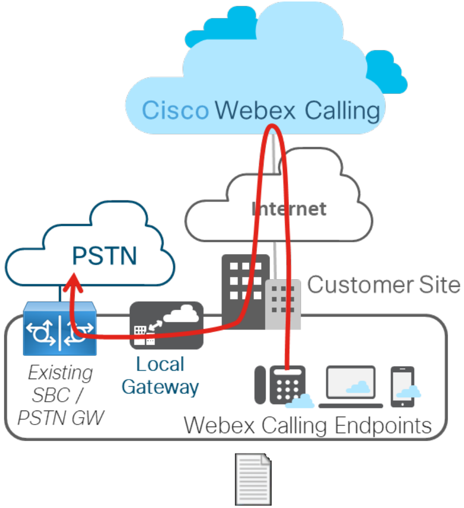
Introducing Always on debugging with VOIP Trace

- VOIP Trace is CUBE's new serviceability framework developed to log SIP Signalling and events without explicitly enabling traditional debugs like debug ccsip messages
- Traces are collected in system memory and can be stored in local buffer or on an external syslog server, which ever is configured
- VOIP Trace is enabled by default from IOS-XE 17.4.1, 17.3.2 (helps troubleshoot intermittent issues)
- VOIP Trace captures:
  - SIP messages for SIP Trunk to Trunk calls
  - Events and API calls from SIP layer to other layers in CUBE
  - SIP Errors
  - Call Control (unified communication call flows processed by CUBE)
  - FSM (Finite State Machines) states and events
  - Dial peer matched
  - RTP ports allocated

```
voice service voip
  trace
```

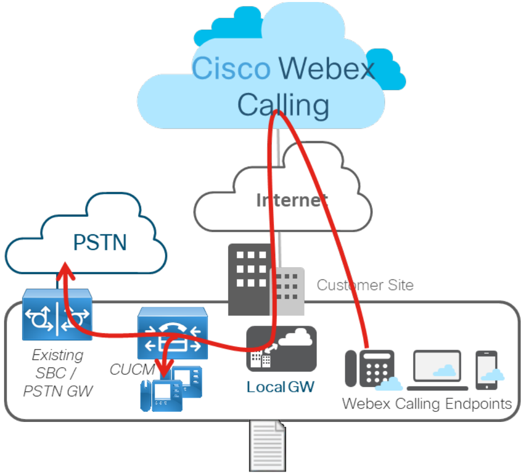
- For more details visit https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/voip-trace-for-cube.html

Local Gateway configuration templates



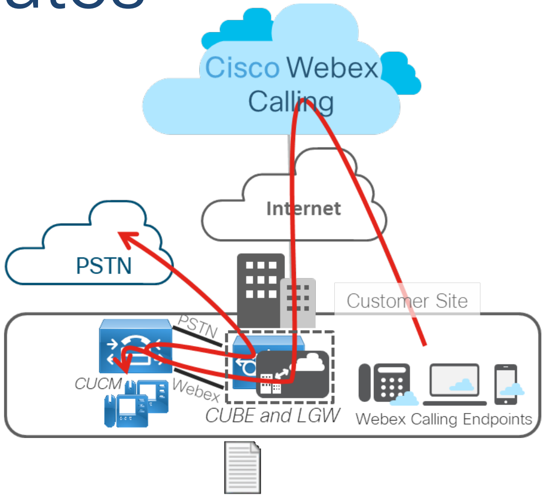
LGW Config Template - IP PSTN based - IOS-XE 16.12.4.txt

```
! 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
- Cisco Live Labs
  - Cisco Webex Calling v2 [HOLCOL-2006] – Hands-on offered on April 6th and 7th, 12 to 4 pm US EST.
  - Transitioning from Unified CM to Webex Calling [HOLCOL-3000] – April 8th, 12 to 4 pm US EST

Thank you

TURN

IT

UP

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