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The bridge to possible

Local Gateway Design and Deployment for Webex Calling

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BRKCOL-2169

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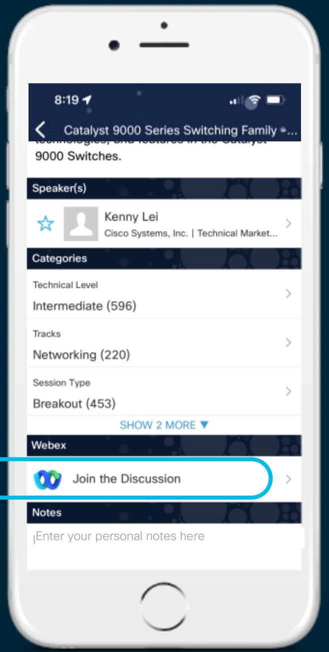

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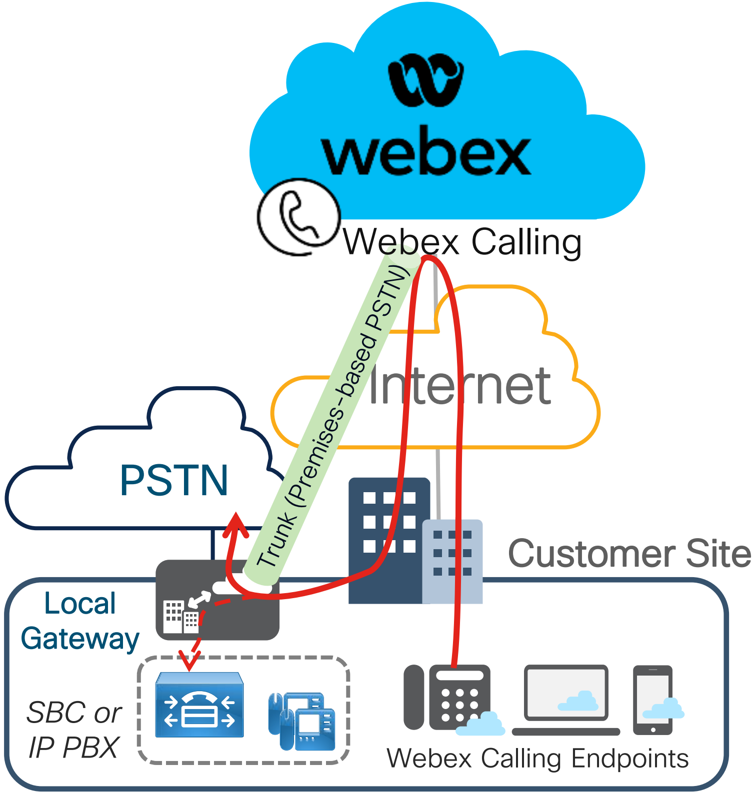
Agenda

- Local Gateway overview and sizing
- Key configuration updates required
- Multiple LGWs on a single CUBE
- Resources

Local Gateway
Overview and Sizing

Webex Calling Trunk - Local Gateway

(Premises-based PSTN) Deployment

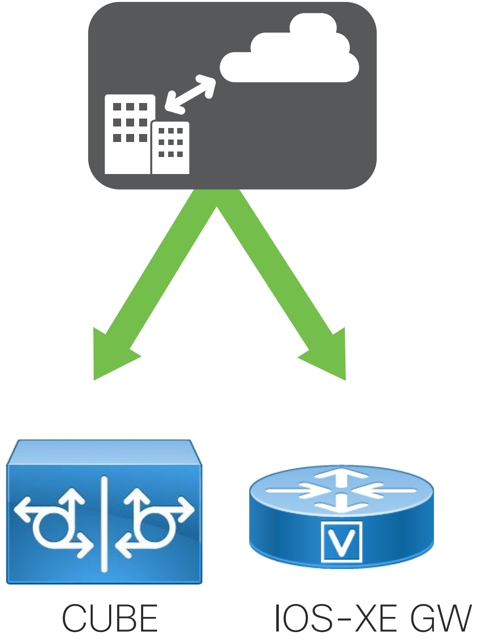


- Provides connectivity to a customer-owned premises-based PSTN service
- May also provide connectivity to an on-premises IP PBX or dedicated SBC/PSTN GW
- Enables on-prem to Webex Calling transition
- **Endpoint registration is NOT proxied through Local Gateway, unlike CUBE Lineside.** Endpoints directly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

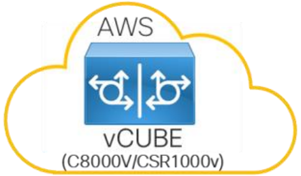
Local Gateway

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 17.6.3)
  - vCUBE in AWS, Azure



  - Catalyst 8200/8300 series (IOS XE 17.6.3)

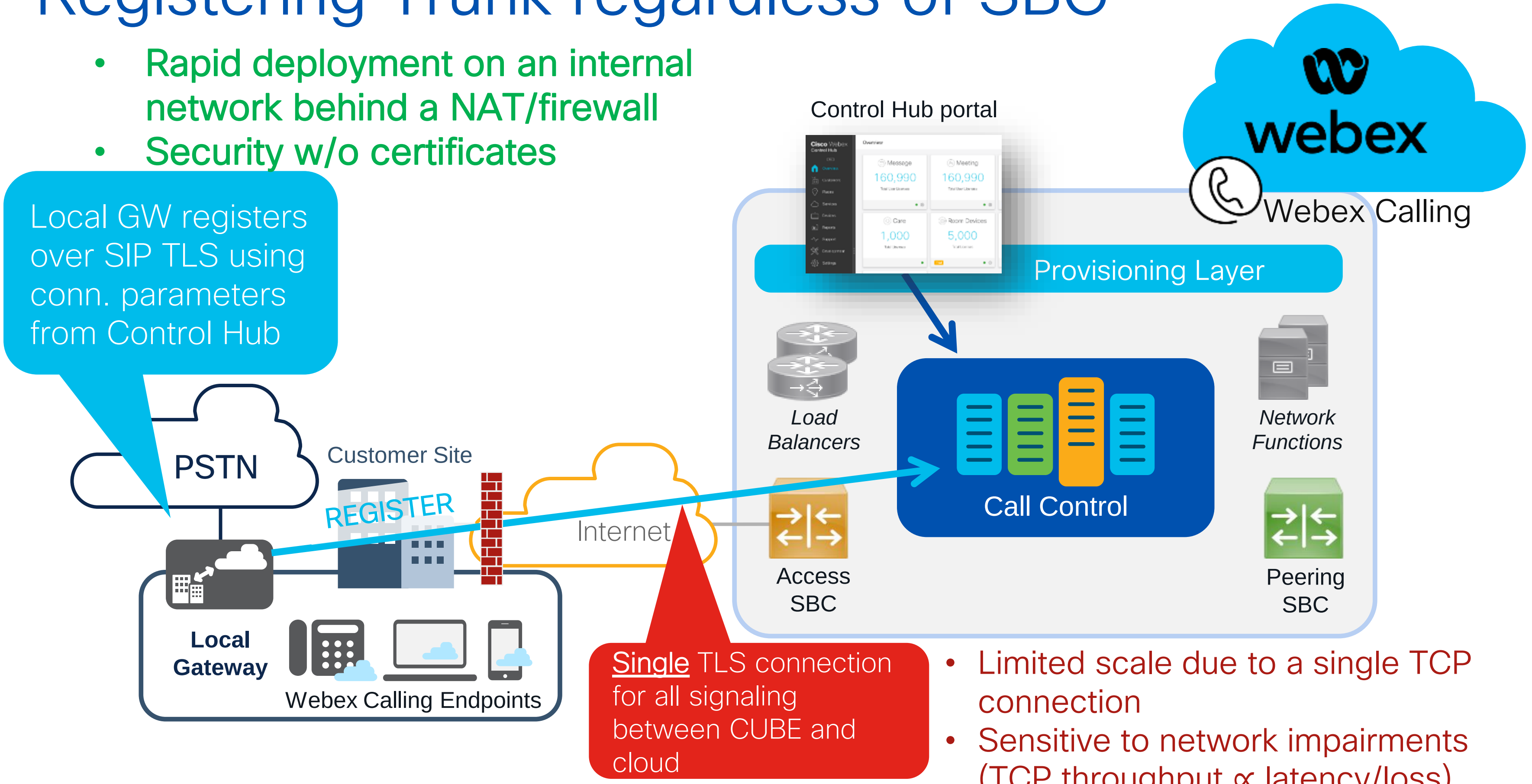
  - CSR 1000v (vCUBE) (16.12.5 or later – 17.3.5 latest) –
  - Catalyst 8000v Edge (vCUBE) (IOS XE 17.6.3)
    - CSR 1000v licenses are not included in Webex Calling Flex and need to be purchased separately
    - Estimate 200 kbps total data throughput for every audio call
  - ISR 1100 (IOS-XE 16.12.5 or later)

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
12.5.0	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 – 16.12.6 – 16.12.7
12.7.1	17.1.1	Nov 2019	-
12.8.0	17.2.1r	March 2020	17.2.3
14.0	17.3.1a	July 2020	17.3.5
14.1	17.3.2*	Oct 2020	17.3.5
14.2	17.4.1a	Nov 2020	17.4.2
14.3	17.5.1	March 2021	17.5.1a
14.4	17.6.1a	July 2021	17.6.3a
14.4	17.7.1a	Nov 2021	
14.4	17.8.1a	March 2022	
TBD	17.9.1	July 2022	

Registering Trunk regardless of SBC

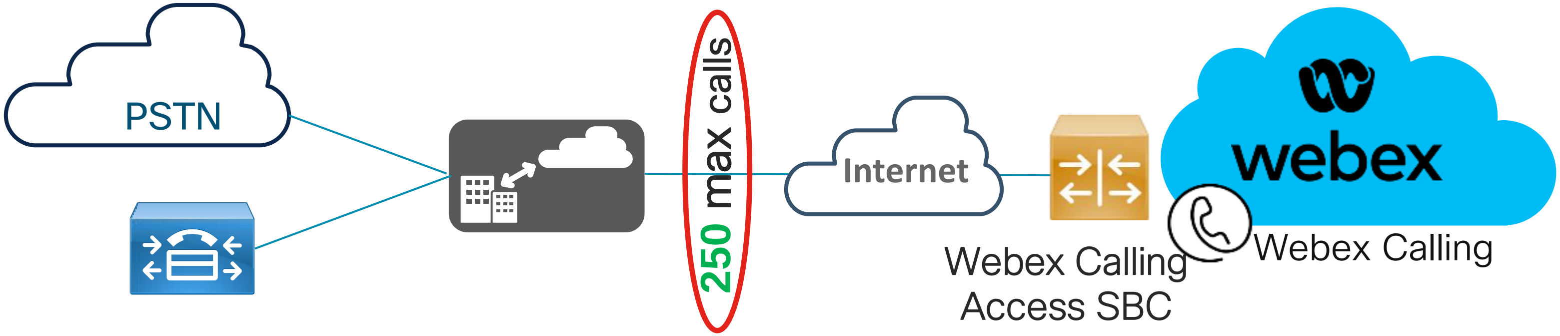
- Rapid deployment on an internal network behind a NAT/firewall
- Security w/o certificates



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

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*

Agenda

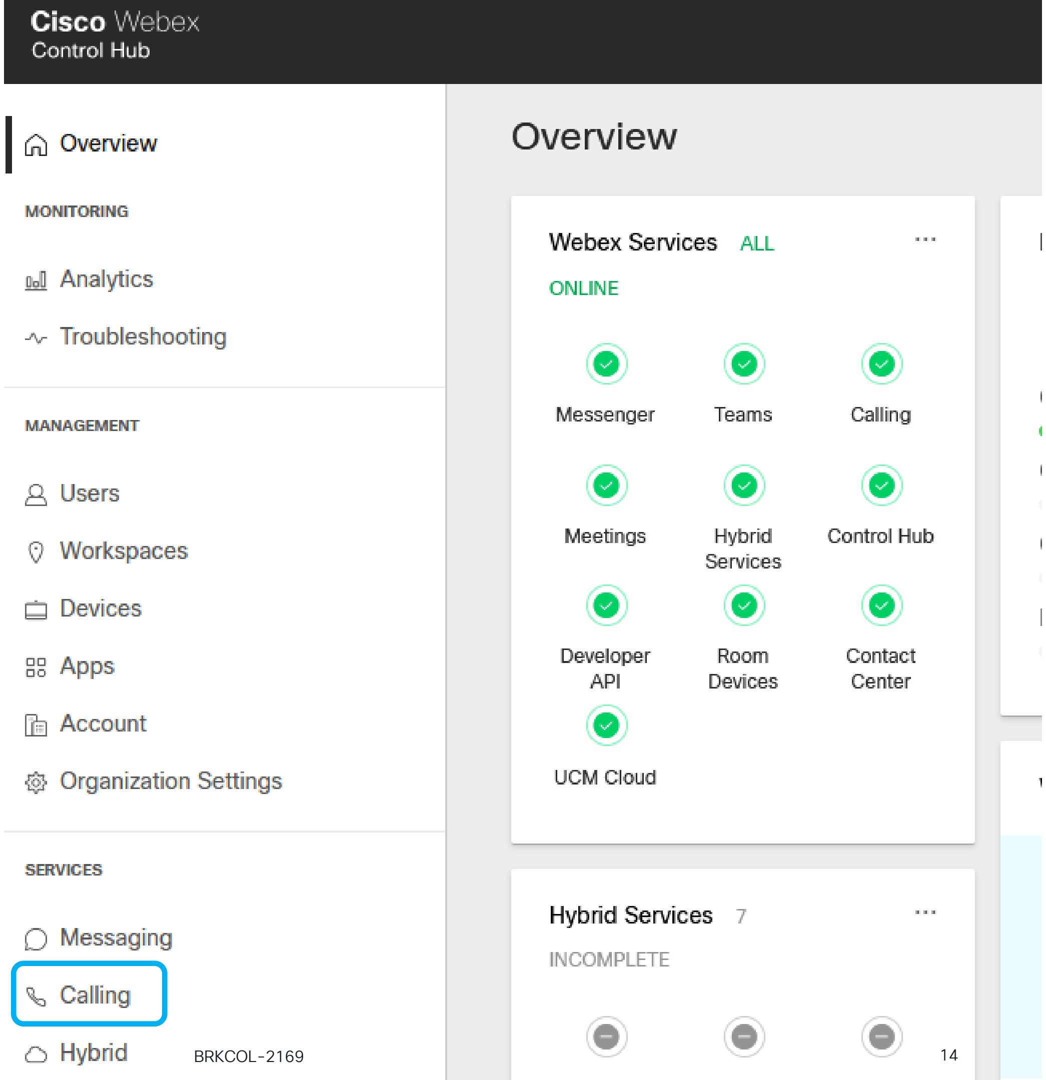
- Local Gateway overview and sizing
- Key configuration updates required
- Multiple LGWs on a single CUBE
- Templates, Troubleshooting, and Resources

Key Configuration Updates Required

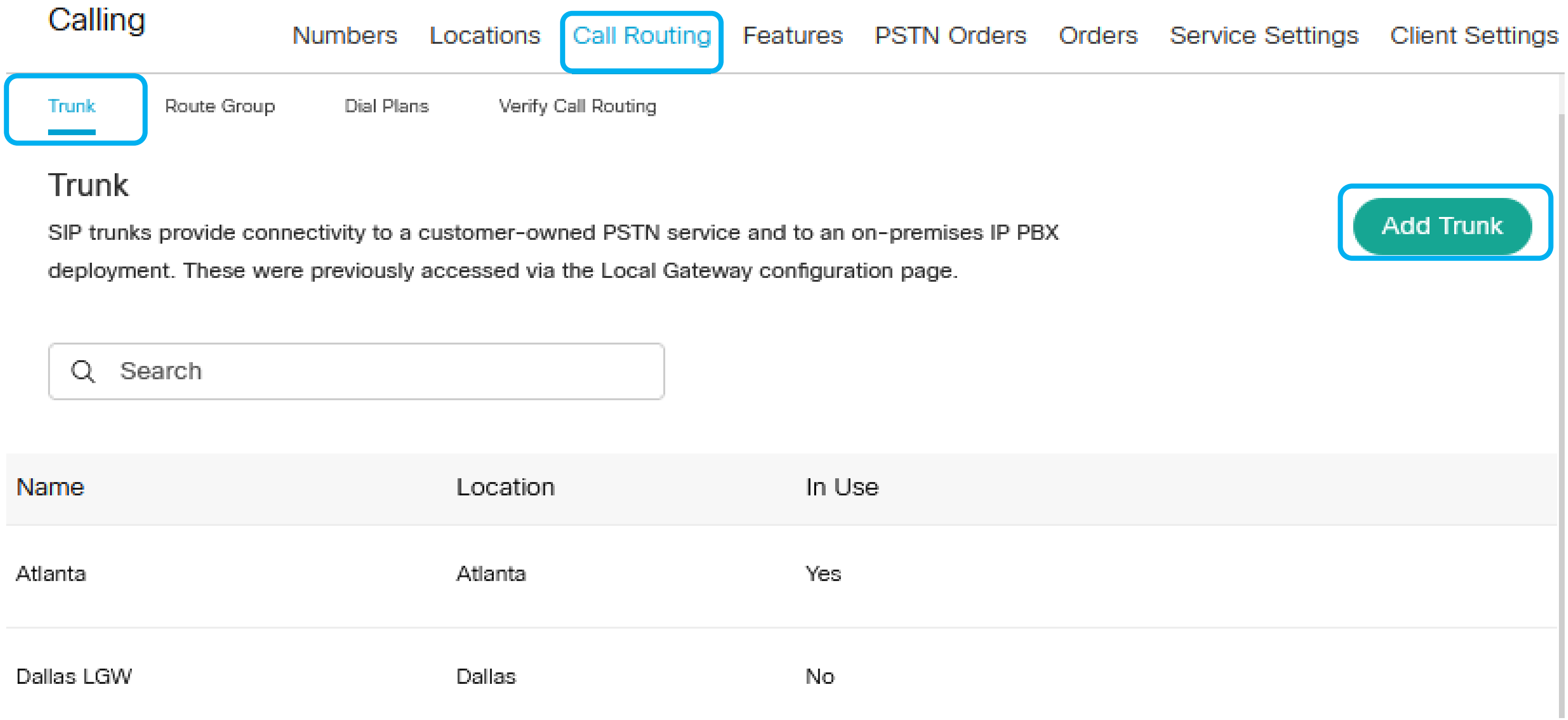
Update

Onboarding Process Webex Calling Trunk

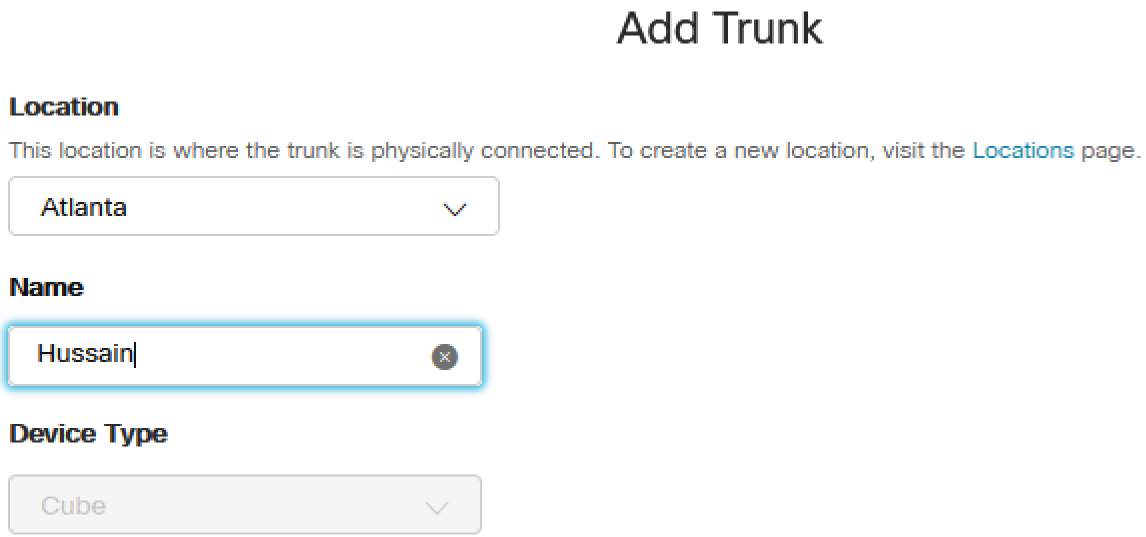
1a. Log in to customer portal and navigate to Services – Click Calling



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

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

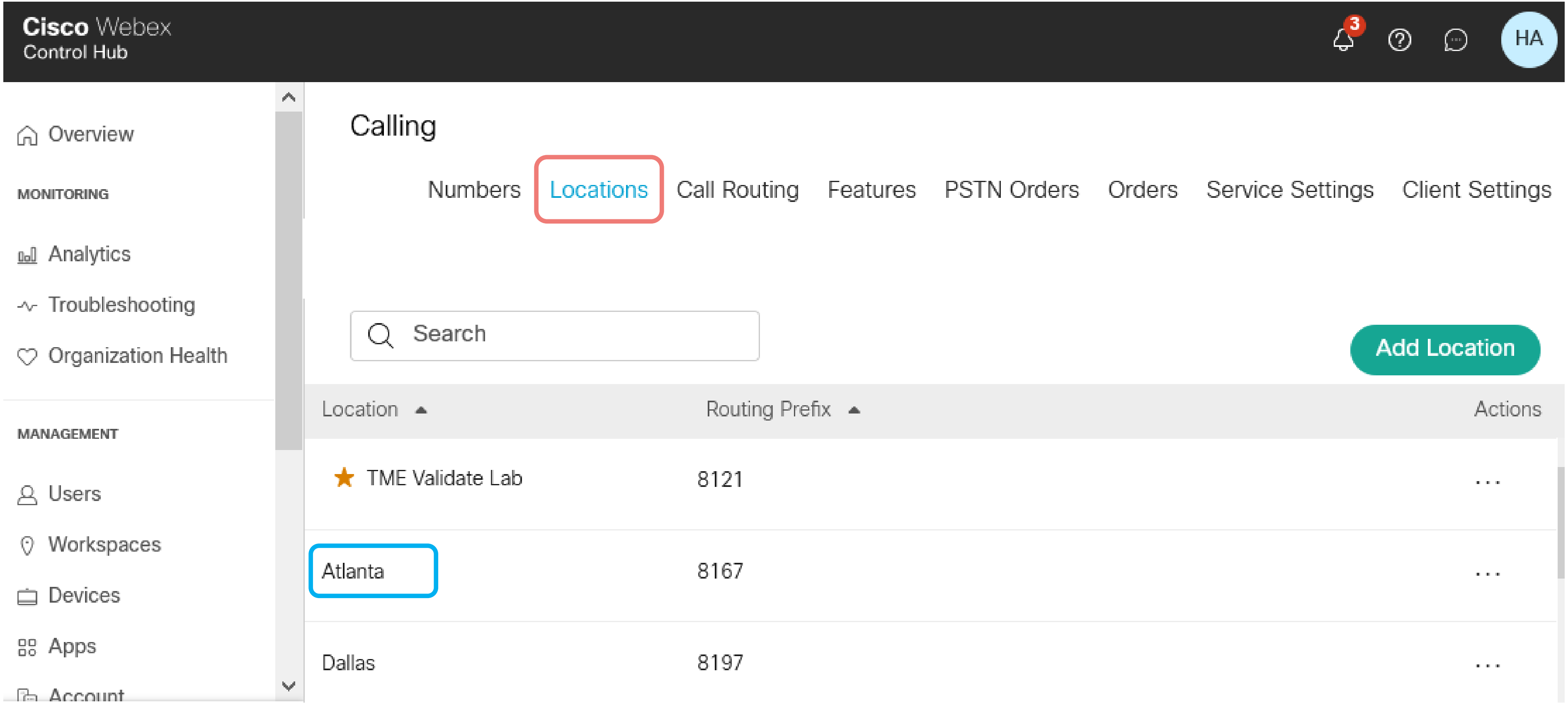
Authentication Information

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

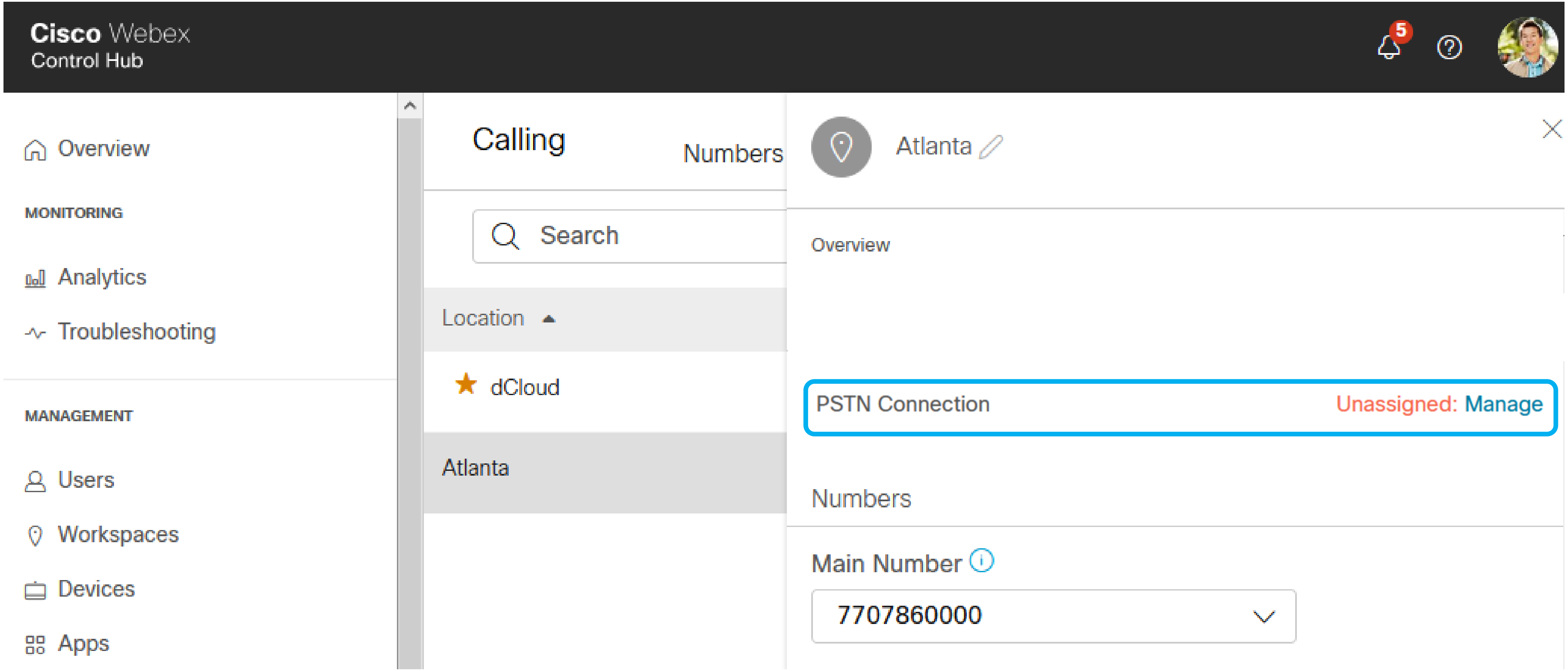
Username: Hussain2572_LGU

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

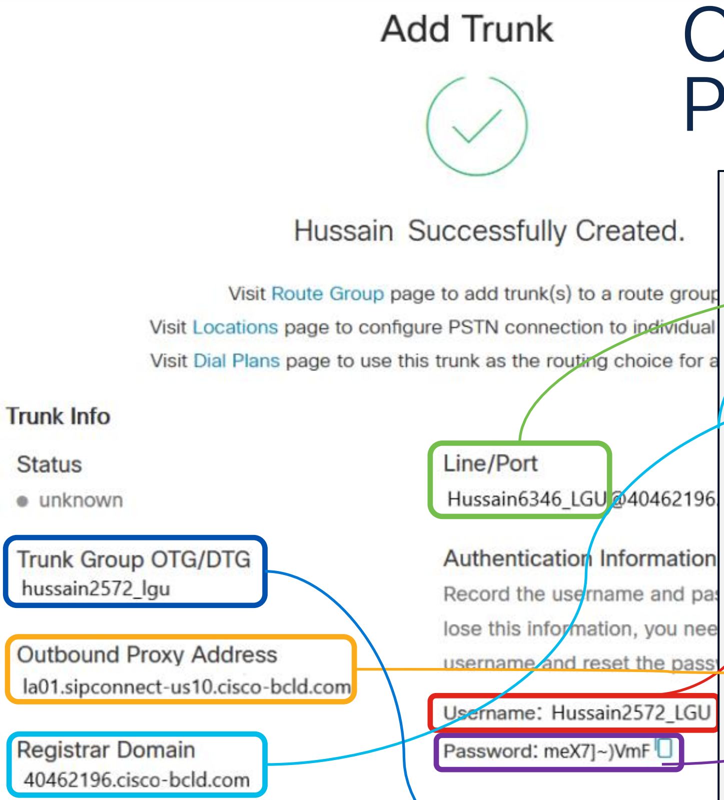
This trunk is located in **Atlanta**.

* 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

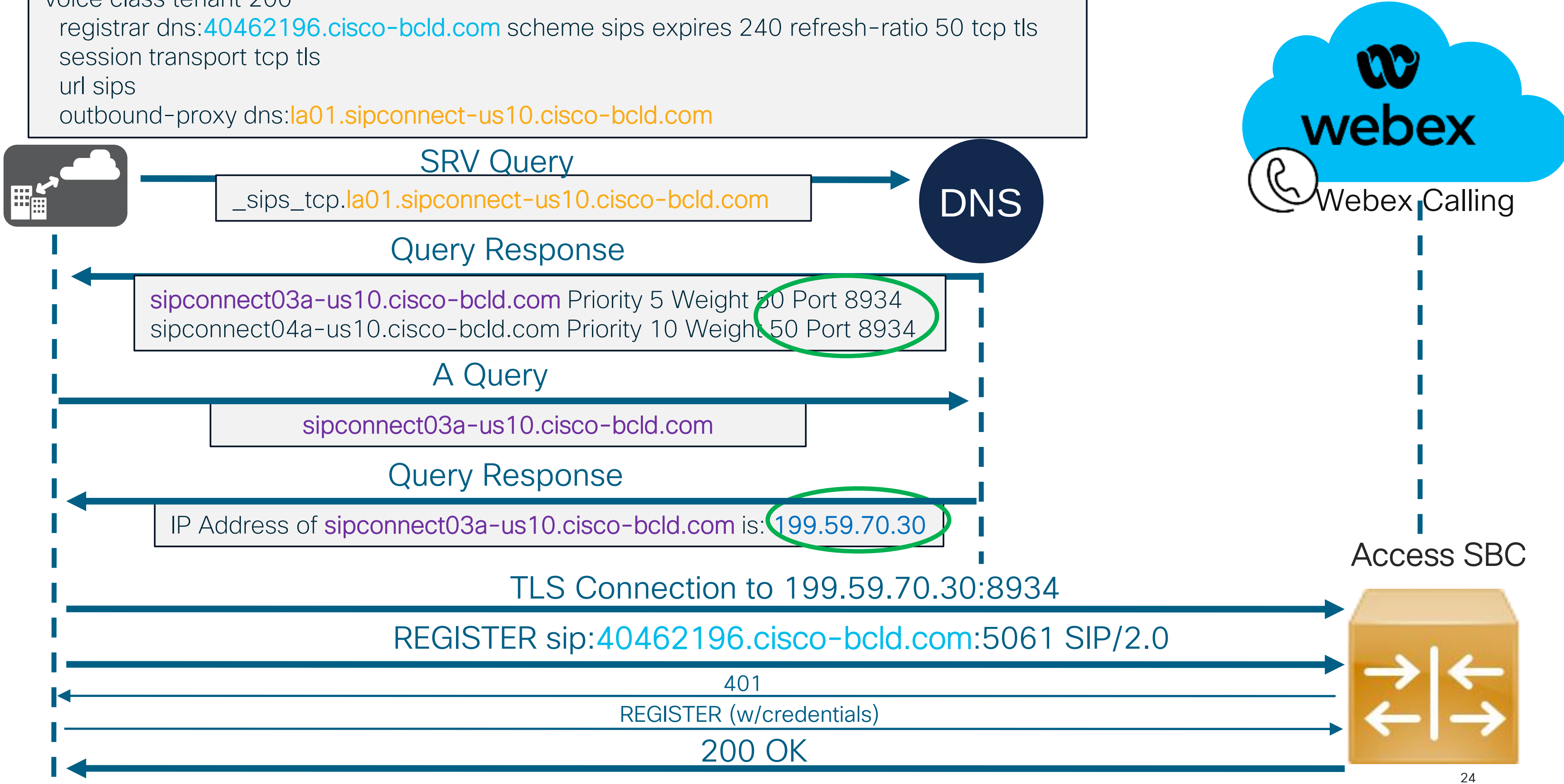
Control Hub Trunk Info Connection Parameters → LGW CLI Config




```
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:"
```

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

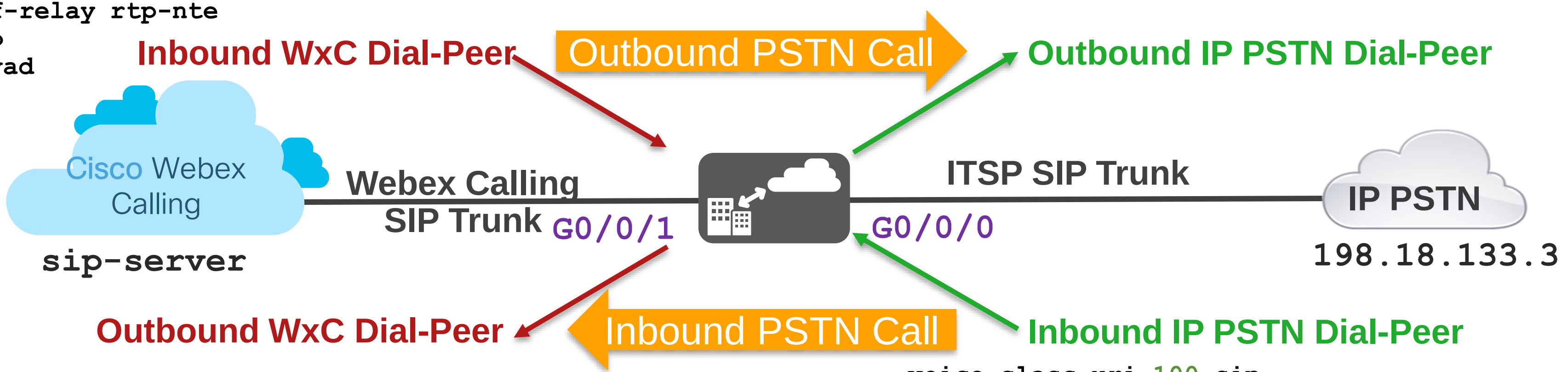
Single Dial-peer facing Webex Calling for Inbound/Outbound Calls

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
```

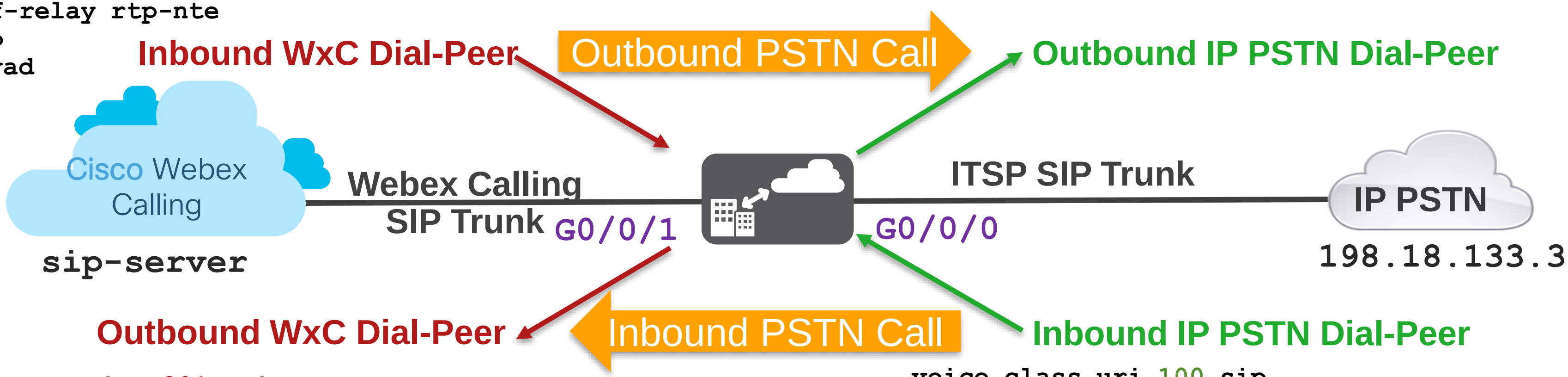
```
voice class uri 100 sip
  host ipv4:198.18.133.3

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
```

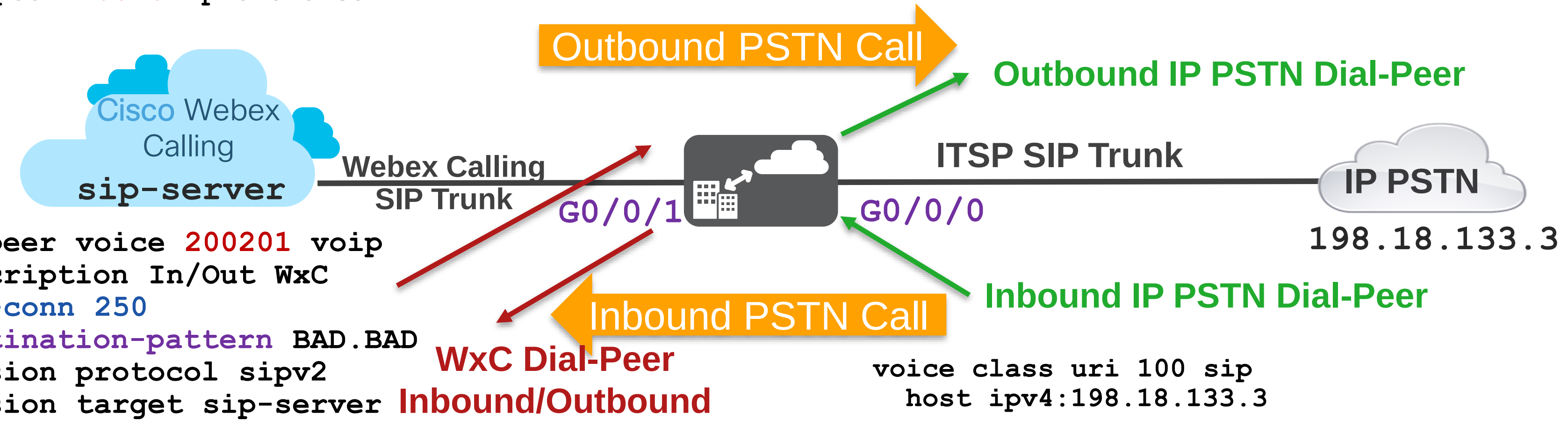
New Dial-peer structure

```
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
```

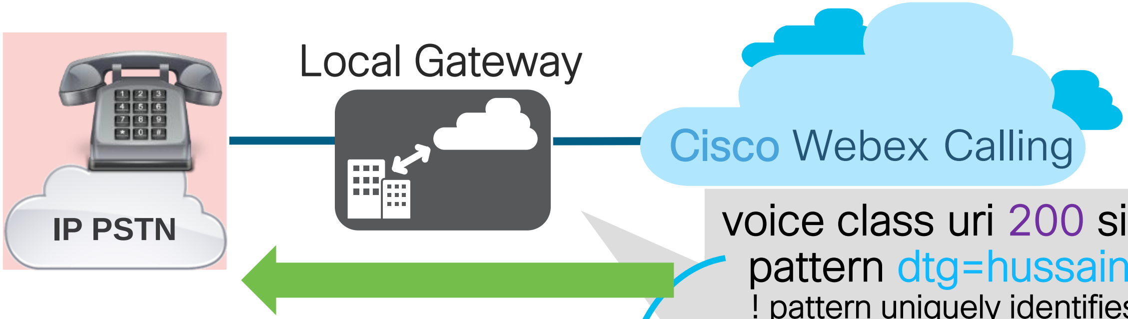



```
dial-peer voice 200201 voip
  description In/Out WxC
  max-conn 250
  destination-pattern BAD.BAD
  session protocol sipv2
  session target sip-server
  destination dpg 100
  incoming uri request 200
  voice-class codec 99
  voice-class stun-usage 200
  no voice-class sip localhost
  voice-class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

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

Opus Codec Transcoding

- Pre 17.6 releases
  - CUBE/LGW didn't support transcoding calls using Opus codec
  - Only calls where Opus codec was used on both legs were supported
  - Calls that required Opus transcoding were dropped in the release before 17.6.1
- 17.6.1 release and later
  - Interworking between Opus and other codecs is supported
  - Transcoding between Opus & other codecs is supported for both secure and non-secure calls
    - RTP-to-RTP, SRTP-to-SRTP, SRTP-to-RTP, and RTP-to-SRTP, SRTP interworking calls, HA setup
  - Opus transcoding is supported only with PVDM4 DSP cards in ISR 4K and Catalyst 82/8300 series platforms

DSPFARM Profile configuration –

```
Active(config)#dspfarm profile 1 transcode
Active(config-dspfarm-profile)#codec g729abr8
Active(config-dspfarm-profile)#codec g729ar8
Active(config-dspfarm-profile)#codec g711alaw
Active(config-dspfarm-profile)#codec g711ulaw
Active(config-dspfarm-profile)#codec g729r8
Active(config-dspfarm-profile)#codec opus
Active(config-dspfarm-profile)#maximum sessions 12
Active(config-dspfarm-profile)# associate application CUBE
Active(config-dspfarm-profile)#no shut
Active(config-dspfarm-profile)#exit
```

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- Multiple LGWs on a single CUBE
- Templates, Troubleshooting, and Resources

Multiple LGWs on a single CUBE

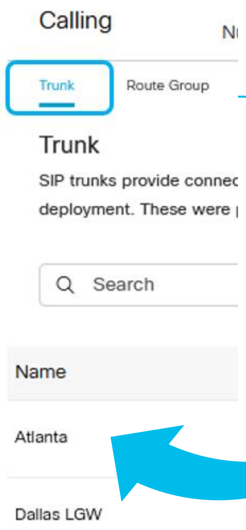
What constitutes a LGW within a CUBE config?

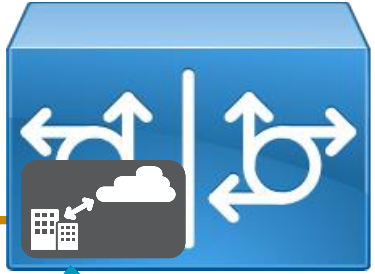
```
voice class sip-profiles 200
  rule 20 request ANY sip-header From modify ">" ";otg= hussain3847_lgu >"
```

```
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
  sip-server dns:XXXXXX
  session transport tcp tls
  url sips
  bind control source-interface GigabitEthernet1
  bind media source-interface GigabitEthernet1
  sip-profiles 200
  outbound-proxy dns:XXXXXX
```

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

```
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
```





Single CUBE instance with two LGWs – Total 500 calls

Any supported CUBE Release

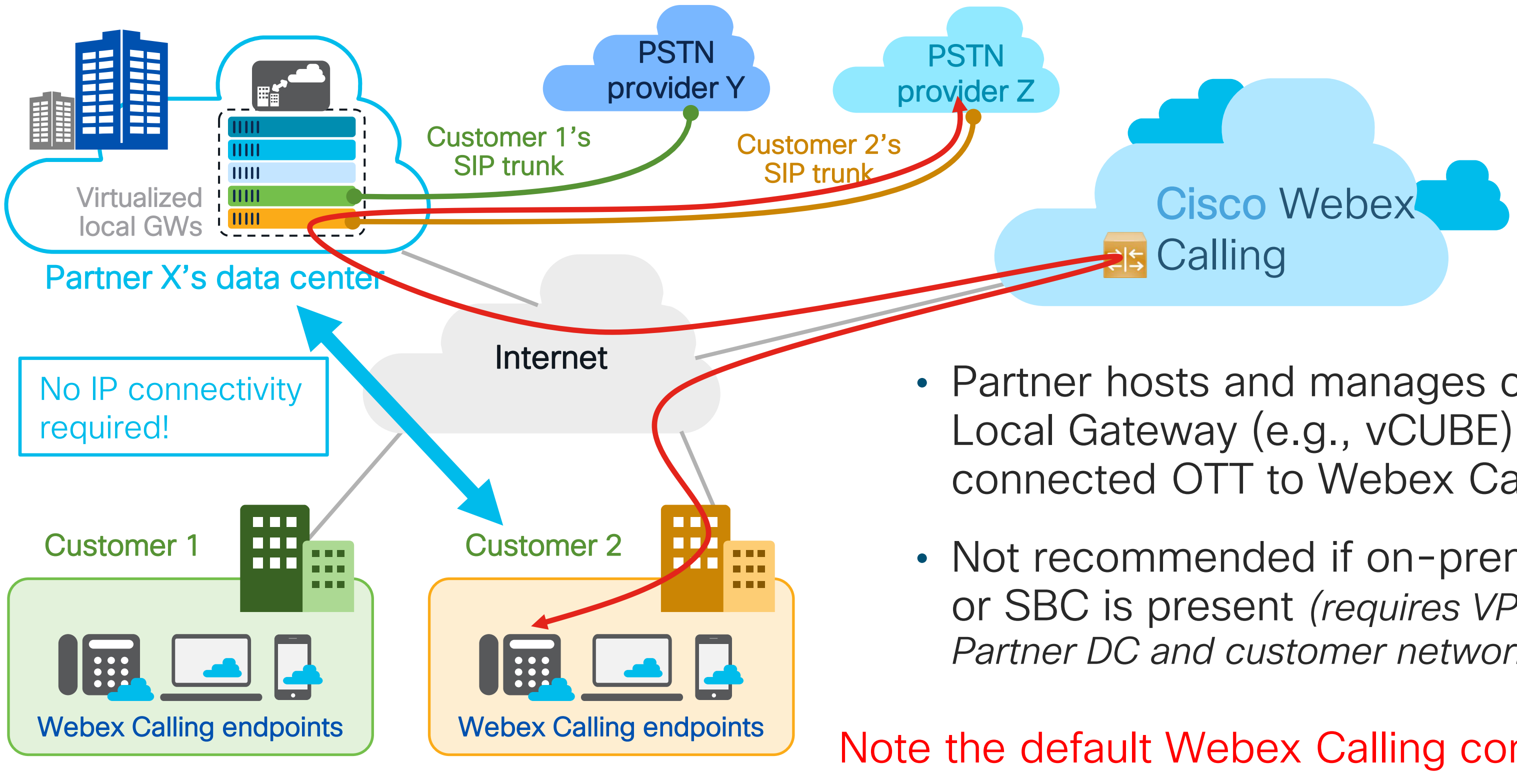
Trunk1 – LGW1=250 calls

```
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
```

Webex Calling PSTN option:
Partner hosted Local Gateway

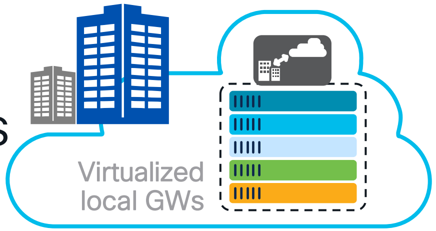


- Partner hosts and manages customer's Local Gateway (e.g., vCUBE) - connected OTT to Webex Calling
- Not recommended if on-premises PBX or SBC is present *(requires VPN between Partner DC and customer network)*

Note the default Webex Calling concurrent call limits if planning for Partner hosted LGW option

Partner Hosted LGW

- Can run on any supported CUBE platform
- Single (v)CUBE instance for multiple tenants
- Stacking of dial-peer configuration
  - Tenant specific dial-peers
- From Webex Calling
  - voice class URI 2xx sip matching on tenant specific DTG obtained from Control Hub
- From PSTN
  - voice lass URI 1xx sip matching on SIP trunk peer address
  - .. or matching on called number, if the SIP trunk is shared



SIP Listening Port on CUBE before IOS-XE 17.8.1a

- Default SIP Listen ports are 5060 (UDP/TCP) and 5061 (TLS)

```
voice service voip
  sip
    listen-port non-secure 2000 secure 2050
```

Introducing multiple SIP listen ports

Multi Tenant – SIP listen ports

- From 17.8.1, inbound listen ports can be configured under each tenant. Previously it was only a global configuration
- Multiple inbound TLS (secured) and TCP/UDP (non-secured) connection can be established on different listen ports.
- Each listen ports can be configured with their own TLS-Profile and their own validation criteria
- Listen port based multi tenancy is supported on both IPv4 and IPv6 on all TLS/TCP/UDP transport protocols

SIP listen ports

Considerations

- It is mandatory to configure bind at the tenant level to support multi tenancy based on listen ports. Without interface bind at the tenant level, listen port will not be opened.
- Listen port and bind interface must be unique across:
    - Global and tenant level
    - Secure and non-secure
  - i.e., we cannot configure the same TLS/TCP/UDP listen port across multiple tenants or globally. CUBE will report an error as “Port number already in use”
- Tenant level listen port configuration cannot be modified while there are active calls using the dial-peer to which the tenant is associated.

Multi Tenant – SIP listen ports
Design Considerations

- The non-secure port range is 5000 – 5500. This is to avoid overlap between UDP/RTP port range
- TLS port range is 1 to 65535
- For listen-port based multi tenant support on CUBE, you **MUST** configure
  - Tls profile under voice class tenant configuration mode
  - Listen port under voice class tenant configuration mode
- Functionality:
- The inbound dial-peers are filtered based on the tenant tag associated to the dial-peer
  - Inbound call to Listen port → tenant → Inbound dial-peers

Multi Tenant – SIP listen ports - Configuration

- Configure voice class tls-profile

```
Router(config)#voice class tls-profile <tag>
Router(config-class)#cn-san validate ?
  bidirectional  Enable CN/SAN validation for both client and server certificate
  client         Enable CN/SAN validation for client certificate
  server         Enable CN/SAN validation for server certificate
```

- Configure voice class tenant, listen port and associate TLS profile to the tenant

```
Router(config)#voice class tenant <tag>
Router(config-class)#listen-port ?

     non-secure  Change UDP/TCP SIP listen port (have bind configured under this
                 tenant for the config to take effect)
     secure      Change TLS SIP listen port (have bind configured under this
                 tenant for the config to take effect

Router(config-class)#tls-profile <tag>
  <1-10000>  Specify the tls-profile tag number
```


Single vCUBE instance with two LGWs – Total 500 calls

IOS-XE 17.8.1a or later

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 GigabitEthernet1
  bind media source-interface GigabitEthernet1
  listen-port secure 5062
  tls-profile 2

voice class tls-profile 2
  trustpoint CUBE-TLS
```

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 GigabitEthernet1
  bind media source-interface GigabitEthernet1
  listen-port secure 5070
  tls-profile 3

voice class tls-profile 3
  trustpoint CUBE-TLS
```


Multi Tenant – SIP listen ports – Performance/Scale

Platform	No. of Tenants	No. TLS Conn.	CPS	Concurrent calls	Max Calls per Tenant
C8KV Medium	200	400	6	1080	6
ASR1006x	200	400	20	3600	18
ISR4461	200	400	39	7000	35
C8300-2N2S-4T2X	200	400	39	7020	36

Platform	No. of Tenants	No. TLS Conn.	CPS	Concurrent calls	Max Calls per Tenant
C8KV Medium	0	1	6	1080	NA
ASR1006x	0	1	20	3600	NA
ISR4461	0	1	39	7000	NA
C8300-2N2S-4T2X	0	1	39	7020	NA

Agenda

- Local Gateway overview and sizing
- Key configuration updates required
- Multiple LGWs on a single CUBE
- Resources

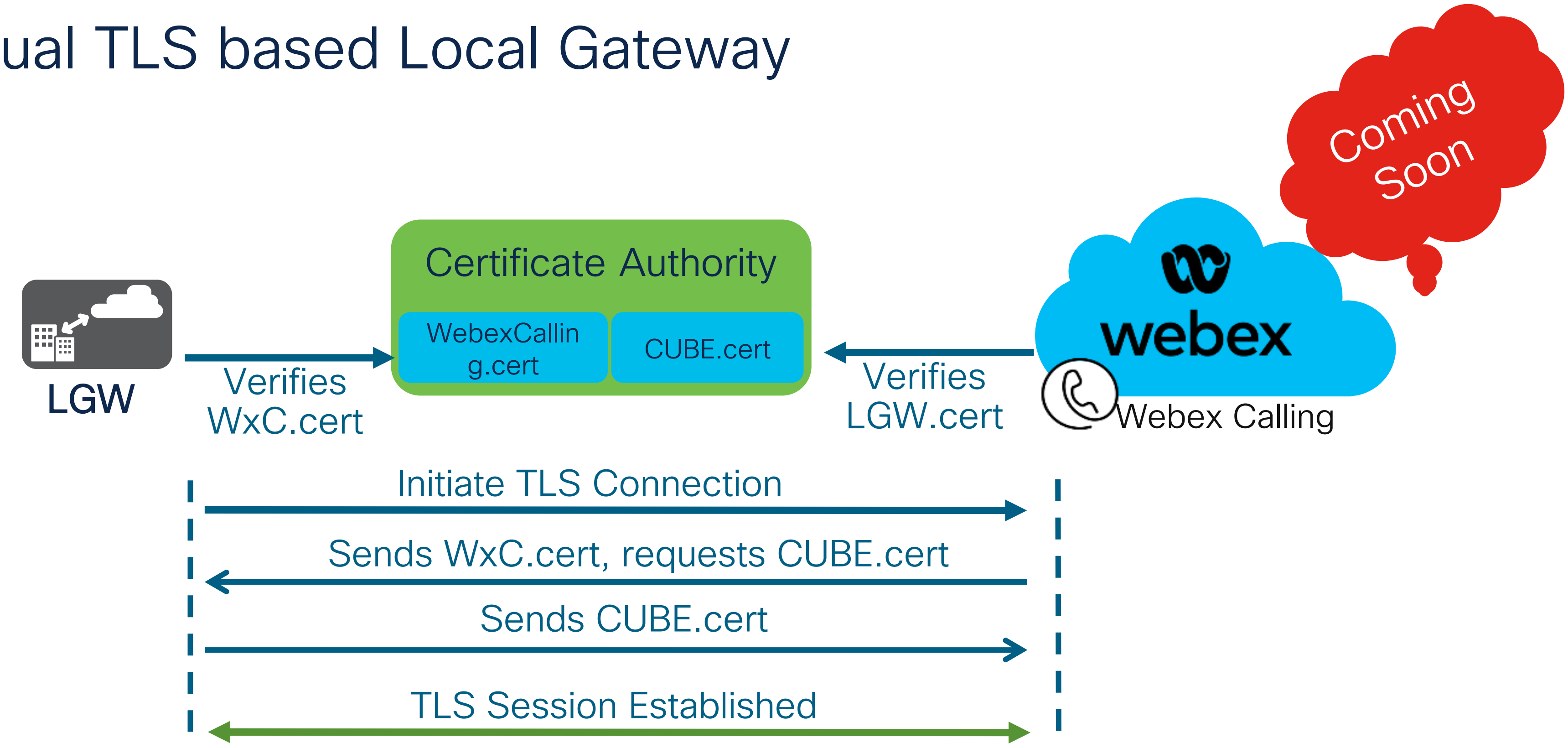
Introducing Always on debugging with VOIP Trace

- 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
- 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

```
voice service voip
  trace
```

Resources

Mutual TLS based Local Gateway



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
- vCUBE support on Azure
- Labs
  - Cisco Webex Calling
  - Transitioning from Unified CM to Webex Calling

Technical Session Surveys

- Attendees who fill out a minimum of four session surveys and the overall event survey will get Cisco Live branded socks!
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Cisco Learning Network
Resource community portal for certifications and learning

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Cisco Training Bootcamps
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Cisco Learning Partner Program
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- Visit the On-Demand Library for more sessions at www.CiscoLive.com/on-demand

The bridge to possible

Thank you

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