Enabling Cloud and Premise Deployments with CUBE

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Cisco Webex App

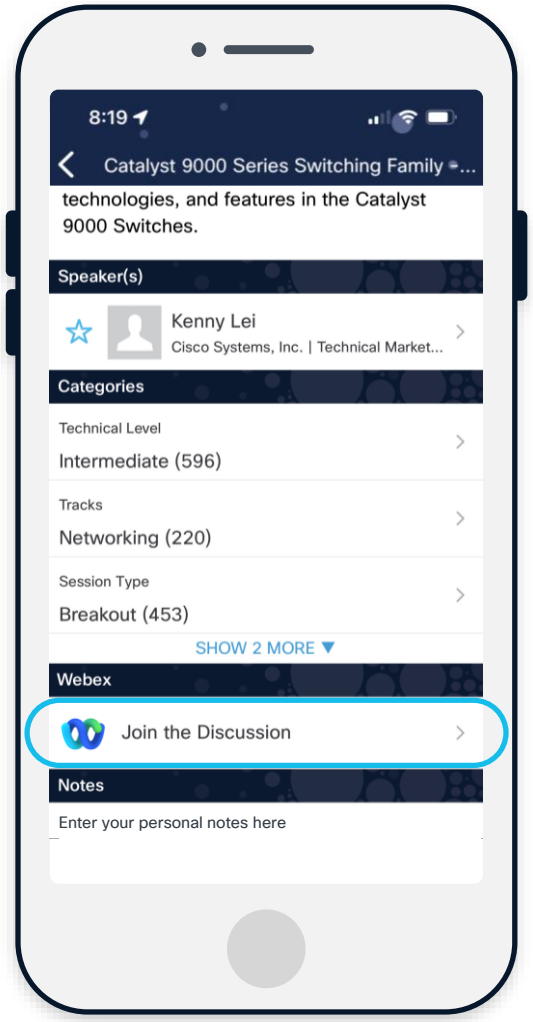
Questions?

Use Cisco Webex App to chat
with the speaker after the session

How

1. Find this session in the Cisco Live Mobile App
2. Click “Join the Discussion”
3. Install the Webex App or go directly to the Webex space
4. Enter messages/questions in the Webex space

Webex spaces will be moderated by the speaker until June 13, 2025.

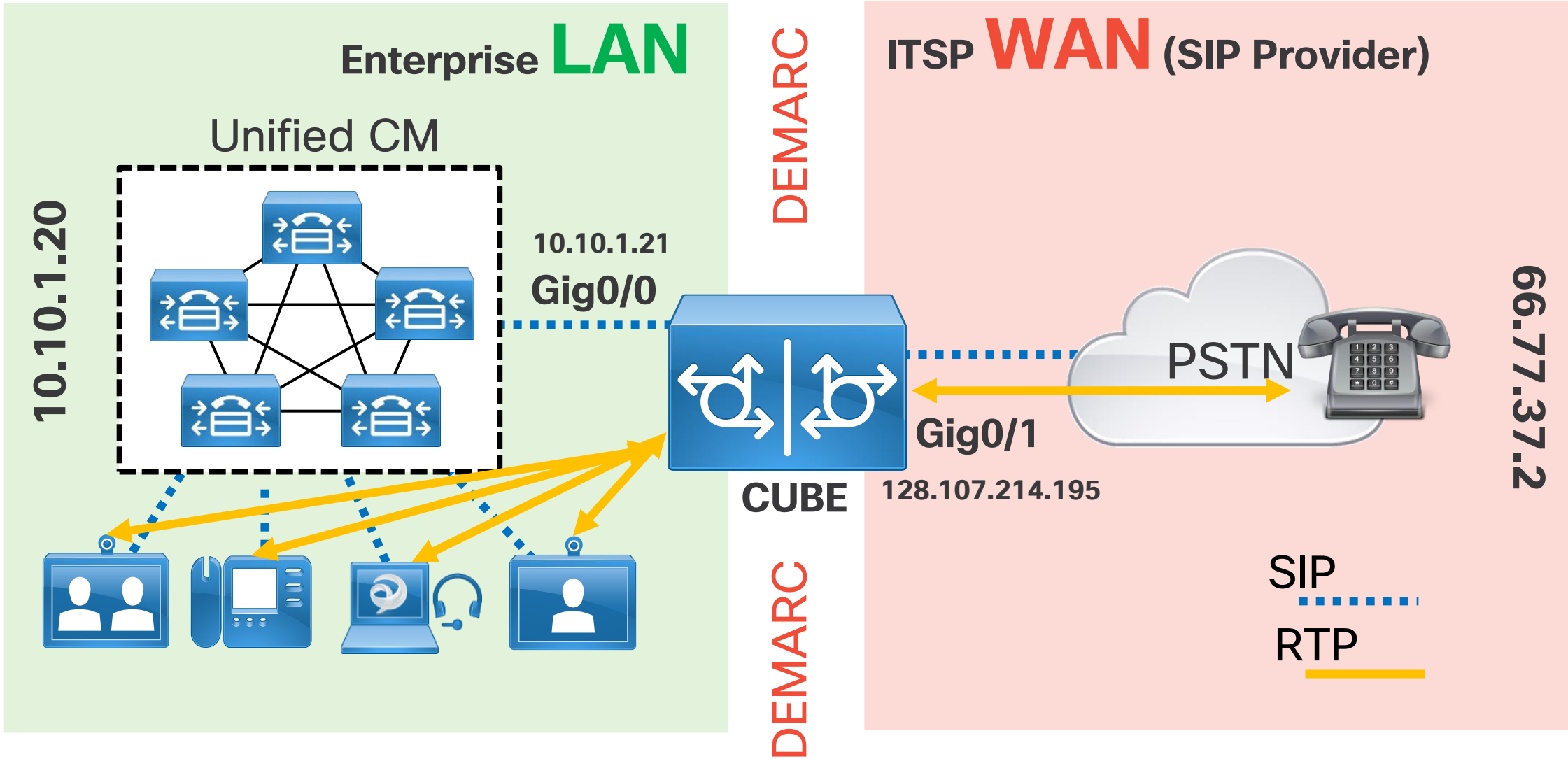


Agenda

- CUBE Recap
  - CUBE Configuration Steps
- Version 14 Updates
- Local Gateway (LGW) for Webex Calling
- Survivability Gateway (SGW) for Webex Calling
- CUBE with ThousandEyes
- References

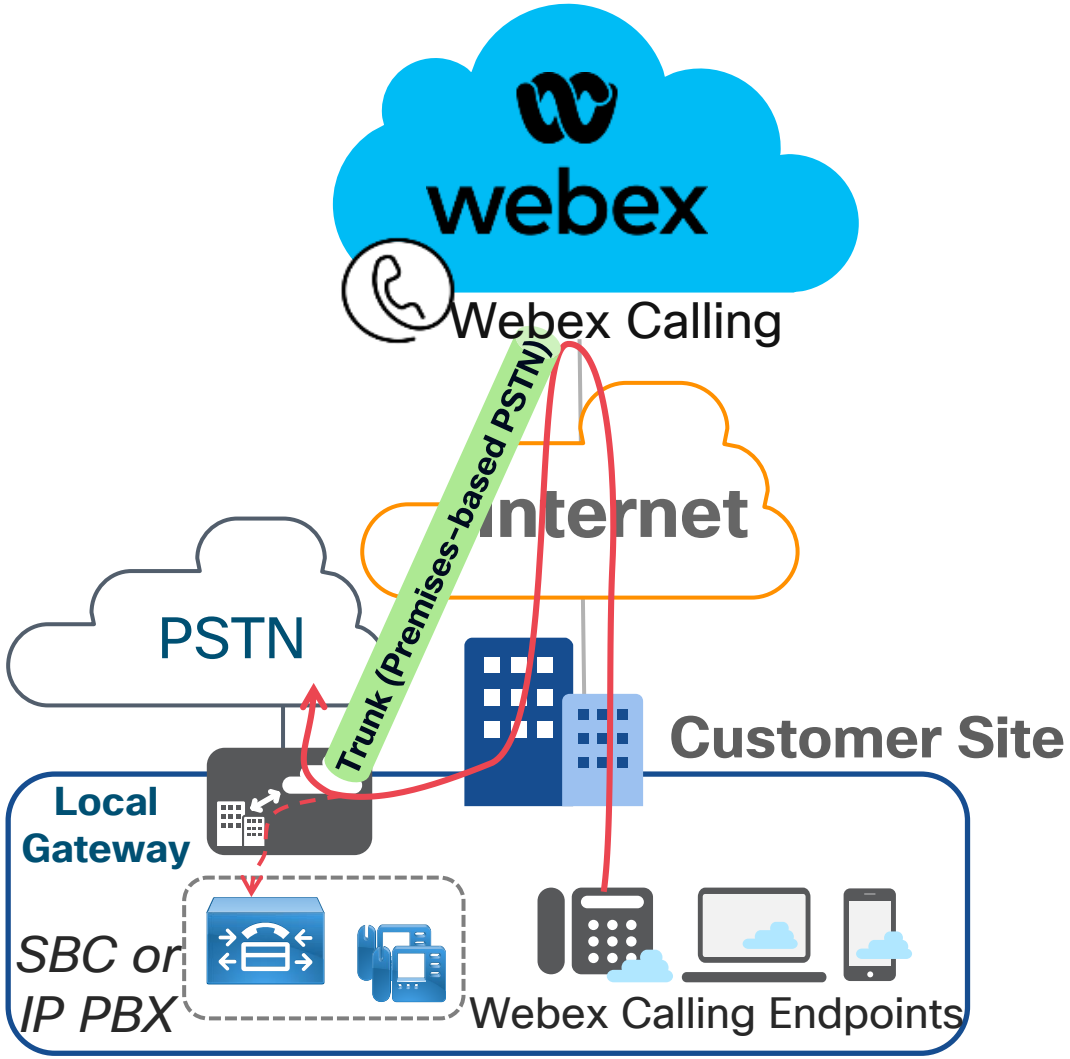
CUBE Recap

CUBE as an SBC for an on-premises Collaboration Deployment



Webex Calling Trunk - Local Gateway

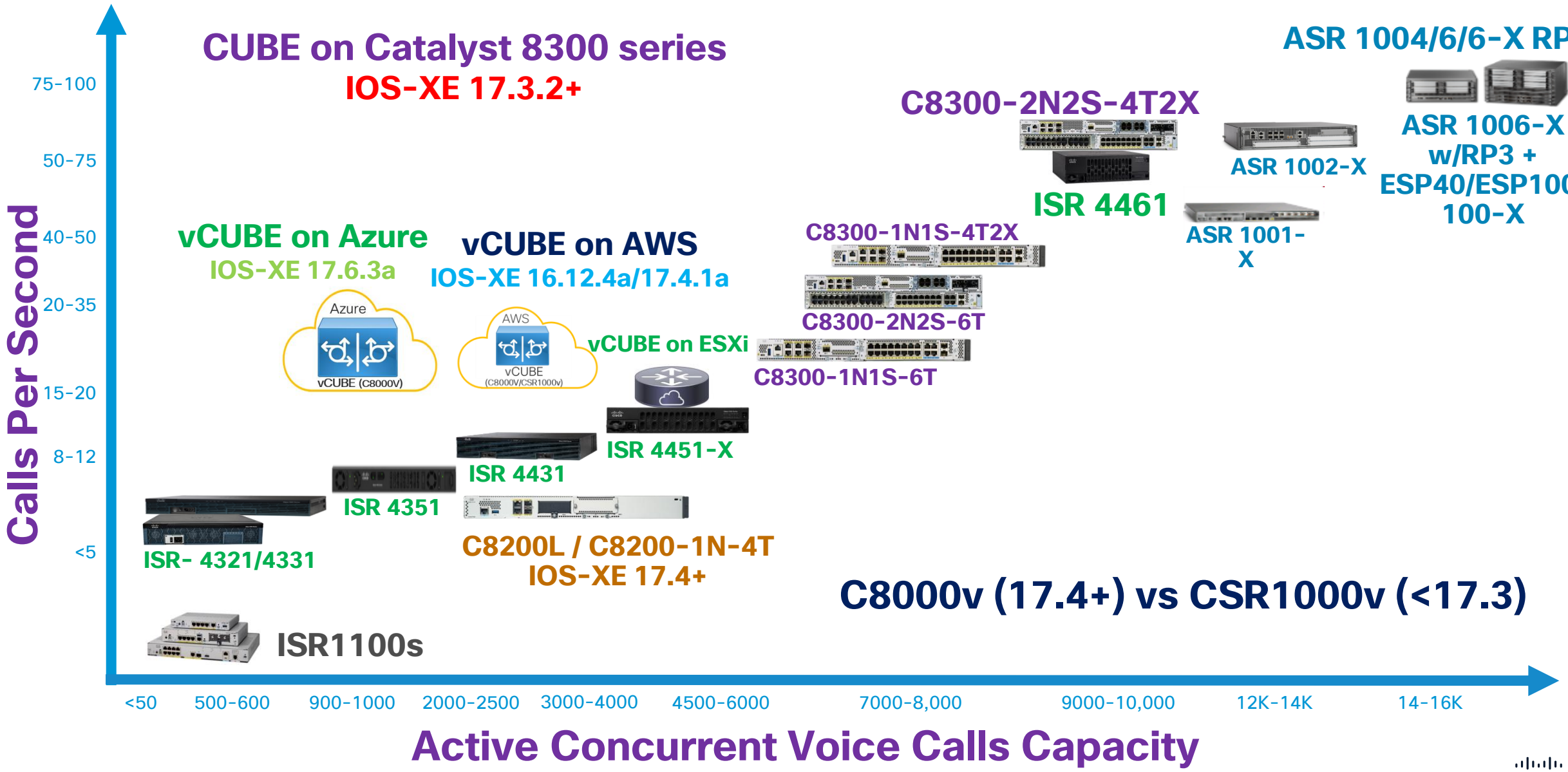
(Premises-based PSTN) Deployment



- 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.** Endpoints directly register to Webex Calling over the Internet.

Platforms for CUBE/vCUBE

CUBE (Enterprise) Product Portfolio [Not to Scale]



CUBE/IOS-XE Software Release Mapping

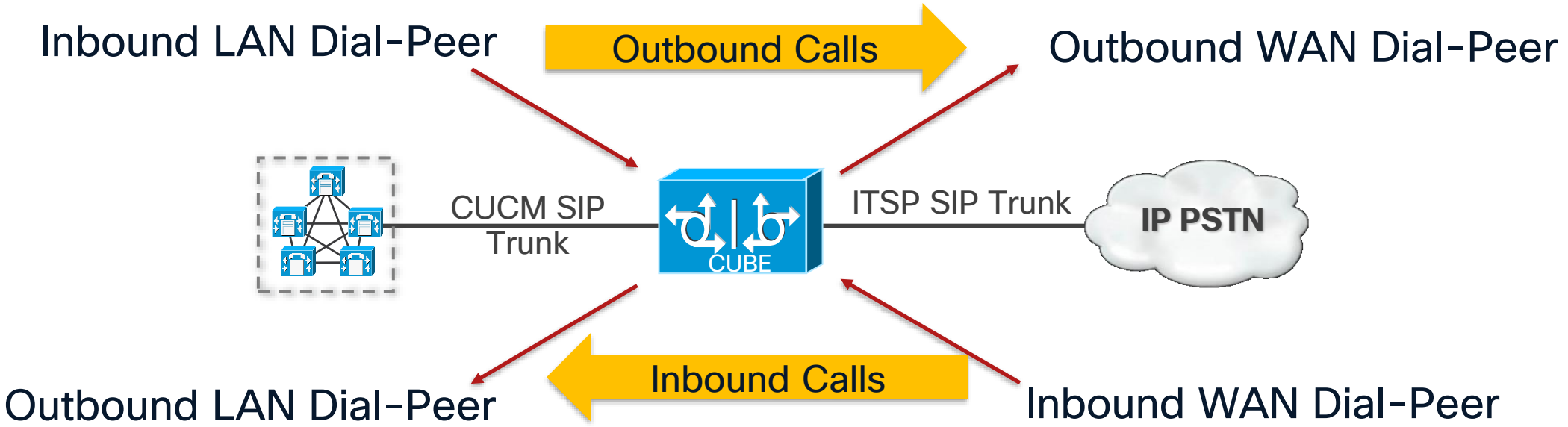
CUBE Version	Initial IOS-XE Release for this CUBE version and Release date		Subsequent IOS-XE Release for this CUBE version
14.4	17.6.1a	July 2021	17.6.8a
14.4	17.7.1a	Nov 2021	17.7.2
14.5	17.8.1a	March 2022	
14.6	17.9.1a	July 2022	17.9.7a
14.6	17.10.1a	Nov 2022	
14.6	17.11.1a	March 2023	
14.7	17.12.1a	July 2023	17.12.5b
14.8	17.13.1a	Nov 2023	
14.9	17.14.1a	March 2024	
14.10	17.15.1a	July 2024	17.15.3a
14.11	17.16.1a	Nov 2024	
TBD	17.18.1a	August 2025	

Last release for ISR4K except ISR4461

Dial-Peer Classifications

LAN and WAN Dial-Peers

- LAN Dial-Peers – Dial-peers that are facing towards the IP PBX for sending and receiving calls to & from the PBX. Should be bound to the LAN interface(s) of CUBE to ensure SIP/RTP is sourced from the LAN IP(s) of the CUBE.
- WAN Dial-Peers – Dial-peers that are facing towards the SIP Trunk provider for sending & receiving calls to & from the provider. Should be bound to WAN interface(s) of CUBE.



CUBE Configuration

Suggested CUBE Config Steps

Enable CUBE

`voice service voip` configuration, e.g. IP Trust list, SIP Parameters, Allow connection

Enable CUBE Application on the platform

1. Enable CUBE Application

```
voice service voip
 allow-connections sip to sip
```

→ By default IOS/IOS-XE voice devices do not allow an incoming VoIP leg to go out as VoIP

2. Configure any other global settings

```
voice service voip
   sip
     early-offer forced
     audio forced
```

3. Create a trusted list of IP addresses to prevent toll-fraud

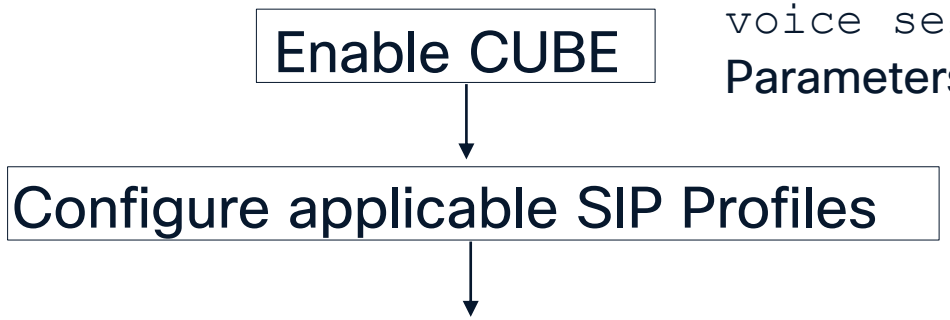
```
voice service voip
  ip address trusted list

  ipv4 66.77.37.2
```

→ Applications initiating signaling towards CUBE, e.g. CUCM, CVP,

→ Service Provider's SBC. IP Addresses from dial-peers with "session target ip" or session server group need not be added here

Suggested CUBE Config Steps



`voice service voip` configuration, e.g. IP Trust list, SIP Parameters, Allow connection

Regular expressions to modify SIP/SDP headers

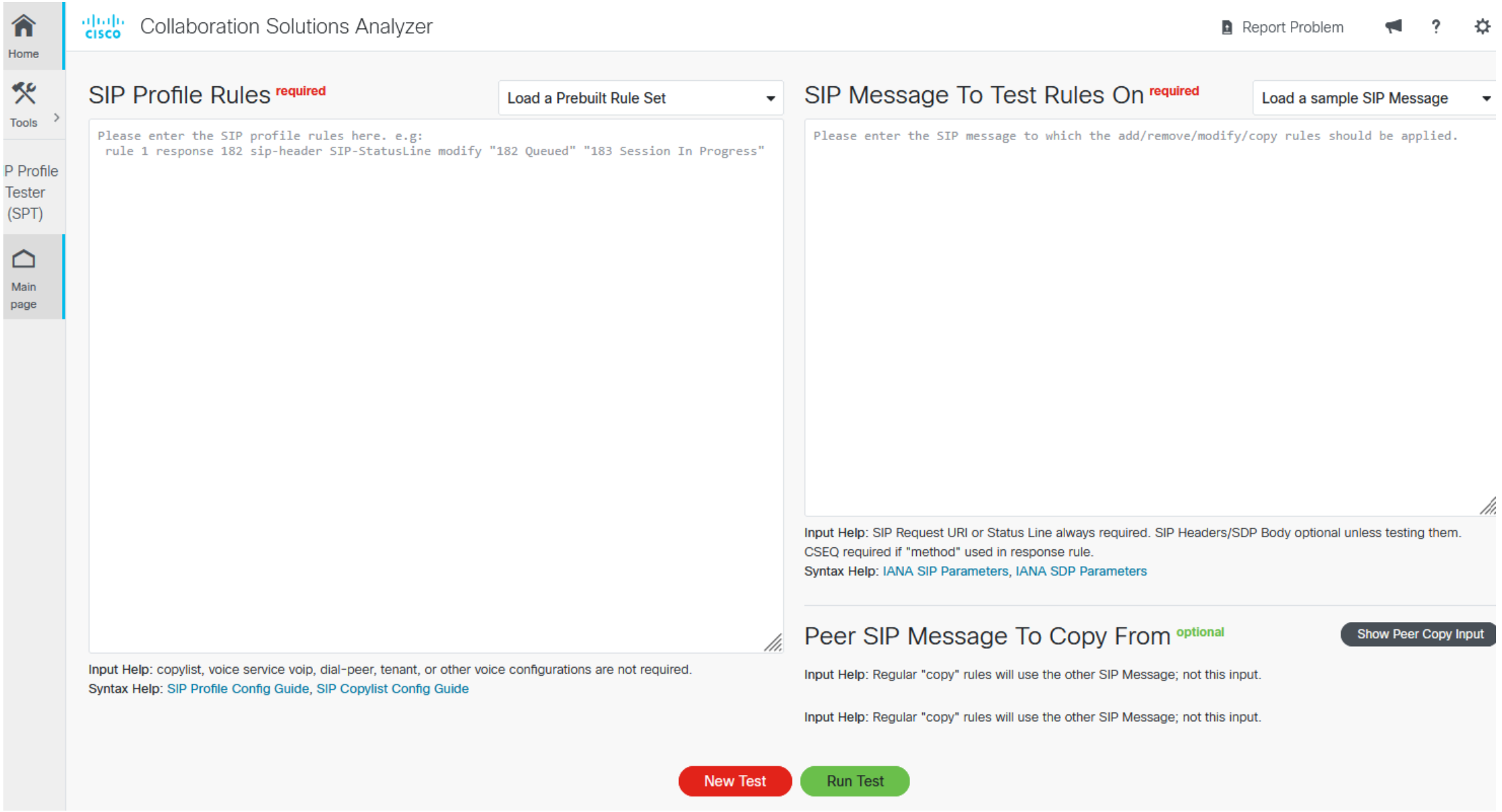
SIP Normalization

SIP profiles are a mechanism to normalize or customize SIP/SDP headers on CUBE to provide interop between incompatible devices

- SIP Profiles can be applied to both incoming and outgoing SIP messages on CUBE
- SIP Profiles can be applied
  - Globally (`voice service voip`)
  - On a dial-peer (`dial-peer voice <tag> voip`)
  - Tenant (`voice class tenant <tag>`)
- SIP Profiles are also supported on non-standard headers

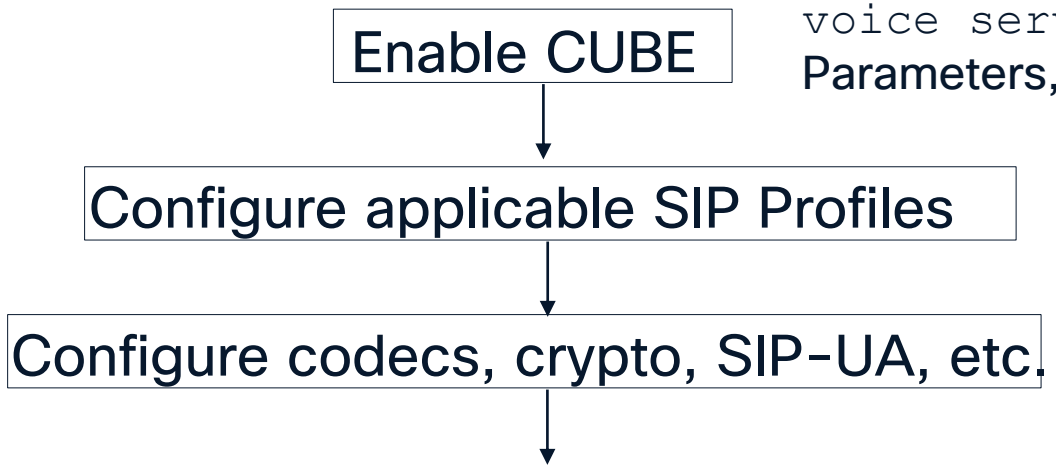
```
voice class sip-profiles 1
 rule 1 request INVITE sip-header Contact Modify "(.*)" "\1;temp=xyz"
 rule 2 request INVITE sip-header Supported Add "Supported: "
```

SIP Profile Test Tool available on Collaboration Solutions Analyzer



https://cway.cisco.com/csa-new/#/home

Suggested CUBE Config Steps



`voice service voip` configuration, e.g. IP Trust list, SIP Parameters, Allow connection

Regular expressions to modify SIP/SDP headers

`voice class codec, srtp-crypto`

Configure Codecs, Crypto, SIP-UA, etc.

```
voice class codec 1
 codec preference 1 opus
 codec preference 2 g711ulaw
```

- List of codecs to be supported on a call leg

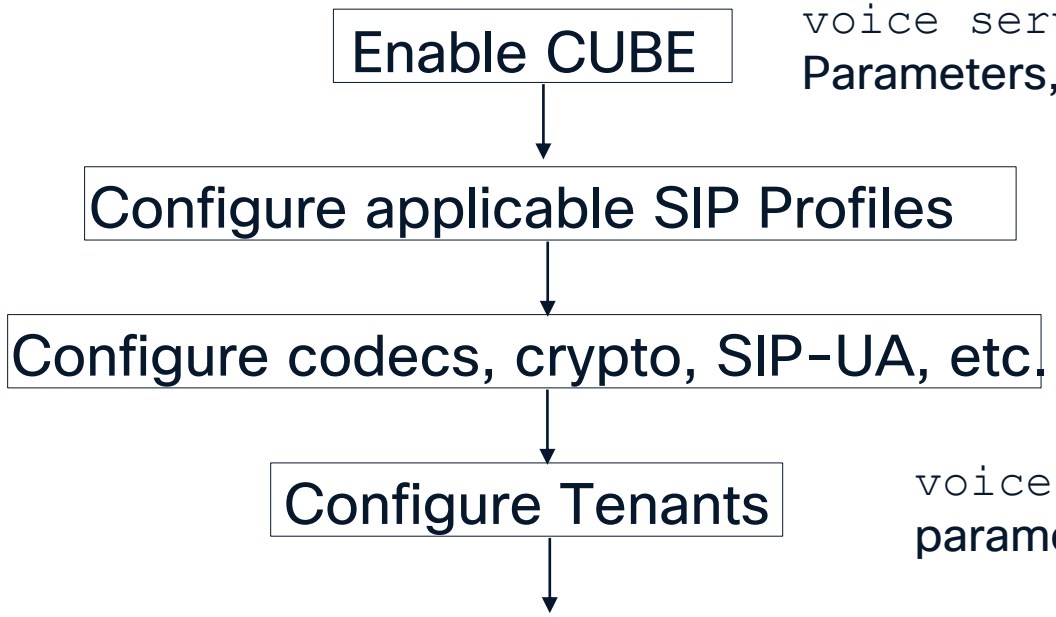
```
sip-ua
 transport tcp tls v1.2
 crypto signaling default trustpoint CUBE_CA_CERT
```

- TLS version, default trustpoint, etc.
- **TLS 1.3 supported from IOS-XE 17.15+**

```
voice class srtp-crypto 100
 crypto 1 AES_CM_128_HMAC_SHA1_80
```

- Used to set the crypto cipher for TLS based SIP Trunks

Suggested CUBE Config Steps



`voice service voip` configuration, e.g. IP Trust list, SIP Parameters, Allow connection

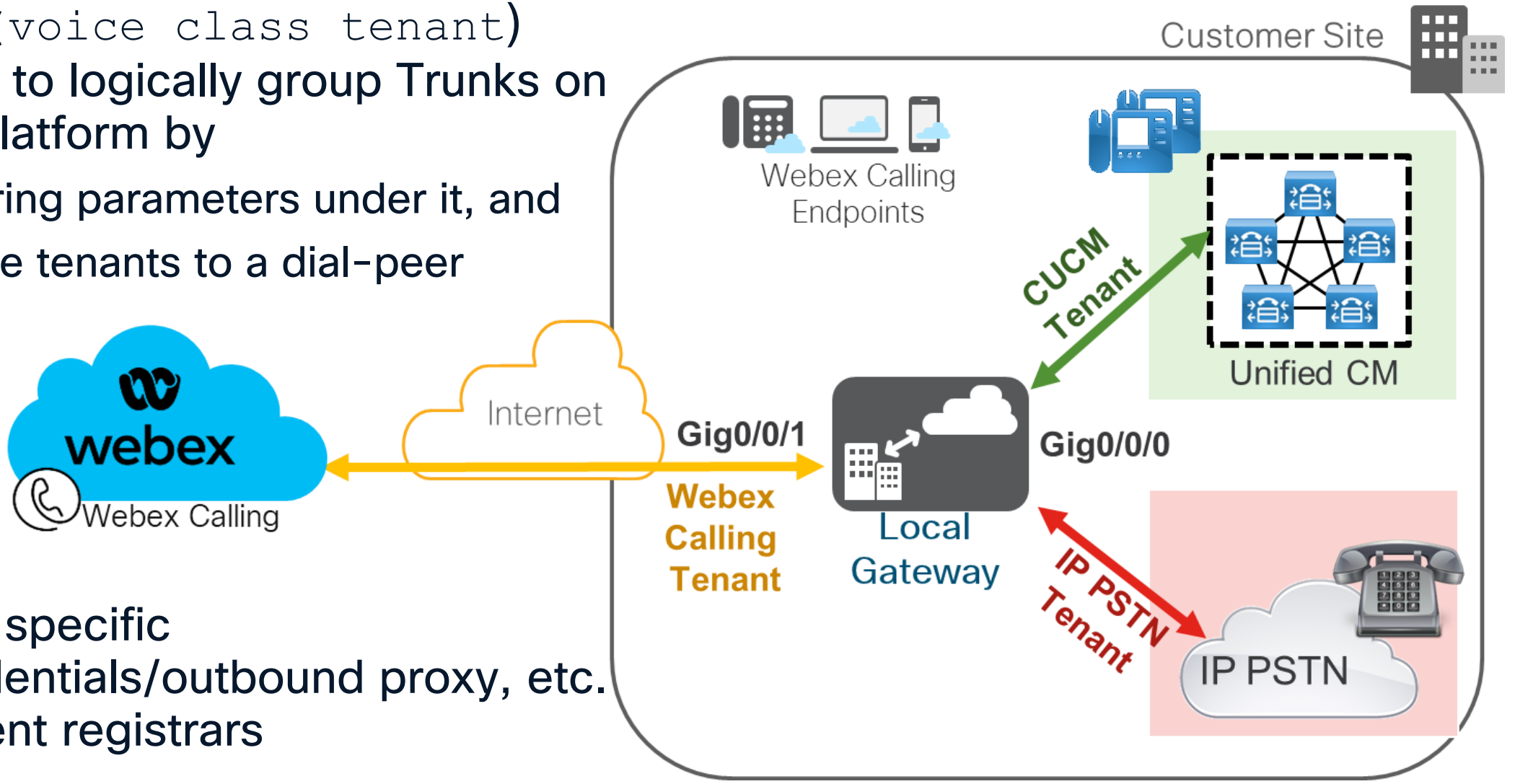
Regular expressions to modify SIP/SDP headers

`voice class codec, srtp-crypto`

`voice class tenant,` Individual Trunks with respective parameters

Logical group of Trunks with Tenants

- Tenants (`voice class tenant`) allow you to logically group Trunks on a CUBE platform by
  - Configuring parameters under it, and
  - Apply the tenants to a dial-peer



- Provision specific bind/credentials/outbound proxy, etc. for different registrars

Configuring Voice Class Tenant

- Configure voice class tenant

```
voice class tenant 1
  registrar 1 ipv4:10.64.86.35:9052 expires 3600
  credentials username aaaa password 7 06070E204D realm aaaa.com
  bind control source-interface GigabitEthernet0/0
  bind media source-interface GigabitEthernet0/0
  copy-list 1
  outbound-proxy ipv4:10.64.86.35:9055
  early-offer forced
```

Add new voice class tenant

- Apply tenant to the desired dial-peer

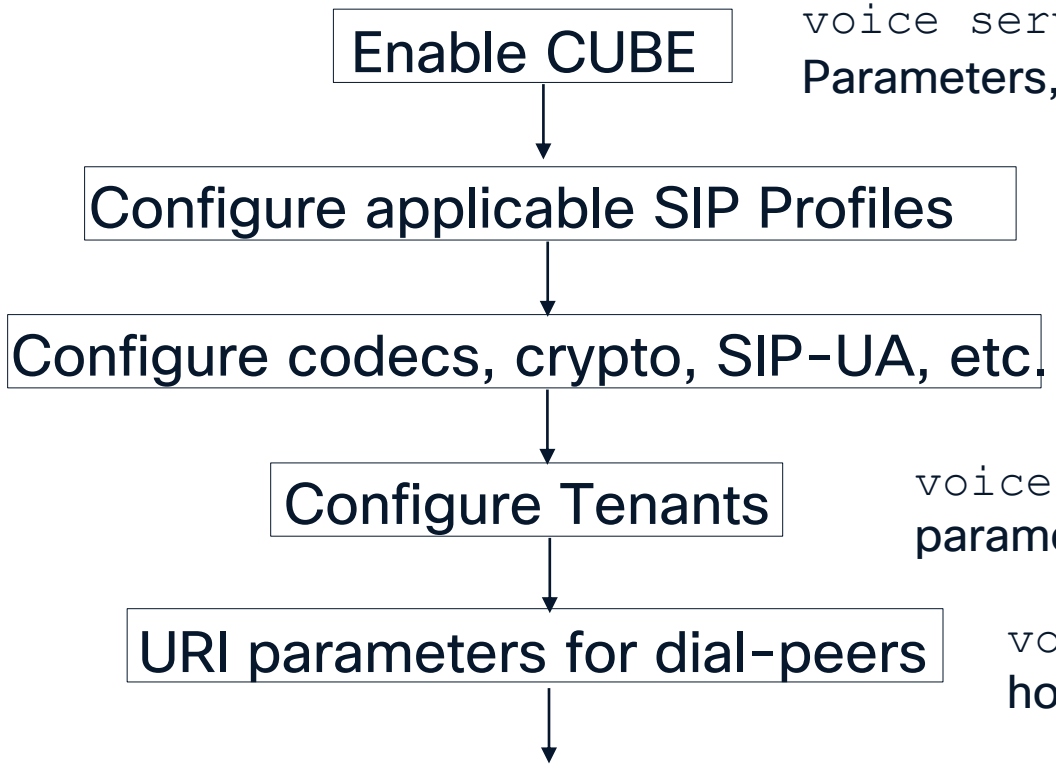
```
dial-peer voice 1 voip
 destination-pattern 111
 session protocol sipv2
 session target ipv4:10.64.86.35:9051
 session transport udp
 voice-class sip tenant 1
```

Configuration Preference Order

1. dial-peer (overrides tenant and global configs)
2. tenant (overrides the global config)
3. global (voice service voip OR sip-ua)

Apply Tenant to a Dial-peer

Suggested CUBE Config Steps



`voice service voip` configuration, e.g. IP Trust list, SIP Parameters, Allow connection

Regular expressions to modify SIP/SDP headers

`voice class codec, srtp-crypto`

`voice class tenant`, Individual Trunks with respective parameters

`voice class uri` match based on IP address, hostname, etc.

Voice class URIs

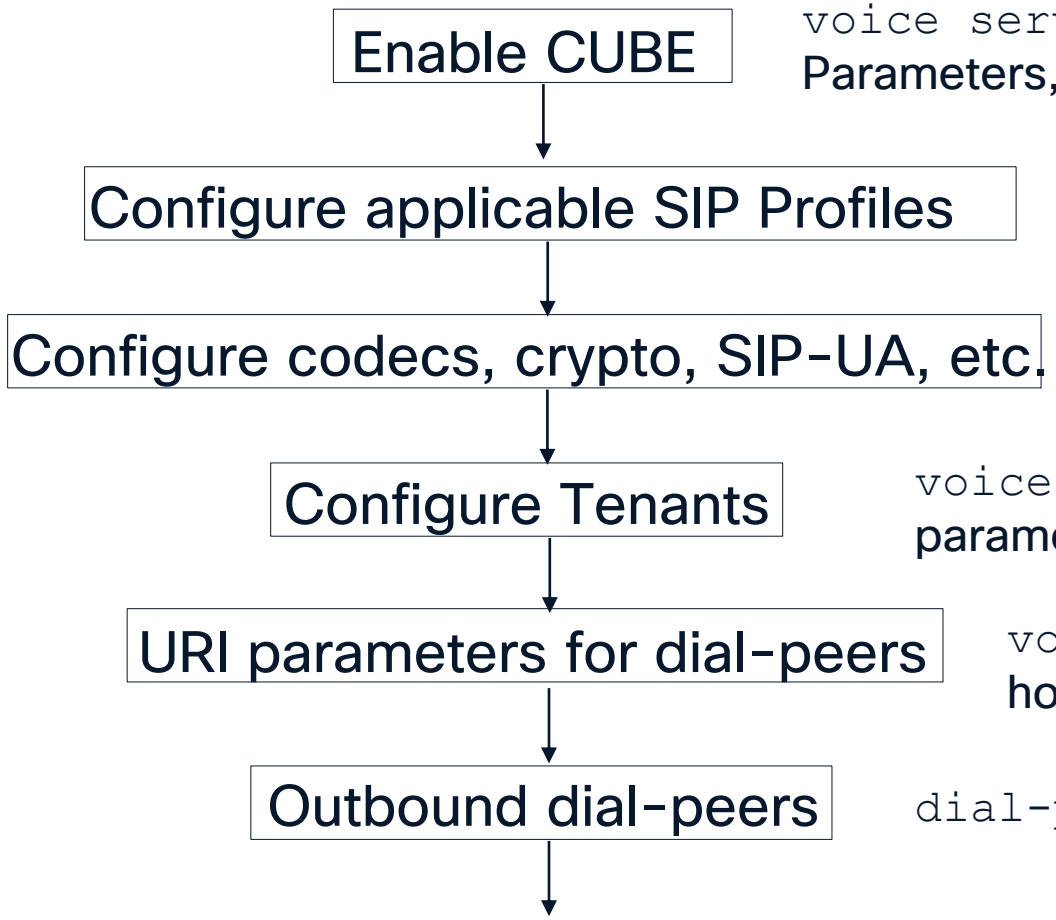
```
voice class uri 1
 host ipv4:10.1.1.1
 host dns:cisco.com
 pattern 654321@10.2.1.1
 user-id 654321
```

- allows you to configure the selection of inbound dial peers by matching parts of the URI sent by a remote SIP entity
- different parts of the URI like username, IP address, and DNS can be specified

Received:

```
INVITE sip:654321@10.2.1.1 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:5060;x-route-tag="cid:orange@10.1.1.1";;branch=z9hG4bK-
23955-1-0
From: "555" <sip:555@10.1.1.1:5060>;tag=1
To: ABC <sip:654321@10.2.1.1:5060>
Call-ID: 1-23955@10.1.1.1
CSeq: 1 INVITE
Contact: sip:555@10.1.1.1:5060
```

Suggested CUBE Config Steps



`voice service voip` configuration, e.g. IP Trust list, SIP Parameters, Allow connection

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`voice class codec, srtp-crypto`

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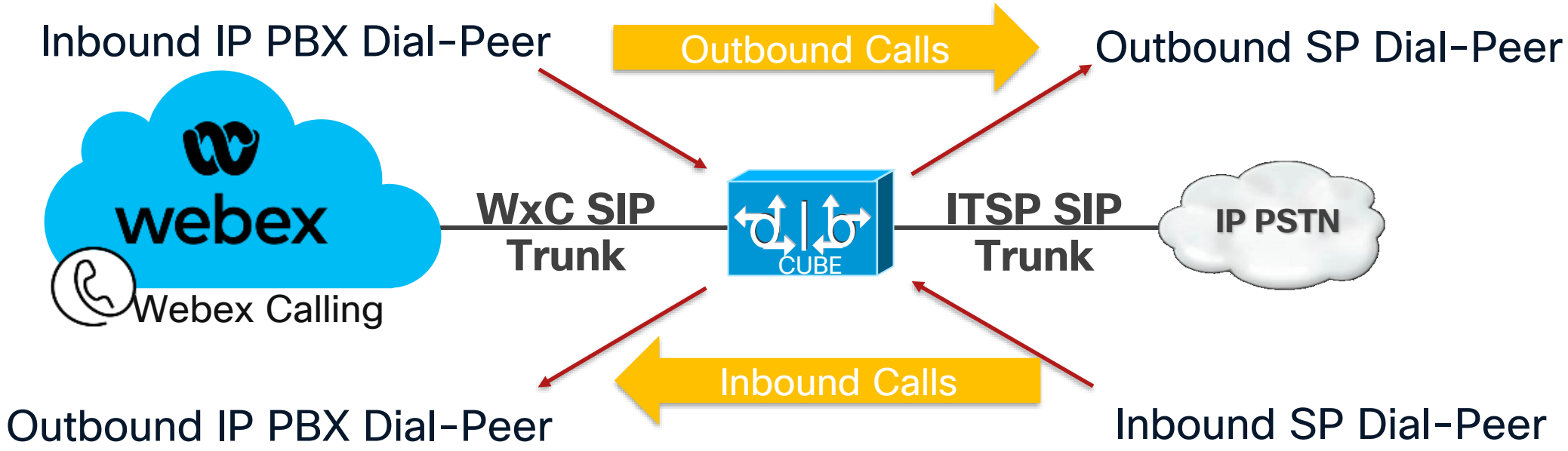
`voice class uri` match based on IP address, hostname, etc.

`dial-peer voice` to deliver outbound call legs

Understanding Dial-Peer Matching and Classification

IP PBX and ITSP Dial-Peers

- **IP PBX Dial-Peers** – Dial-peers that are facing towards the IP PBX for sending and receiving calls to and from the PBX (e.g. Webex Calling, CUCM).
- **Service Provider (ITSP) Dial-Peers** – Dial-peers that are facing towards the SIP Trunk provider for sending and receiving calls to and from the provider.



Outbound Dial-Peer Configuration

SIP Service Provider facing OUTBOUND dial-peer

```
dial-peer voice 101 voip
 description Outgoing dial-peer to IP PSTN
 destination-pattern +1T
 session protocol sipv2
 session target ipv4:A.B.C.D
 voice-class codec 1
 voice-class sip tenant 1
 dtmf-relay rtp-nte
 no vad
```

Meaningful description on the dial-peer

Phone number match criteria

Destination IP Address to deliver the call leg. E.g., SIP service provider

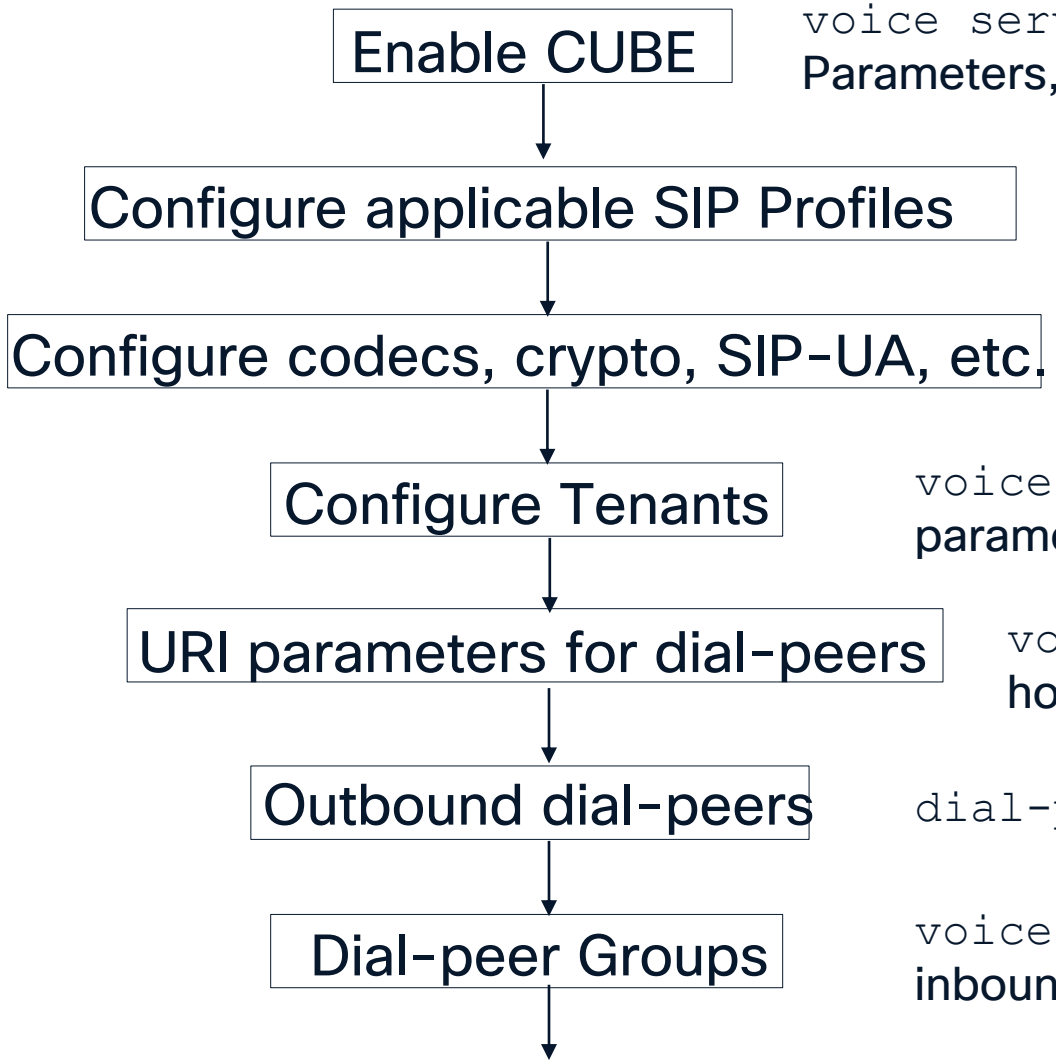
Codec List supported for this call leg

Inherit parameters from the tenant

Outbound SIP Dial-Peer Matching

Priority	Match Criteria	Dial-peer Commands
1	Dial-Peer Group Dial-Peer	destination dpg <dpg-tag> (DPG configured on inbound dial-peer)
2	Dial-Peer Provision Policy URI	destination uri-from <uri-tag> destination uri-to <uri-tag> destination uri-via <uri-tag> destination uri-diversion <uri-tag> destination uri-referred-by <uri-tag> (DPP configured on inbound dial-peer)
3	ILS Route String	destination route-string <route-string-tag>
4	URI and Carrier-ID	destination uri <uri-tag> AND carrier-id target <string>
5	Called Number & Carrier-ID	destination-pattern <number-string> AND carrier-id target <string>
6	URI	destination uri <uri-tag>
7	Called Number	destination-pattern <DNIS-number> destination e164-pattern-map <pattern-map-number> dnis-map <dnis-map-number>
8	Calling Number	destination calling e164-pattern-map <pattern-map-number>

Suggested CUBE Config Steps



`voice service voip` configuration, e.g. IP Trust list, SIP Parameters, Allow connection

Regular expressions to modify SIP/SDP headers

`voice class codec, srtp-crypto`

`voice class tenant,` Individual Trunks with respective parameters

`voice class uri` match based on IP address, hostname, etc.

`dial-peer voice` to deliver outbound call legs

`voice class dpg` to invoke outbound dial-peer(s) from inbound dial-peer

Destination Dial-peer Group

```
voice class dpg 10000
  description Voice Class DPG for SJ
  dial-peer 1002 preference 1
  dial-peer 1003
!
dial-peer voice 100 voip
 description Inbound DP
 incoming called-number 1341
 destination dpg 10000
```

```
dial-peer voice 786 voip
 destination-pattern 1341
 session protocol sipv2
 session target ipv4:10.1.1.1
!
dial-peer voice 1002 voip
 destination-pattern 3333
 session protocol sipv2
 session target ipv4:10.1.1.2
!
dial-peer voice 1003 voip
 destination-pattern 4444
 protocol sipv2
 4:10.1.1.3
```

3. Outbound DP is selected

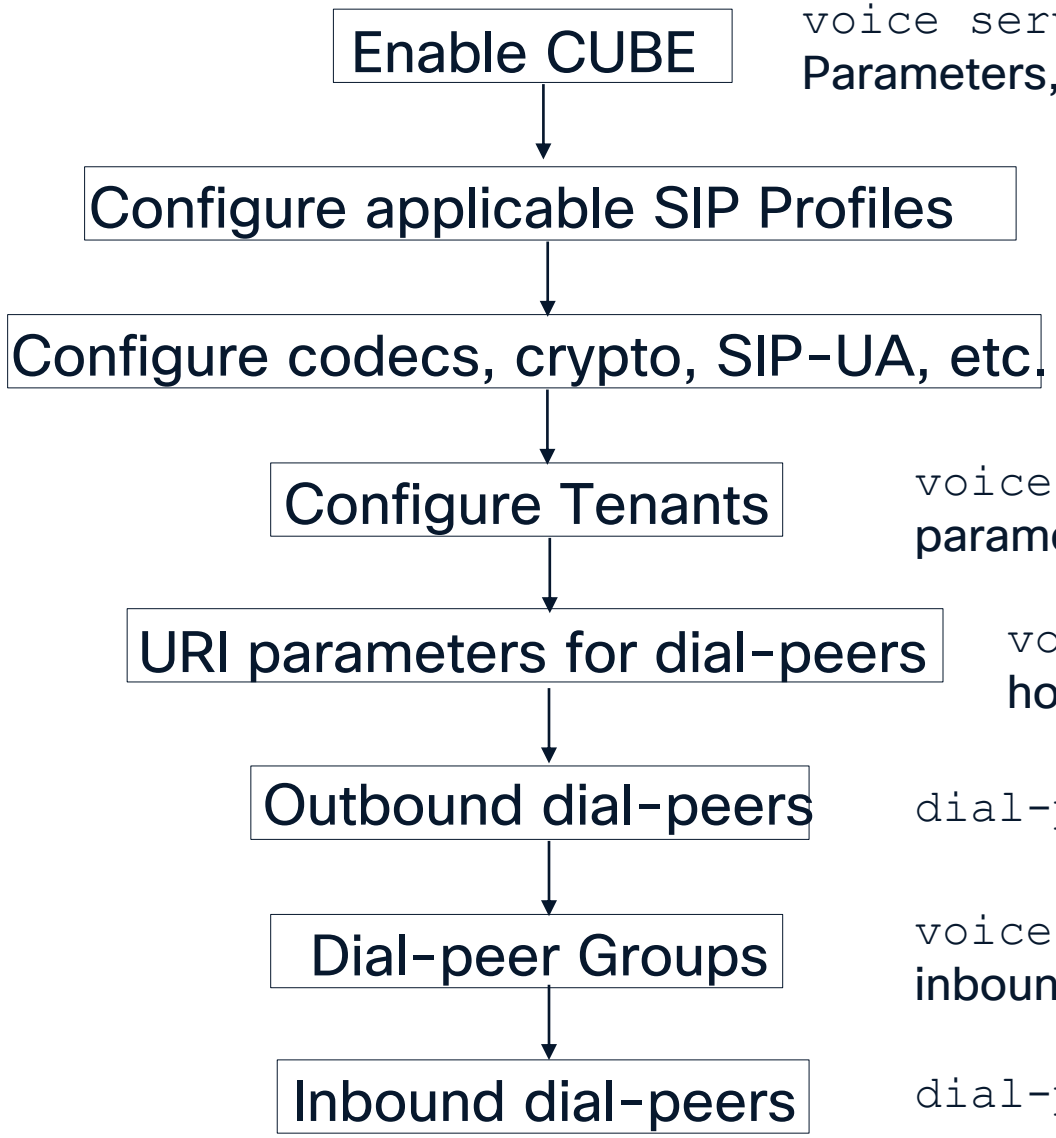
Received:
INVITE sip:1341@<CUBE-IP> SIP/2.0

1. Incoming Dial-peer is first matched

Sent:
INVITE sip:1341@10.1.1.3 SIP/2.0

2. the DPG associated with INBOUND DP is selected

Suggested CUBE Config Steps



`voice service voip` configuration, e.g. IP Trust list, SIP Parameters, Allow connection

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`voice class uri` match based on IP address, hostname, etc.

`dial-peer voice` to deliver outbound call legs

`voice class dpg` to invoke outbound dial-peer(s) from inbound dial-peer

`dial-peer voice` To accept incoming call legs

Inbound Dial-Peer Configuration

SIP Service Provider facing INBOUND dial-peer

```
dial-peer voice 786 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
```

Meaningful description on the dial-peer

Bypass default outbound dial-peer matching and invoke outbound dial-peer(s) of this DPG

Inbound match criteria

Inherit parameters from the tenant

Codec List supported for this call leg

Inbound SIP Dial-Peer Selection Preference

Preference	Match Criteria	Dial-peer Commands
1	URI	incoming uri via <uri-tag>
2		incoming uri request <uri-tag>
3		incoming uri to <uri-tag>
4		incoming uri from <uri-tag>
5	Called Number	incoming called-number <number-string> incoming called e164-pattern-map <pattern-map-number>
6	Calling Number	incoming calling e164-pattern-map <pattern-map-number> answer-address <number-string>
7	Destination-pattern (ANI)	destination-pattern <number-string>
8	Carrier-ID	carrier-id source <string>

Suggested CUBE Config Steps

Enable CUBE

`voice service voip` configuration, e.g. IP Trust list, SIP Parameters, Allow connection

Configure applicable SIP Profiles

Regular expressions to modify SIP/SDP headers

Configure codecs, crypto, SIP-UA, etc.

`voice class codec, srtp-crypto`

Configure Tenants

`voice class tenant,` Individual Trunks with respective parameters

URI parameters for dial-peers

`voice class uri` match based on IP address, hostname, etc.

Outbound dial-peers

`dial-peer voice` to deliver outbound call legs

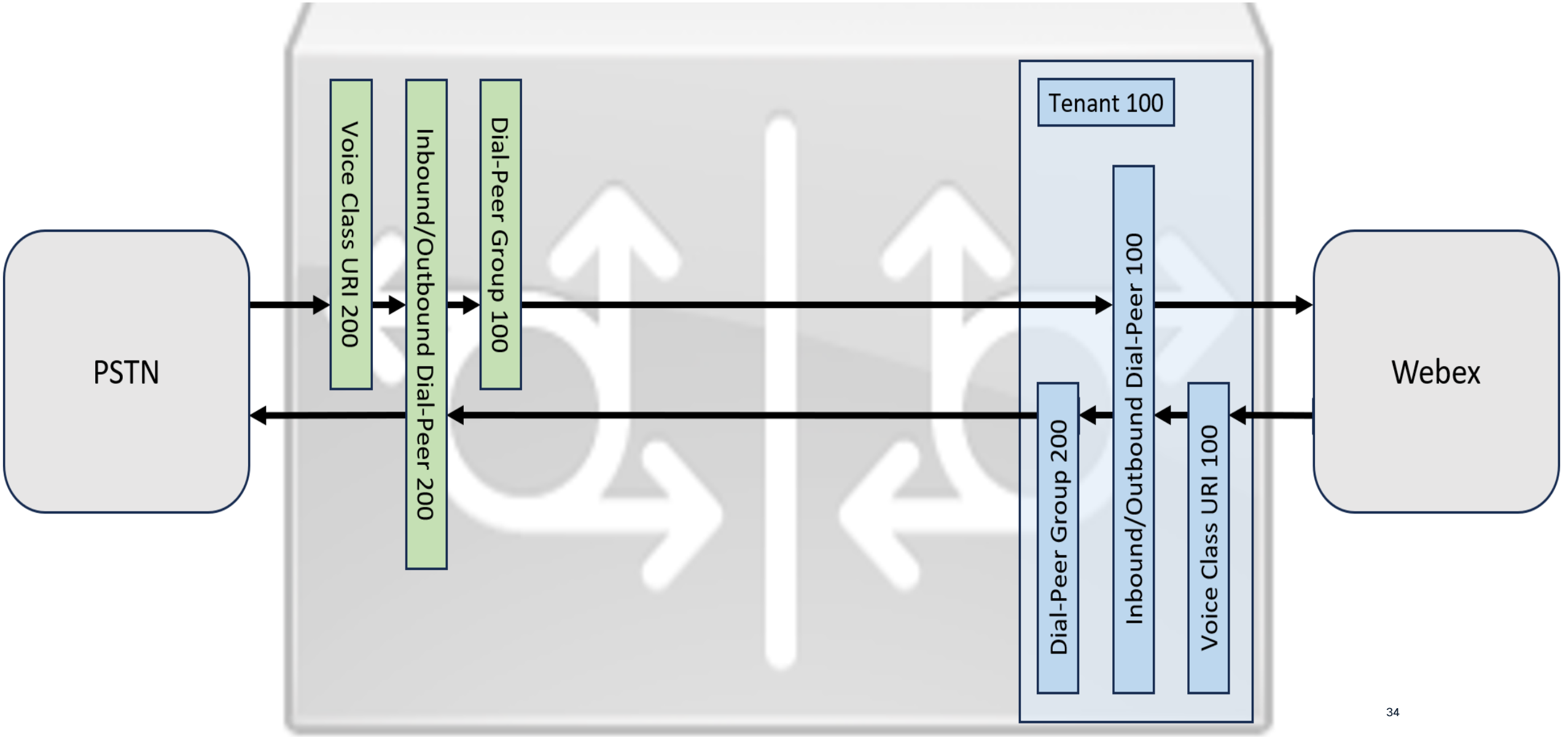
Dial-peer Groups

`voice class dpg` to invoke outbound dial-peer(s) from inbound dial-peer

Inbound dial-peers

`dial-peer voice` To accept incoming call legs

Grouping trunks with voice class tenants



Call Admission Control (CAC)

- Call processing capacity for any CUBE instance will be influenced by several considerations, including software version, features configured and the platform itself
- To ensure that calls continue to be processed reliably, configure Call Admission Control as follows to reject calls when use of system resources exceeds 80%[1]. Refer to the CUBE Configuration Guide for further details

```
enable
conf t
 call threshold global cpu-avg low 75 high 80
 call threshold global total-mem low 75 high 80
 call treatment
end
```

[1] For an ISR4461, starting IOS-XE 17.3.2 or later, testing benchmark is at 80% memory utilization. Configure the memory threshold CAC accordingly.

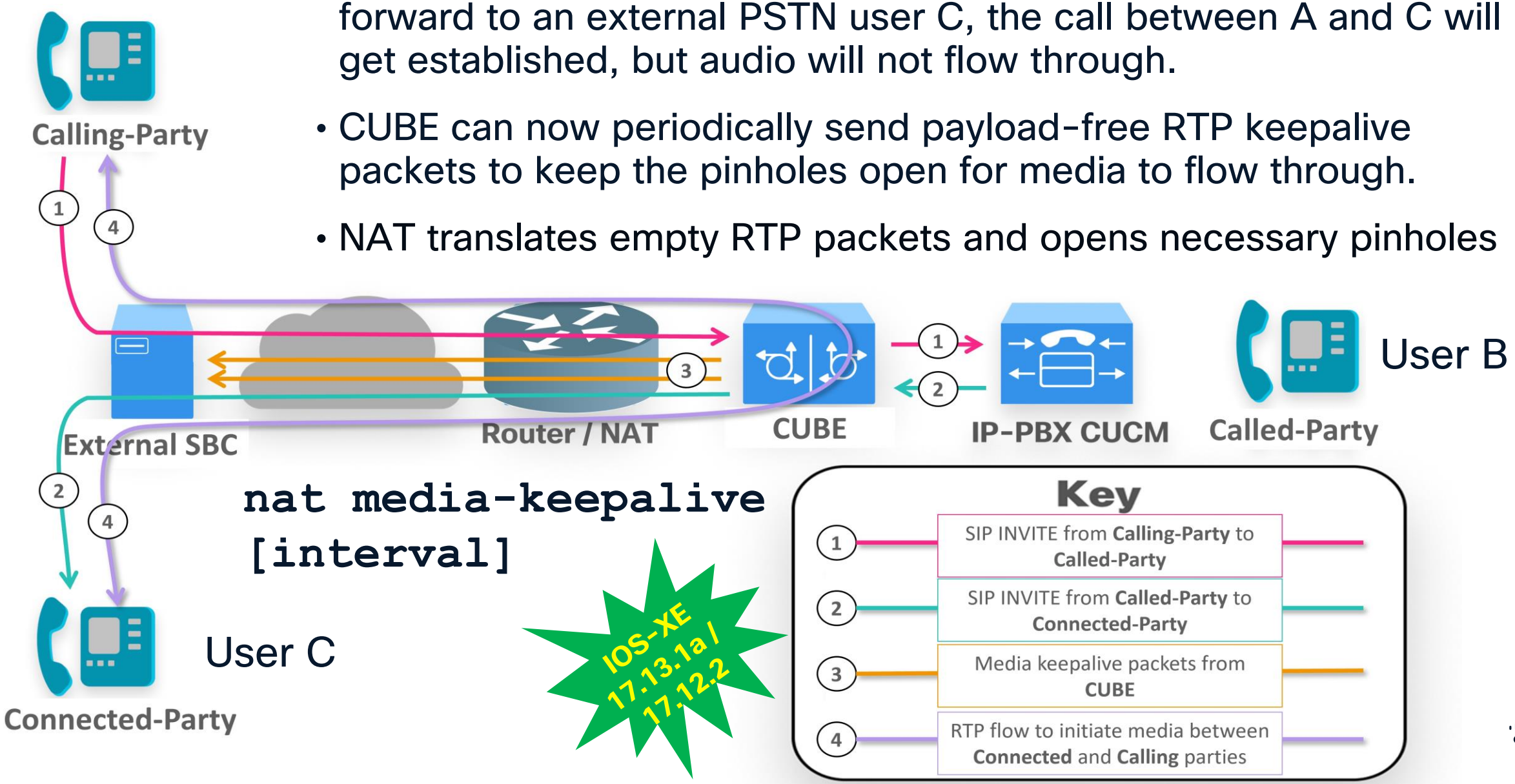
```
call threshold global total-mem low 80 high 85
```

Version 14 Updates

NAT Traversal using RTP keepalives

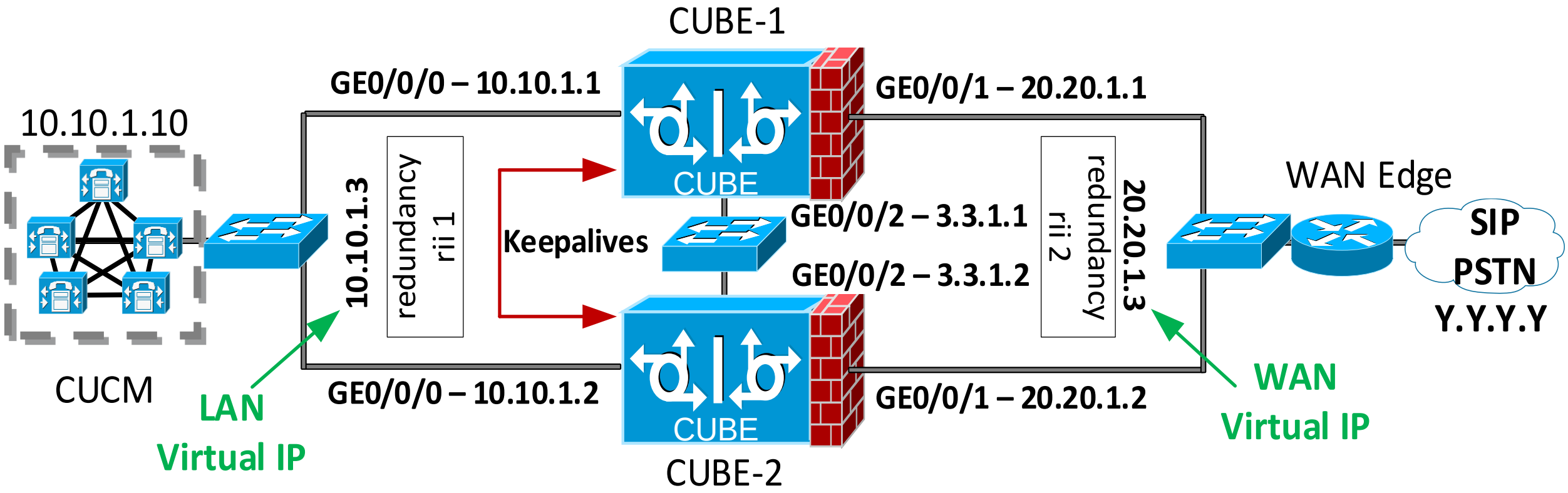
Introducing NAT Traversal using RTP keepalives

- When a PSTN user A calls an enterprise user B who has setup call forward to an external PSTN user C, the call between A and C will get established, but audio will not flow through.
- CUBE can now periodically send payload-free RTP keepalive packets to keep the pinholes open for media to flow through.
- NAT translates empty RTP packets and opens necessary pinholes



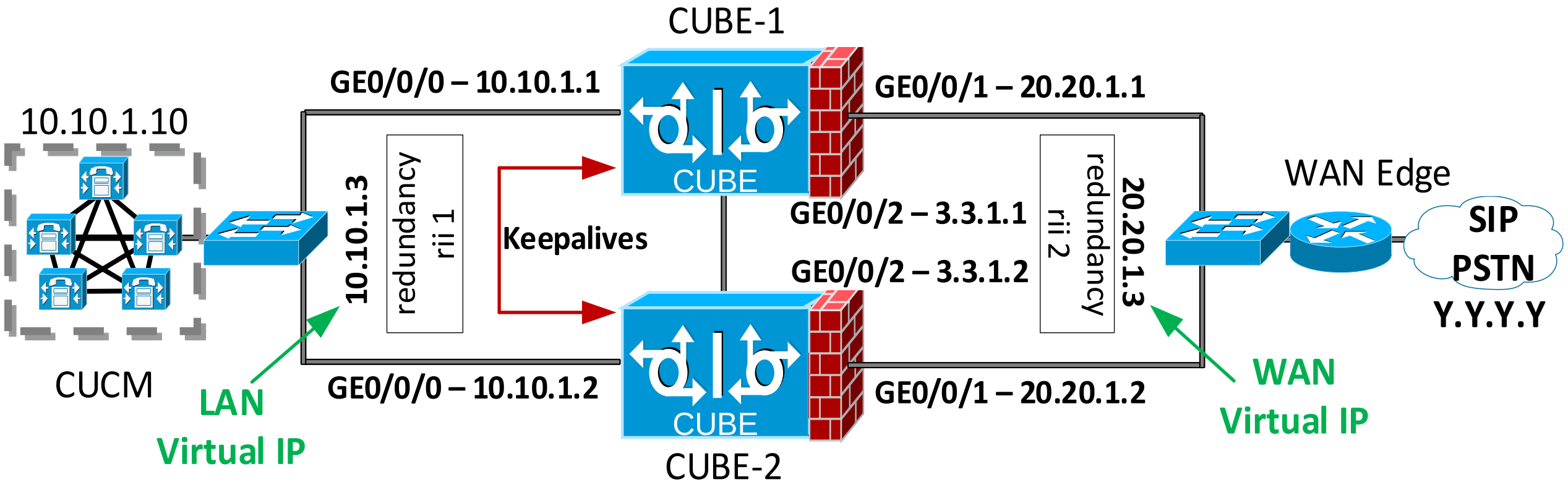
CUBE High Availability (HA) Updates

CUBE HA for Call Preservation



- RG Control/Data interface (G0/0/2) had to be connected via a physical switch
- It can now be connected via a back-to-back cable

Crossover cable for Keepalive interface



- RG Control/Data interface (G0/0/2) had to be connected via a physical switch
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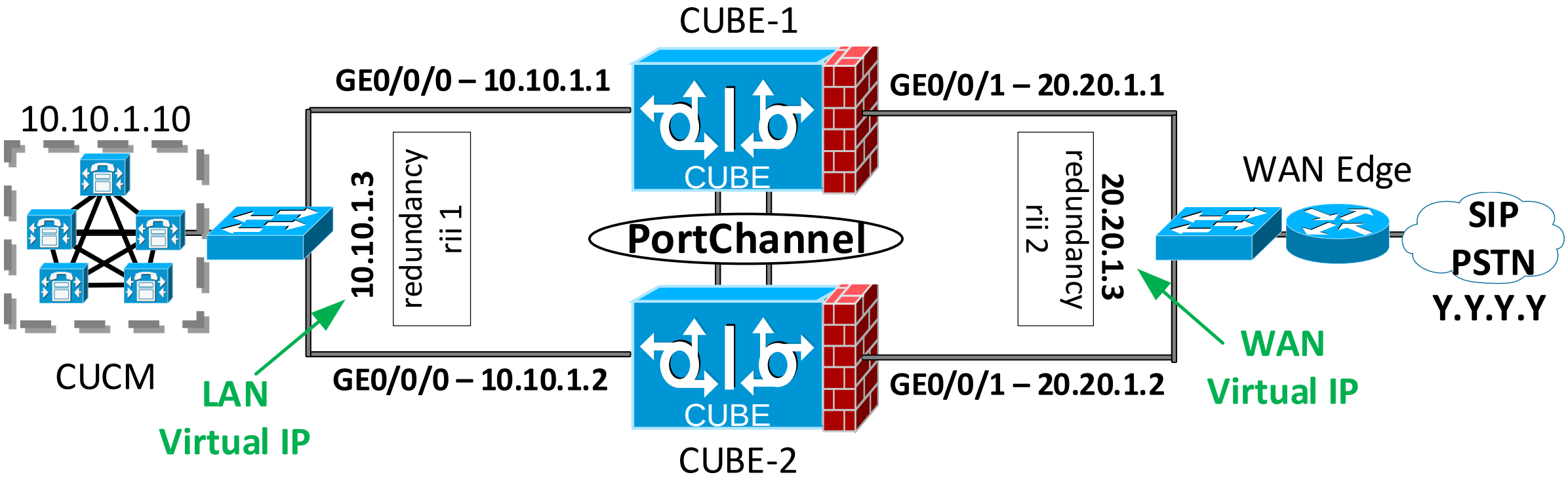
Port channel for keepalive interfaces



- RG Control/Data interface (G0/0/2) had to be connected via a physical switch
- It can now be connected via a back-to-back cable

DNS SRV based load balancing on Dial-peers

Introducing DNS SRV based OPTIONS ping

OPTIONS ping for DNS SRV hosts

- **IOS-XE 17.9.1a or later**
  - CUBE attempts OPTION keepalives with the all the hosts to determine their status and use it for routing the calls.
  - This feature can be used by configuring a dial-peer target with an FQDN that resolves to a set of DNS SRV records.
  - A DNS SRV lookup results in multiple targets (A records), each with its own weight, priority.
  - CUBE performs DNS lookup against each record (obtained through SRV lookup) to determine the IPv4/IPv6 addresses and triggers OOD SIP Option message to each destination to monitor the status.

Pre-requisites

- FQDN must be used as `session target` in dial-peers
  - Example: `session target dns:webex.com`
- DNS server with SRV records
- Configuring `voice-class sip options-keepalive profile <tag>` under dial-peer is must

```
CUBE(config)# voice class sip-options-keepalive 1
CUBE(config-class)#description keepalive-webex
CUBE(config-class)#transport tcp
CUBE(config-class)#down-interval 15
CUBE(config-class)#up-interval 5
CUBE(config-class)#retry 1

dial-peer voice 786 voip
 session target dns:webex.com
 session transport tcp
 voice-class sip options-keepalive profile 1
```

DNS SRV Based Option Ping : Status of Hosts

- Each SRV node associated with the DNS target is monitored. It provides the option of marking a dial-peer and host's status as below:
  - **Active** – When all the hosts from DNS SRV resolution are reachable and send 200 OK for the OPTION message
  - **Busyout** (inactive) – When all the hosts from the DNS SRV resolution DO NOT respond or send an Error response
  - **Partial** – This is a new state, if one of the hosts from the DNS SRV resolution fails to send a 200 OK response
- CUBE will check the status of the nodes when routing a call. If the node is in a busyout state, CUBE will not send the call to that node and move to the next node in the host list

Feature Operation

Dial-peer configured with session target DNS (FQDN)

```
dial-peer voice 786 voip
 session target dns:webex.com
 voice-class sip options-keepalive profile 1
```

DNS

Is SIP OPTION keepalive profile configured?

NO → Feature will not work

YES

DNS SRV query for webex.com →

← Successful DNS response with host list

DNS lookup gets resolved to 3 hosts
example1.webex.com
example2.webex.com
example3.webex.com

A Record resolution for the hosts →

← IP Address resolution

IP Address Resolution of all hosts

Send OPTIONs to all hosts →

OPTIONS → example1.webex.com
← 200 OK **Active**

OPTIONS → example2.webex.com
← 200 OK **Active**

OPTIONS → example3.webex.com
Busyout

Is OPTIONs successful for ALL hosts?

YES → Dial-peer state Active

NO

Is OPTIONs successful to ANY host?

YES → Dial-peer state Partial

NO → Dial-peer state Busyout

One node is down

Dial-peer show output - Active

`show dial-peer voip keepalive status <tag> | <tenant>`

- New CLI introduced from 17.9.1, this will list all the hosts that are being monitored using the sip options keepalive profile.

- **Case 1 - Dial-peer is active as all the hosts are reachable**

```
CUBE#show dial-peer voip keepalive status 786

 TAG        TENANT    DESTINATION                OOD-SessID    PRI    WT     STATUS
 786        -         dns:webex.com                                                active
                      example3.webex.com         46            10     50     active
                      ipv4:10.64.86.70:5880
                      example2.webex.com         45            10     50     active
                      ipv4:10.65.105.59:5060
                      example1.webex.com         44            10     50     active
                      ipv4:10.65.105.58:5060
```

Dial-peer show output - Partial

`show dial-peer voip keepalive status <tag> | <tenant>`

- **Case 2 - Dial-peer is partially active as one of the hosts is not reachable**

```
CUBE#show dial-peer voip keepalive status 786

 TAG        TENANT    DESTINATION                 OOD-SessID    PRI    WT      STATUS
 786        -         dns:webex.com                                                partial
                      example3.webex.com          46            10     50      busyout
                      ipv4:10.64.86.70:5880
                      example2.webex.com          45            10     50      active
                      ipv4:10.65.105.59:5060
                      example1.webex.com          44            10     50      active
                      ipv4:10.65.105.58:5060
```

Dial-peer show output - Busyout

`show dial-peer voip keepalive status <tag> | <tenant>`

- **Case 3 - Dial-peer is busyout as all the hosts are not reachable**

```
CUBE#show dial-peer voip keepalive status 786
```

TAG	TENANT	DESTINATION	OOD-SessID	PRI	WT	STATUS
786	-	dns:webex.com				**busyout**
		example3.webex.com ipv4:10.64.86.70:5880	46	**10**	**50**	busyout
		example2.webex.com ipv4:10.65.105.59:5060	45	**10**	**50**	busyout
		example1.webex.com ipv4:10.65.105.58:5060	44	**10**	**50**	busyout

OPTIONs Keepalive profile show output

```
CUBE#show voice class sip-options-keepalive 1
Voice class sip-options-keepalive: 1 AdminStat: Up
 Description: keepalive-webex
 Transport: tcp Sip Profiles: 0
 Interval(seconds) Up: 5 Down: 15
 Retry: 1
  Peer Tag    Server Group OOD SessID  OOD Stat    IfIndex
  --------    ------------ ----------  --------    -------
  786                                  None          31
  OOD SessID: 46           OOD Stat:   Busy
    Target: ipv4:10.64.86.70:5880
    Transport: tcp          Sip Profiles: 0
  OOD SessID: 47           OOD Stat:   Active
    Target: ipv4:10.65.105.59:5060
    Transport: tcp          Sip Profiles: 0
  OOD SessID: 48           OOD Stat:   Active
    Target: ipv4:10.65.105.58:5060
    Transport: tcp          Sip Profiles: 0
```

Dial-peer voice summary show output

```
CUBE#show dial-peer voice summary

dial-peer hunt 0

                    AD                                      PRE PASS SESS-SER-GRP\  OUT
TAG     TYPE   MIN  OPER PREFIX DEST-PATTERN    FER THRU SESS-TARGET    STAT PORT   KEEPALIVE
3001    voip   down down                        0   syst
3002    voip   down down         6022           0   syst ipv4:10.65.105.25
12851   voip   up   up           +48203577....$  0   syst ipv4:10.48.53.54           busyout
786     voip   up   up           50785          0   syst dns:webex.com              partial
 For server-grp details please execute command:show voice class server-group <tag_id>
 To see complete session target for ipv6 use 'sh running-config | section dial-peer <tag>
```

DNS SRV based Call Routing

DNS SRV Based Call Routing

- The usage of DNS SRV as the target for CUBE helps in load balancing of the outbound SIP call traffic across a trunk.
- CUBE distributes calls across the SRVs based on the **priority, weight,** and **status (only active hosts) of the DNS SRV records**.
- If the priority and weight are the same, then the node will be selected in round-robin fashion.
- If CUBE receives a 503 response or no response for the INVITE, CUBE then marks that node as “Busyout” and attempts the call on the next node that is marked as active. The call is rejected if CUBE does not receive any response from any of the elements.

Call distribution based on DNS SRV lookup

Case 1 : Record routes with same priority and same weight

Configuration:

```
_sip._tcp.example1.webex.com    524   IN    SRV   10  50  5060 example1.webex.com
_sip._tcp.example2.webex.com    524   IN    SRV   10  50  5060 example2.webex.com
_sip._tcp.example3.webex.com    524   IN    SRV   10  50  5880 example3.webex.com
```

Total calls:- 3284

Call Counts on all 3 User Agent Server

example1.webex.com : 1083

example2.webex.com : 1099

example3.webex.com : 1102

Call distribution based on DNS SRV lookup

Case 2 : Record routes with same priority but different weights

Configuration:

```
_sip._tcp.example1.webex.com    524   IN    SRV   10  80  5060 example1.webex.com
_sip._tcp.example2.webex.com    524   IN    SRV   10  50  5060 example2.webex.com
_sip._tcp.example3.webex.com    524   IN    SRV   10  50  5880 example3.webex.com
```

Total calls:- 3004

Call Counts on all 3 User Agent Server

example1.webex.com : 2391

example2.webex.com : 321

example3.webex.com : 292

Call distribution based on DNS SRV lookup

Case 3 : One Record Route with lower priority and the other two Record Routes with higher priority and same weight across

Configuration:

```
_sip._tcp.example1.webex.com    524   IN    SRV   70 50 5060 example1.webex.com
_sip._tcp.example2.webex.com    524   IN    SRV   10 50 5060 example2.webex.com
_sip._tcp.example3.webex.com    524   IN    SRV   10 50 5880 example3.webex.com
```

Total calls:- 1000

Call Counts on all 3 User Agent Server

example1.webex.com : 0

example2.webex.com : 499

example3.webex.com : 501

Call distribution based on DNS SRV lookup

Case 4 : Record Routes with different priorities but same weight across

Configuration:

```
_sip._tcp.example1.webex.com    524   IN    SRV   10  50  5060 example1.webex.com
_sip._tcp.example2.webex.com    524   IN    SRV   20  50  5060 example2.webex.com
_sip._tcp.example3.webex.com    524   IN    SRV   30  50  5880 example3.webex.com
```

Total calls:- 1000

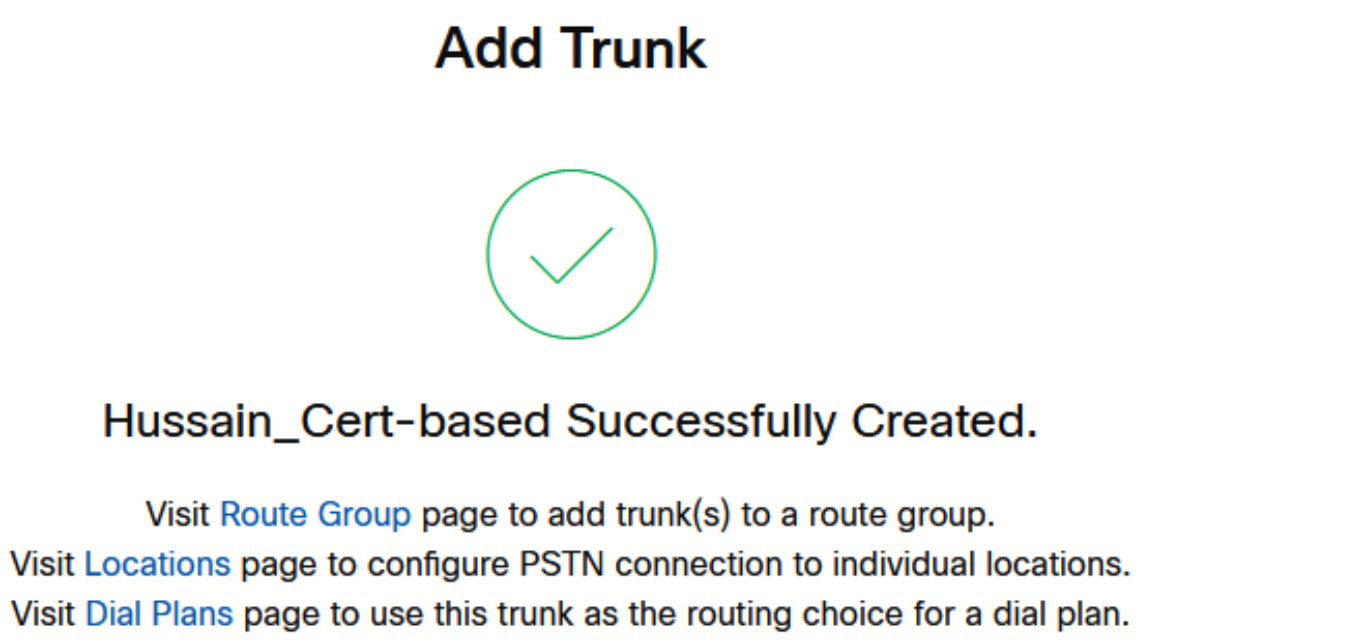
Call Counts on all 3 User Agent Server

example1.webex.com : 1000

example2.webex.com : 0

example3.webex.com : 0

Certificate-based LGW




```
dial-peer voice 2000 voip
 description To WxC Edge Proxy SRV Address
 session protocol sipv2
 session target dns:us18.sipconnect.bcld.webex.com
```

Note: CSCwm08791 Dial-peer never recovers an active state if it goes partial or busyout

Symptom: Dial-peer status is shown as partial/busyout with FQDN, whereas all the nodes in the SRV or IP address are marked as up.

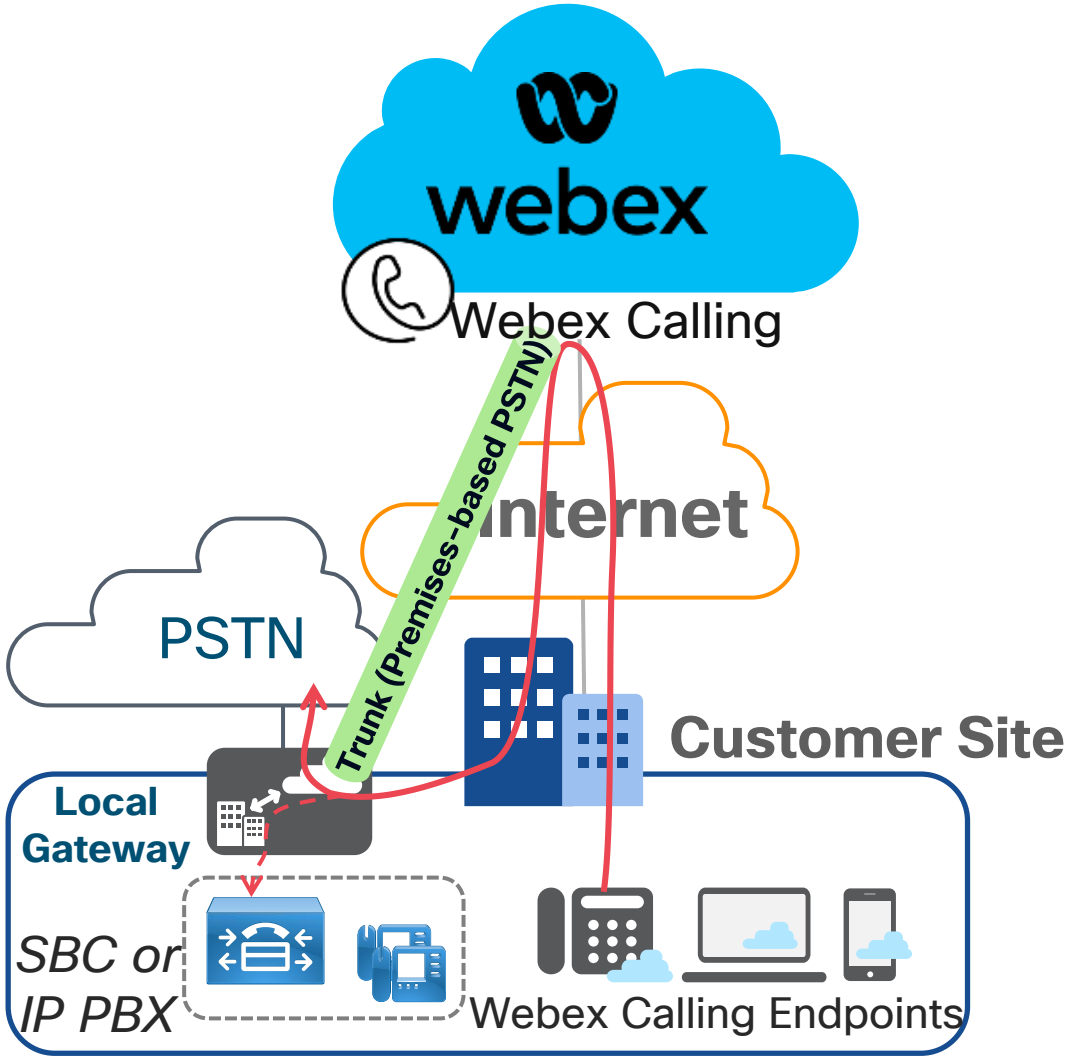
Fixed Release: IOS-XE 17.9.7a, 17.12.5a/5b, 17.15.3a/3b, 17.16.1a

Local Gateway for Webex Calling

Premises-based PSTN Trunking models

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.** Endpoints directly register to Webex Calling over the Internet.

Premises-based PSTN Trunking Models

- There are two types of Premises-based PSTN trunking models:
  - Registration-based trunks
  - Certificate-based trunks
- Both models provide similar functionality, but they differ in scale and device support

Comparing Local Gateway trunking models

Functionality	Registration-based	Certificate-based
Concurrent Calls	Concurrent calls of up to 250 per trunk (OTT Internet)	Up to 6500 concurrent calls per trunk
Device Type	Supports only CUBE (except ASR1000 series)	Supports all CUBE and 3rd party SBCs
Authentication model	Digest-based authentication model, which relies on a shared username and password used to authenticate registration and calls.	Certificate-based authentication model
Public DNS service requirements	None	Domain claims required. A DNS A or SRV record must be configured in public DNS server

Network, firewall, and NAT requirements

Registration-based	Certificate-based
Any NAT or Public IP is supported. • Dynamic NAT is preferred since it's easier for setup and requires less firewall configs For ingress traffic, inbound pinholes(from WxC to LGW) are opened by the firewall based on outbound registration messages Pinhole opening is recommended for all Webex Calling IP addresses and ports.	Public internet-facing network including a public IP or Static NAT.

Both requires firewall to allow both ingress and egress traffic (Webex calling to Local Gateway and vice versa).

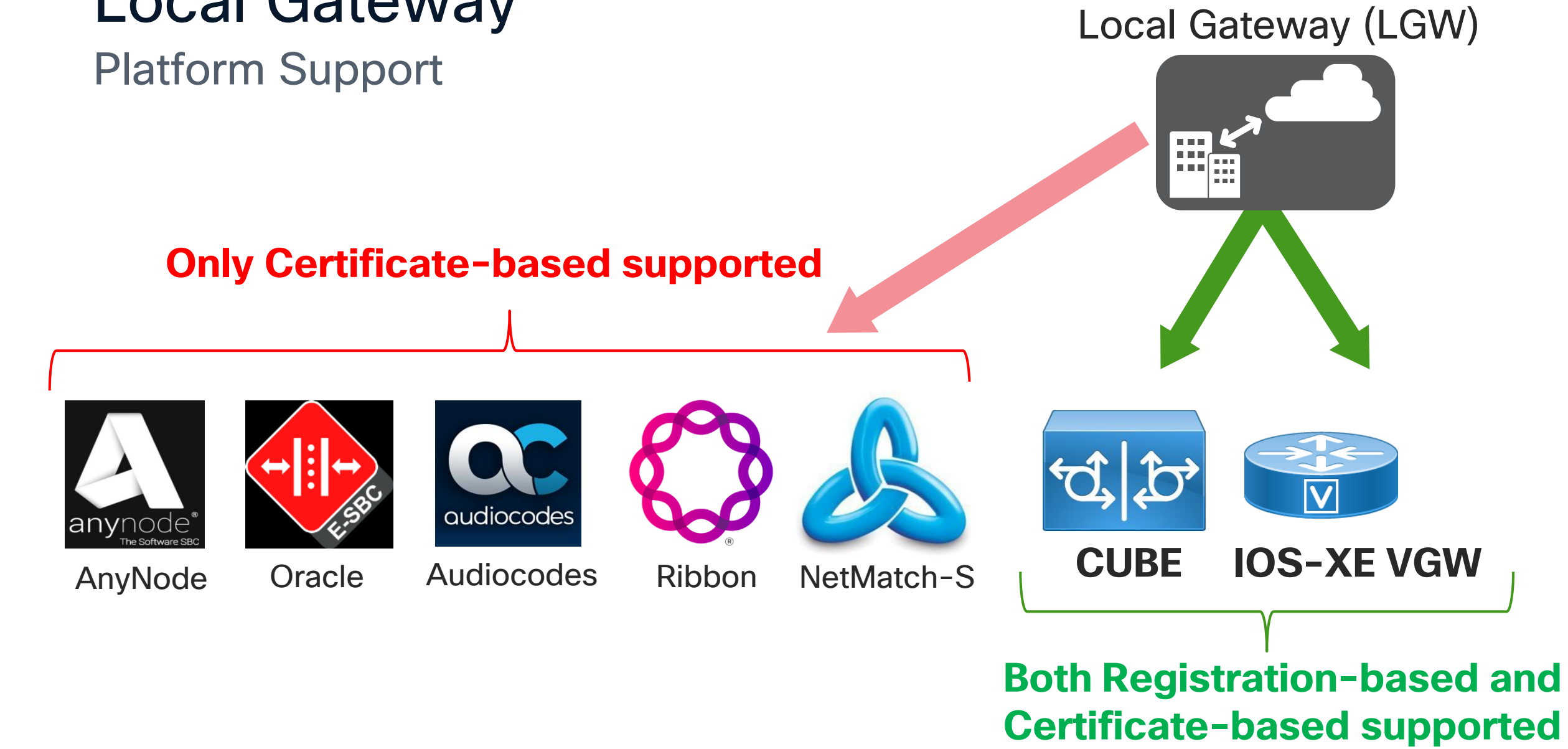
CA and certificate requirements

Registration-based	Certificate-based
	Local gateway must have a signed certificate using one of the certificate authorities listed in Root Certificate Authorities. • Wild-card certificates are not supported • Certificates must be signed per guidelines as mentioned in Configure Trunks, Route Groups, and Dial Plans for Webex Calling

CA bundle that signed the Webex service's certificate has to be uploaded to the Local Gateway.

Local Gateway

Platform Support



Calling Capacity requirements

- Registration-based and Certificated-based trunking models have different concurrent call capacities as shown below

Concurrent calls per local gateway / trunk	Trunk type Preference	Minimum Link Quality
~ 2000–6500	Certificate-based	Interconnect
250 to ~ 2000	Certificate-based	Over the top Internet (OTT)
up to 250	Registration-based	OTT

Connection qualifications

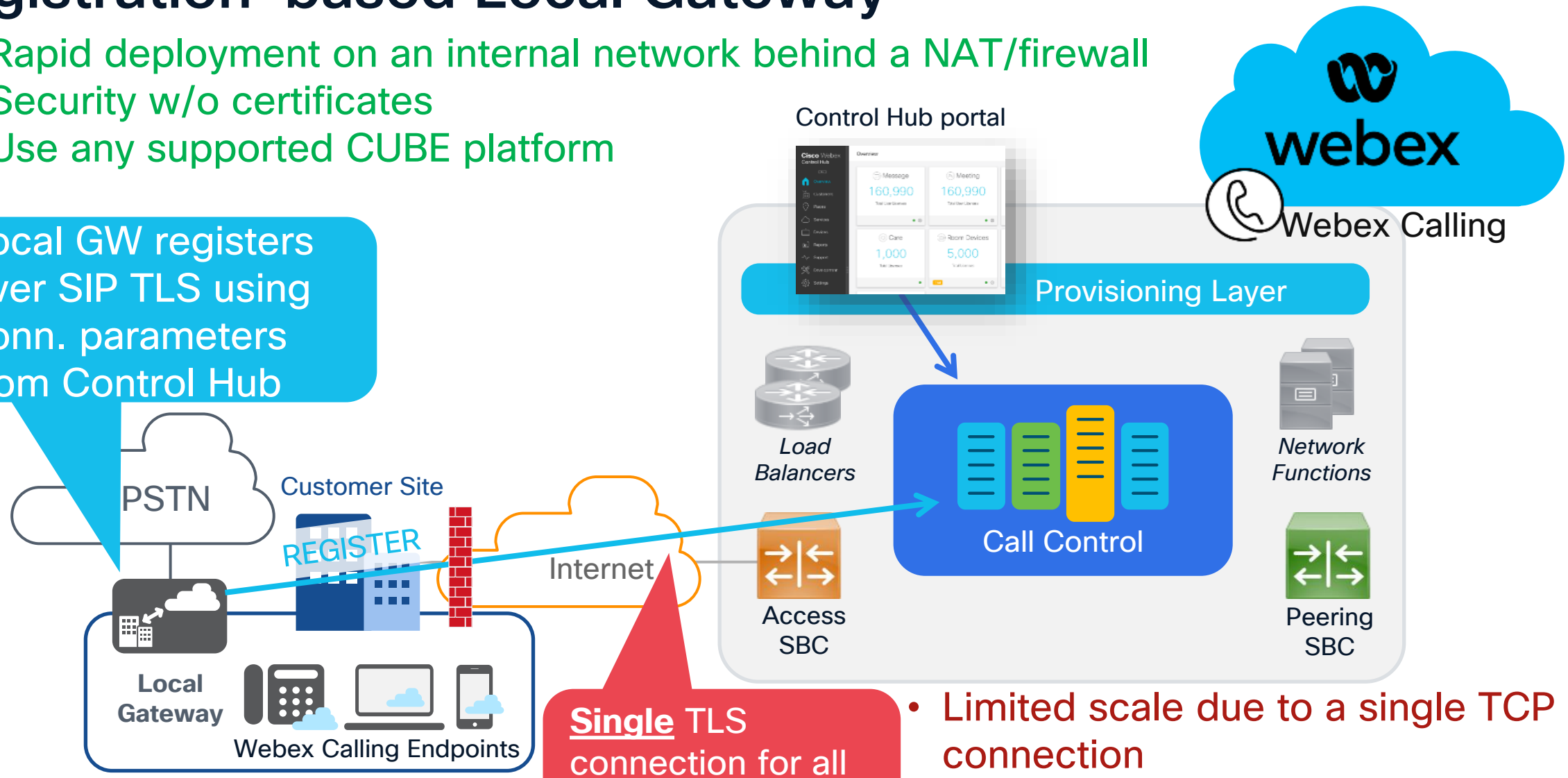
- Over the top (OTT) Internet and interconnect (e.g. Webex Edge Connect) must meet the following link quality conditions

Connection Type	Latency	Jitter	Packet loss
OTT	100 ms (max)	100 ms (max)	0.2%
Interconnect	30 ms	5 ms	Zero packet loss

Registration-based LGW on a CUBE

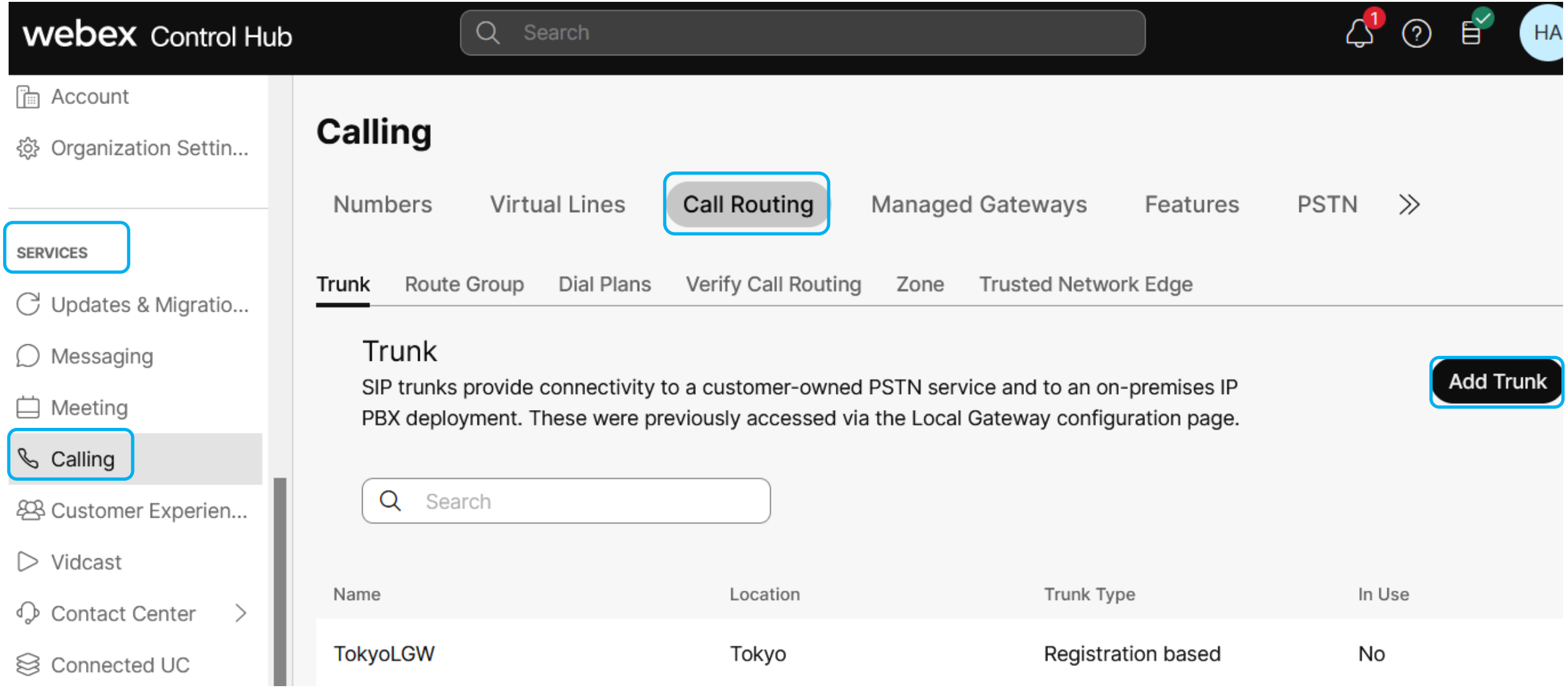
Registration-based Local Gateway

- Rapid deployment on an internal network behind a NAT/firewall
- Security w/o certificates
- Use any supported CUBE platform

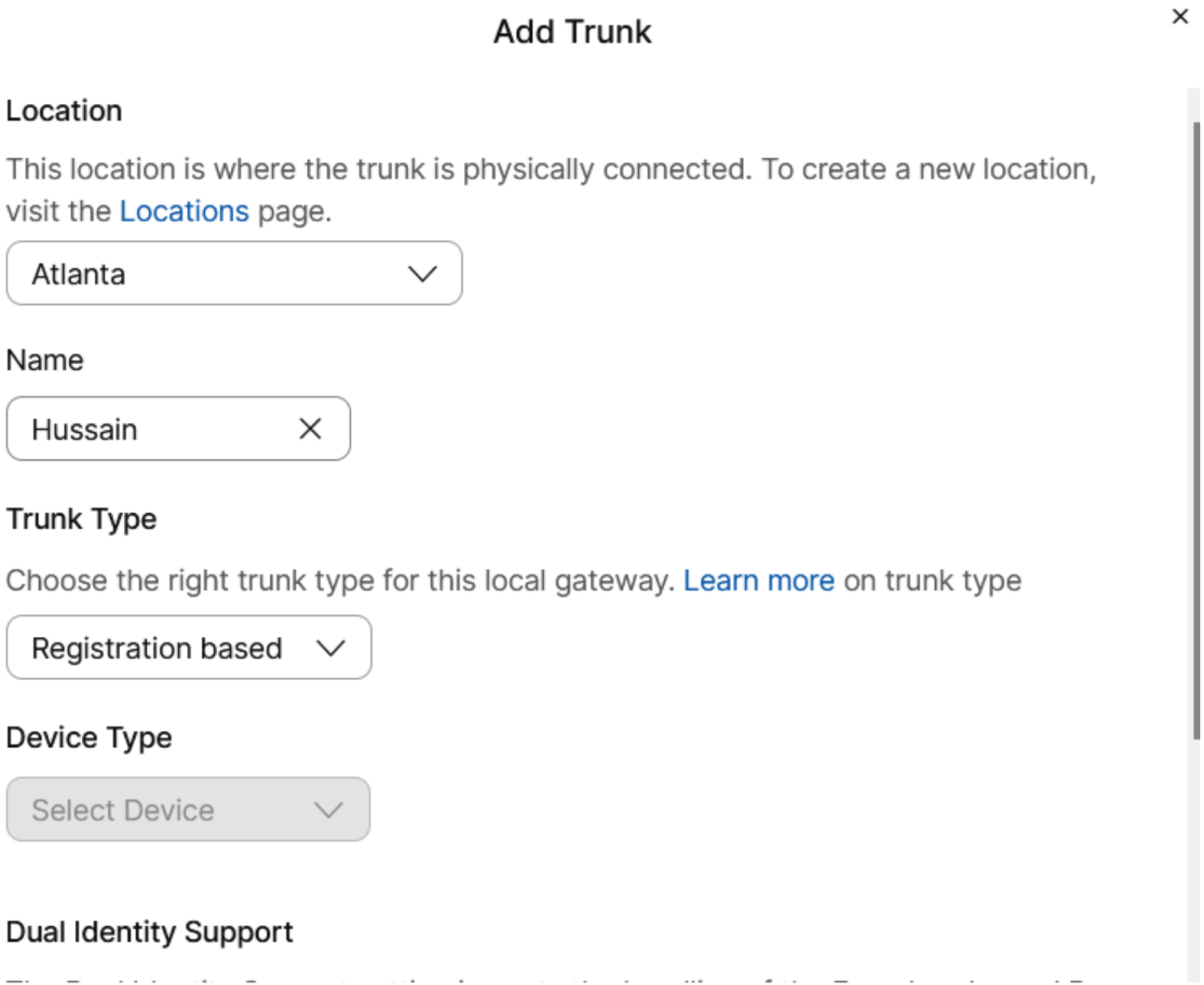


- Limited scale due to a single TCP connection
- Sensitive to network impairments (TCP throughput ∝ latency/loss)

Log in to Control Hub. Navigate to **Services** – Click **Calling** and then go to the **Call Routing** Tab. Click **Add Trunk**.



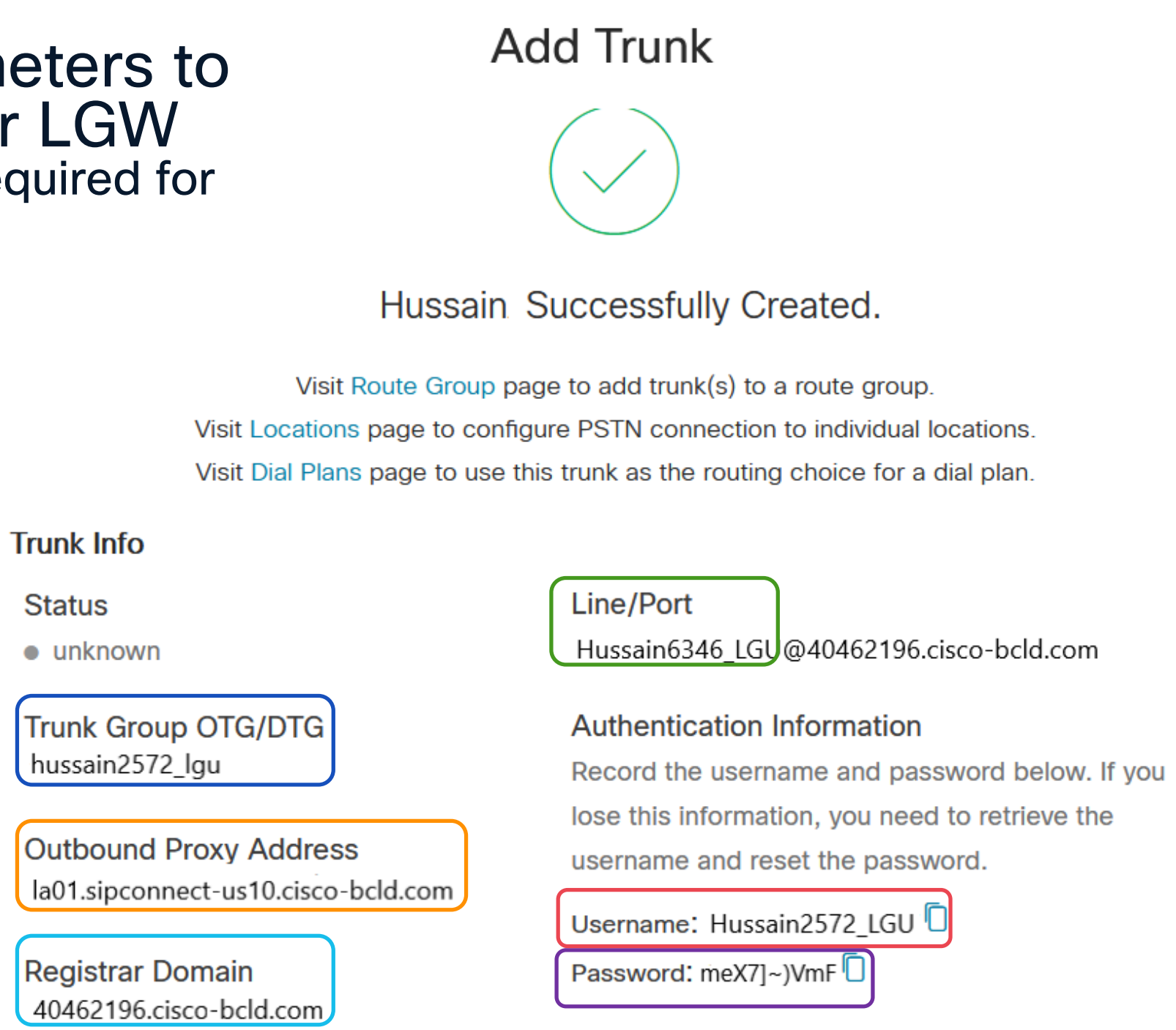
Add a new Trunk for the desired Location



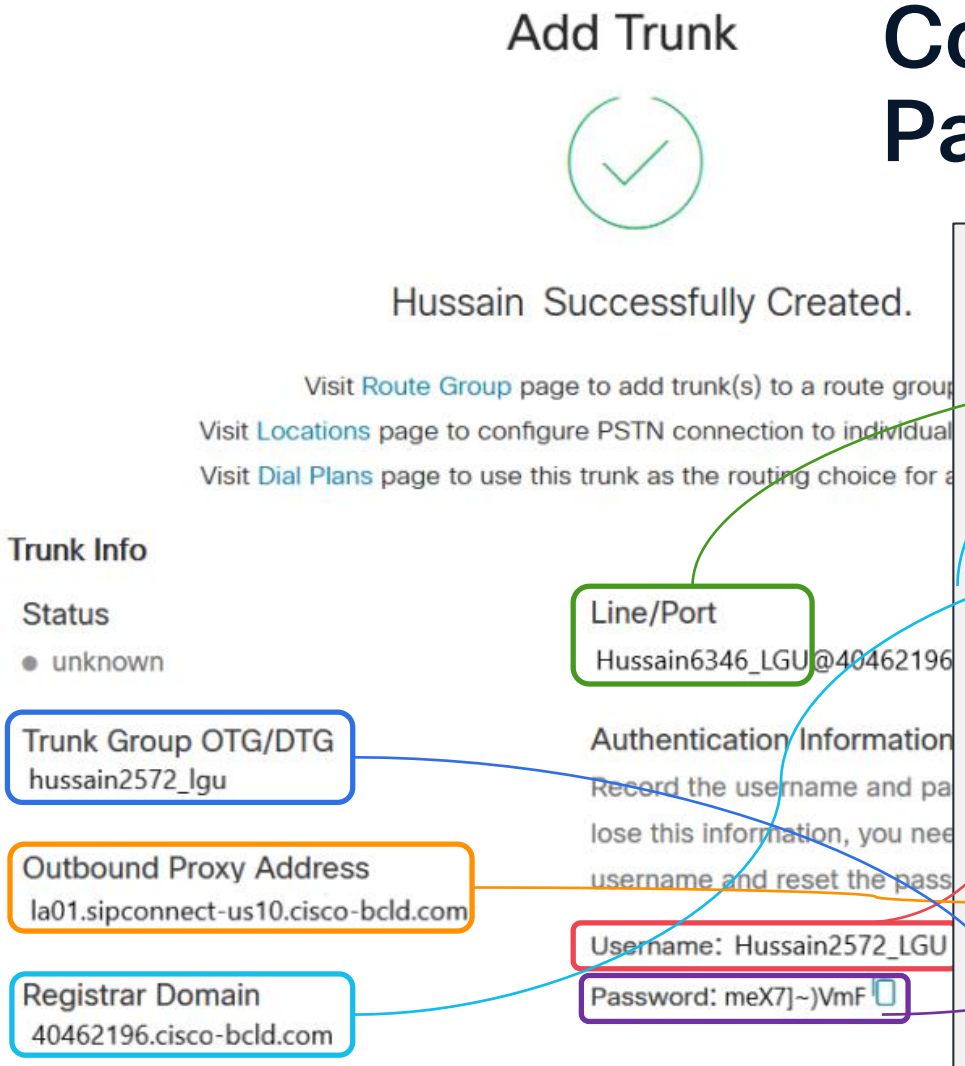
- Trunk name is limited to 24 characters

Save the Trunk parameters to build the CUBE CLI for LGW

Parameters on this display required for building LGW CLI



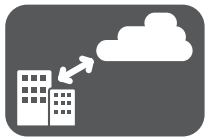
Control Hub Trunk Info Connection Parameters → LGW CLI Config




```
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
```



SRV Query

_sips_tcp.la01.sipconnect-us10.cisco-bcld.com

DNS

webex

Webex Calling

Query Response

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

A Query

sipconnect03a-us10.cisco-bcld.com

Query Response

IP Address of **sipconnect03a-us10.cisco-bcld.com** is: **199.59.70.30**

Access SBC

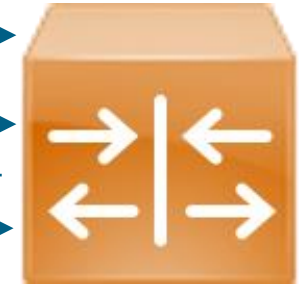
TLS Connection to 199.59.70.30:8934

REGISTER sip:40462196.cisco-bcld.com:5061 SIP/2.0

401

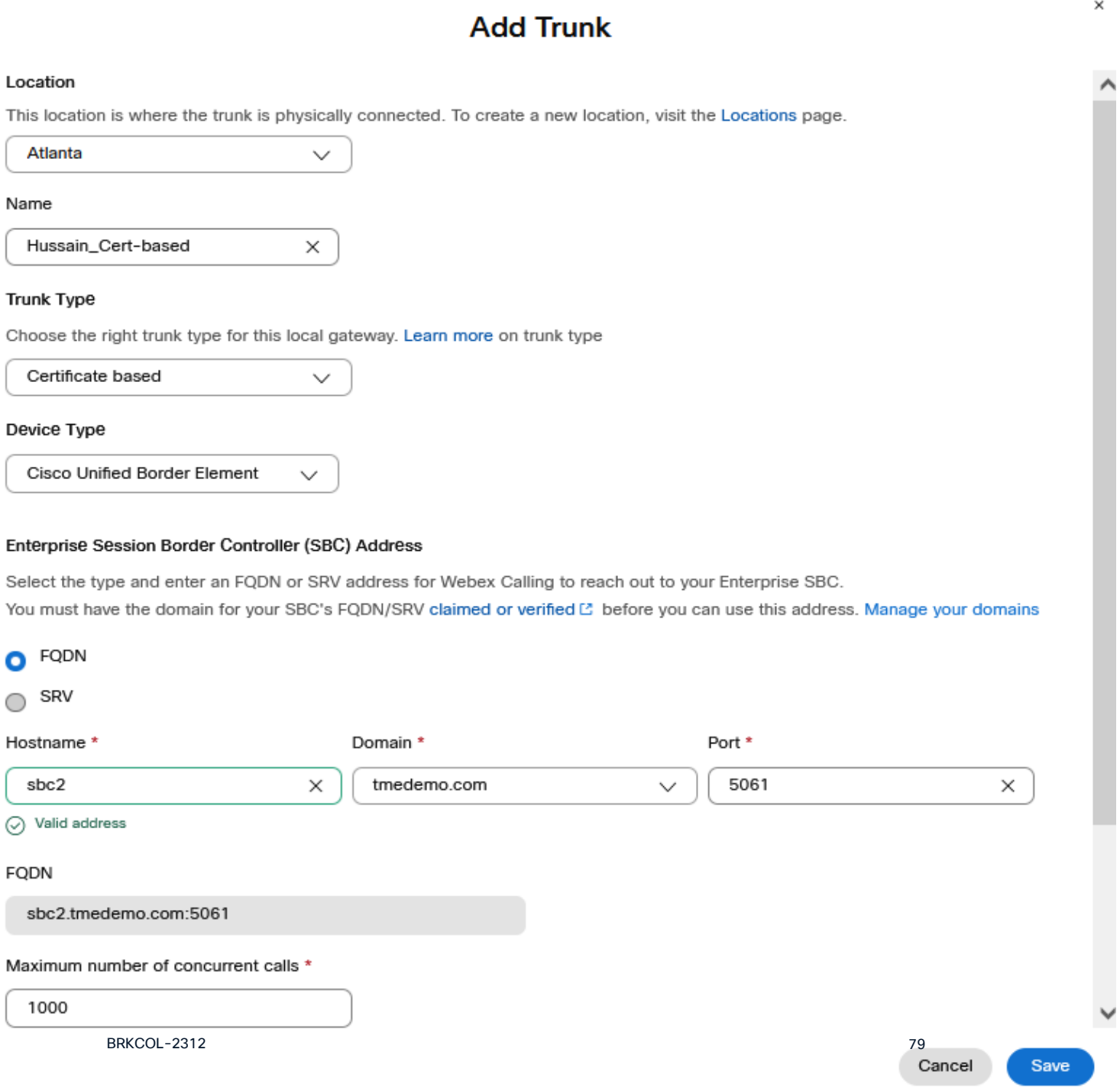
REGISTER (w/credentials)

200 OK

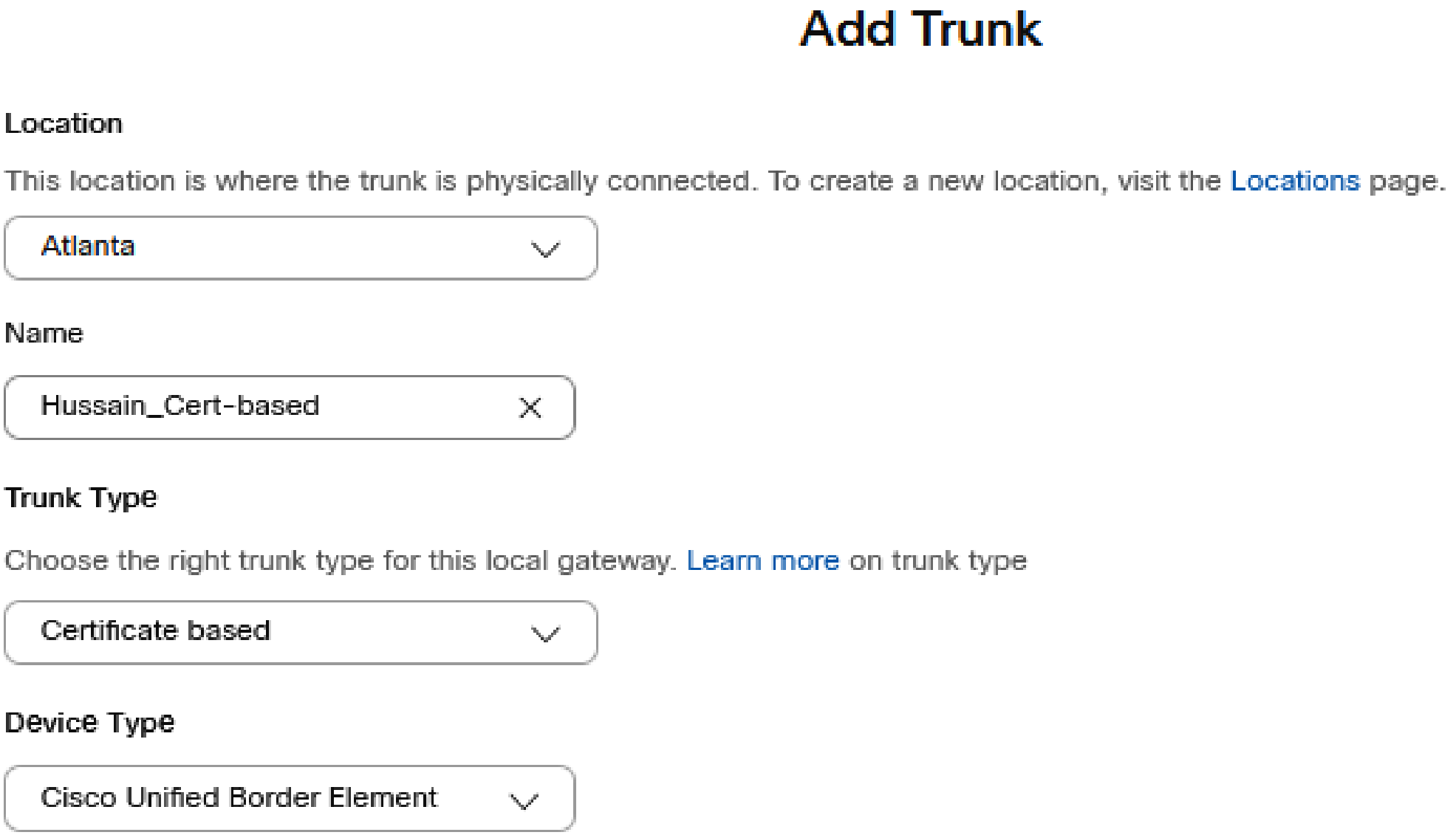


Certificate-based Local Gateway

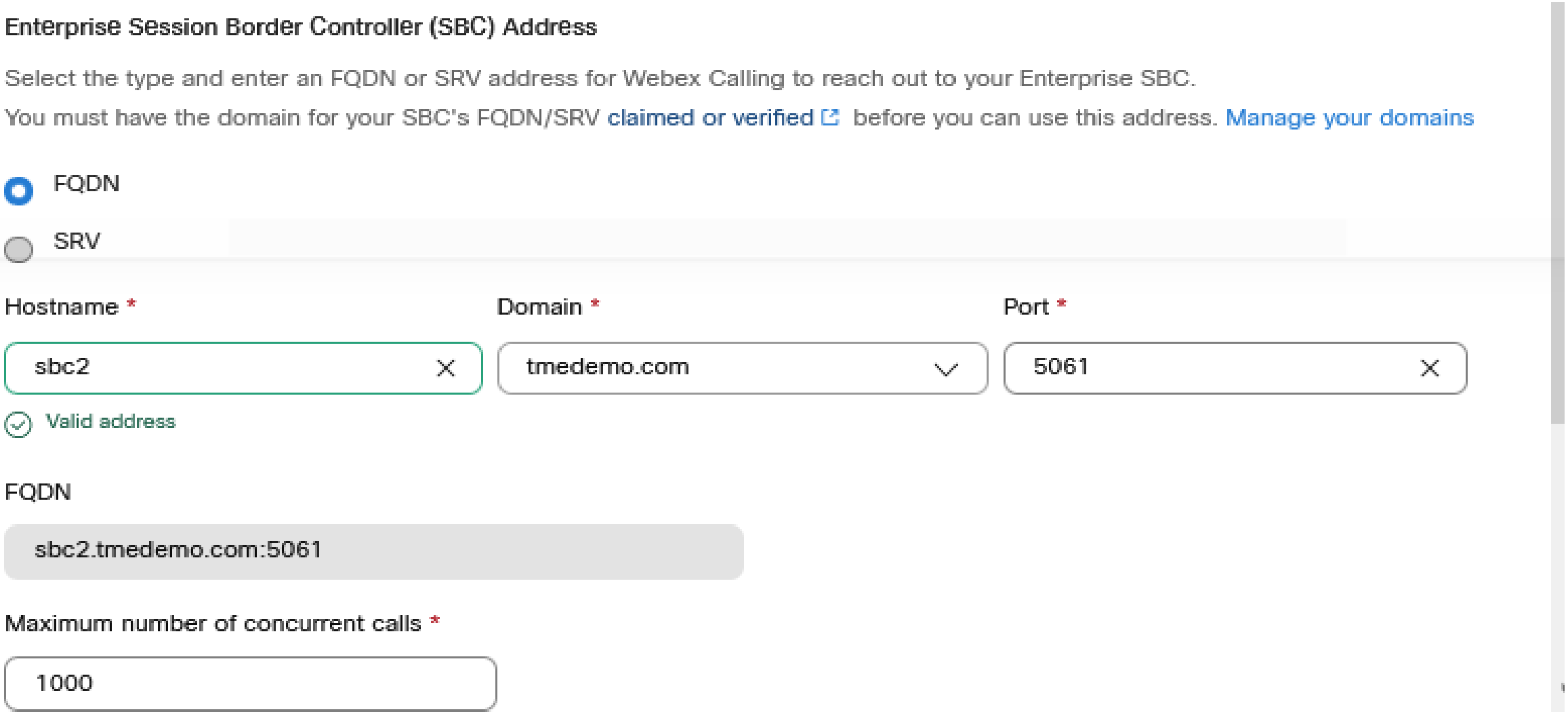
Add a Certificate-based Trunk to a Location



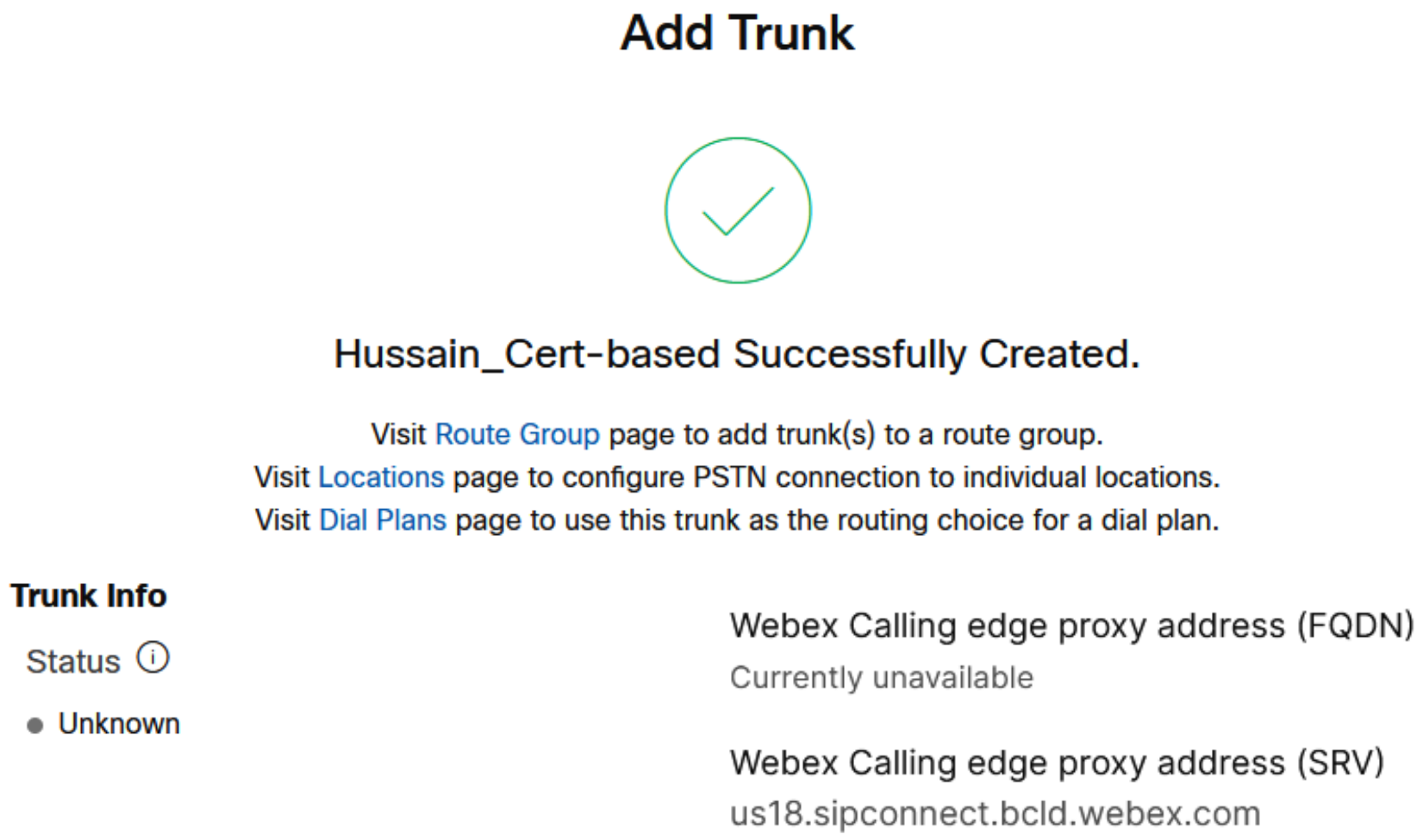
Adding a Trunk



Define the LGW hostname and select to resolve the LGW through an FQDN or an SRV



Save the Webex Calling Edge Proxy Address displayed



Webex Calling Trunk – Local Gateway (Certificate-based)

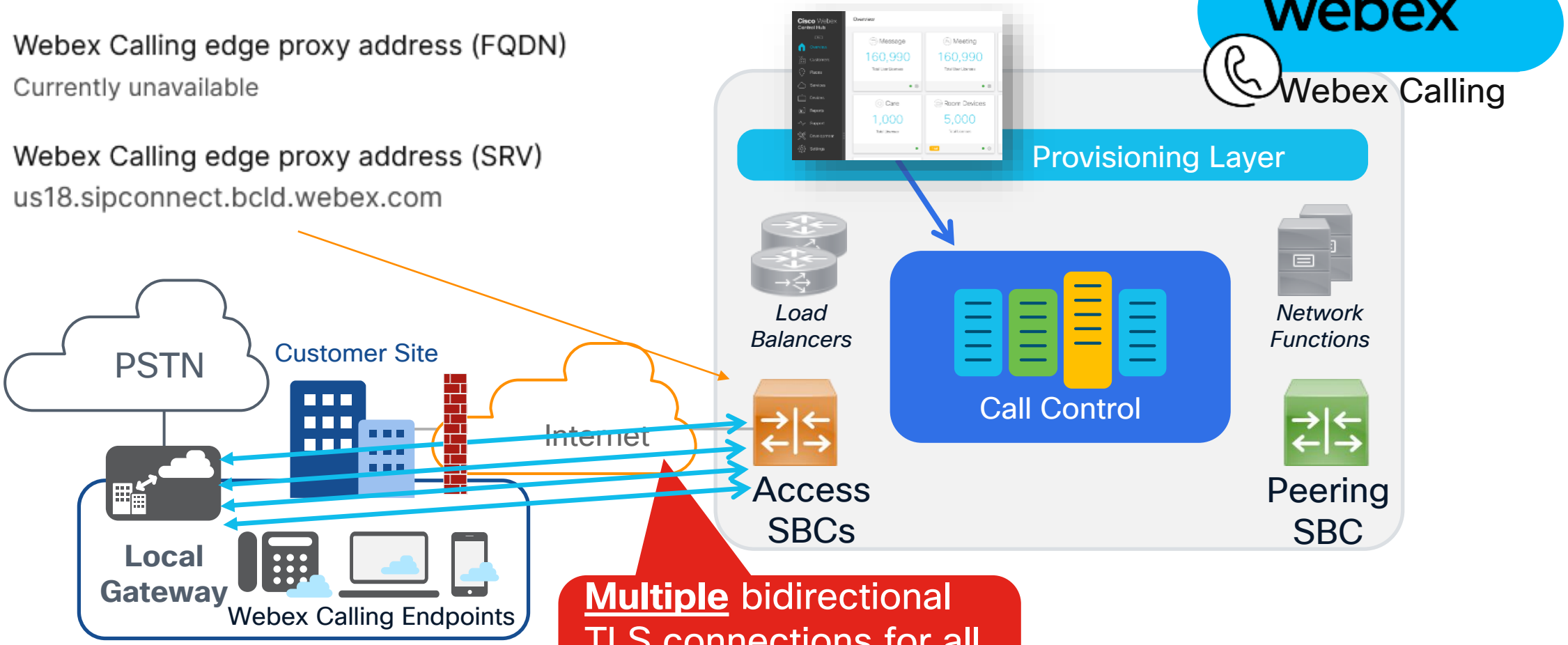
Customer DNS/FQDN SRV's configured in CH

Webex Calling edge proxy address (FQDN)

Currently unavailable

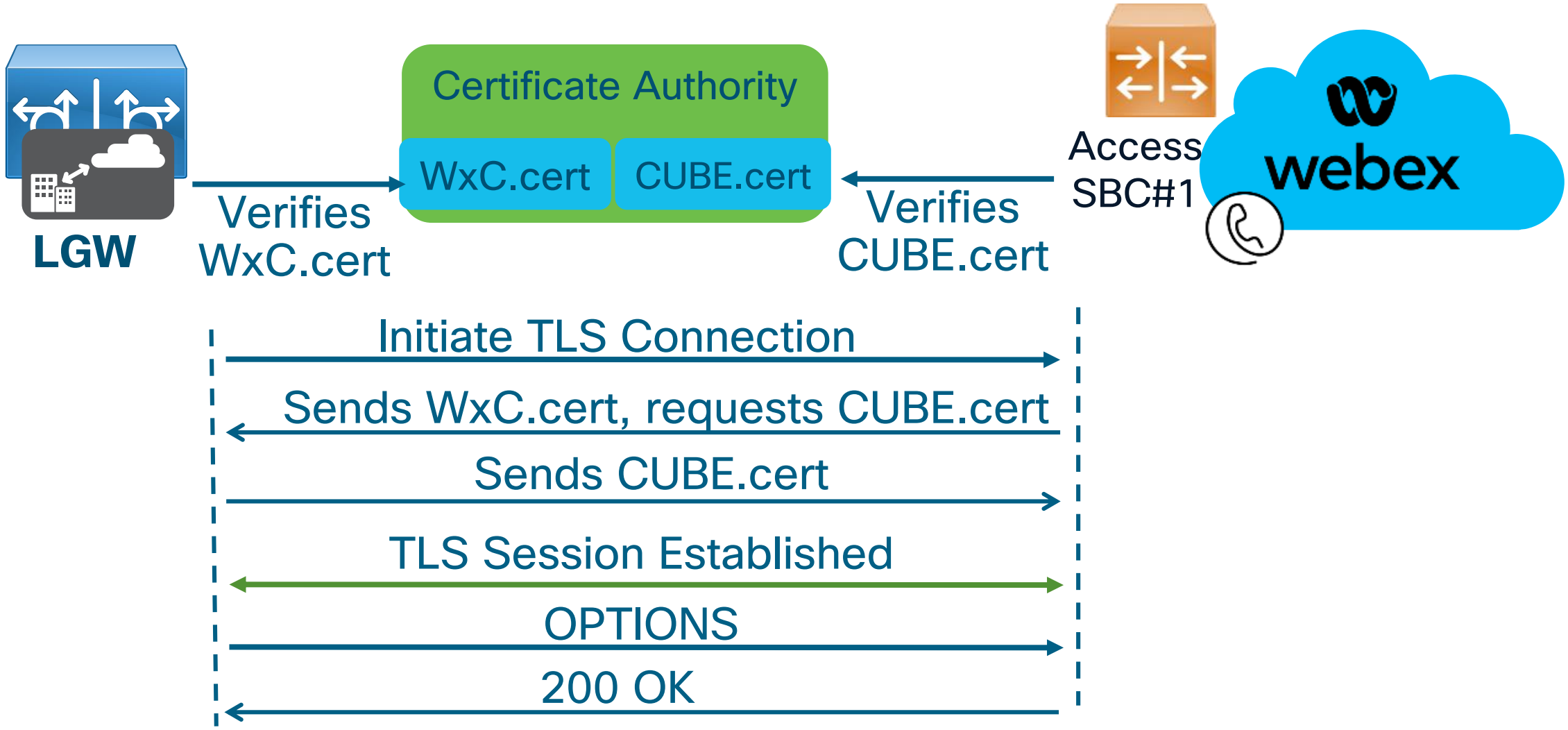
Webex Calling edge proxy address (SRV)

us18.sipconnect.bcld.webex.com

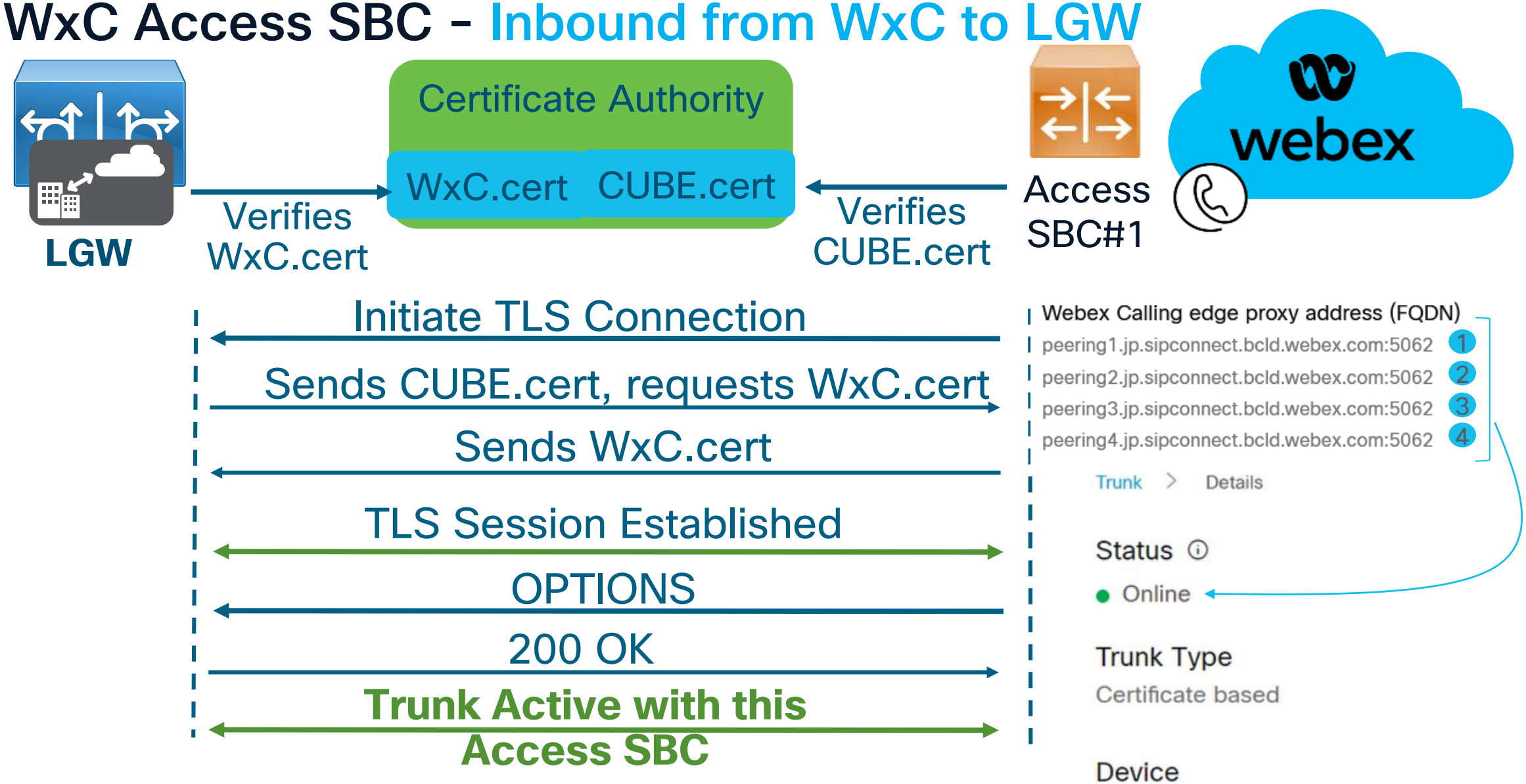


Multiple bidirectional TLS connections for all signaling between LGW and cloud

Certificate-based Local Gateway (Trunk Establishment) – 1st WxC Access SBC – Outbound from LGW to WxC



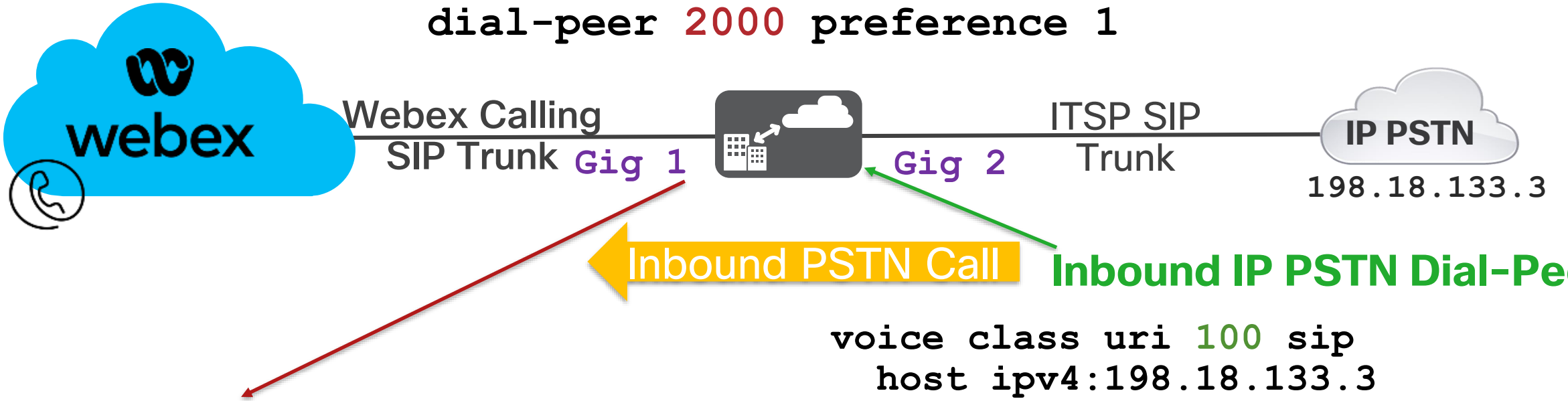
Certificate-based Local Gateway (Trunk Establishment) – 1st WxC Access SBC – Inbound from WxC to LGW



Now repeat the process with the 2nd, the 3rd, and the 4th WxC Access SBC

Inbound LGW PSTN Call

```
voice class dpg 200
 description Incoming IP PSTN(DP100) to WxC(DP2000)
 dial-peer 2000 preference 1
```



Outbound WxC Dial-Peers

```
dial-peer voice 2000 voip
 description Outbound dial-peer
 ! to Webex Calling Proxy SRV
 session target dns:us01.sipconnect.bcld.webex.com
```

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

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

Outbound LGW PSTN Call

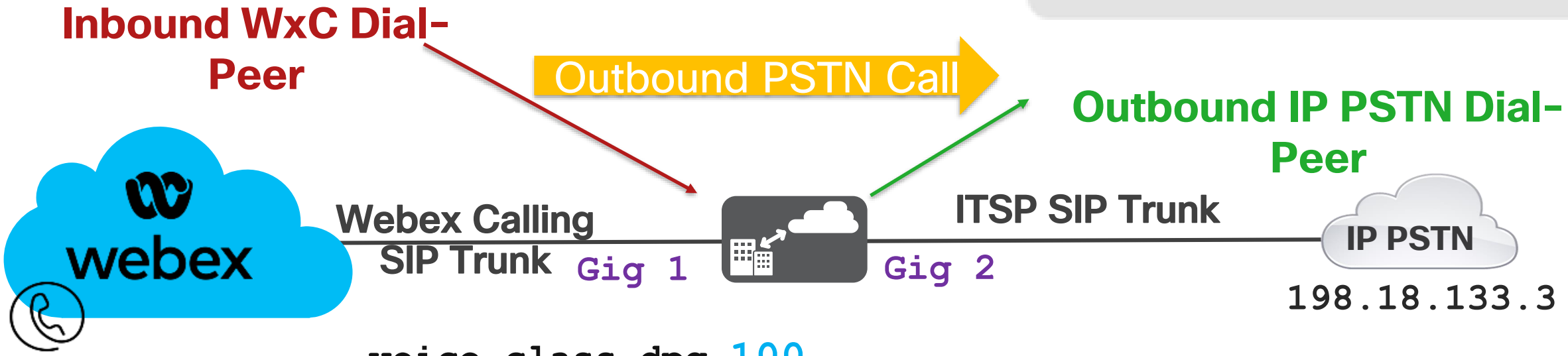
```
voice class uri 200 sip
  pattern sbc2.tmedemo.com
```

```
dial-peer voice 101 voip
 description Outgoing to IP PSTN
 destination-pattern BAD.BAD
 session target ipv4:198.18.133.3
```

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

FQDN

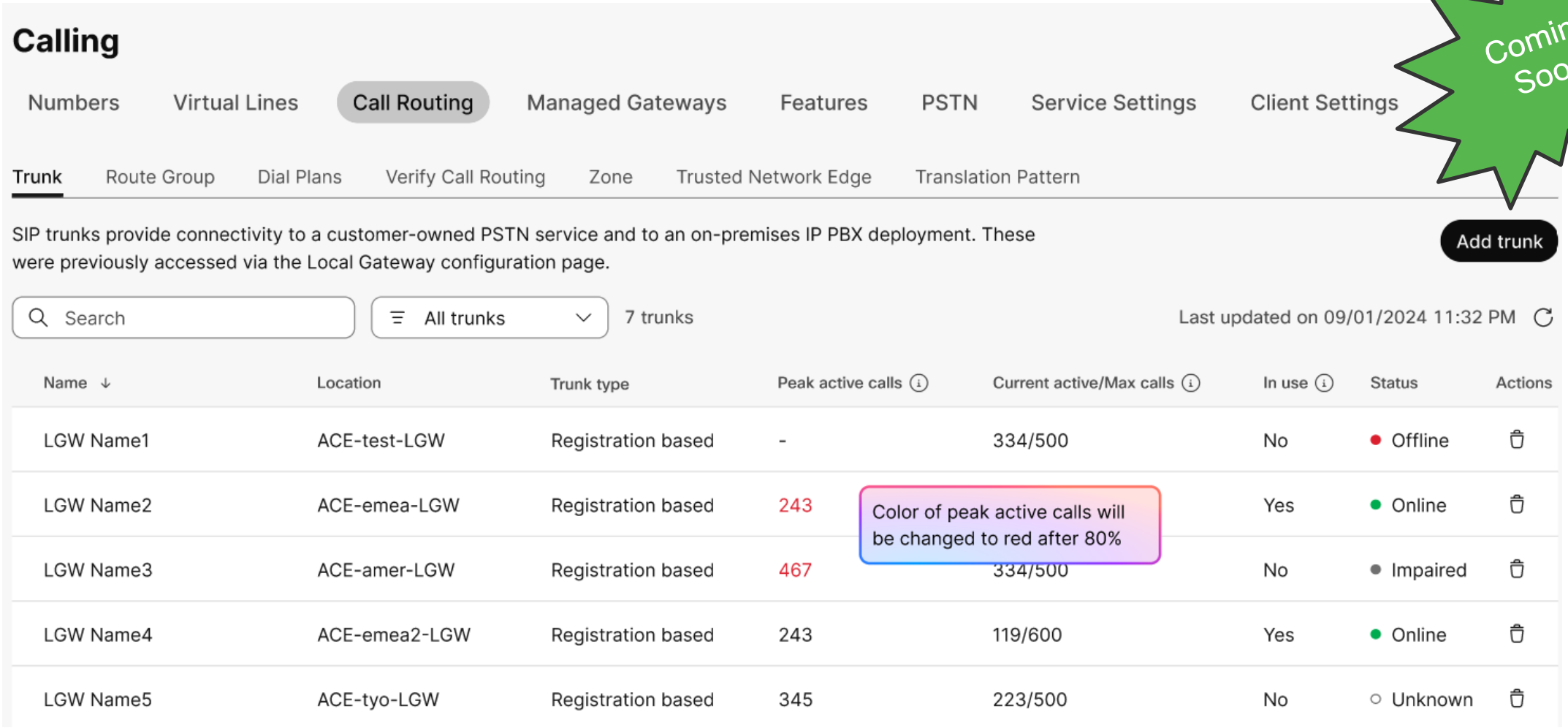
sbc2.tmedemo.com:5061




```
voice class dpg 100
 description Incoming WxC(DP200) to IP PSTN(DP101)
 dial-peer 101 preference 1
```

Local Gateway Status Improvements in Control Hub

Peak, current active, and Max calls will be displayed



Coming Soon

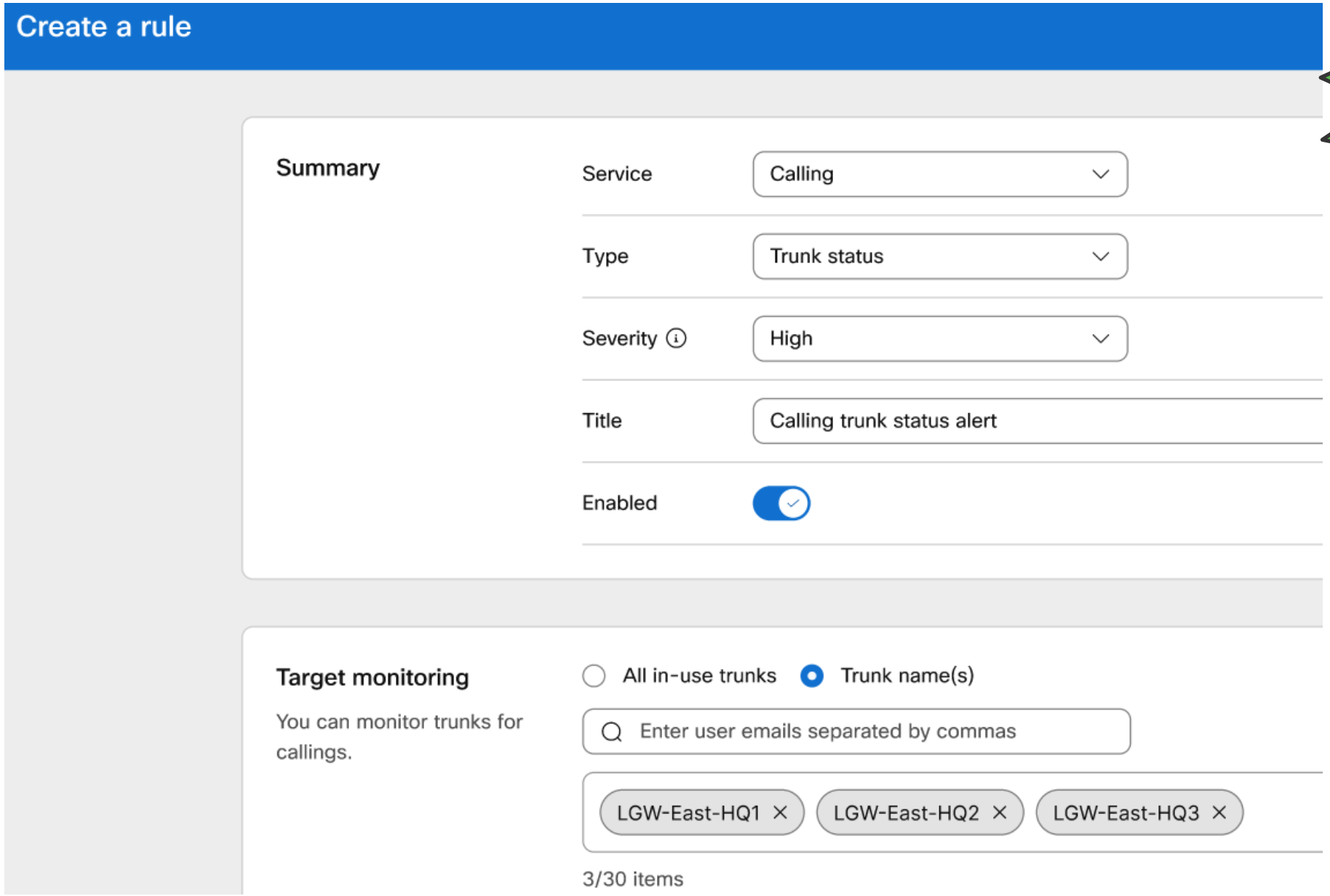
- Peak number of calls handled by the trunk in the last 72 hours. Trunks exceeding 80% capacity are shown in red.
- Active concurrent calls handled by the trunk
- Max configured calls

Local Gateway Alerts in Control Hub

Real time Alerts

- Alerts will trigger based on LGW status changes and will include:
  - Trunk status Online/Offline events (Real Time).
  - Ability to choose one or more trunk(s)
  - Certificate expiry (Pre expiry and post expiry alerts)
    - Pre-expiry alert before predefined days at specific days at 60 days, 30 days, 15 days, 7 days.
      Post-expiry, an alert just after the certificate has expired. Single alert.
  - Concurrent call limit alerts for trunks (based on sum of both inbound and outbound calls)
    - alerts based on threshold percentages [e.g., alert if 80% of peak active call had reached for Trunk(s)]
      - Assists to identify call concurrency for a busy day, busy hour, etc.

Create a rule – Trunk Status Alert



Certificate Expiry Alert

Summary

Service	Calling
Type	Call trunks certificate expiry
Severity ⓘ	High
Title	Call trunks certificate expiry alert
Enabled	✓

Delivery Method

By default, all alerts appear in Control Hub.

- ☑ Email
  - reyskywalker@theforce.com ×
  - yoda@jedimaster.com ×
  - caradune@rebels.com ×
  - Enter user emails seperated by commas
- ☑ Webex app space
  - ○ Space name ⓘ
  - ○ Space ID ⓘ
  - ◉ 1 to 1 bot ⓘ

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Trunk Concurrent call limit Alert

Summary

Service	Calling
Type	Trunk concurrent call limit
Severity	High
Title	Trunk concurrent call limit alert
Enabled	

Set the sampling interval to determine how often the system checks the percentage of concurrent calls against the threshold. If the percentage exceeds the specified limit (e.g., 70%) during the selected time range, an alert will be triggered.

Target monitoring

You can monitor trunks for callings.

○ All in-use trunks ◉ Trunk name(s)

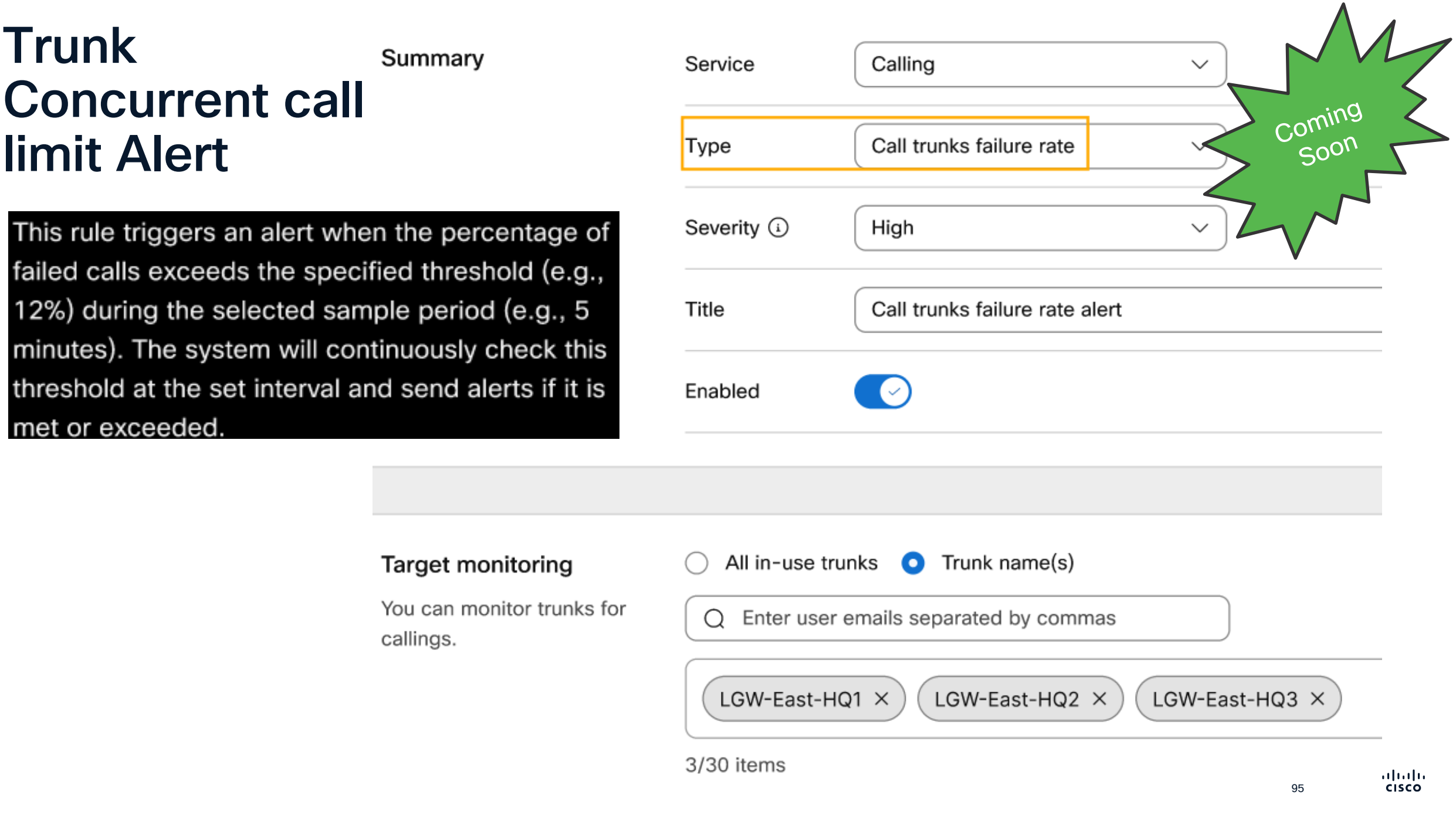
Enter user emails separated by commas

LGW-East-HQ1 × LGW-East-HQ2 × LGW-East-HQ3 ×

3/30 items

Trunk Concurrent call limit Alert

This rule triggers an alert when the percentage of failed calls exceeds the specified threshold (e.g., 12%) during the selected sample period (e.g., 5 minutes). The system will continuously check this threshold at the set interval and send alerts if it is met or exceeded.

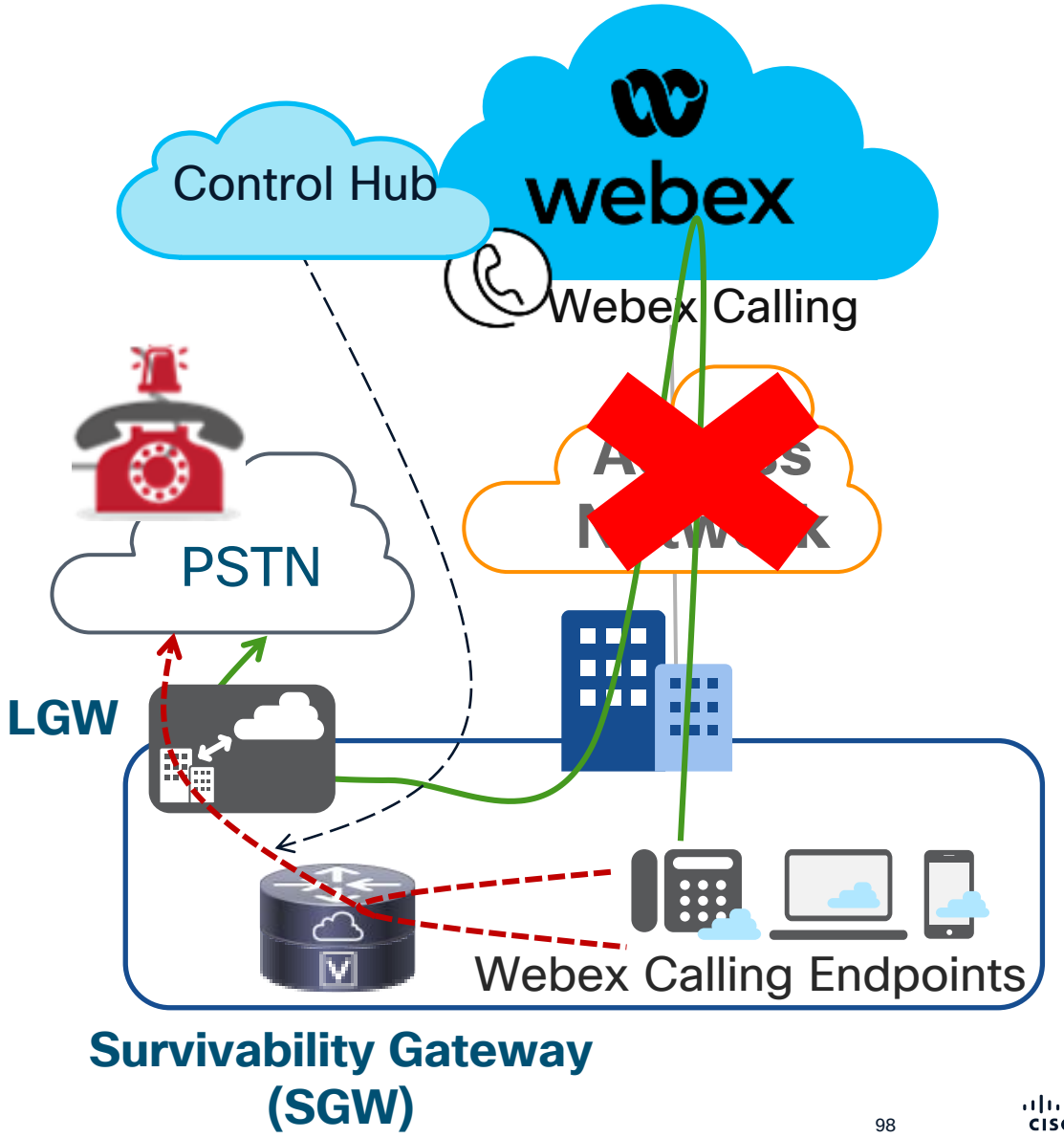


Site Survivability for Webex Calling

Site Survivability
Solution Overview

Site Survivability for Webex Calling

- A Survivability Gateway (SGW) is installed at a customer site
- The SGW is managed by and gets configuration details from the Control Hub (Webex Cloud)
- In the event of a network outage:
  - Internal/external calls are routed via the SGW
  - Emergency calls are routed via the SGW



Active Mode

Survivability Mode

Endpoint Support

Type	Model	Version
Desk Phones	6821, 6841, 6851, 6861, 6861 Wi-Fi, 6871, 7811, 7821, 7841, 7861, **7832, 8832** 8811, 8841, 8851, 8861, 8845 (audio only), 8865 (audio only)	**12.0(1)**
Webex App	Windows, Mac	**43.2**
Analog Endpoints	**VG400, VG410, and VG420**	**17.16.1a**
	ATA191 and ATA192	**11.3(1)**

Cisco Desk Phone 9800 series, ATA, VG400 series now supported

Survivability Gateway (SGW)

Platform Support

Survivability Gateway (SGW)

- Hardware and software requirements:
  - ISR 4321, 4331, 4351, 4431, 4451 **(IOS-XE 17.12.5b)**
  - ISR4461 **(IOS-XE 17.12.5b)**
  - Catalyst 8200/8300 series **(IOS-XE 17.12.5b)**

  - Catalyst 8000v Edge (vCUBE) **(IOS-XE 17.12.5b)**

IOS-XE Platform Scale support

Model	Max Registrations
Integrated Services Router 4321	50
Integrated Services Router 4331	100
Integrated Services Router 4351	700
Integrated Services Router 4431	1200
Integrated Services Router 4451-X	2000
Integrated Services Router 4461	2000
Catalyst Edge 8200L-1N-4T	1500
Catalyst Edge 8200-1N-4T	2500
Catalyst Edge 8300-1N1S-6T	2500
Catalyst Edge 8300-2N2S-6T	2500
Catalyst Edge 8300-1N1S-4T2X	2500
Catalyst Edge 8300-2N2S-4T2X	2500
Catalyst Edge 8000V Software – Small/Medium/Large	500 / 1000 / 2000

IOS-XE Software Release Mapping

CUBE Version	Initial IOS-XE Release for this CUBE version and Release date		Subsequent IOS-XE Release for this CUBE version
14.7	17.12.1a	July 2023	17.12.5b
14.8	17.13.1a	Nov 2023	
14.9	17.14.1a	March 2024	
14.10	17.15.1a	July 2024	17.15.3a
14.11	17.16.1a	Nov 2024	
TBD	17.18.1a	August 2025	

Last release for ISR4K except ISR4461

Note: CSCwk44855 Cannot dial to/from phone after it reboots in survivability mode

Symptom: When phones register to the Survivability Gateway (SGW), phones are successfully registered and can make and receive calls. If a phone gets rebooted, it will register back to the SGW but after 5minus the SGW deletes the AOR entry for the phone. This affects the call routing as the virtual dial-peer is deleted, causing a 404 Not Found SIP Error message since there is no route to the destination extension. Phone cannot dial-out indicating "no line" status.

Fixed Release: IOS-XE 17.12.5a/5b, 17.15.3a/3b, 17.16.1a

Port Reference Information for SGW

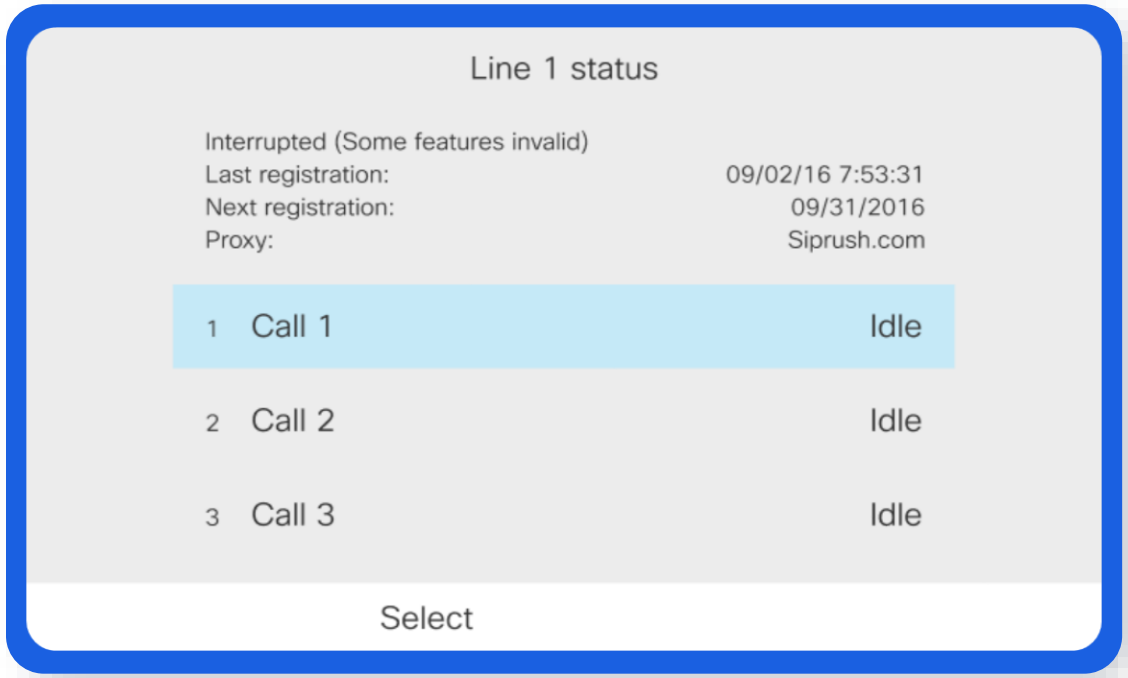
Connection Purpose	Source Addresses	Source Ports	Protocol	Destination Addresses	Destination Ports
Call signaling to SGW (SIP TLS)	Devices	5060-5080	TLS	SGW	8933
Call media to SGW (SRTP)	Devices	19560-19660	UDP	SGW	8000-14198 (SRTP over UDP)
Call signaling to PSTN gateway (SIP)	SGW	Ephemeral	TCP or UDP	Your ITSP PSTN gateway	5060
Call media to PSTN gateway (SRTP)	SGW	8000-48198	UDP	Your ITSP PSTN gateway	Ephemeral
Time synchronization (NTP)	SGW	Ephemeral	UDP	NTP server	123
Name resolution (DNS)	SGW	Ephemeral	UDP	DNS server	53
Cloud Management	Connector	Ephemeral	HTTPS	Webex services	443, 8433

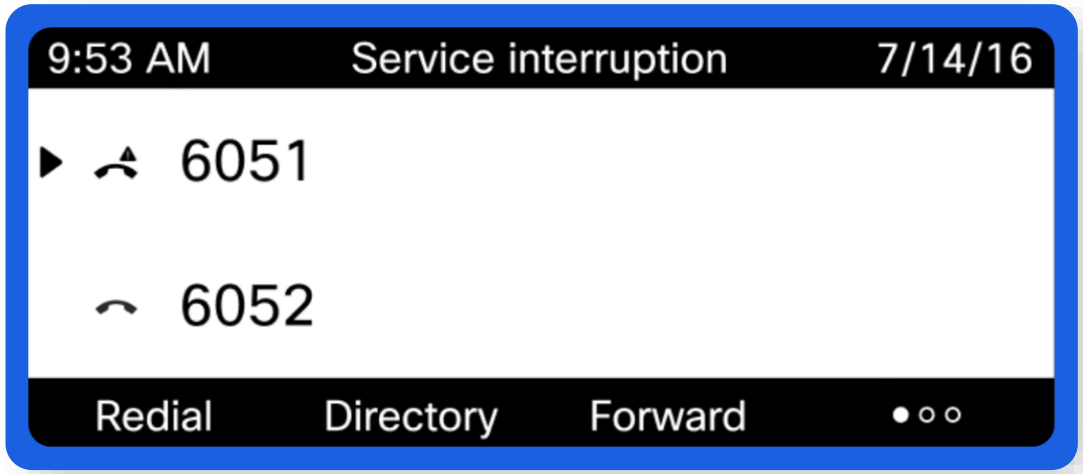
Survivability zones

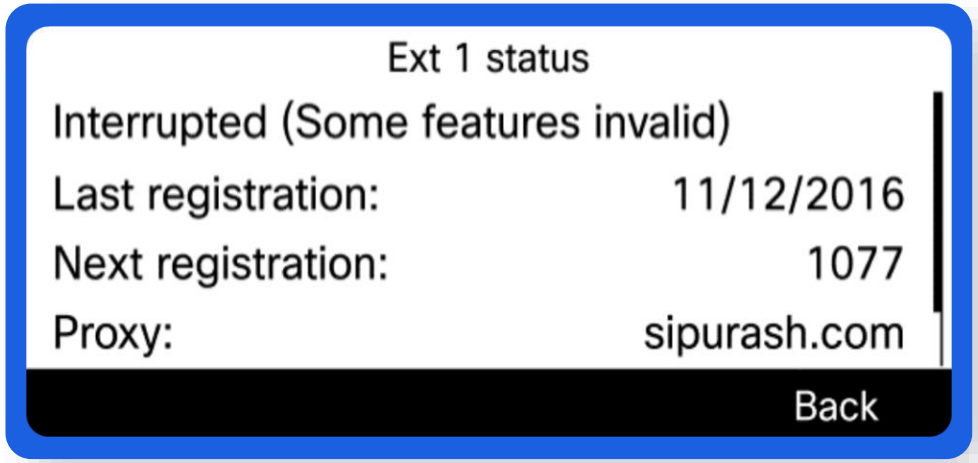
- Currently, survivability sites are aligned with “locations” in Webex Calling
- Each Survivability Gateway is associated to a specific location (1:1)
- All supported devices added to that location are configured to use that gateway, in the event of an outage
- The SGW is automatically provisioned with registration details for users/devices for its location
- No manual assignment of users to survivability zone is required
- **Multiple Locations per SGW coming soon**

Webex App and endpoint changes

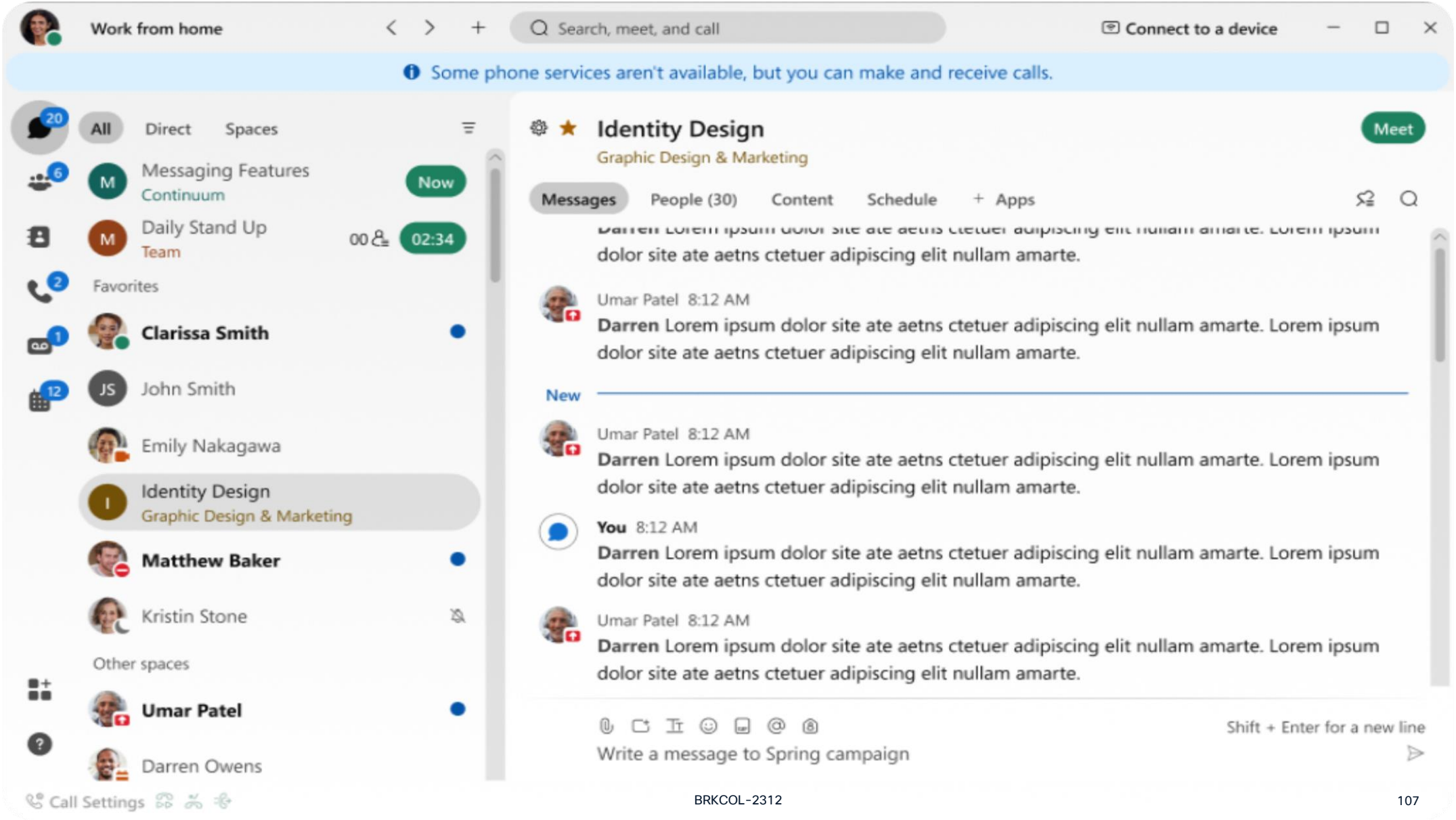
Interface Design – IP Phones



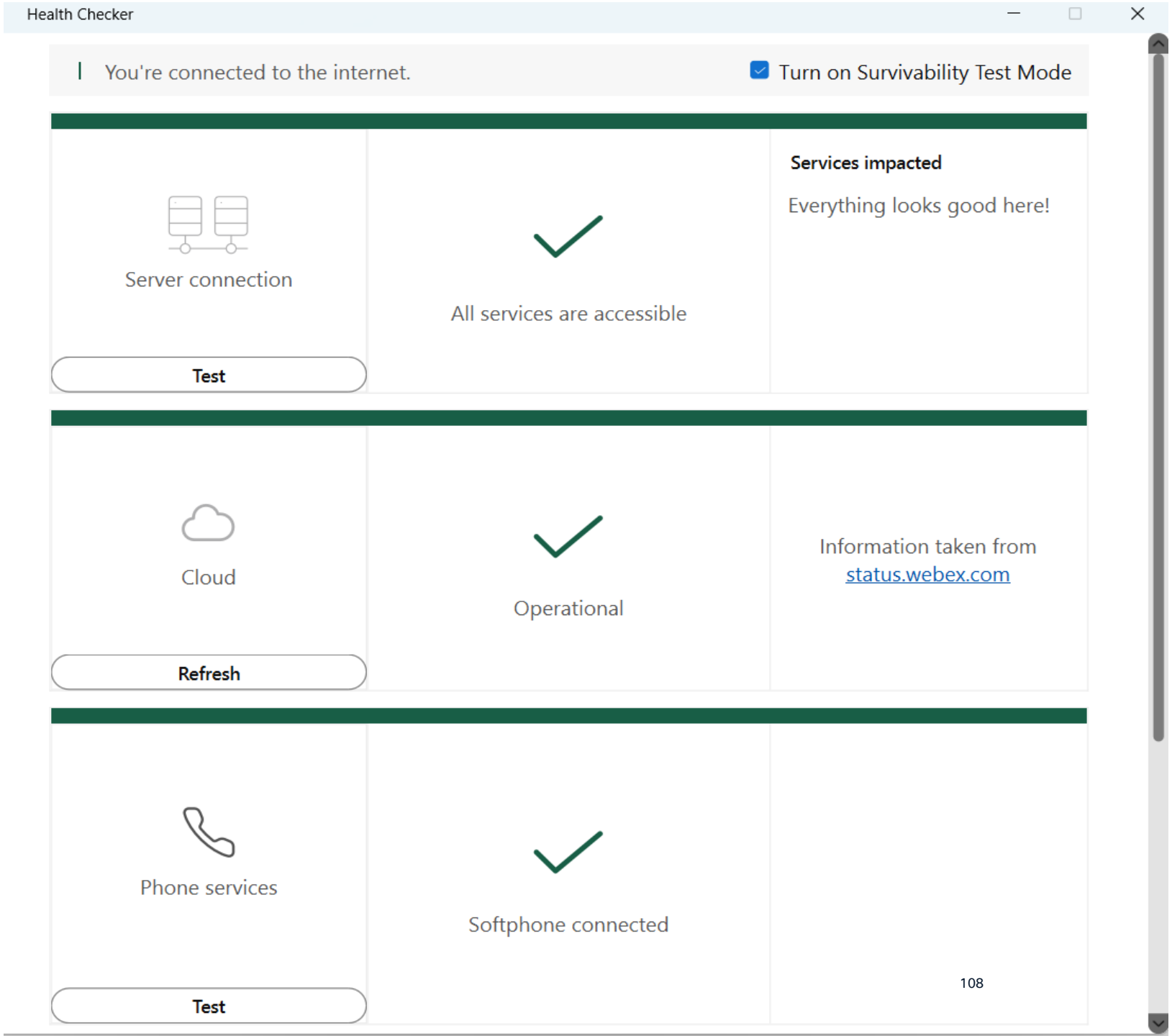




Interface Design – Webex Desktop App

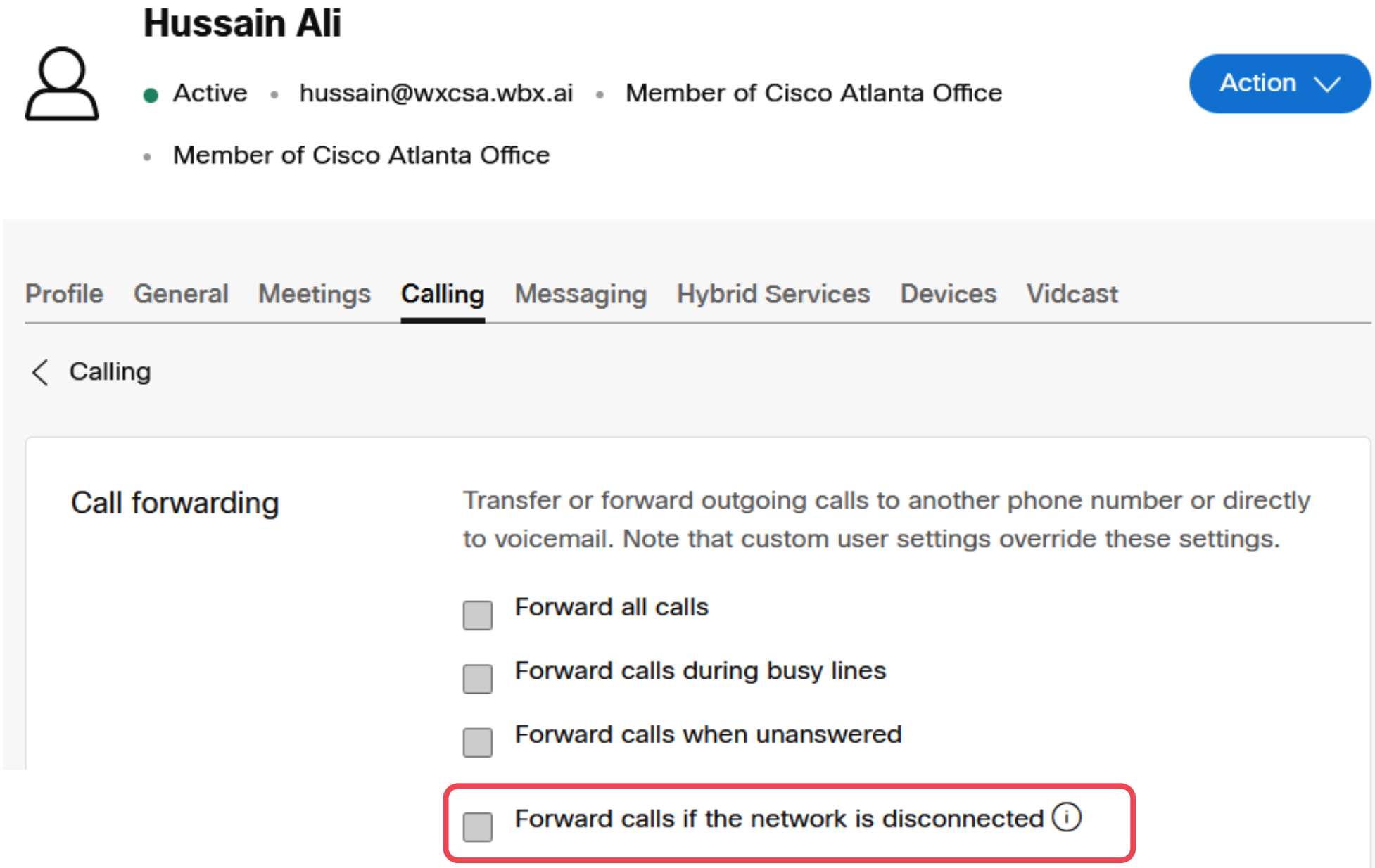


Test mode in the Webex App

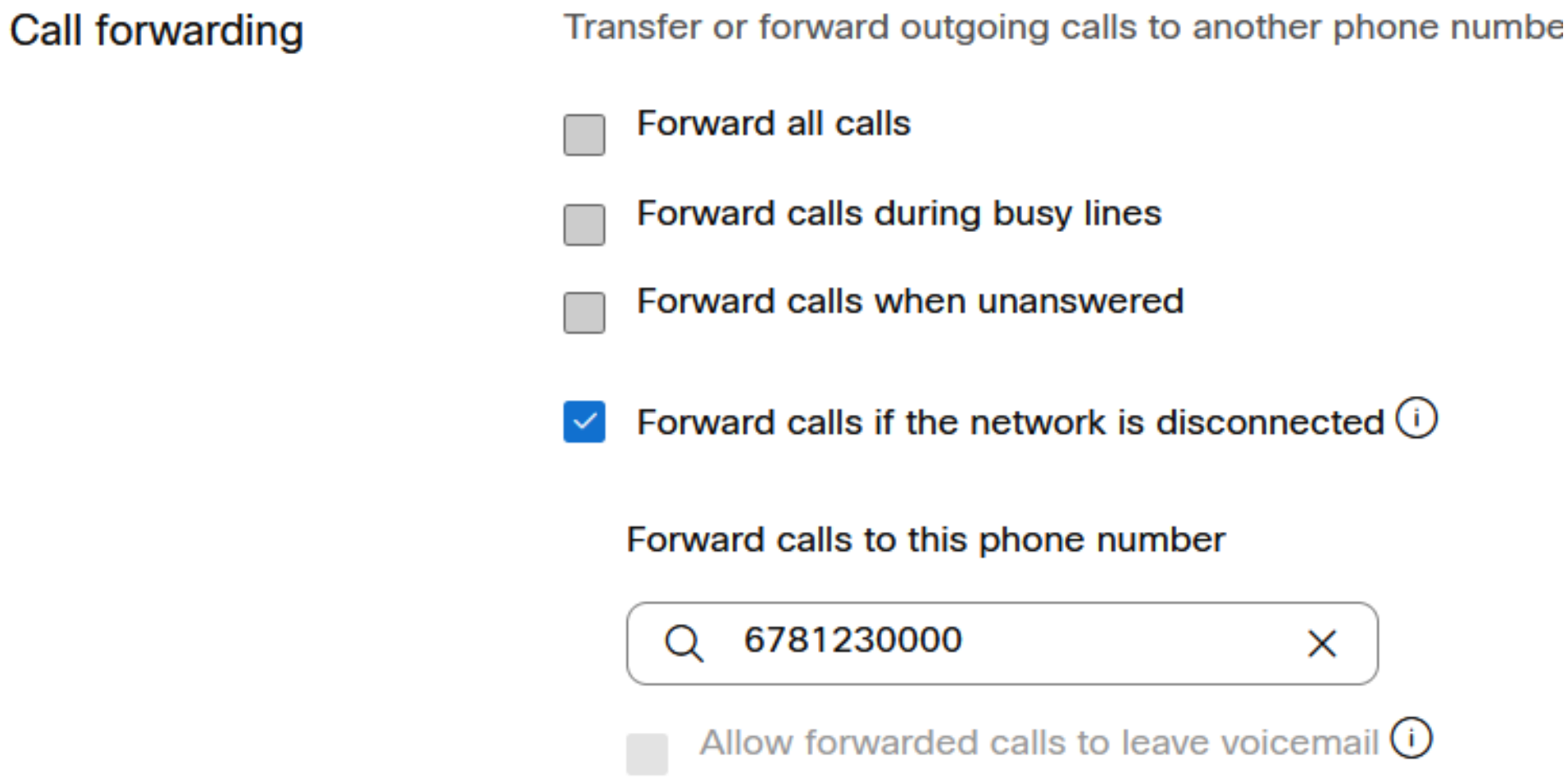


Call Forward Unreachable Setting

Updated Call Forward Unreachable option

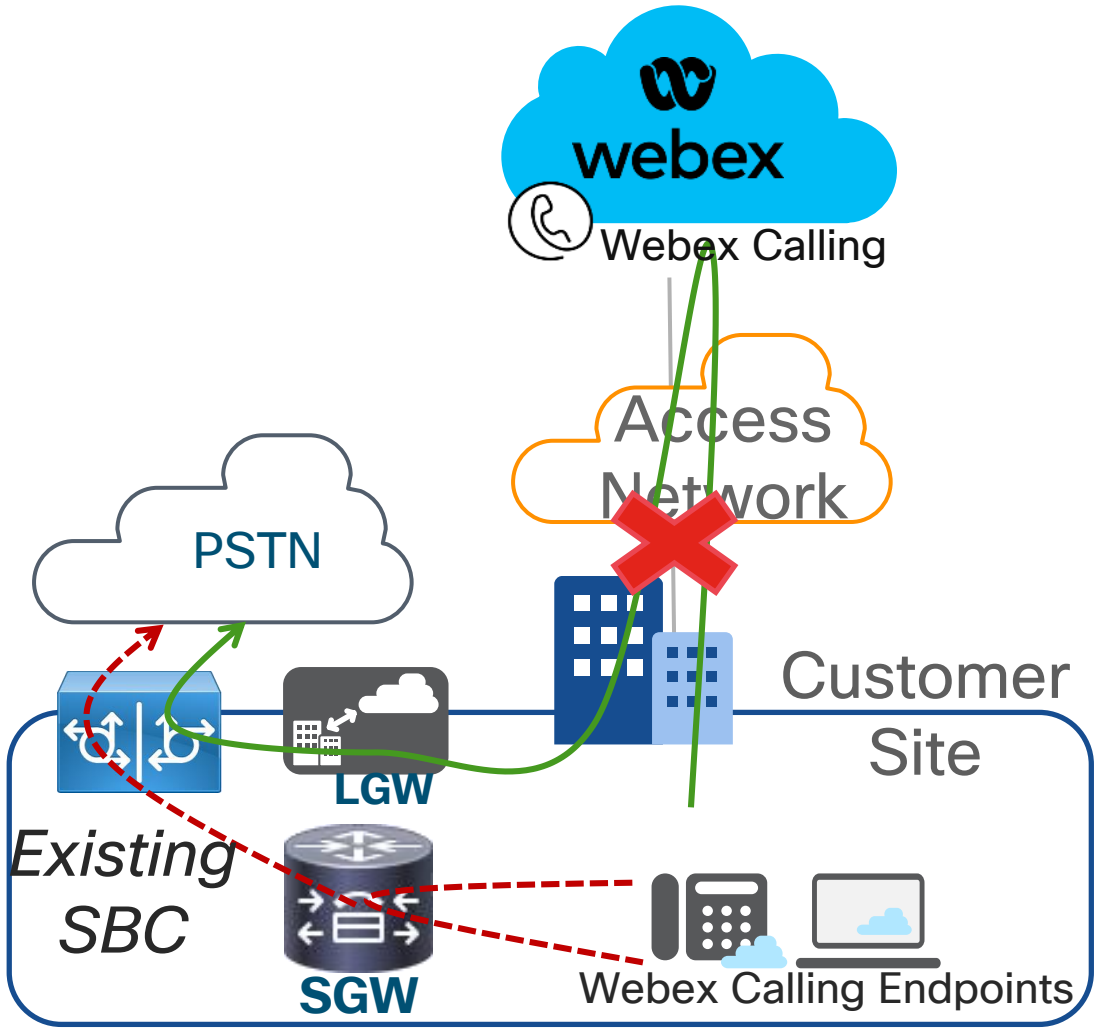


Updated Call Forward Unreachable option



Survivability Call Flows

Site Survivability for Webex Calling – Single Site

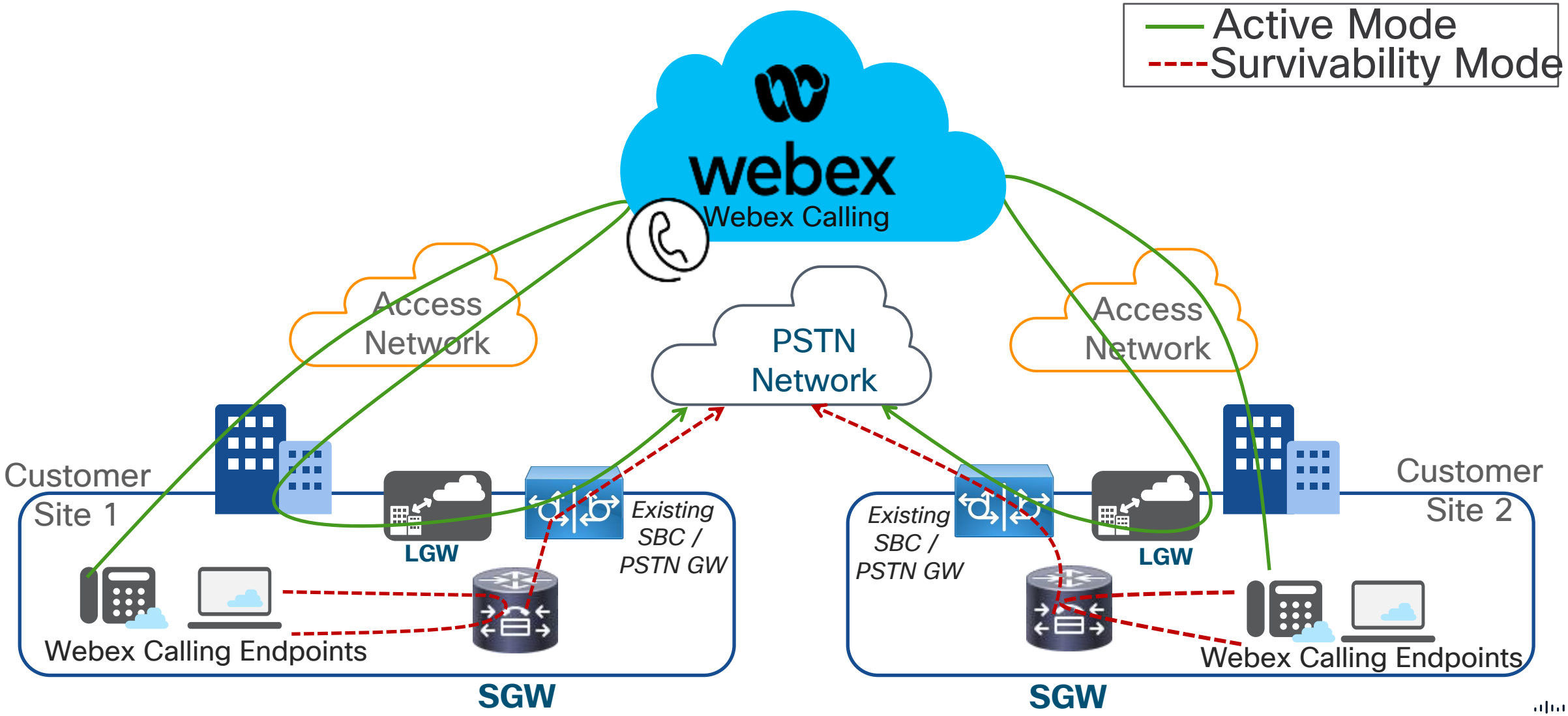


Survivability Mode

- Extension basic dialing
- Outbound/Inbound calls via local PSTN Gateway
- Config default trunk towards PSTN gateway

Active Mode
Survivability Mode

Site Survivability for Webex Calling – Multi Site

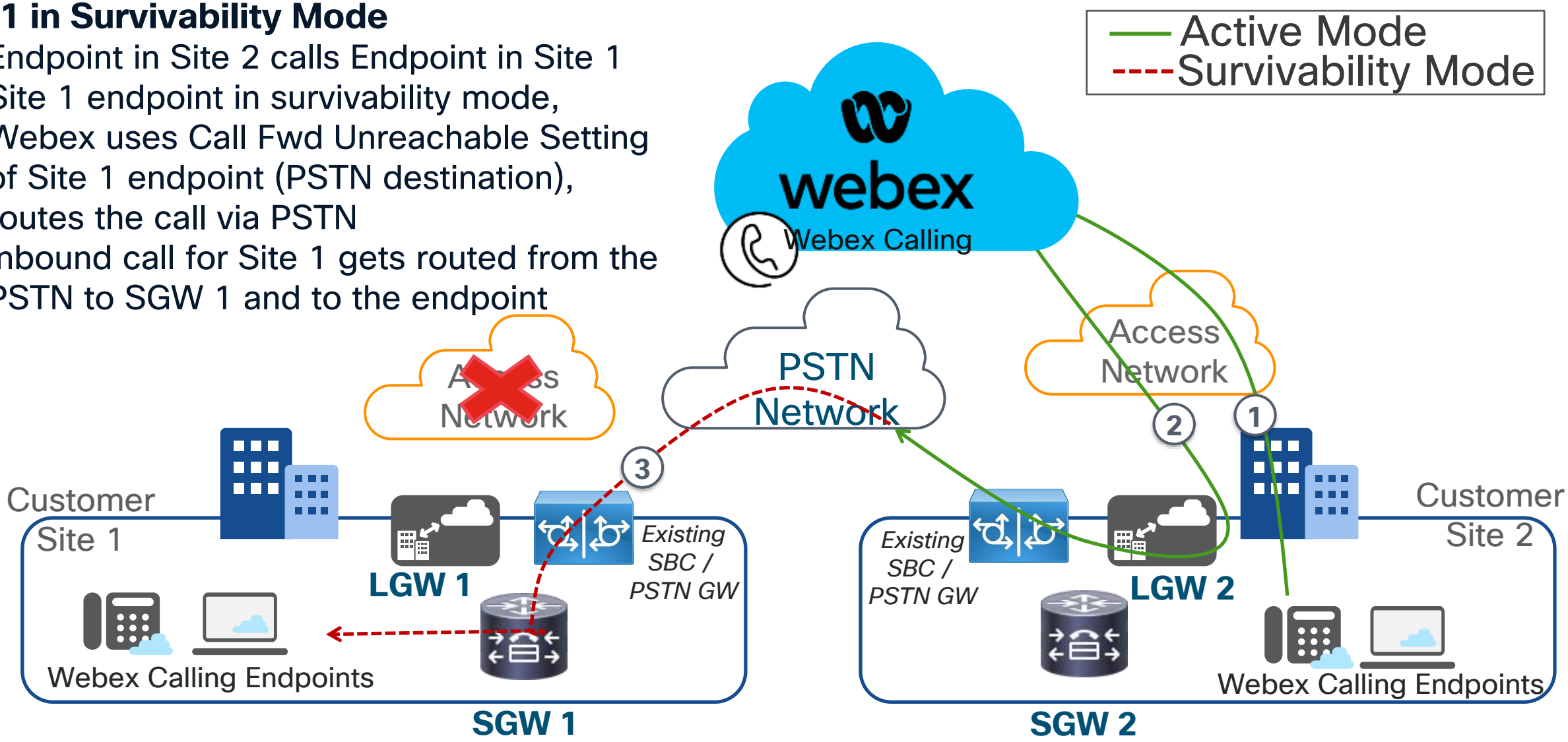


Site Survivability for Webex Calling

Inter site call routing Active Mode Site to Survivability Mode Site

Site 1 in Survivability Mode

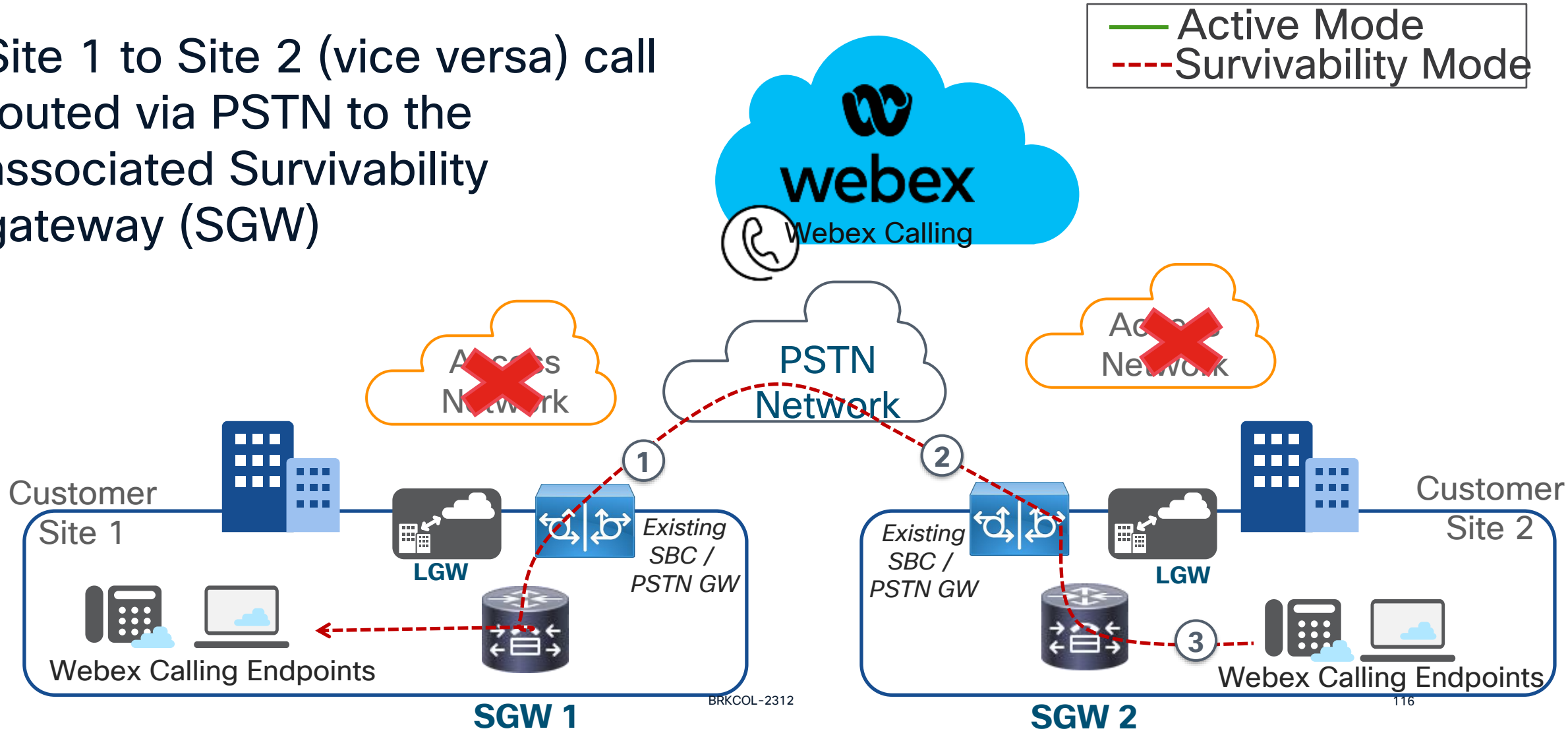
1. Endpoint in Site 2 calls Endpoint in Site 1
2. Site 1 endpoint in survivability mode, Webex uses Call Fwd Unreachable Setting of Site 1 endpoint (PSTN destination), routes the call via PSTN
3. Inbound call for Site 1 gets routed from the PSTN to SGW 1 and to the endpoint



Site Survivability for Webex Calling

Inter site call routing Active Mode Site to Survivability Mode Site

Site 1 to Site 2 (vice versa) call routed via PSTN to the associated Survivability gateway (SGW)

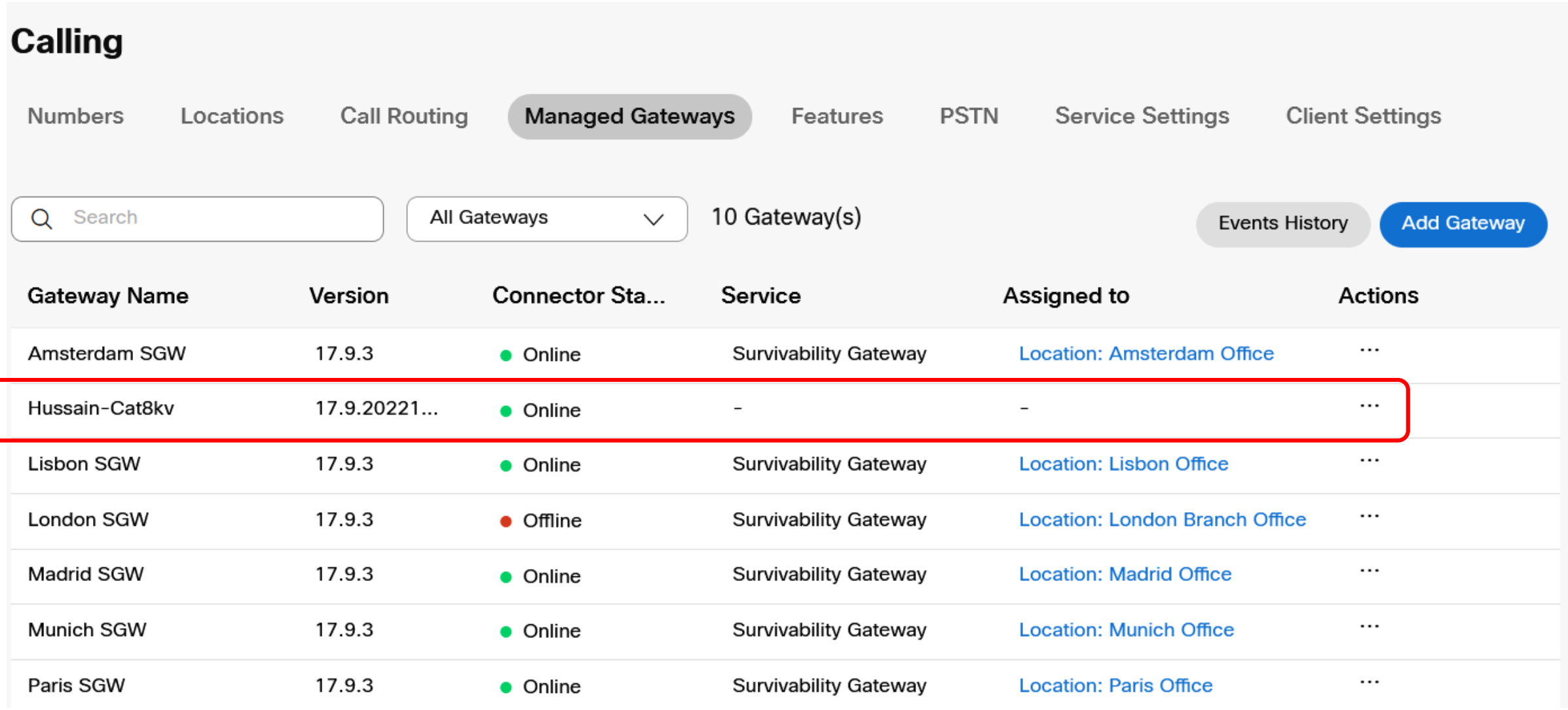


Site Survivability
Deployment Workflow

Site Survivability Prerequisite

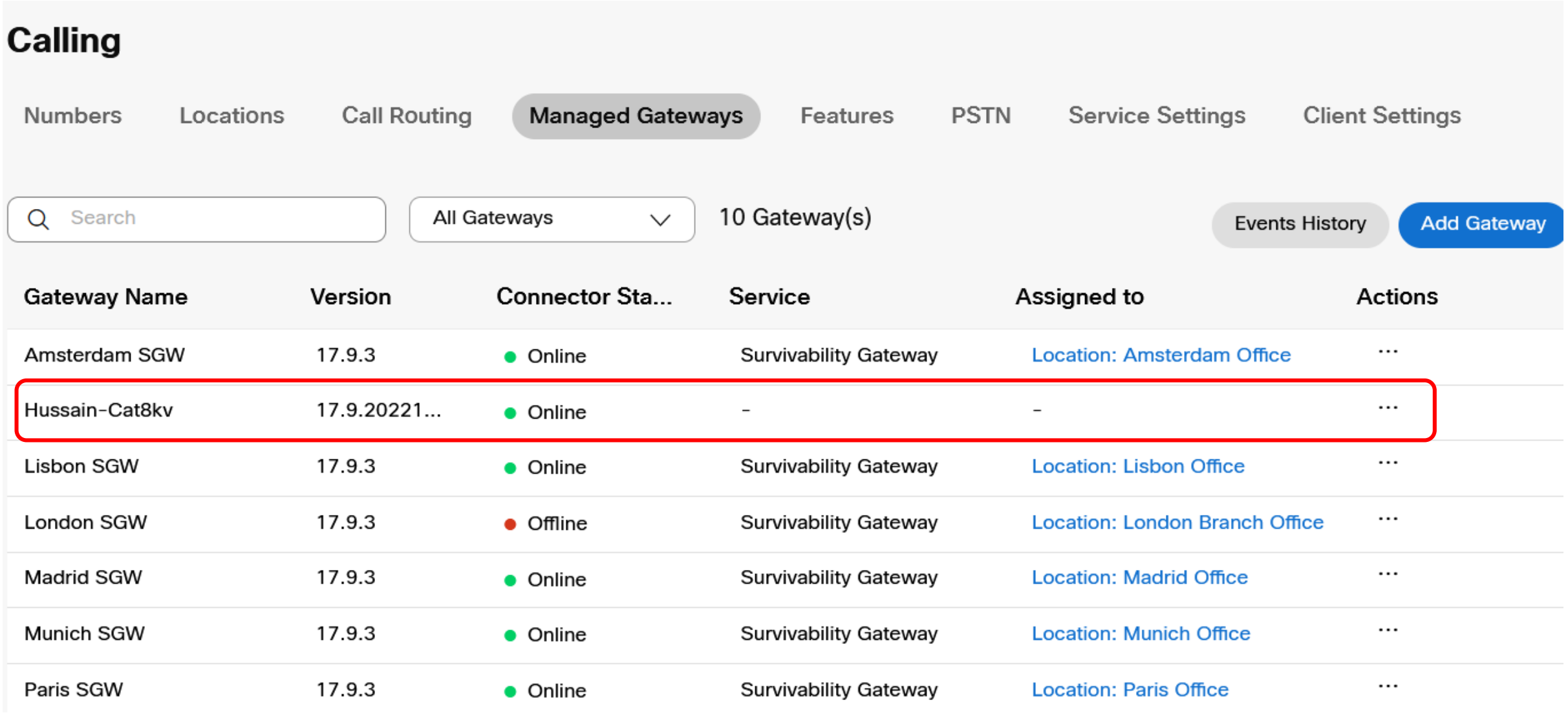
- IOS-XE Gateway, that is, SGW should be connected to Cisco Webex Control Hub as mentioned in the below onboarding connector document
- https://help.webex.com/en-us/article/xftgfc/Enroll-Cisco-IOS-Managed-Gateways-to-Webex-Cloud

Managed Gateway is a must for deploying an SGW

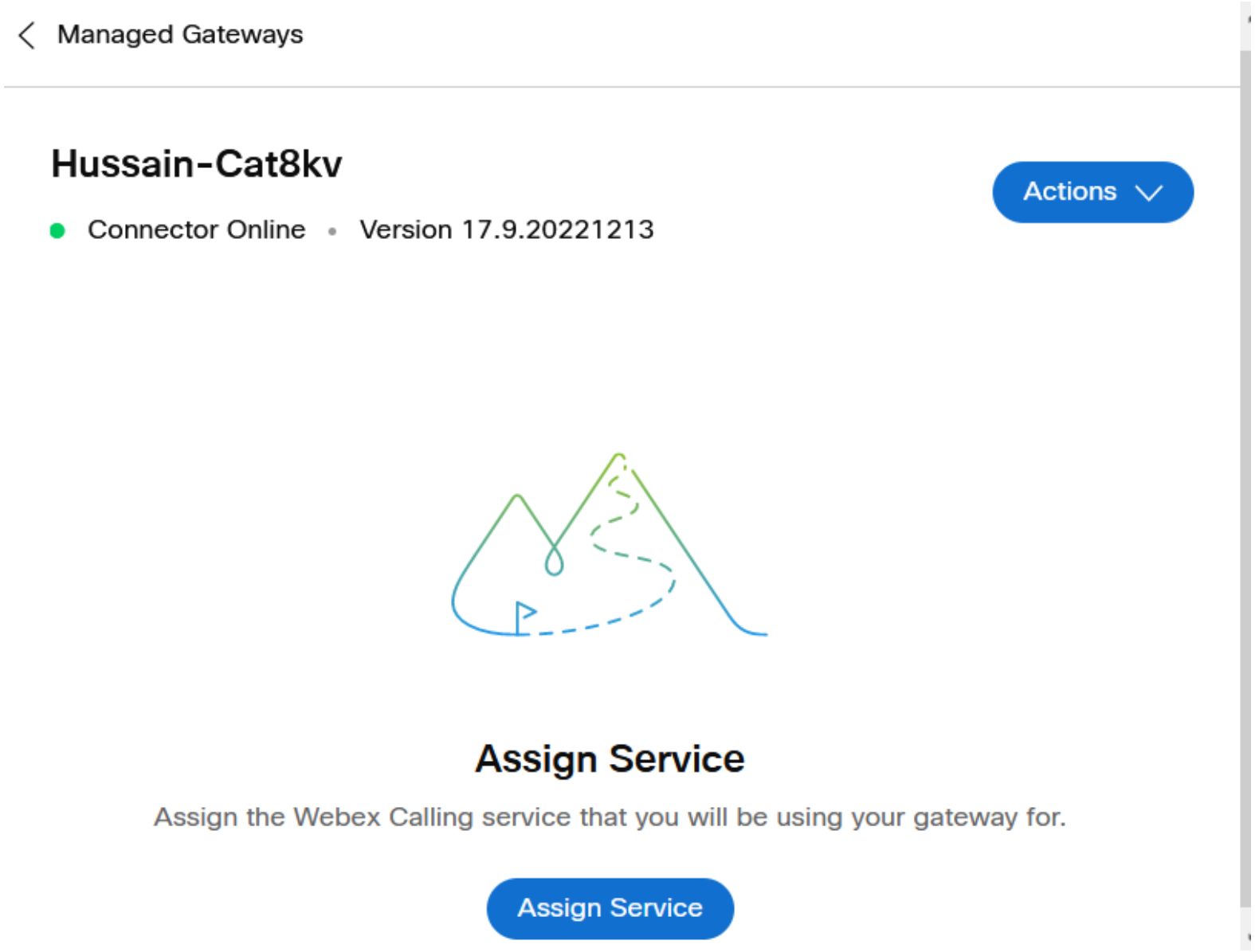


Assign Survivability Service to the Gateway from within the Control Hub

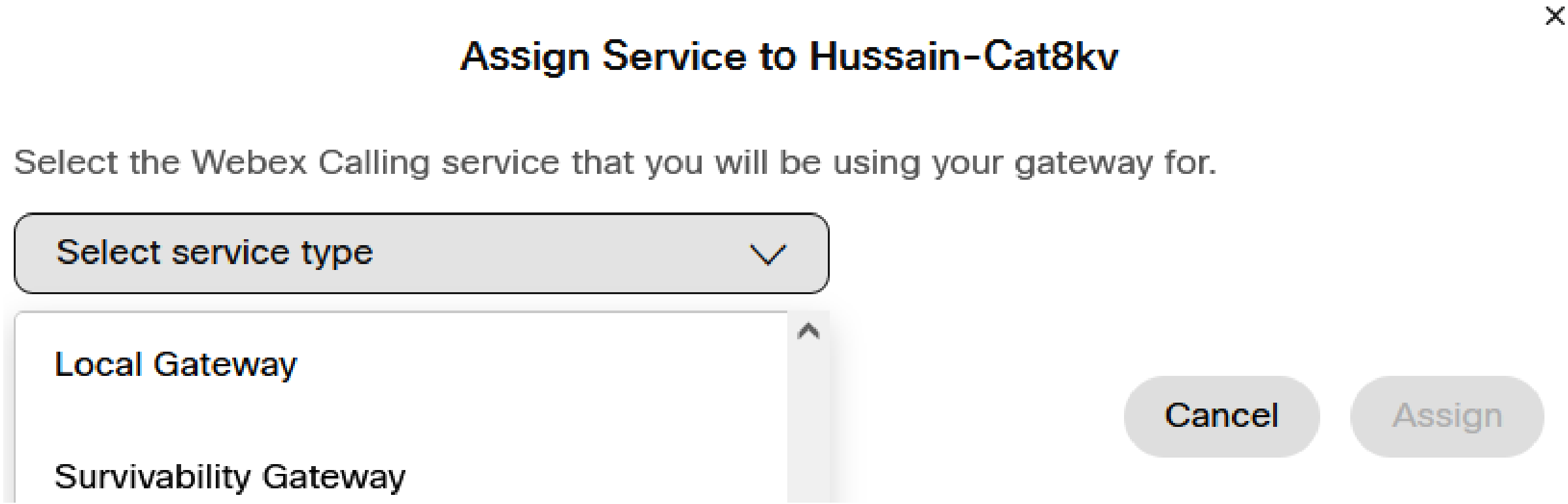
Managed Gateway now **Online**



Assign a Service to the Managed Gateway

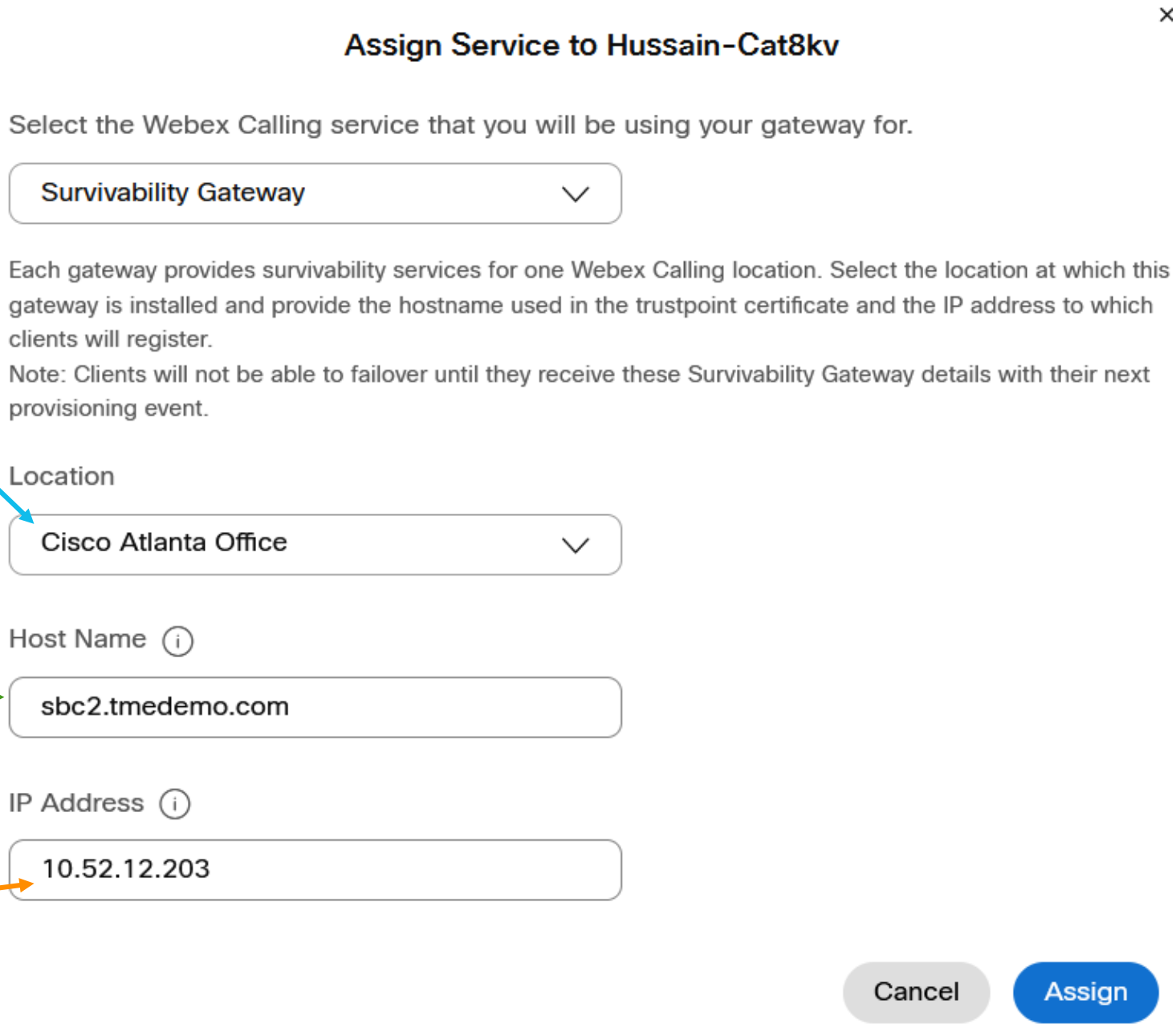


Service Type: LGW or SGW

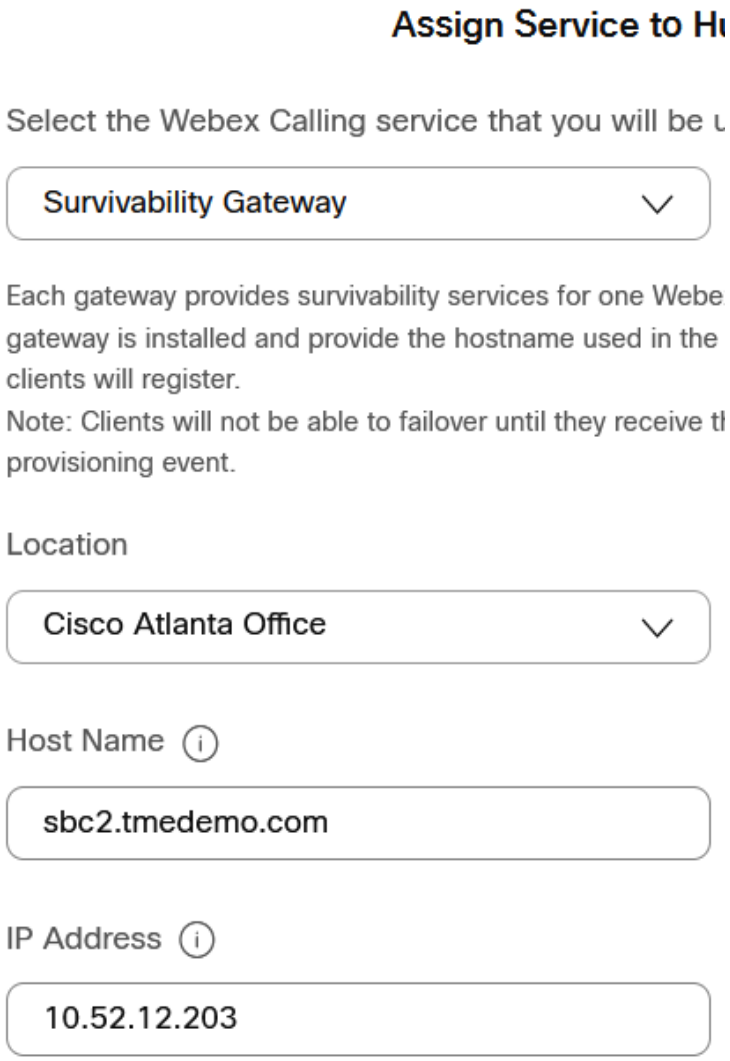


Select Survivability Gateway as the Service

- Endpoints belonging to this Location will map to this SGW
- Host Name: This would be the hostname / FQDN used in the certificate required for establishing the TLS connection with clients
- IPv4 address of the gateway interface where the endpoints will register



Endpoint Config update



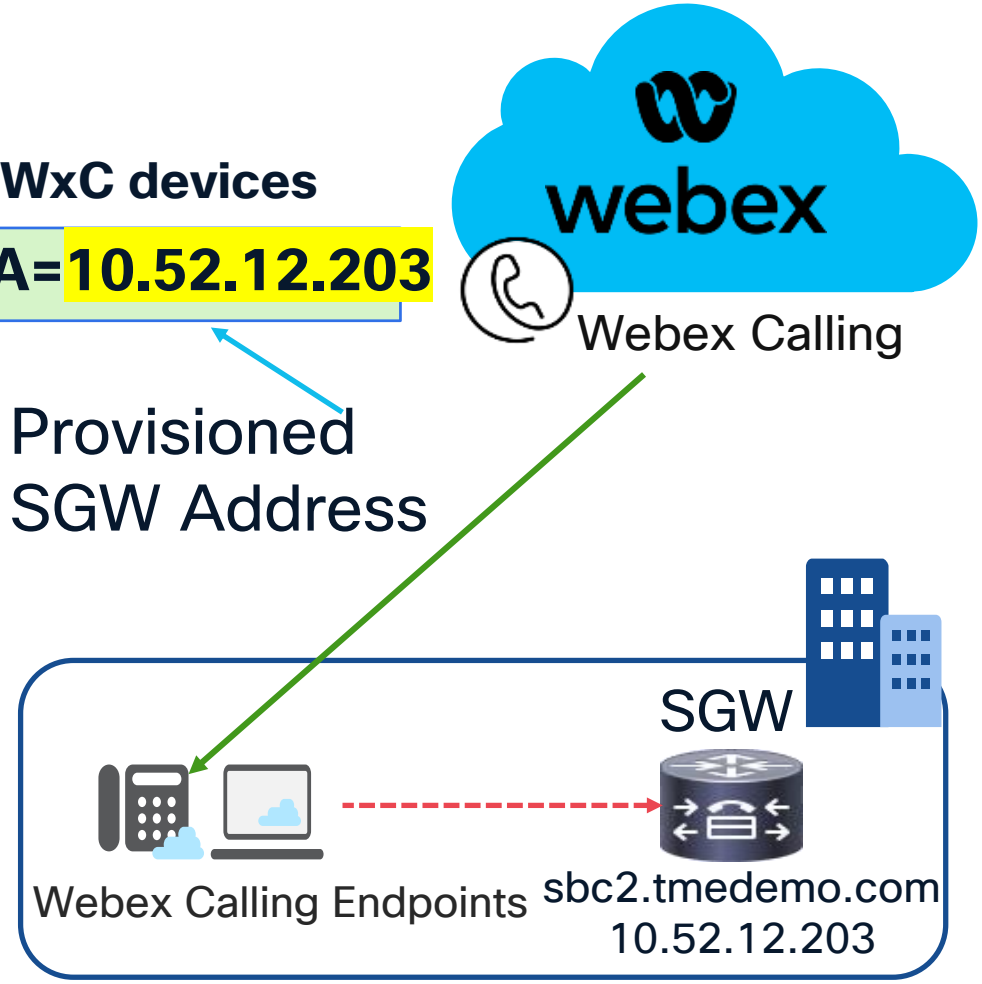
Survivability Proxy Value for WxC devices

sbc2.tmedemo.com:8933:A=10.52.12.203

Provisioned SGW hostname (FQDN)

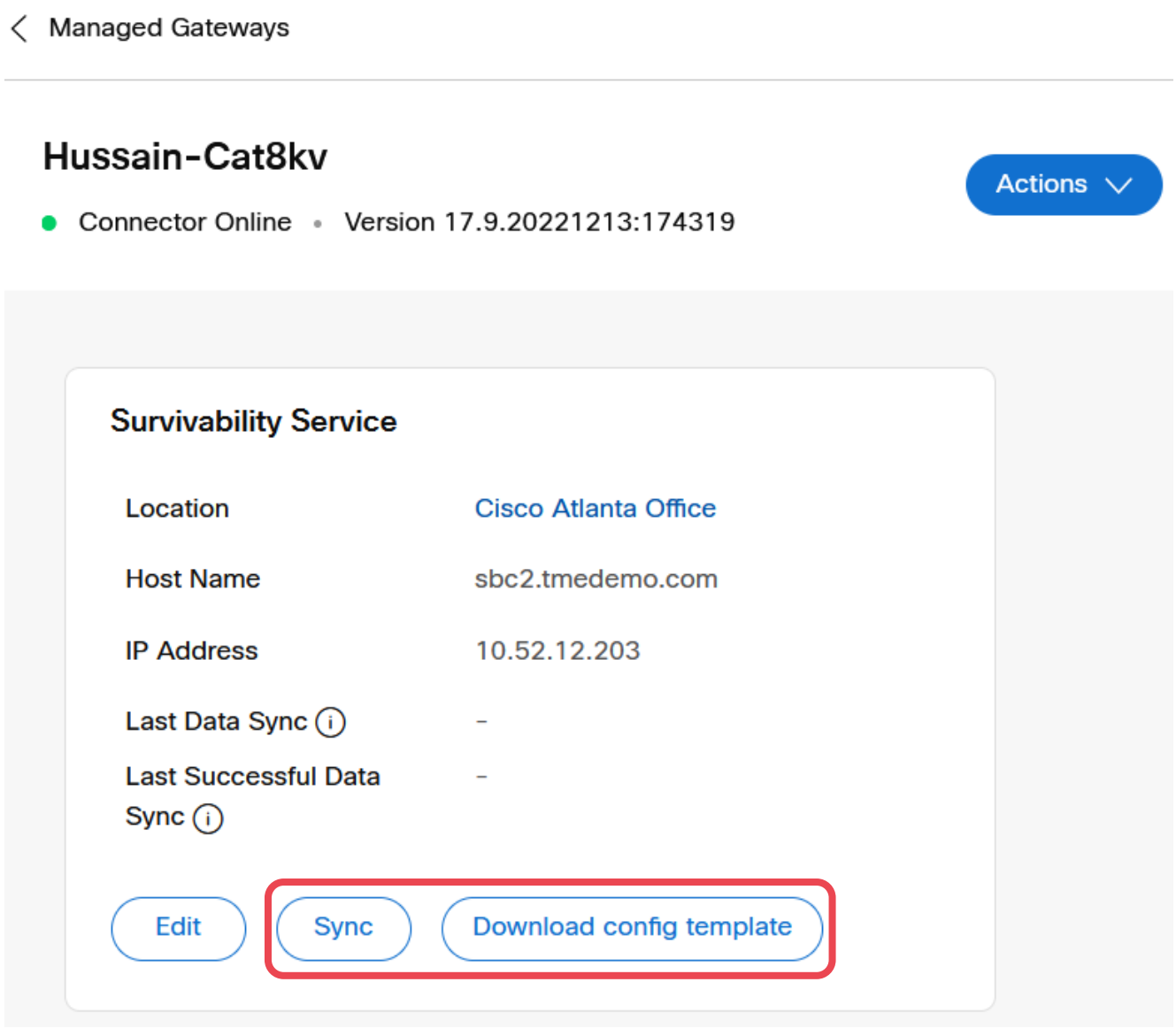
SGW Port (Fixed)

Provisioned SGW Address



SGW Sync Operation

- Sync option manually triggers a data download.
  - Start / finish can be verified in the gateway log
  - Status card can take up to 10 mins to update following a manual Sync.
  - Data is downloaded automatically every night by the gateway.



Dial-peer voice summary

```
SGW#show dial-peer voice summary
dial-peer hunt 0
                    AD                                      PRE  PASS  SESS-SER-GRP\   OUT
TAG     TYPE  MIN   OPER PREFIX  DEST-PATTERN               FER  THRU  SESS-TARGET     STAT
100     voip  up    up                                       0   syst
101     voip  up    up           +1408944....$              0   syst  ipv4:198.18.133.3
201     voip  up    up           81[2-9]..[2-9]..-          0   syst  ipv4:10.1.40.11
1050    voip  up    up           999                        0   syst  ipv4:198.18.133.186
40001   voip  up    up           +14085551074$              0   syst  ipv4:198.18.133.39:5
40002   voip  up    up           +14085557700$              0   syst  ipv4:198.18.133.38:5
```

Dynamic (Virtual) dial-peers are setup for SIP SRST/SGW endpoints when they register to the SRST/SGW platform

```
dial-peer voice 40001 voip
 destination-pattern pjkj1ffadg$
 redirect ip2ip
 session target ipv4:198.18.133.39:5074
 session protocol sipv2
 dtmf-relay rtp-nte
 digit collect kpml
 voice-class codec 1
  after-hours-exempt    FALSE
```

Co-locating Survivability Gateway (SGW) and Local Gateway (LGW)

Destination Dial-peer Group Limitation

```
voice class dpg 10000
  description Voice Class DPG for SJ
  dial-peer 1002 preference 1
  dial-peer 1003
!
dial-peer voice 100 voip
 description Inbound DP
 incoming called-number 1341
 destination dpg 10000
```

Outbound DP is selected based on preference listed in DPG only

```
dial-peer voice 786 voip
 destination-pattern 1341
 session protocol sipv2
 session target ipv4:10.1.1.1
!
dial-peer voice 1002 voip
 destination-pattern 3333
 session protocol sipv2
 session target ipv4:10.1.1.2
!
dial-peer voice 1003 voip
 destination-pattern 4444
 session protocol sipv2
 session target ipv4:10.1.1.3
```

Received:

INVITE sip:1341@<CUBE-IP> SIP/2.0

Sent:

INVITE sip:1341@10.1.1.3 SIP/2.0

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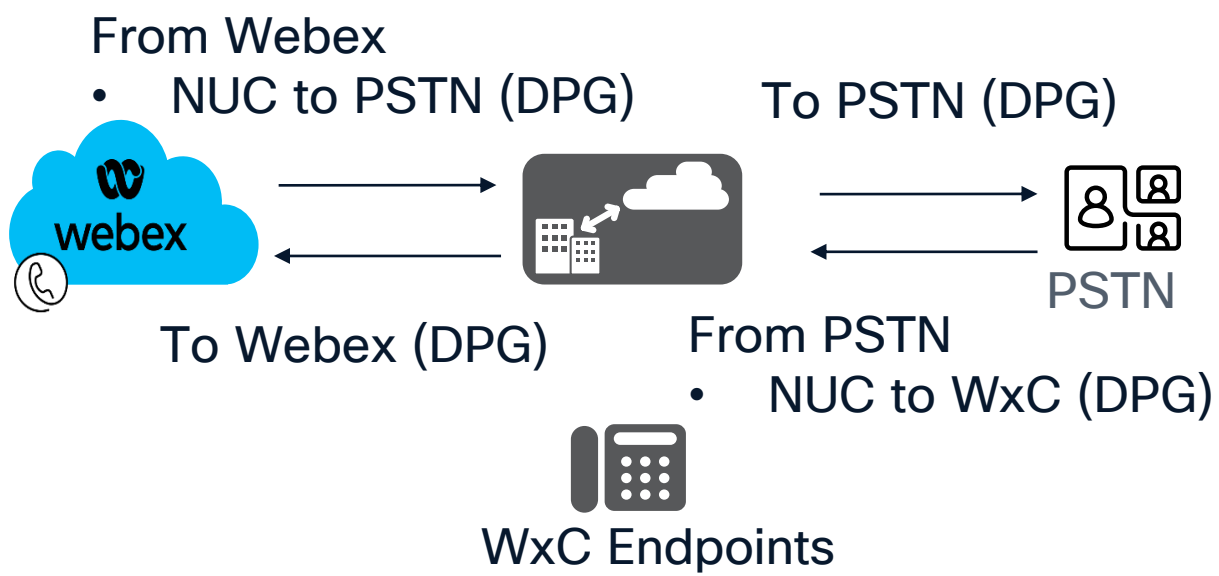
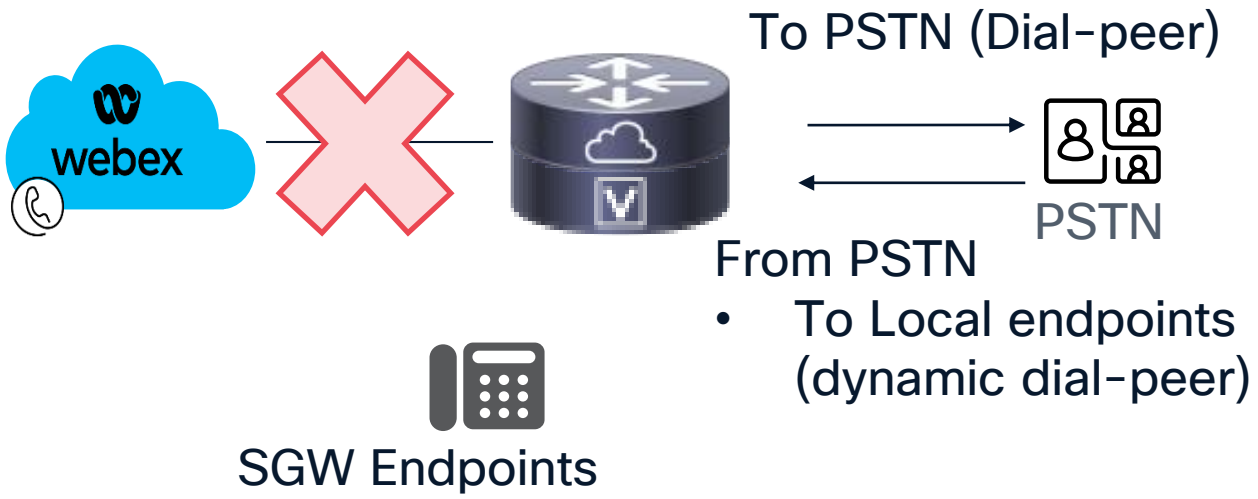
Call Routing Overview – Existing LGW and SGW Operation





Standalone Local Gateway in Cloud mode

- Nailed up connections (NUC) using Dial peer groups
- Simplistic approach, does not require admins to have knowledge of Webex Calling or PBX dial plans or configure these patterns on the LGW
- Challenge: No way to exit DPG and match system (dynamic) dial peers when Lines register locally on the router

Site Survivability Gateway during Cloud/WAN outage

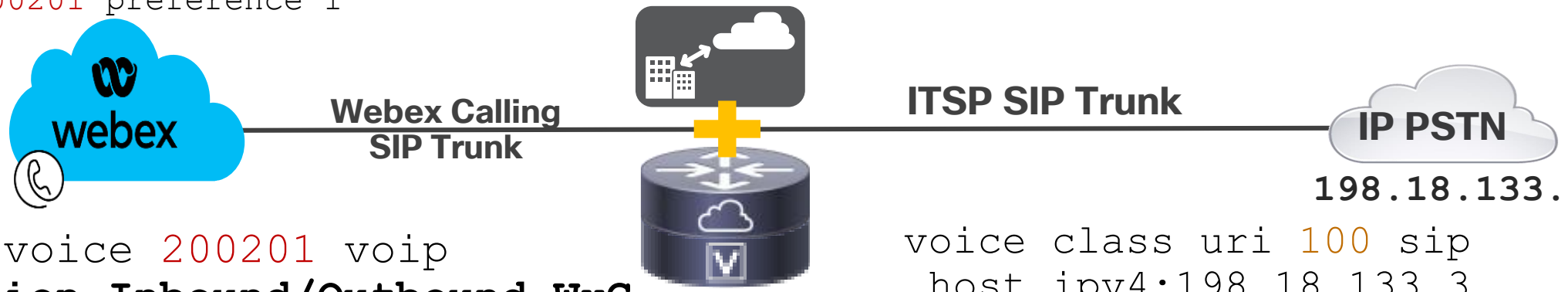
- Local extension calling – Webex Calling SGW Phones use system (dynamic) dial-peers (no config needed)
- Outbound line to PSTN use Dial-peer based routing
- Inbound PSTN routes to local lines first. Lines (dynamic dial-peers) always have precedence

Co-located LGW and SGW config summary

```
voice class uri 200 sip
  pattern dtg=hussain2572_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 to IP PSTN
 destination-pattern .T
 preference 3
```

→ **Replace** `BAD.BAD` **with** `.T`
→ **Add** `preference 3`




```
dial-peer voice 200201 voip
  description Inbound/Outbound WxC
  destination dpg 100
  destination-pattern .T
  preference 2
  incoming uri request 200
  voice-class sip tenant 200
```

→ **Replace** `BAD.BAD` **with** `.T`
→ **Add** `preference 2`

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

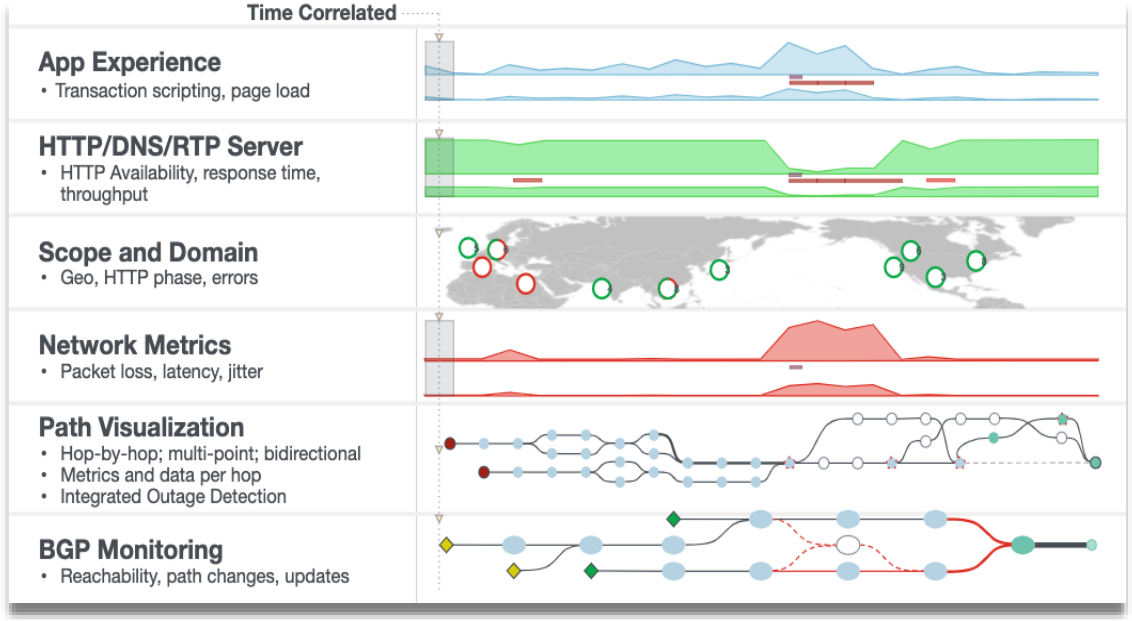
```
dial-peer voice 100 voip
 description Incoming from IP PSTN
 incoming uri via 100
 destination dpg 200
```

→ **Remove** `destination dpg 200` **from inbound ITSP dial-peer to allow operation in SGW mode**

Thousand Eyes

ThousandEyes – Delivering Network Intelligence

A SaaS based I/internet and cloud intelligence solution



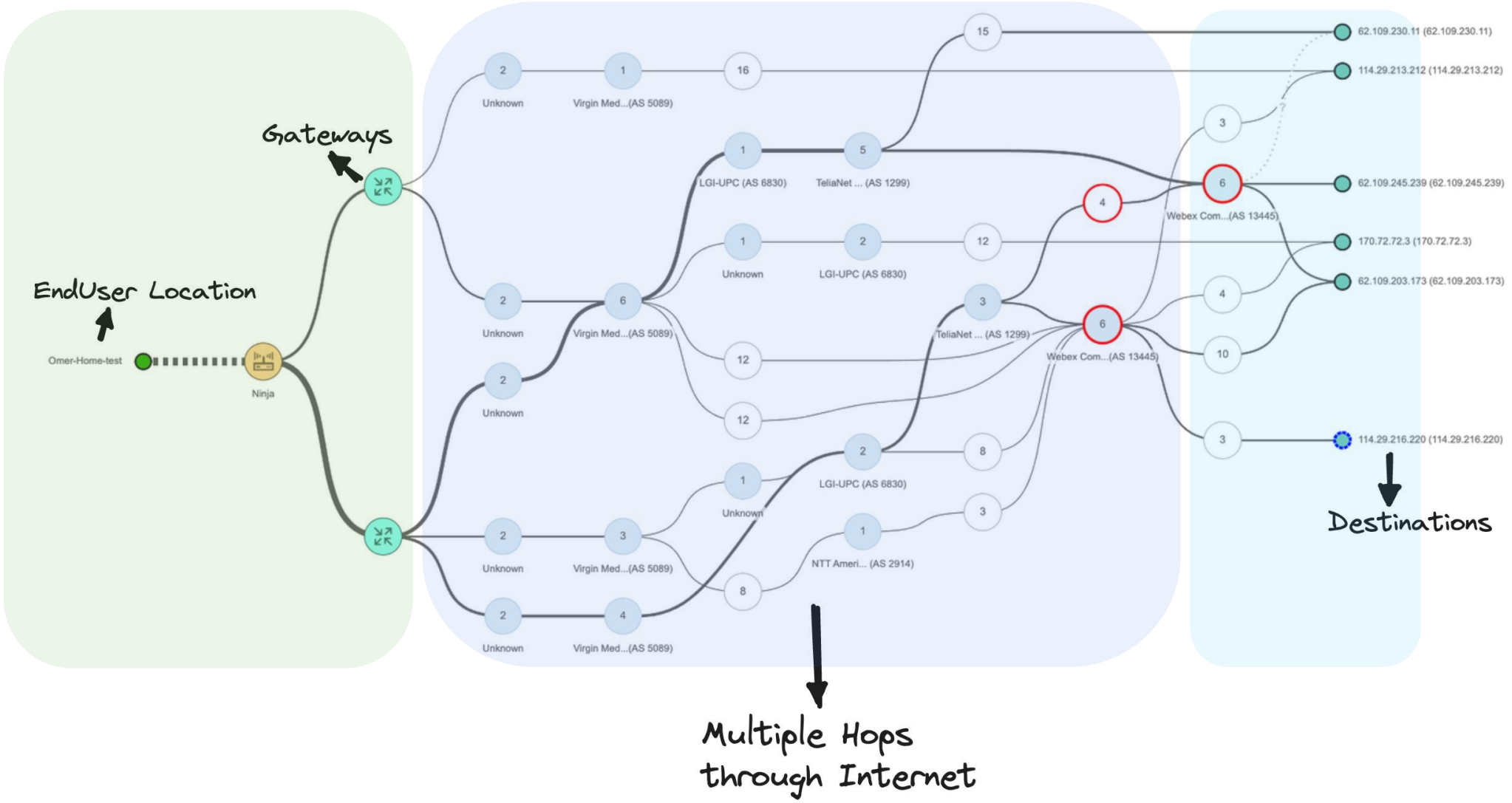
Superior Network Experience

Collective Intelligence

Interactive Collaboration

ThousandEyes – Delivering Network Intelligence

A SaaS based I/internet and cloud intelligence solution



ThousandEyes - Agent Types

ThousandEyes Agents

ThousandEyes

Cloud Agents

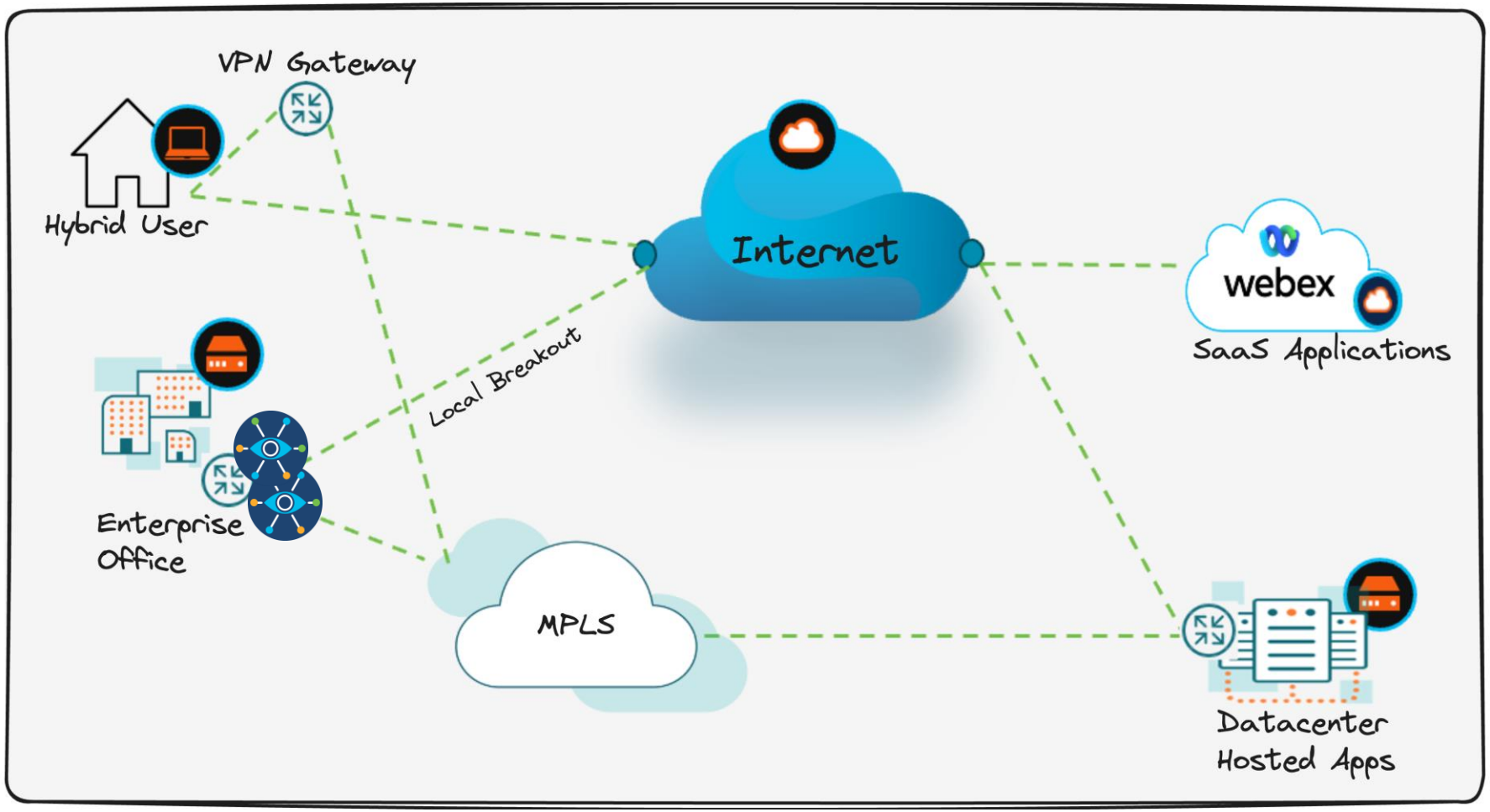
Enterprise Agents

Endpoint Agents

Agents / Vantage Points

Enterprise Agents

Enterprise Agents – Monitoring Internal and External Networks



Enterprise Agents + Cube

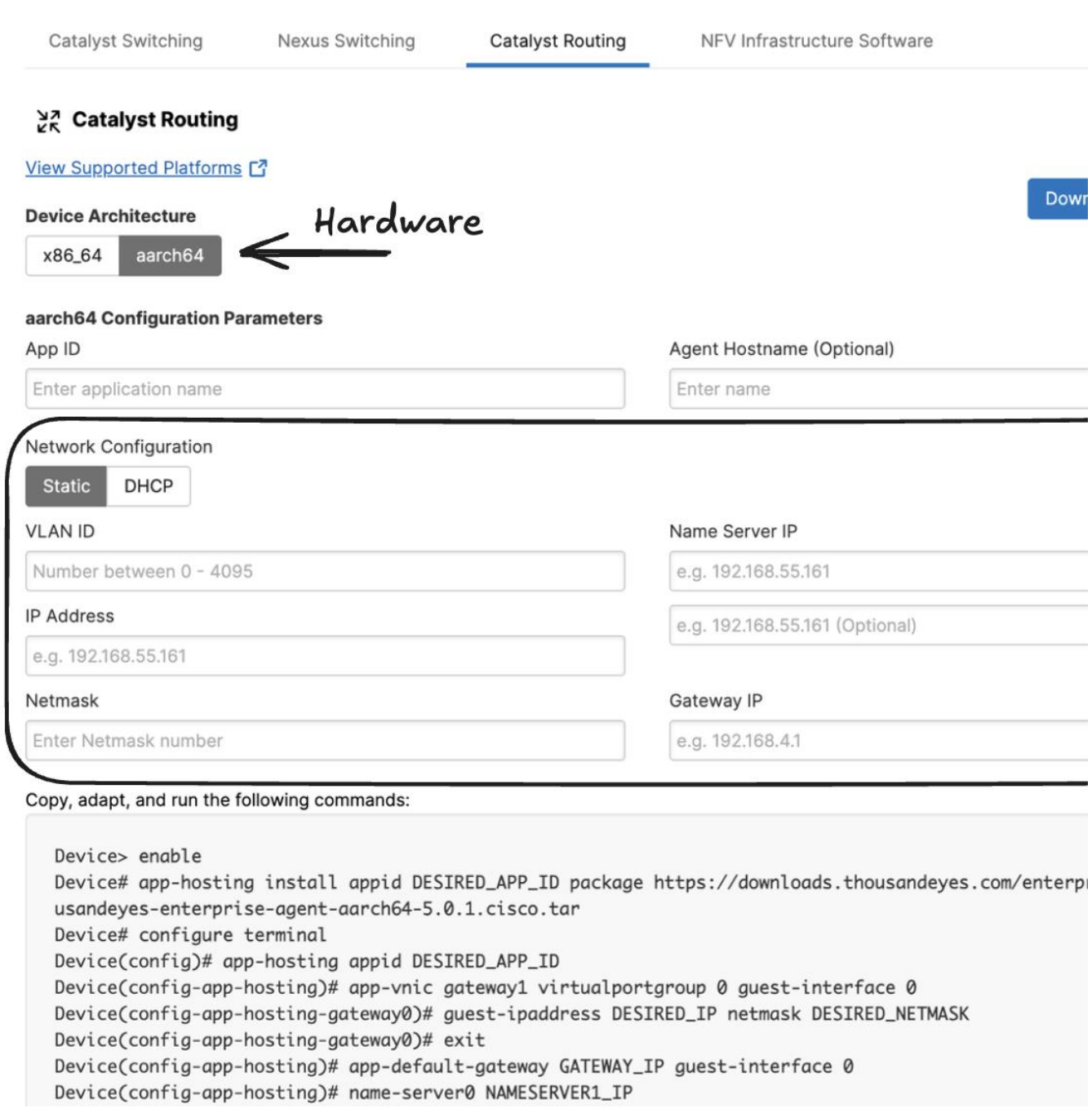
Pre-Requisites

- Enterprise Agent can be installed on the Cisco router, using one of three available methods
  1. SDWAN Manager Feature Template
  2. SDWAN Manager TE workflow
  3. Using Docker
- Both SD-WAN and Autonomous mode are supported.
- Min IOS version IOS-XE 17.6.1 or later
- The filename structure for ThousandEyes agent images is:

thousandeyes-enterprise-agent-aarch64-5.0.1.cisco.tar

Architecture

Version



Verification of Agents Installation

```
router# sh app-hosting list
App id                                   State
thousandeyes_enterprise_agent            RUNNING
```

```
router#sh app-hosting detail appid thousandeyes_enterprise_agent
App id : thousandeyes-enterprise-agent
Owner : iox
State : RUNNING
Application
  Type : docker
  Name : ThousandEyes Enterprise Agent
  Version : 5.0.1
  Description : Perform active synthetic measurements to your business-critical
  Author : ThousandEyes support@thousandeyes.com
  Path : https://downloads.thousandeyes.com/enterprise-agent/thousandeyes-enter
  URL Path : bootflash:./thousandeyes-enterprise-agent-x86_64-5.0.1.cisco.tar
Activated profile name : custom

Resource reservation
  Memory : 500 MB
  Disk : 1 MB
  CPU : 1850 units
  CPU-percent : 9 %
  VCPU : 1

Platform resource profiles
  Profile Name                  CPU(unit) CPU(percent)  Memory(MB)  Disk(MB)
  ------------------------------------------------------------------------

Attached devices
  Type              Name               Alias
  ---------------------------------------------
  serial/shell     iox_console_shell   serial0
  serial/aux       iox_console_aux     serial1
  serial/syslog    iox_syslog          serial2
  serial/trace     iox_trace           serial3

Network interfaces
   ---------------------------------------
eth0:
   MAC address         : 52:54:dd:e9:9e:9f
   IPv4 address        : 172.29.1.11
   IPv6 address        : ::
   Network name        : VPG0
```

References

CUBE Resources

- CUBE Configuration Guide Through IOS-XE 17.5
- CUBE Configuration Guide – IOS-XE 17.6 Onwards
- vCUBE support on Azure
- vCUBE on AWS
- CUBE Interop Portal including Direct Routing Application Note
- 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

SGW and LGW Resources

For more information take a look at the following resources:

- Webex Calling Trunks
- Local Gateway Configuration Guide
- Enroll Cisco IOS Managed Gateways to Webex Cloud
- Assign Services to Managed Gateways
- Site Survivability for Webex Calling
- Colocation of LGW and SGW

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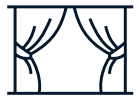

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