

July 2021

Walkthrough Wednesday

Flexible PSTN options for your
Webex Calling deployment

Hussain Ali

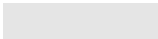
Technical Marketing Engineer

July 2021

Welcome to the
Webex Community!

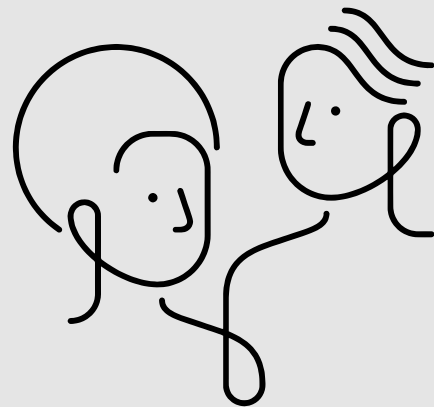
Subscribe to our Announcements Board

The screenshot shows the Cisco Community Announcements board interface. At the top, there is a navigation bar with a dropdown menu set to "This board", a search bar labeled "Search Announcements", and an "Options" button. Below this is a category bar with icons for "Technology & Support", "For Partners", "Customer Connection", and "Webex". The breadcrumb trail reads "Cisco Community / Webex / Webex User Community / Announcements". A green message states "Success! Subscription removed." A large black button with white text displays the URL "cs.co/CommunityAnnouncements". Below the button, the heading "Announcements" is followed by a paragraph: "Subscribe to the board to stay in the know with Webex announcements, community events, feature updates, and...". A context menu is open on the right side, listing various actions: "Community Admin", "Board Admin", "Mark all as New", "Mark all as Read", "Float this item to the top", "Subscribe" (highlighted with a blue background and a black arrow pointing to it), "Bookmark", "Subscribe to RSS Feed", "Invite a Friend", and "Threaded format".



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

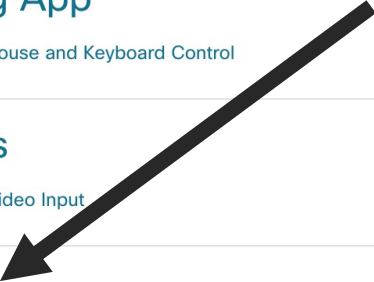
- Connect with 100,000+ other Webex users like you
- Ask questions and share knowledge
- Attend live events with esteemed guests
- Submit product ideas that go straight our Product team
- Participate in contests



Webex User Community

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

	Posts	Replies	Latest Post
<h2>Announcements</h2> <p>LATEST POST - Re: Announcing the Webex Ambassador Program!</p>	54	252	07-20-2021 10:44 AM
<h2>Webex Meetings and Webex App</h2> <p>LATEST POST - Audio Notifications</p>	1824	3618	07-21-2021 05:35 AM
<h2>Webex Events App</h2> <p>LATEST POST - Re: host role - unmuting an attendees during Q&A session (Events classic)</p>	182	306	07-20-2021 11:04 AM
<h2>Webex Training App</h2> <p>LATEST POST - Re: Assign Mouse and Keyboard Control</p>	162	294	07-16-2021 02:40 AM
<h2>Webex Devices</h2> <p>LATEST POST - Reorganize Video Input</p>	141	189	07-20-2021 03:24 PM
<h2>Webex Calling</h2> <p>LATEST POST - Re: Do Not Disturb/Out of Office Sync</p>	84	134	07-21-2021 06:54 AM



Agenda

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

Webex Calling PSTN Options

Integrated and flexible calling plans

Customers can consume the way they want



Local Gateway

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

Cloud Connected Calling Provider

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

Service Provider (Cisco Cloud Calling + PSTN)

Service Provider offer that bundles their Calling solution with Webex services

Cisco Calling Plans

New

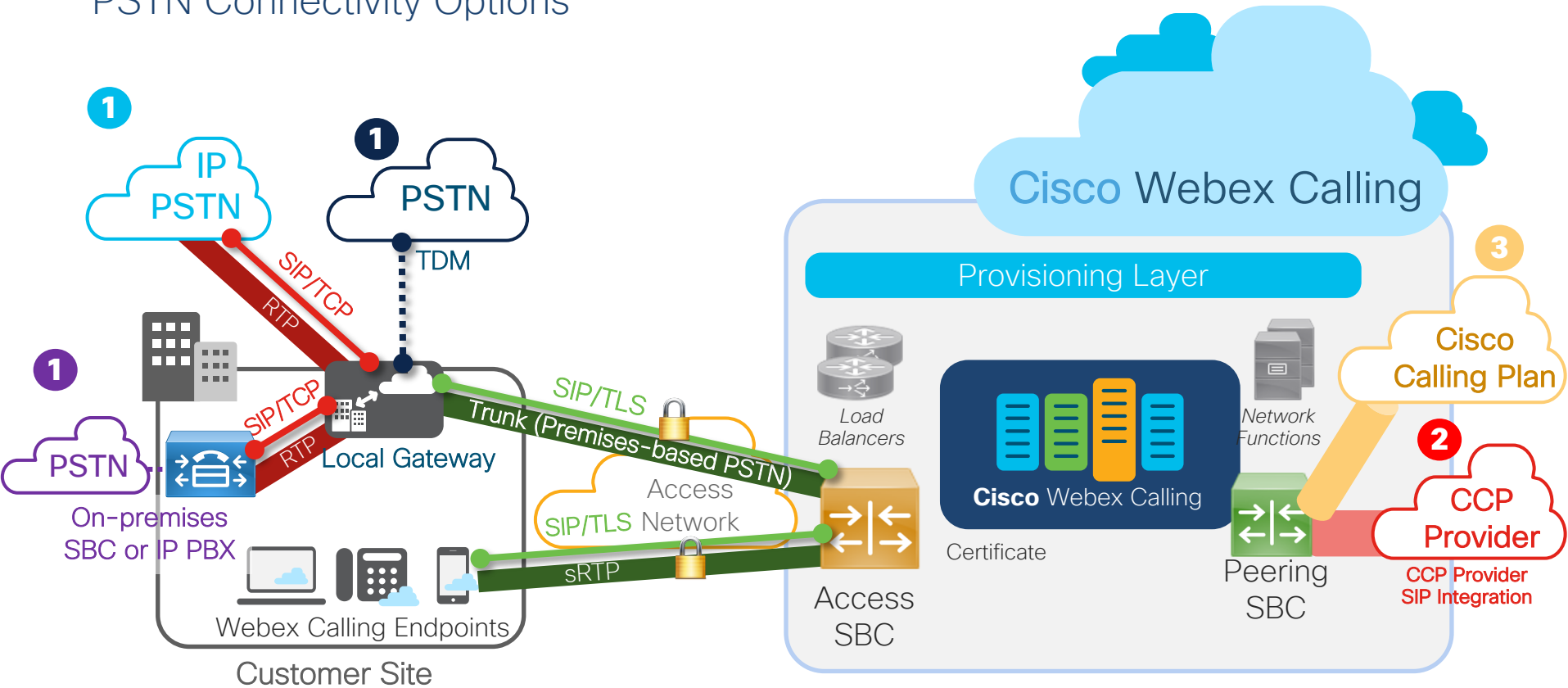
- Intended for VARs ONLY
- Cisco provided Cloud Calling integration into our Webex services
- Easy automated setup & centralized management in Control Hub
- Single order, single bill to customer from partners

Partner Calling Plans Available Today

Cisco Calling Plans

Webex Calling

PSTN Connectivity Options



Setting up Cisco Calling Plan

How to Provision (Control Hub)

1. Setting up Cisco PSTN

Managing PSTN for a Location

The screenshot displays the Cisco Webex Control Hub interface. The top navigation bar includes the Cisco Webex Control Hub logo, a 'Select Customer' dropdown menu, and notification icons. The left sidebar contains navigation options categorized into Overview, MONITORING (Analytics, Troubleshooting), MANAGEMENT (Users, Workspaces, Devices, Apps, Account, Organization Settings), and SERVICES (Messaging, Meeting, Calling, Hybrid, Zeus Trial B). The main content area is titled 'Calling' and features a search bar and a list of locations. The 'Cisco Charlotte' location is selected and highlighted. A modal window for 'Cisco Charlotte' is open, showing the 'PSTN Connection' section. In this section, the 'Main Number' is set to 'Not Selected', and a warning message states: 'You will not be able to make or receive calls until this number is added'. A large blue arrow points from the 'PSTN Connection' section to the text 'Unassigned: Manage'.

Calling

Search

Location Routing Prefix

- Main Location
- Cisco Charlotte**
- Leawood Office
- Secondary office
- Shawnee Office
- Third Office

Cisco Charlotte

Overview

PSTN Connection

Numbers

Main Number

Not Selected

You will not be able to make or receive calls until this number is added

Dialing

- Internal Dialing
- External Dialing

Call Settings

- Scheduling
- Voice Portal
- Advanced Call Settings

Unassigned: Manage

Adding Cisco PSTN for a Location

Connection Type

Choose the connection type for all phone numbers associated with Cisco Charlotte

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.



Cloud Connected PSTN

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



Registering Contract Info

Contract Information Emergency Services Address Done

Contract Information

Provide information of the person who will sign the legal contract with Cisco.

Company Name

First Name

Last Name

Email Address

Confirm Email Address

Registration for Cisco PSTN Complete



Contract Information



Emergency Services Address



Done

✔ PSTN Connection Saved

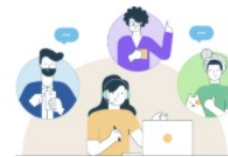
Your Information

Contract Information

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

Emergency Services Address

1900 South Blvd
Charlotte, NC 28203
United States of America



Recommended

Next, add numbers to your trial

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

How many?

1



Get Numbers

[I prefer to search and select my own numbers](#)

2. Ordering Telephone Numbers (TNs)

Adding TNs

Select a Location Select Numbers Done

Location **Connection Type**

Cisco Charlotte ▾ Cisco PSTN

Order New Numbers

Add an order for new numbers directly from Cisco.

Select

Port Numbers Over Available with paid subscription

Transfer numbers from your current carrier to Cisco.

Select

Searching for New TNs

Add Numbers (Cisco Charlotte)

Select a Location

Select Numbers

Done

Specify the numbers you want to order

What area should these numbers be from?

We'll find you numbers in the area code or city of your choice.

Country

Select

State

North Carolina

Search by

Area Code

Prefix ⓘ

Area Code

980

Any

How many numbers do you want auto-selected for you?

We can choose up to 10 non-consecutive numbers for you. You will be able to see and change the numbers before submitting the order.

5

Search

Selecting New TNs

Add Numbers (Cisco Charlotte)



Specify the numbers you want to order

85 numbers found in **North Carolina** with area code **980** and prefix **Any**.

[Search again?](#) Selected numbers will be saved in your cart.

- (980) 550-1547
- (980) 842-0037
- (980) 842-0059
- (980) 842-0249
- (980) 842-0266
- (980) 842-0364
- (980) 907-8179
- (980) 907-8286
- (980) 907-8341
- (980) 485-0063
- (980) 494-6525
- (980) 483-0319
- (980) 705-5456
- (980) 705-5533
- (980) 705-5564
- (980) 745-0333
- (980) 745-0581
- (980) 745-0615
- (980) 274-2040
- (980) 274-2187
- (980) 274-2321
- (980) 458-0245
- (980) 458-0569
- (980) 737-8075
- (980) 737-8138
- (980) 737-8220
- (980) 737-8281
- (980) 737-8343
- (980) 737-8358
- (980) 737-8398
- (980) 892-0257
- (980) 892-0265

Selected Numbers

[Clear All](#)

- (980) 550-1547 X
- (980) 842-0037 X
- (980) 842-0059 X
- (980) 842-0249 X
- (980) 842-0266 X

Total: 9/10

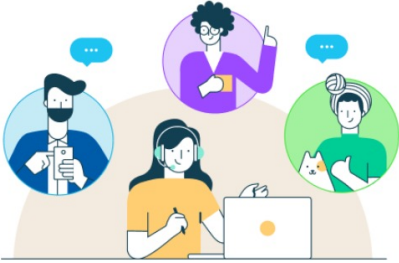
[Back](#)

[Order](#)

Submitting a TN Order

Add Numbers ✕

Select a Location Select Numbers Done



Order Submitted
You can check your order status in [PSTN Orders](#)

Phone Numbers (5)

- (980) 550-1547
- (980) 842-0037
- (980) 842-0059
- (980) 842-0249
- (980) 842-0266

Close View orders

Porting Existing TNs

Enter numbers you want to port

To port numbers from one or more carriers, enter up to 1000 numbers. Include area codes but no country codes, plus signs, or leading zeros. Dashes and parentheses are acceptable, e.g. 4507832223, (450) 783-2223, or 450-783-2223.

(240) 307-8998 × (240) 307-8999 × (240) 307-8990 × (240) 307-8991 × (240) 307-1921 ×

(240) 439-1212 × (240) 352-4623 × (240) 439-1328 × (240) 412-4511 × (240) 251-1124 ×

(240) 532-3811 × (240) 212-4212 × (240) 567-2124 × (240) 463-1246 × (240) 211-5124 ×

(240) 642-2134 × (240) 135-3281 × (240) 335-8832 × (240) 212-2163 × (240) 641-2451 ×


(240) 353-2622 × (240) 362-6423 × (240) 374-2475 × (240) 232-6433 × (240) 251-8631 ×

(240) 333-8634 × (240) 335-8832 × (240) 352-4623 × (240) 362-3678 × (240) 361-3613 ×

Enter phone numbers seperated by commas

186/1000 phone numbers ⊗ 6 numbers are not portable. Clear non-portable numbers before proceeding.

 Clear all

 Clear errors

3. Assigning Outbound Calling Plans (OCPs)

Assigning an OCP to a User

Users

Search: All 2 Administrators 1 External Administrators 1

First Name	Last Name	Display Name	Email
badmaev+zeusb	badmaev+zeusb	badmaev+zeusb@gmail.com	badmaev+zeusb@gmail.com
Boris	Badmaev	Boris Badmaev	badmaev+zeusc@gmail.com

User Profile: Boris Badmaev
badmaev+zeusc@gmail.com

Services [Edit](#)

- Messaging: Cisco Webex Teams Free Messaging
- Meeting: Cisco Webex Teams Free Meetings
- Calling: Webex Calling Enterprise >


Hybrid Services

- Calendar Service: Off >
- Message Service: Off >

Roles and Security

- Administrator Roles >
- Security >

Devices [Add Webex Rooms Device](#)





Assigning an OCP to a User

Users

Search: All 2 Administrators 1 External Administrators 1

First Name	Last Name	Display Name	Email
badmaev+zeusb	badmaev+zeusb	badmaev+zeusb@gmail.com	badmaev+zeusb@gmail.com
Boris	Badmaev	Boris Badmaev	badmaev+zeusc@gmail.com


Boris Badmaev  

badmaev+zeusc@gmail.com

User > Calling > Advanced
> Outgoing and Incoming Permissions > Outgoing Calls

Cisco Calling Plan

This user is assigned to a Cisco PSTN location with Unlimited Outbound Calling Plan. Enable this user to utilize a plan and allow them to make outbound calls.



Outgoing Call Settings Override

Turn on Outgoing Call Settings for this user to override the calling settings from Location Main Location that are used by default.

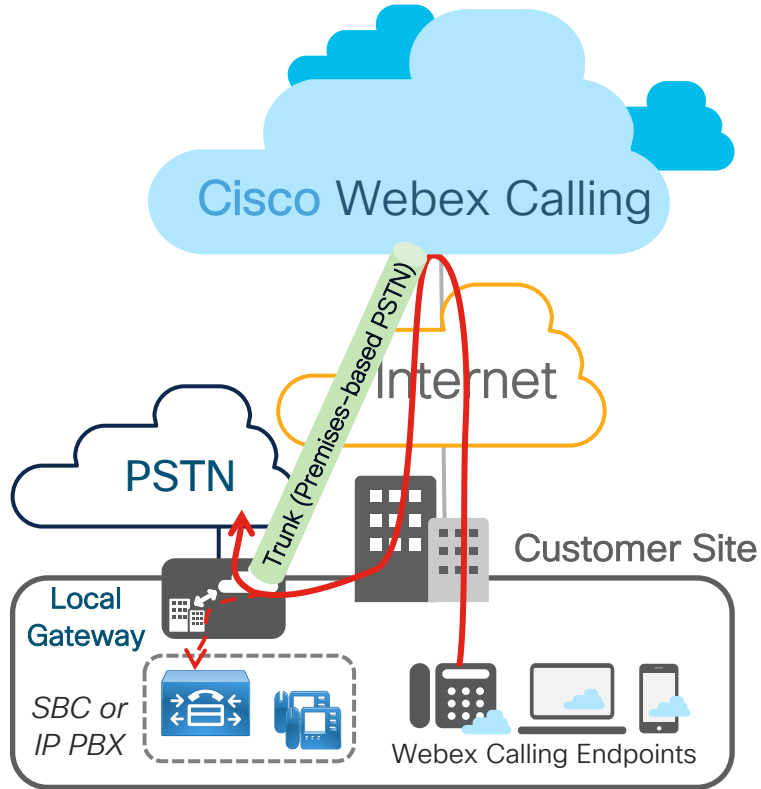
Call Types

Internal	Allow
Local	Allow
Toll Free	Allow
Long Distance	Allow
International	Block

Local Gateway Updates

Webex Calling Trunk - Local Gateway

(Premises-based PSTN) Deployment

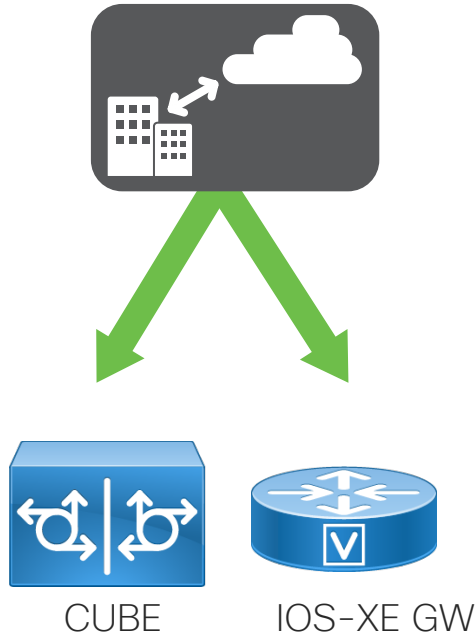


- Provides connectivity to a customer-owned premises-based PSTN service
- May also provide connectivity to an on-premises IP PBX or dedicated SBC/PSTN GW
- Enables on-prem to Webex Calling transition
- **Endpoint registration is NOT proxied through Local Gateway, unlike CUBE Lineside.** Endpoints directly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

Local Gateway

Platform Support

Local Gateway (LGW)

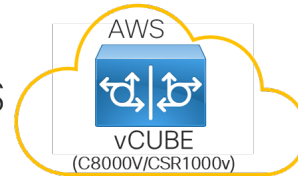


- **Cisco CUBE** (for IP-based connectivity) or Cisco IOS Gateway (for TDM-based connectivity)

- Hardware and software requirements:

- ISR 4321, 4331, 4351, 4431, 4451, 4461 (IOS XE 17.3.3)

- vCUBE in AWS



- Catalyst 8200/8300 series



- CSR 1000v (vCUBE) (16.12.5 or later) –

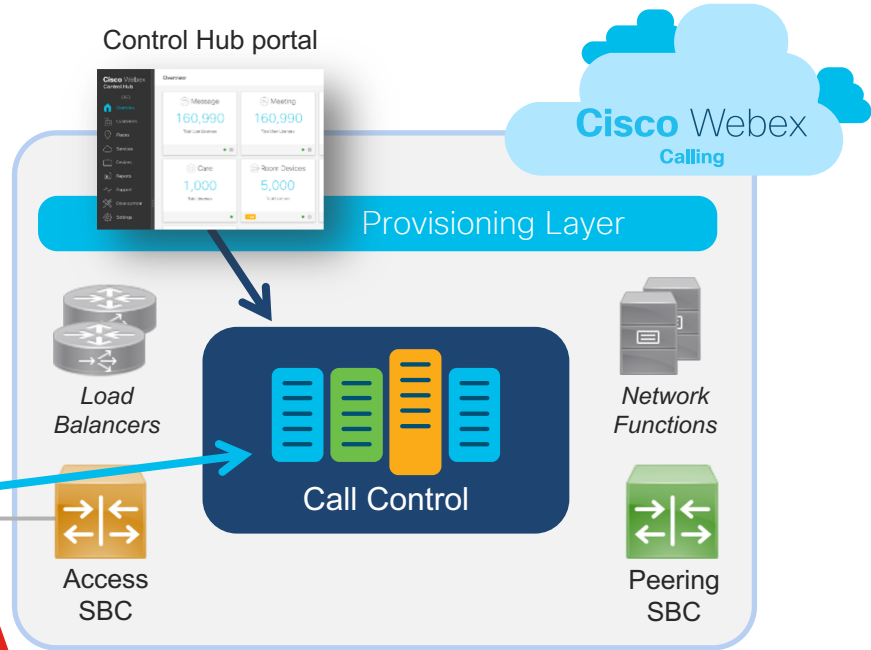
- CSR 1000v licenses are not included in Webex Calling Flex and need to be purchased separately
- Estimate 200 kbps total data throughput for every audio call

- ISR 1100 (IOS-XE 16.12.5 or later)

Registering Trunk regardless of SBC

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

Local GW registers over SIP TLS using conn. parameters from Control Hub



Single TLS connection for all signaling between CUBE and cloud

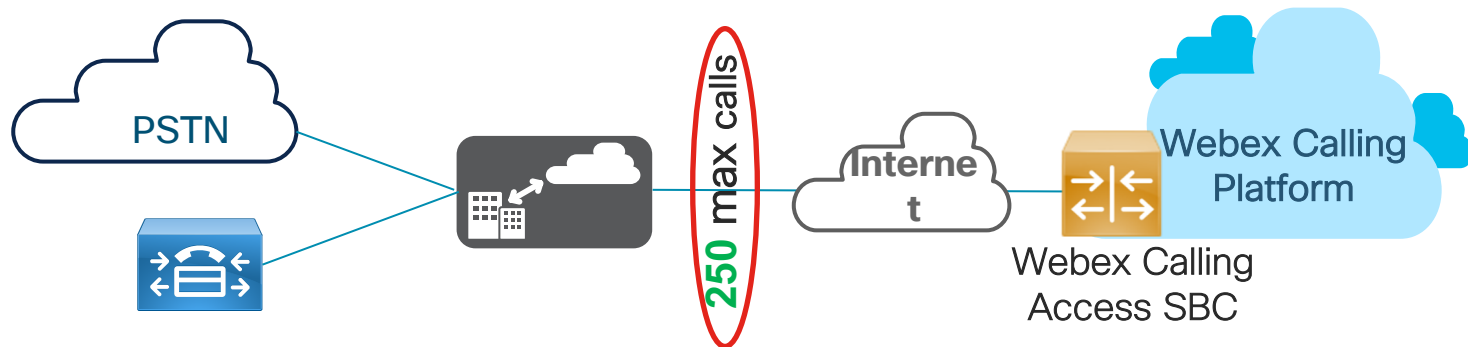
- Limited scale due to a single TCP connection
- Sensitive to network impairments (TCP throughput \propto latency/loss)

Update IP Trust based on latest subnets

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


Webex Calling Trunk – Local gateway Concurrent Call Limits

- Regardless of LGW platform, premises trunks between LGW and Webex Calling cannot exceed **250** concurrent calls when connected over the Internet (OTT).
 - This assumes a maximum of 100ms one-way latency with no more than 10ms jitter, less than 0.5% packet loss
 - Poor network conditions between Local Gateway and Webex Calling access SBC can limit the performance of the signaling connection leading to an even lower concurrent calls limit.
- Multiple LGWs with Trunk and Route groups can be deployed for higher scale:
 - **Premises → cloud calls**: load balancing supported today (e.g., CUCM route groups)
 - **Cloud → premises calls**: Webex Calling Trunk and Route Groups



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

Key Local Gateway Configuration Updates Required

A red, irregular thought bubble shape with a small tail pointing downwards and to the left. The word "Update" is written in white text inside the bubble.

Update

Onboarding Process
Webex Calling Trunk

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

Overview

MONITORING

Analytics

Troubleshooting

MANAGEMENT

Users

Workspaces

Devices

Apps

Account

Organization Settings

SERVICES

Messaging

Calling

Hybrid

Overview

Webex Services ALL

ONLINE



Messenger



Teams



Calling



Meetings



Hybrid Services



Control Hub



Developer API



Room Devices



Contact Center



UCM Cloud

Hybrid Services 7

INCOMPLETE



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

The screenshot shows a web interface for configuring call routing. At the top, there is a navigation bar with the following items: Calling, Numbers, Locations, Call Routing (highlighted with a blue box), Features, PSTN Orders, Orders, Service Settings, and Client Settings. Below this, a sub-navigation bar includes Trunk (highlighted with a blue box), Route Group, Dial Plans, and Verify Call Routing. The main content area is titled 'Trunk' and contains a descriptive paragraph: 'SIP trunks provide connectivity to a customer-owned PSTN service and to an on-premises IP PBX deployment. These were previously accessed via the Local Gateway configuration page.' To the right of this text is a green 'Add Trunk' button (highlighted with a blue box). Below the text is a search bar with a magnifying glass icon and the word 'Search'. At the bottom, there is a table with three columns: Name, Location, and In Use. The table contains two rows of data.

Name	Location	In Use
Atlanta	Atlanta	Yes
Dallas LGW	Dallas	No

1c. Add a new Trunk for the desired Location

Add Trunk

Location

This location is where the trunk is physically connected. To create a new location, visit the [Locations](#) page.

Name

Device Type

- 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

The screenshot shows the Cisco Webex Control Hub interface. The top navigation bar includes the Cisco Webex Control Hub logo, a notification bell with a red '3', a help icon, a chat icon, and a user profile icon labeled 'HA'. The left sidebar contains navigation options: Overview, MONITORING (Analytics, Troubleshooting, Organization Health), and MANAGEMENT (Users, Workspaces, Devices, Apps, Account). The main content area is titled 'Calling' and features a breadcrumb trail: Numbers, Locations (highlighted with a red box), Call Routing, Features, PSTN Orders, Orders, Service Settings, and Client Settings. Below the breadcrumb is a search bar with a magnifying glass icon and the text 'Search'. To the right of the search bar is a green 'Add Location' button. Below these elements is a table with three columns: Location, Routing Prefix, and Actions. The table contains three rows: 'TME Validate Lab' with routing prefix '8121', 'Atlanta' (highlighted with a blue box) with routing prefix '8167', and 'Dallas' with routing prefix '8197'. Each row has a three-dot menu icon in the Actions column.

Cisco Webex Control Hub

Calling

Numbers **Locations** Call Routing Features PSTN Orders Orders Service Settings Client Settings

Search

Add Location

Location	Routing Prefix	Actions
★ TME Validate Lab	8121	...
Atlanta	8167	...
Dallas	8197	...

1g. Click on **Unassigned** under PSTN Connection

The screenshot displays the Cisco Webex Control Hub interface. The top navigation bar includes the Cisco Webex Control Hub logo, a notification bell with a red '5', a help icon, and a user profile picture. The left sidebar contains navigation options: Overview, MONITORING (Analytics, Troubleshooting), and MANAGEMENT (Users, Workspaces, Devices, Apps). The main content area is titled 'Calling' and shows the 'Numbers' section for the 'Atlanta' location. A search bar is present. Under the 'Location' dropdown, 'dCloud' and 'Atlanta' are listed. The 'PSTN Connection' is highlighted with a blue box, and the status 'Unassigned: Manage' is displayed in red text next to it. Below this, the 'Main Number' is shown as '7707860000' with a dropdown arrow.

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.

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

Add Trunk



Hussain Successfully Created.

Visit [Route Group](#) page to add trunk(s) to a route group
Visit [Locations](#) page to configure PSTN connection to individual
Visit [Dial Plans](#) page to use this trunk as the routing choice for a

Trunk Info

Status

● unknown

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.

Authentication Information

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

Username: Hussain2572_LGU

Password: meX7]~)VmF

```
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:"
```

Add Trunk



Hussain Successfully Created.

Visit [Route Group](#) page to add trunk(s) to a route group

Visit [Locations](#) page to configure PSTN connection to individual

Visit [Dial Plans](#) page to use this trunk as the routing choice for a

Trunk Info

Status

● unknown

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

Authentication Information
Record the username and password for this trunk. If you lose this information, you need to use the [Reset Password](#) link to reset the password.

Username: Hussain2572_LGU

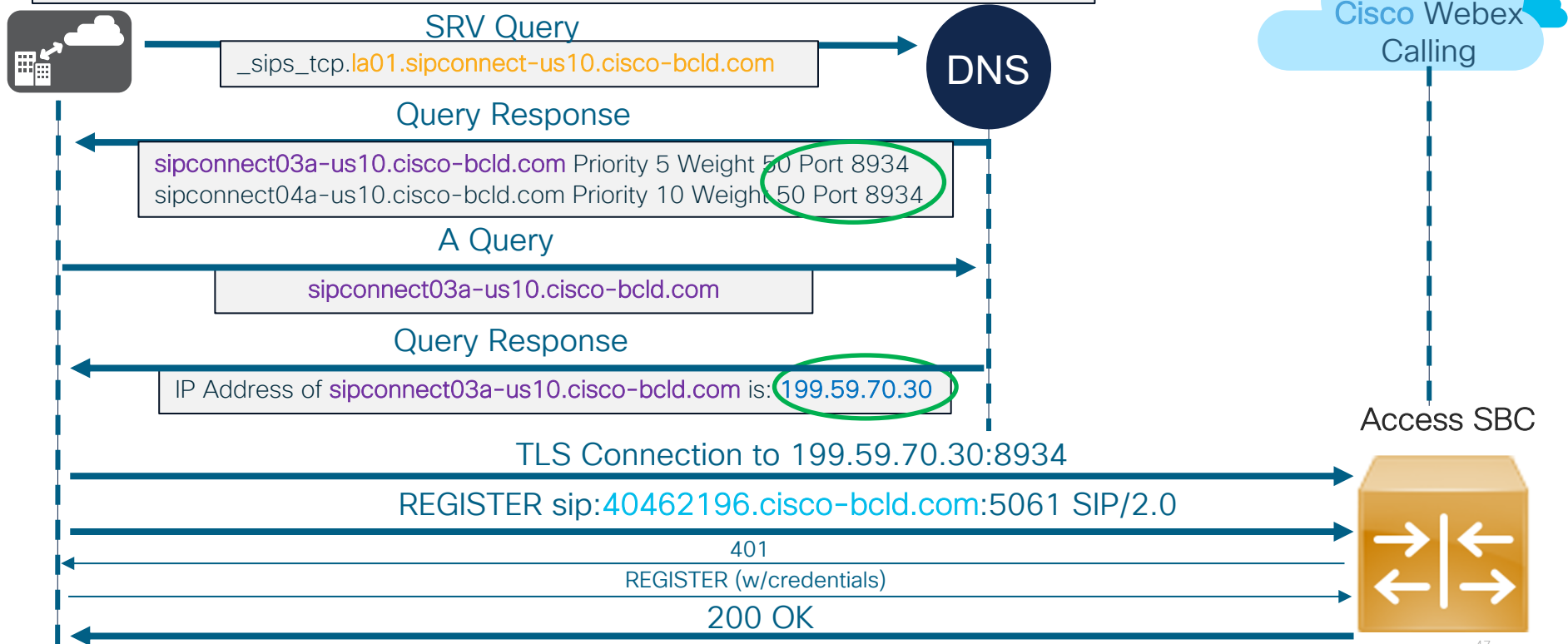
Password: meX7]-)VmF

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

A red, irregular thought bubble shape with a small tail pointing downwards and to the left. The word "Update" is written inside in white text.

Update

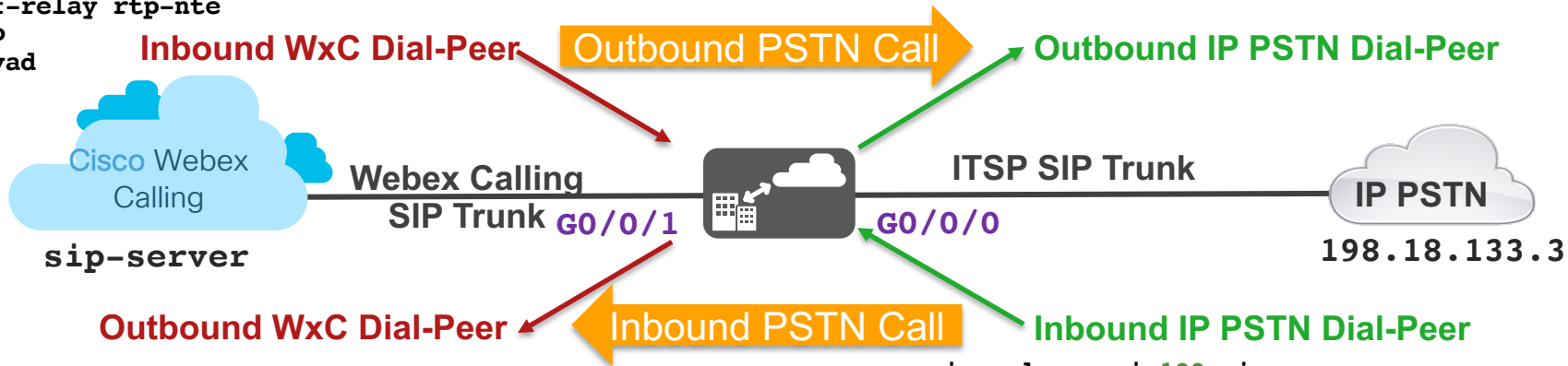
Single Dial-peer facing Webex
Calling for Inbound/Outbound Calls

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
```

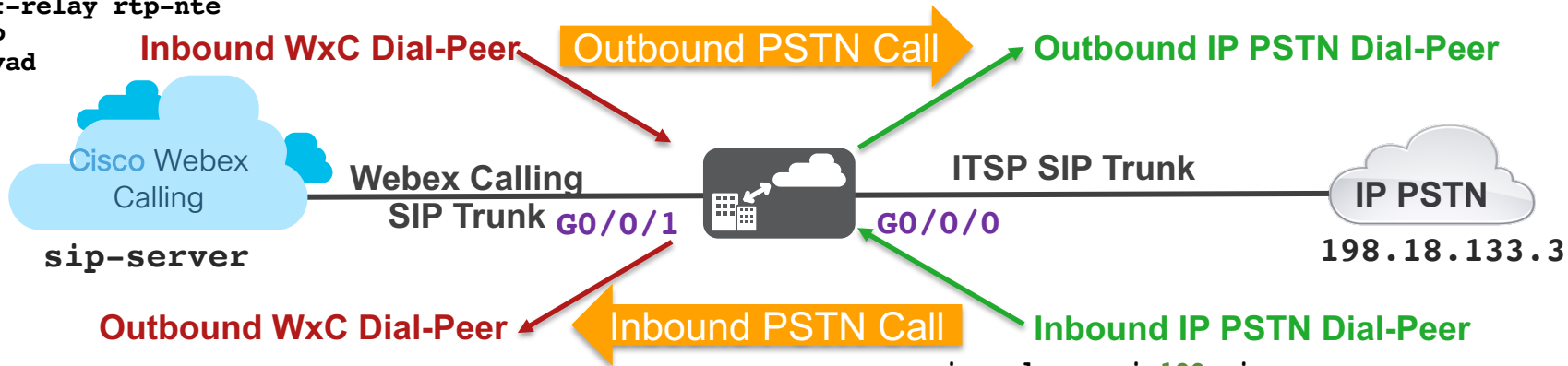
```
voice class uri 100 sip  
host ipv4:198.18.133.3
```

```
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
```

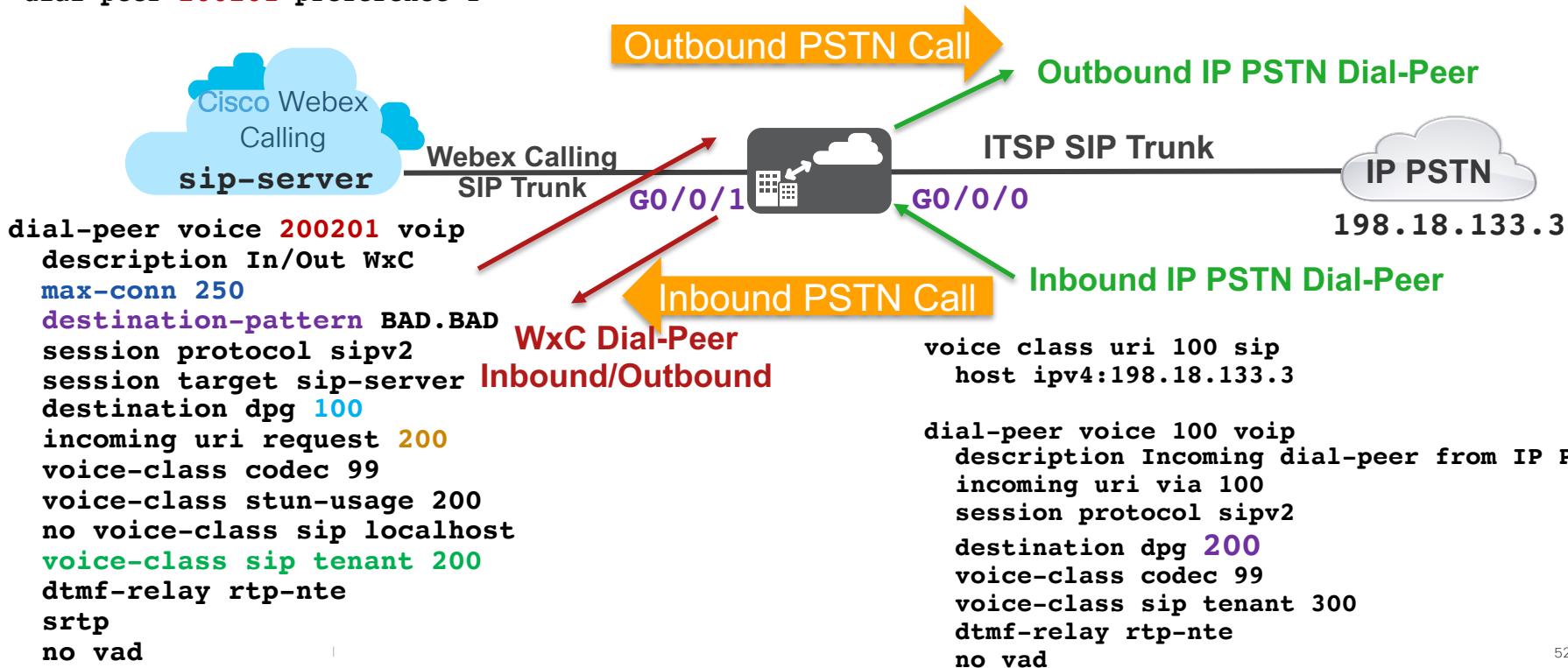
```
voice class uri 200 sip
  pattern dtg=hussain3847.lgu
```

```
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
```

New Dial-peer structure

```
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
```



Single vCUBE instance with two LGWs – Total 500 calls

Trunk1 - LGW1=250 calls

Trunk 2 - LGW2=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
```

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

voice class tenant 300

```
bind control source-interface GigabitEthernet0/0/0
bind media source-interface GigabitEthernet0/0/0
```

Caller ID handling

Caller ID settings

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

The screenshot displays the Cisco Webex Control Hub interface. On the left is a navigation sidebar with sections for Overview, MONITORING (Analytics, Troubleshooting), and MANAGEMENT (Users, Workspaces, Devices, Apps, Account, Organization Settings). The main content area is titled 'Users' and contains a search bar with 'All 8' results, a filter for 'Administrators 1', and another filter for 'External Administrators 1'. Below this is a table of users with columns for First Name, Last Name, and Display Name. The user 'Charles Holland' is highlighted in blue. To the right of the table is a user profile card for Charles Holland (cholland@cb378.dc-01.com) with an edit icon. Below the profile card is a 'Services' section with an 'Edit' link, listing 'Messaging' (Cisco Webex Teams), 'Meeting' (Cisco Webex Team Meetings), and 'Calling' (Webex Calling Enterprise). At the bottom, 'Hybrid Services' are listed: 'Calendar Service' (Off) and 'Message Service' (Off).

First Name	Last Name	Display Name
Anita	Perez	Anita Perez
Charles	Holland	Charles Holland
Eric	Steele	Eric Steele
Kellie	Melby	Kellie Melby
Rebekah	Barretta	Rebekah Barretta
Ricardo	Filice	Ricardo Filice

Services

- Messaging: Cisco Webex Teams
- Meeting: Cisco Webex Team Meetings
- Calling: Webex Calling Enterprise >

Hybrid Services

- Calendar Service: Off >
- Message Service: Off >

Caller ID settings

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

Caller ID settings

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

 Charles Holland 
cholland@cb378.dc-01.com 

User > Calling

User Settings [Edit](#) 

Directory Numbers [Add Number](#)

6021 or 9194745971 Primary >

Call Settings

Voicemail On >

Call Forwarding Off >

Call Waiting On >

Caller ID Direct Line >

Advanced Call Settings >

 Charles Holland 
cholland@cb378.dc-01.com 

User > Calling > Caller ID

Caller ID

Choose which information will be displayed when this User makes an outgoing call.

Caller ID Phone Number

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

Caller ID First Name

Charles 

Caller ID Last Name

Holland 

Caller ID and Local Gateway

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

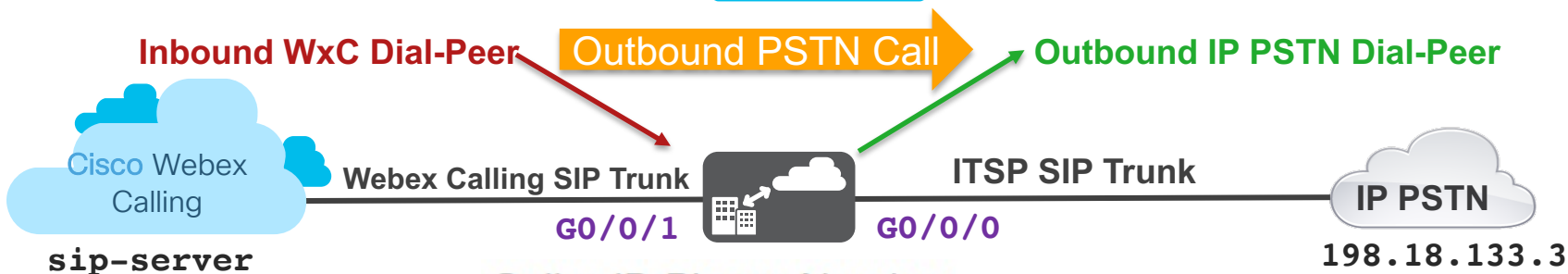
```
voice class tenant 200
  no remote-party-id
  asserted-id pai
  privacy-policy passthru
```

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

Outbound LGW call – Location Number (Main Number)

Received:

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



Caller ID Phone Number

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

Sent:

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

Troubleshooting and Templates

Common LGW/CUBE Commands and Debugs

Command/Debug

show sip-ua register status

show call active voice brief/compact

show run

show log

clear log

debug ccsip non-call

debug ccsip message

debug ccsip error

debug ccsip transport

debug voip ccapi inout

Explanation

display the registration status of a tenant

display parameters of active calls

View running CUBE configuration

View CUBE debug logs, logged to a buffer

Clear CUBE debug logs

Enable non-call context trace (REGISTER, OPTIONS), origination from LGW/CUBE

Enable SIP messaging traces

Enable SIP error debugging trace

Enable SIP Transport layer debugging traces

Enable end to end IOS-XE VoIP call control (dial-peers) debugs


```

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

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

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

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

!%%%%%%%% BEGIN

configure terminal

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

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

configure terminal
crypto pki trustpool import clean url http://www.cisco.com/security/pki/trs/ios_core.n7b

```

```
configure terminal
voice service voip
  ip address trusted list
  !! Verify updated trust list from the Webex Calling Config Guide!!
  ipv4 85.119.56.128 255.255.255.192
  ipv4 85.119.57.128 255.255.255.192
  ipv4 185.115.196.0 255.255.255.128
  ipv4 185.115.197.0 255.255.255.128
  ipv4 128.177.14.0 255.255.255.128
  ipv4 128.177.36.0 255.255.255.192
  ipv4 135.84.169.0 255.255.255.128
  ipv4 135.84.170.0 255.255.255.128
  ipv4 135.84.171.0 255.255.255.128
  ipv4 135.84.172.0 255.255.255.192
  ipv4 199.59.64.0 255.255.255.128
  ipv4 199.59.65.0 255.255.255.128
  ipv4 199.59.66.0 255.255.255.128
  ipv4 199.59.67.0 255.255.255.128
  ipv4 199.59.70.0 255.255.255.128
  ipv4 199.59.71.0 255.255.255.128
  ipv4 135.84.172.0 255.255.255.128
  ipv4 135.84.173.0 255.255.255.128
  ipv4 135.84.174.0 255.255.255.128
  ipv4 199.19.197.0 255.255.255.0
  ipv4 199.19.199.0 255.255.255.0
  ipv4 139.177.64.0 255.255.255.0
  ipv4 139.177.65.0 255.255.255.0
  ipv4 139.177.66.0 255.255.255.0
  ipv4 139.177.67.0 255.255.255.0
  ipv4 139.177.68.0 255.255.255.0
  ipv4 139.177.69.0 255.255.255.0
  ipv4 139.177.70.0 255.255.255.0
  ipv4 139.177.71.0 255.255.255.0
  ipv4 139.177.72.0 255.255.255.0
  ipv4 139.177.73.0 255.255.255.0
  exit
  allow-connections sip to sip
  media statistics
  media bulk-stats
  no supplementary-service sip refer
  no supplementary-service sip handle-replaces
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  stun
  stun flowdata agent-id 1 boot-count 4
```



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

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

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

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

voice class srtp-crypto 200
  crypto 1 AES_CM_128_HMAC_SHA1_80
  exit

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



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

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

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

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

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

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

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

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

```
dial-peer voice 200201 voip
 destination dpg 100
```

end

copy run start

!!%***% END-TEMPLATE

Resources

Resources

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

