



# Cisco Webex Provider SIP Trunk Configuration Guide

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# 1 Introduction

The intended purpose of this document is to provide a general guideline to configure your Cisco Webex Provider SIP Trunk to Portal. This document does not cover advanced SIP configurations that could be used. To access the Portal, follow the link below:

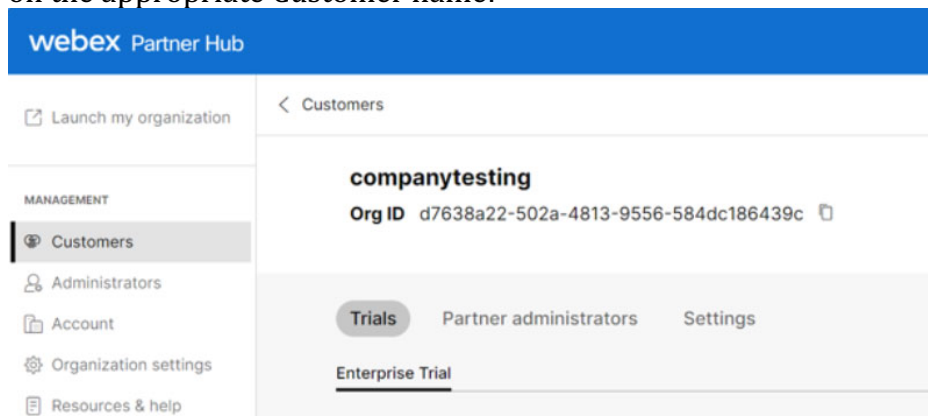
<https://portal.infobip.com/apps/voice-and-webrtc/sip-trunks>

## 2 Requirements & Information

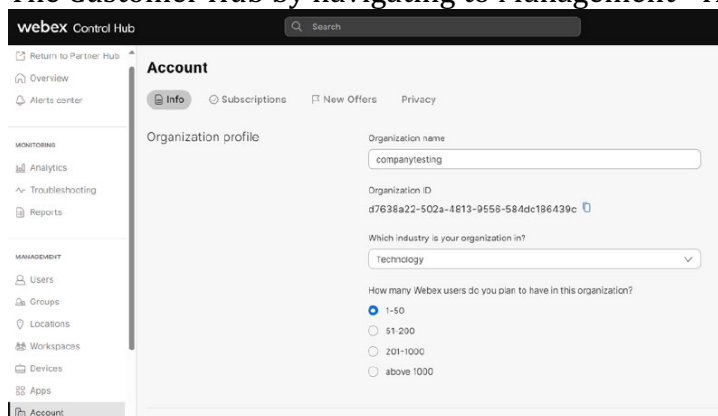
### 2.1 CISCO WEBEX BYOC

- Your Customer Organizational ID (UUID) is unique for each Customer. This UUID is found in one of two places:

- The Partner Hub by navigating to Management->Customers and clicking on the appropriate Customer name:



- The Customer Hub by navigating to Management->Account:



- Incoming calls to BYOC are authenticated by IP Access Control List
- BYOC requires E.164 format phone numbers

## 3 Portal Configuration

We will assume that you have created an Infobip Portal account and will purchase at least one DID.

## 3.1 PORTAL LOGIN

Login to the Portal using your login credentials. <https://portal.infobip.com/apps/voice-and-webrtc/sip-trunks>

## 3.2 ADD SIP TRUNK

- Click the "Create SIP Trunk" button and select "Cisco Webex" trunk type:

The screenshot shows the 'Voice and WebRTC' management interface. At the top, there's a navigation bar with 'Channels and Numbers / Voice and WebRTC / SIP Trunking' and a balance of €0.00. The main heading is 'Voice and WebRTC' with a sub-heading 'Set up voice numbers, SIP trunks and Call Routing, Recording, Call Link and WebRTC settings. [Learn more](#)'. Below this, there are tabs for 'Numbers', 'SIP Trunking', 'Call Routing', 'Number Masking', 'Call Link', 'WebRTC', and 'Recording'. The 'SIP trunks' section is active, showing a table of existing trunks and a 'CREATE SIP TRUNK' dropdown menu.

SIP trunk name	Location	Provider	Type	Status	Action status	Active calls
APAC-demo	Singapore	Infobip	Static	Enabled	Success	0
US2-LDIP-NOOPTIONS	Portland	Infobip	Static	Disabled	Success	0
FR-Office	Frankfurt	Infobip	Static	Enabled	Success	0
dummySIP	Frankfurt	Infobip	Static	Enabled	Success	0

The 'CREATE SIP TRUNK' dropdown menu is open, showing options: 'Infobip', 'Freshworks', 'Genesys Cloud', and 'Cisco Webex'. The 'Cisco Webex' option is highlighted.

- Select the appropriate SIP channel plan:

The screenshot shows the 'Choose plan' dialog for creating a SIP trunk. The dialog has a progress bar with four steps: 'Choose plan', 'Set up channels', 'Configure settings', and 'Review order'. The 'Choose plan' step is currently active. There are two plan options: 'Metered' and 'Unlimited', both marked as 'US only'.

**Metered** (US only)  
Pay for your call minutes only.  
[View detailed pricing](#)  
**€7.99** per month / per channel  
Minutes inbound: Free  
Minutes outbound: €0.01  
**CHOOSE PLAN**  
Competitive channel price  
Pay only for what you use  
Fully automated, real-time SIP Channel management

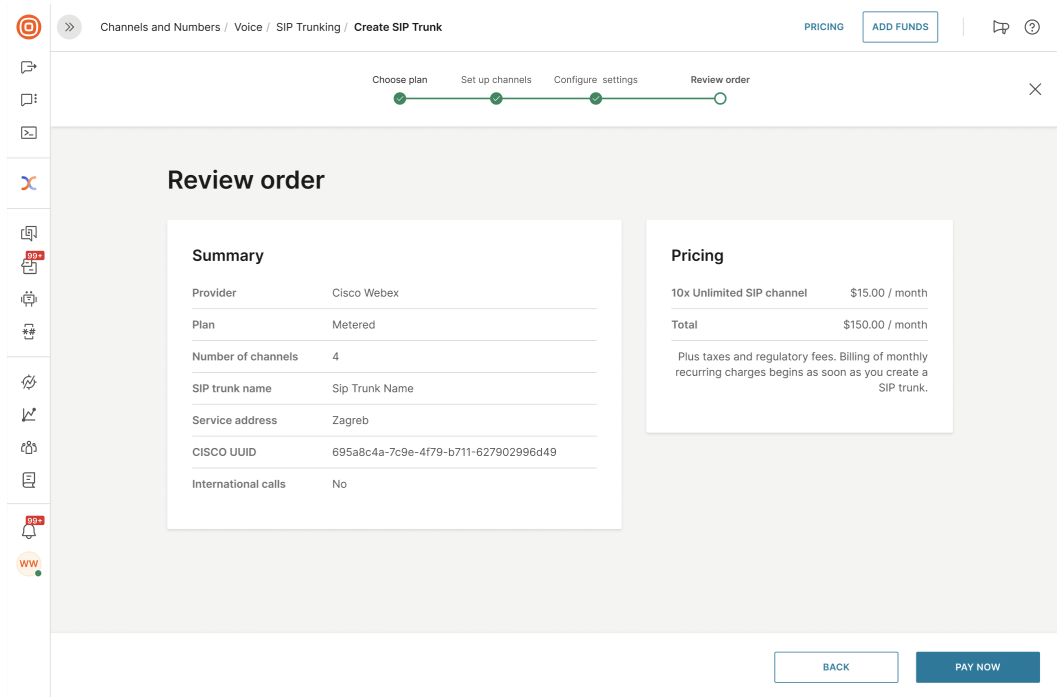
**Unlimited** (US only)  
Unlimited calls within the US.  
[View detailed pricing](#)  
Starting from  
**€15.00** per channel / per month  
Minutes inbound: Free  
Minutes outbound: Free  
**CHOOSE PLAN**  
Predictable billing, every month  
No increased costs as your business grows  
Recommended for small and medium enterprises

**CANCEL**

- Enter the appropriate number of channels for the chosen channel plan:

- Choose the following trunk settings and click the “Next” button
  - A SIP Trunk name
  - Cisco UUID = Your unique Customer Organizational ID (UUID)
  - Service Address = the primary place of use for this trunk. For more information on service addresses and why these should be correctly defined, see <https://www.infobip.com/docs/voice-and-video/sip-trunking#service-address>
  - The Infobip datacenter location (New York or Portland)
  - International Calling = Disabled (Default) or Enabled

- Validate that the trunk name, settings and Charges are the expected values and click the "Pay Now" button.



- Upon returning to the list of SIP trunk, monitor the status of your newly created Cisco Webex trunk until it is in status **Enabled** with an action status **Success**.
- Proceed to the Cisco Webex Admin Portal

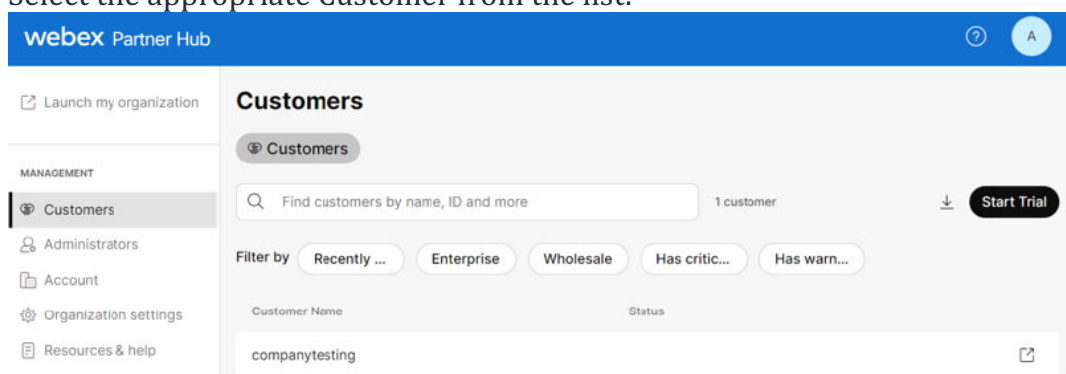
## 4 Cisco Webex BYOC Enterprise Configuration

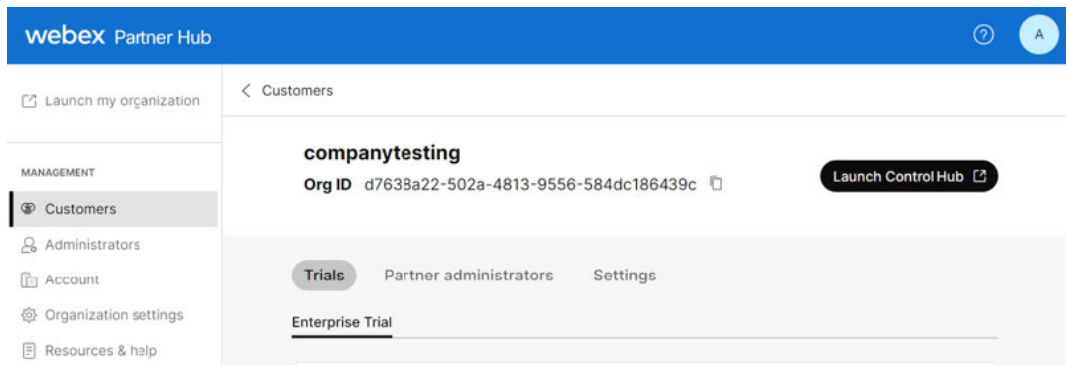
### 4.1 PARTNER HUB

- Login to the Partner Hub using your Admin credentials.
- <https://admin.webex.com/partner/>

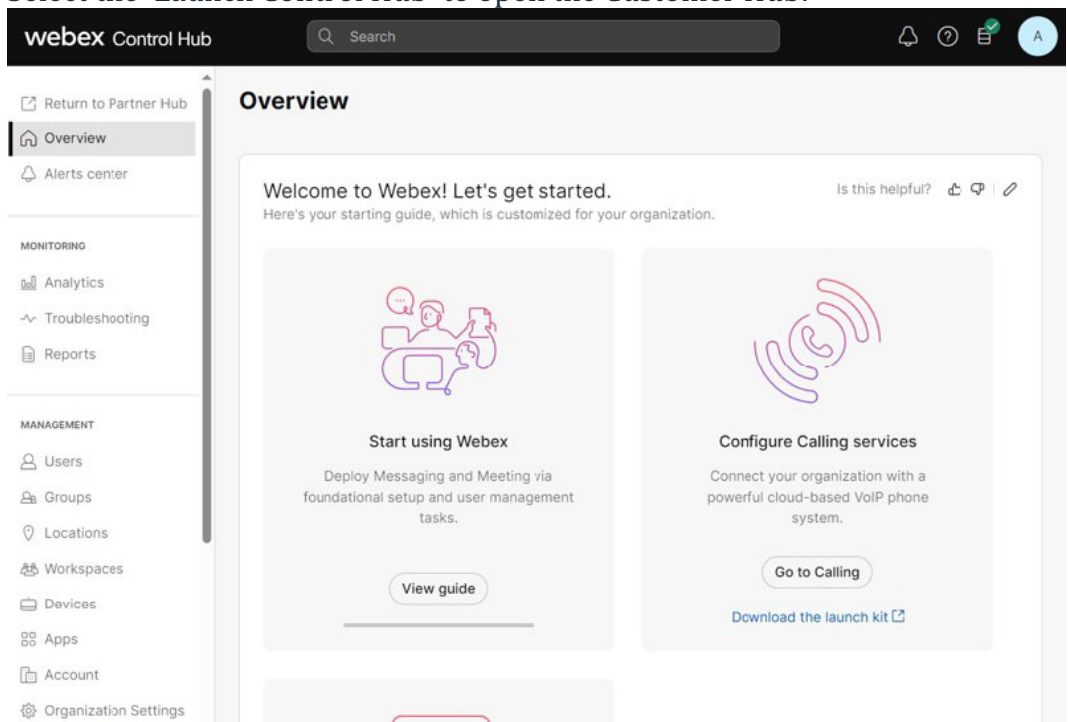
### 4.2 CUSTOMERS

- Navigate to Management->Customers
  - Select the appropriate Customer from the list:



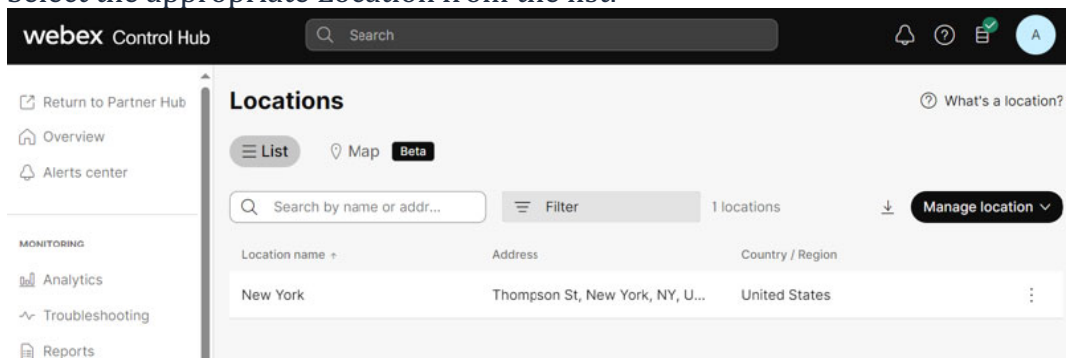


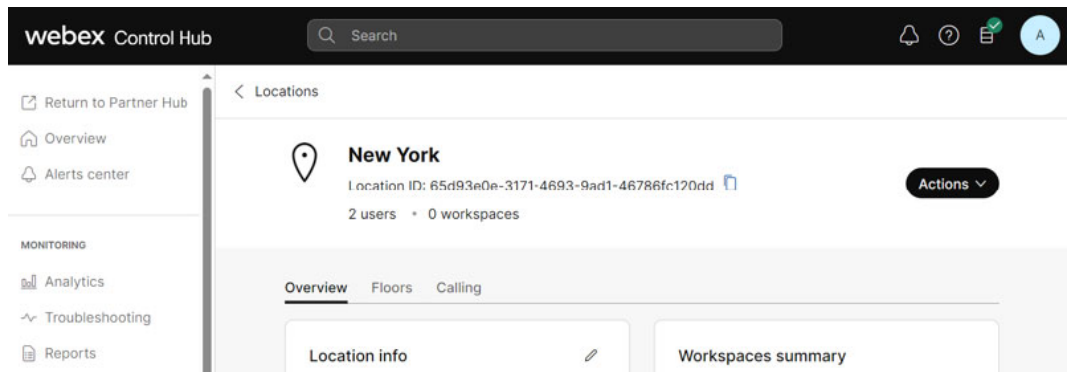
- Select the 'Launch Control Hub' to open the Customer Hub:



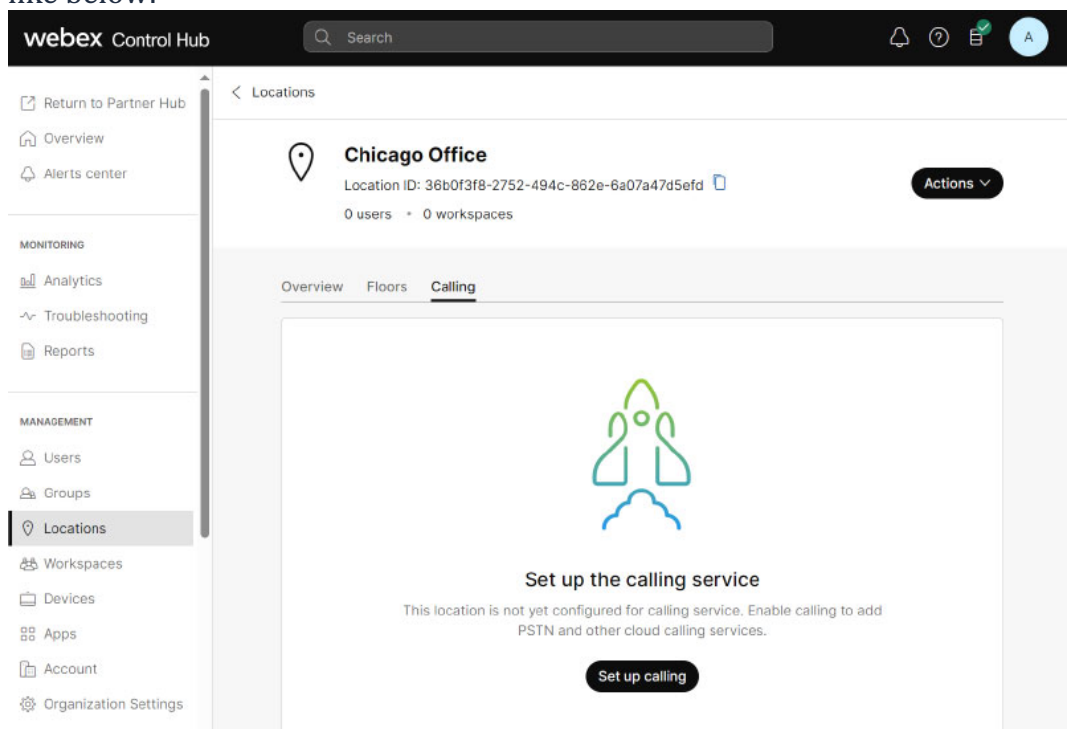
## 4.3 PSTN CONNECTION

- Navigate to Management->Locations
  - Select the appropriate Location from the list:



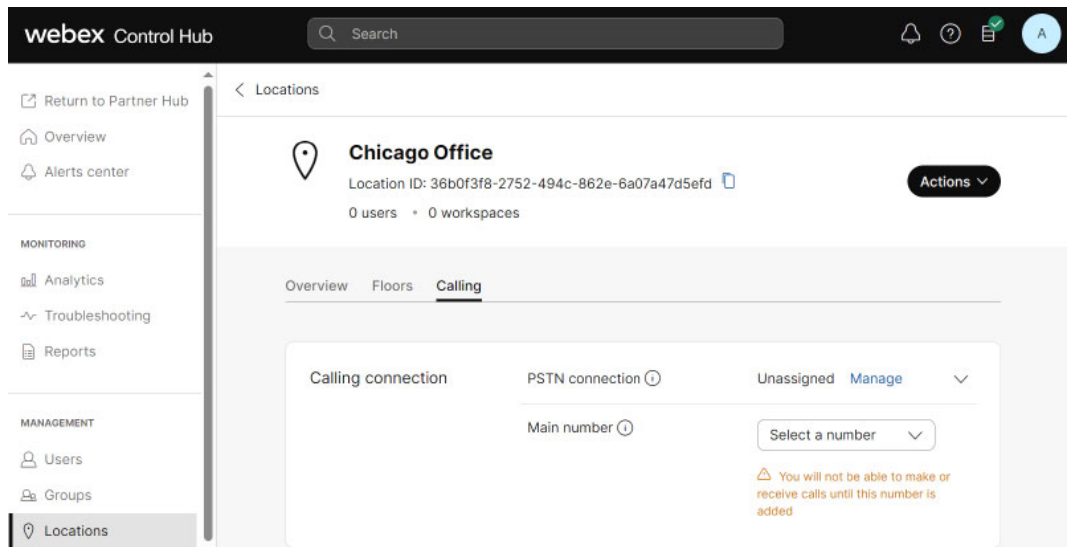


- Select 'Calling' tab from within the Location
- If the Location was newly created, please select the "Set up calling" button like below:

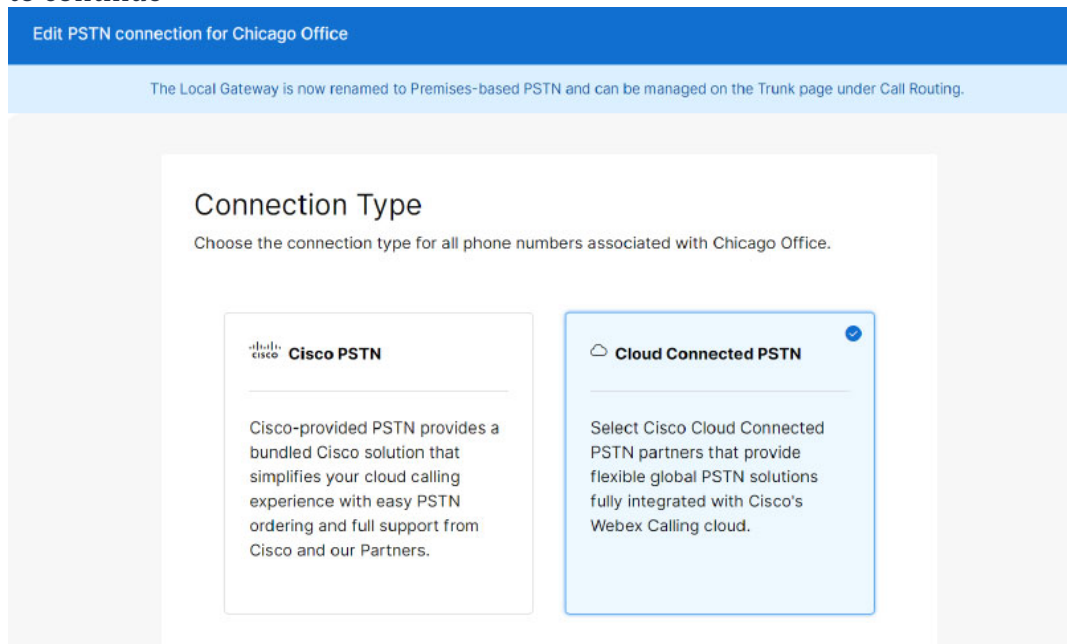


- If this is a previously established Location, please select the 'Manage' for PSTN connection in the Calling connection section like below:

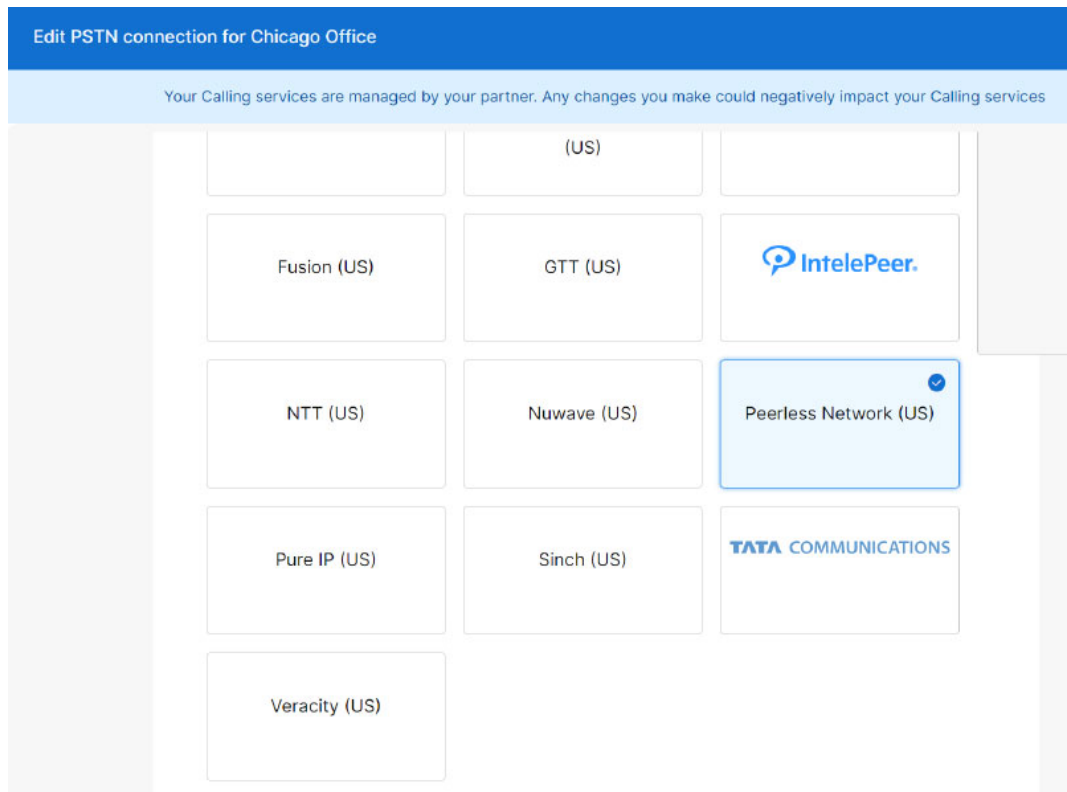




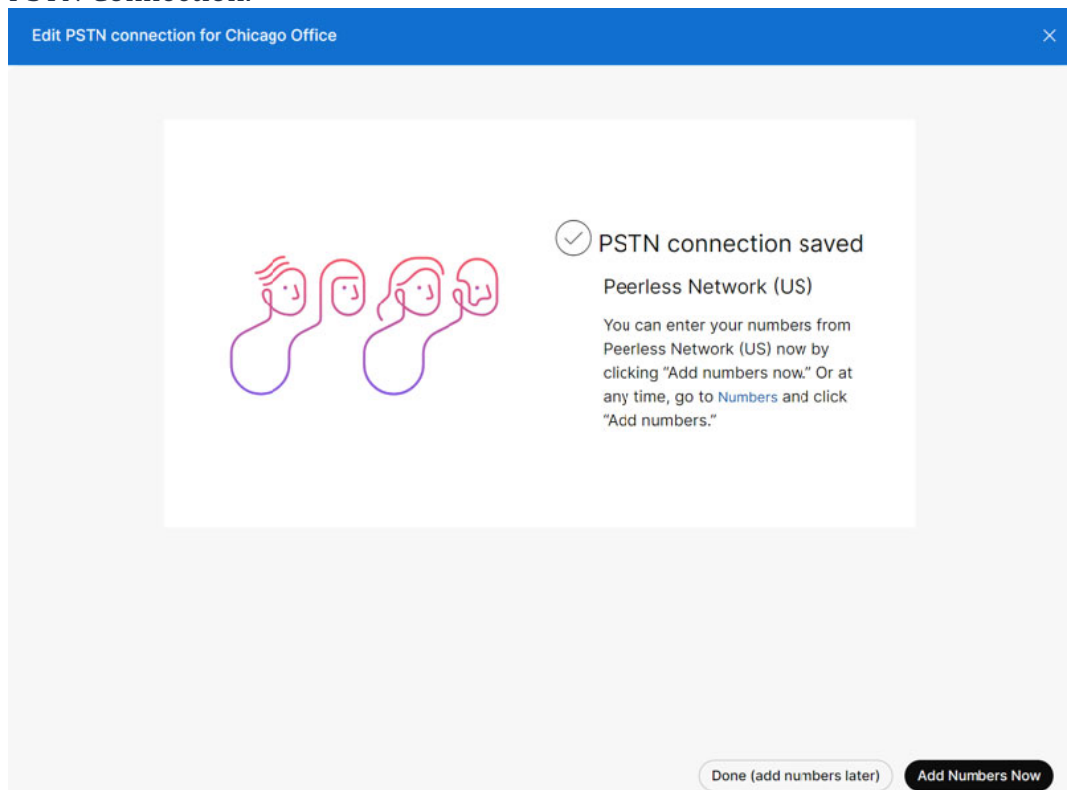
- Select 'Cloud Connected PSTN' as the Connection Type, and select 'Next' to continue



- Select 'Peerless Network (US)' from the Provider list, and select 'Next' to continue (Peerless Networks is an Infobip company)



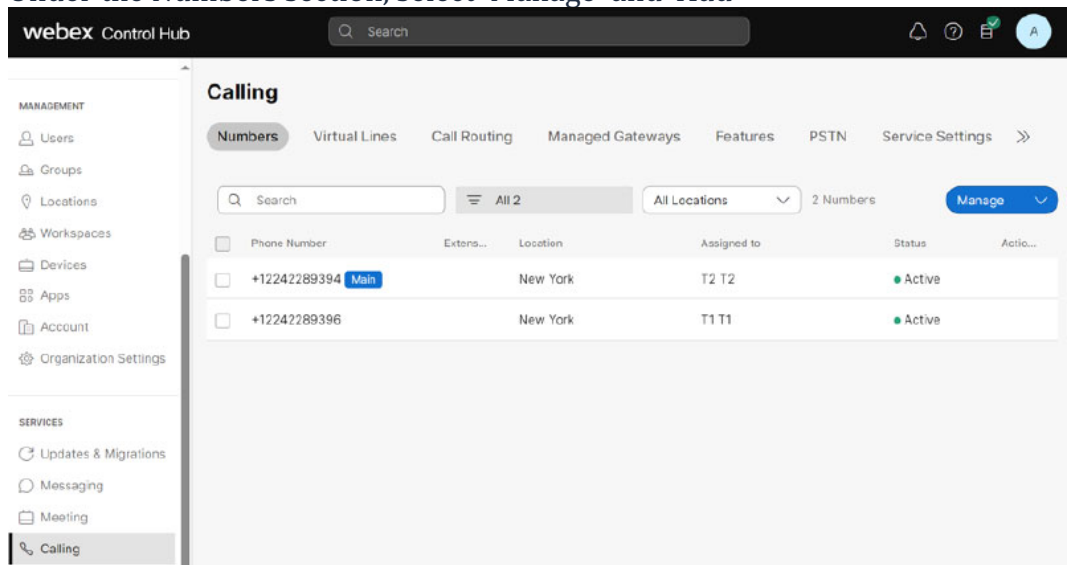
- The following page should be displayed showing Peerless Network as the PSTN Connection.



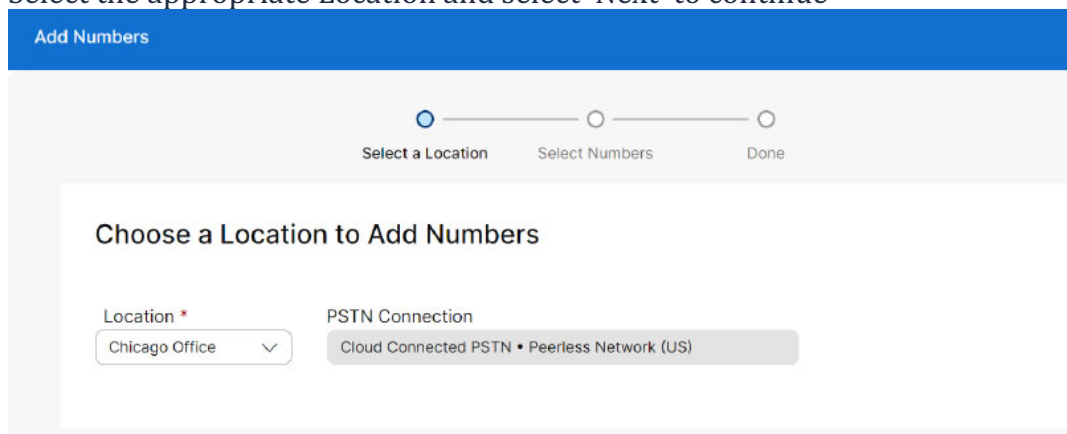
- Select 'Add Numbers Now' to begin number provisioning, or select 'Done' (add numbers later) to add numbers later.

## 4.4 NUMBERS

- Navigate to Services->Calling
  - Under the Numbers section, select 'Manage' and 'Add'



- Select the appropriate Location and select 'Next' to continue



- Enter in all the Numbers/DID's you purchased in Step 3.2 and select 'Save' to continue

**Add Numbers (Chicago Office)**

Progress: **Select a Location** (Completed) | **Select Numbers** (Active) | Done

### Enter numbers you want to add

Input your numbers, with area codes, to add them to this location.  
Country codes, plus signs, dashes, and parentheses are optional.  
Valid examples: 4507832223, (450) 783-2223, 450-783-2223, +1-450-783-2223

Activate Numbers Later ⓘ


Enter phone numbers separated by commas

1/1000 Phone numbers Clear All

- You will see the following screen when the numbers are successfully added. Select 'Close' to continue.

**Add Numbers**

Progress: **Select a Location** (Completed) | **Select Numbers** (Completed) | **Done** (Active)



✔ Successfully saved numbers

Phone Numbers (1)

(312) 867-5309

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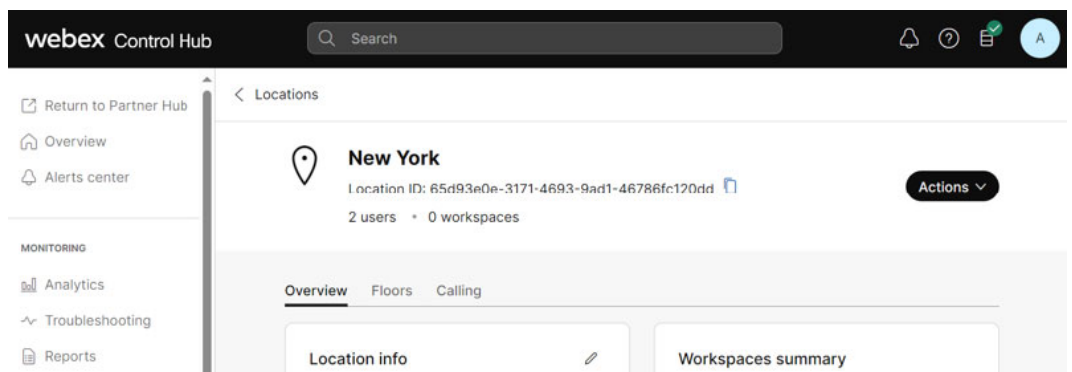
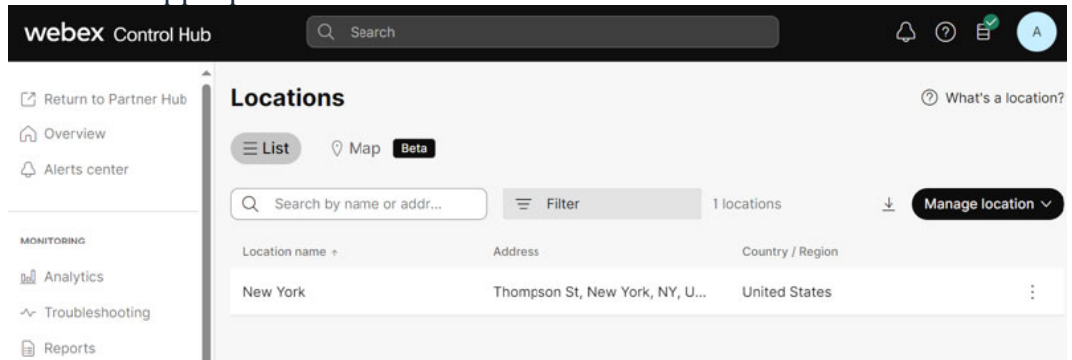
**Do you need to order more new numbers?**  
You'll be able to search for new individual, block, or toll-free numbers to add.

[Order more numbers](#)

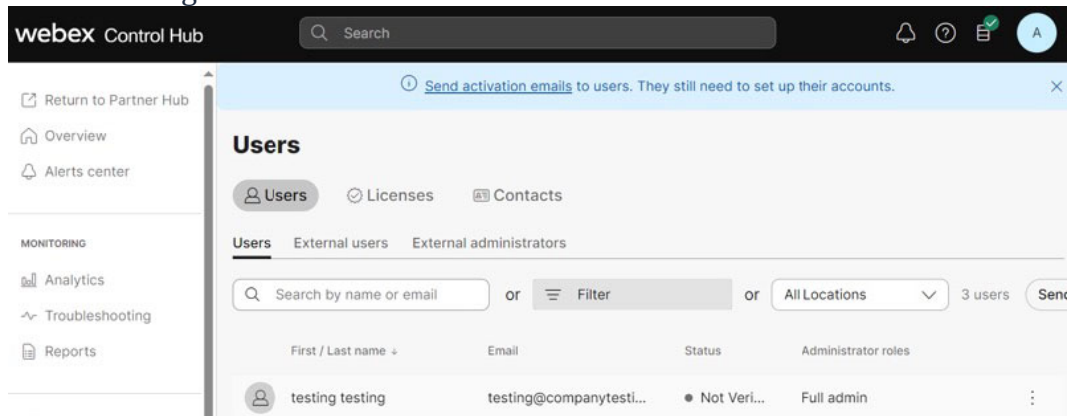
## 4.5 MAIN LINE

- Navigate to Management->Locations

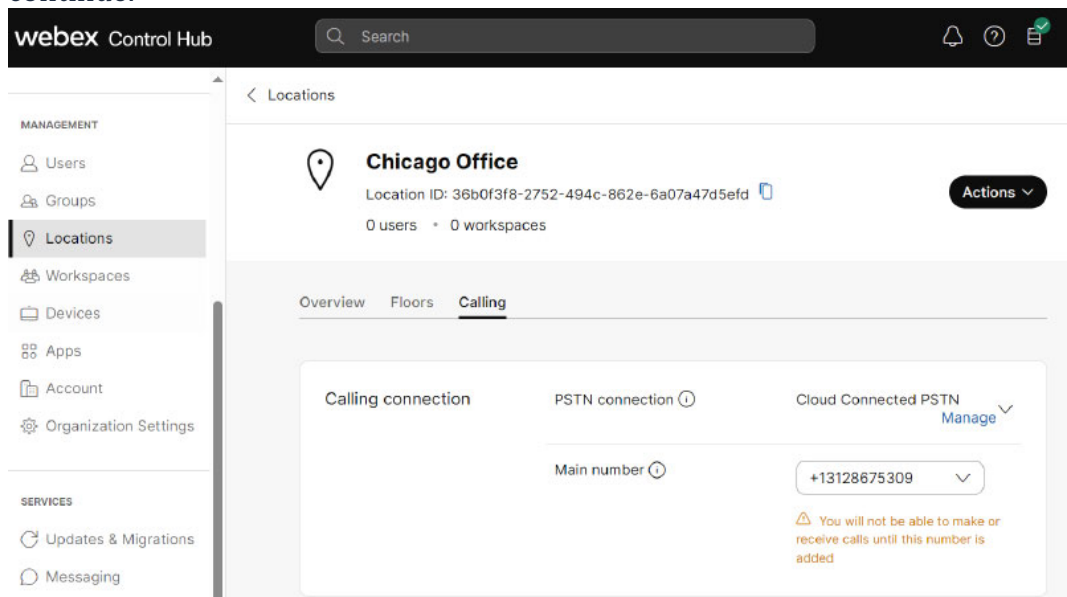
- Select the appropriate location from the list:



- Select 'Calling' tab from within the Location

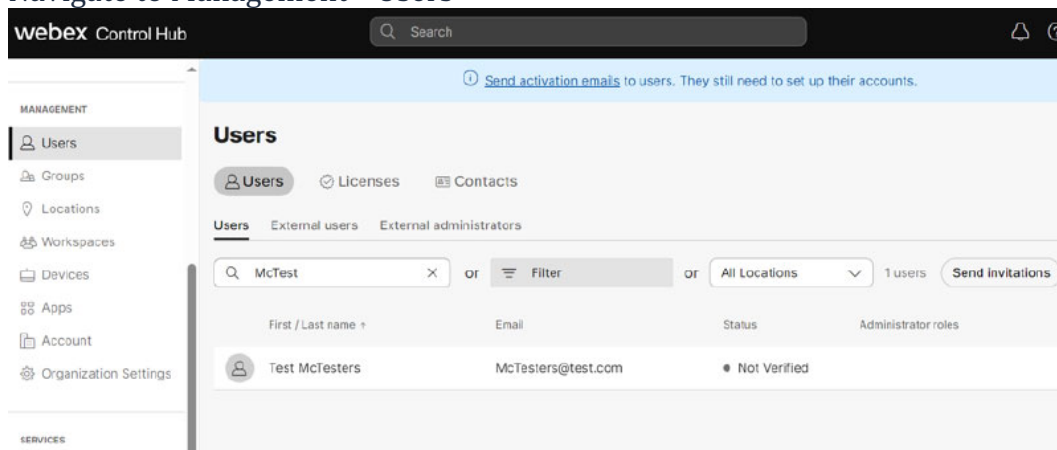


- Select a previously added number as the Main number and select 'Save' to continue.

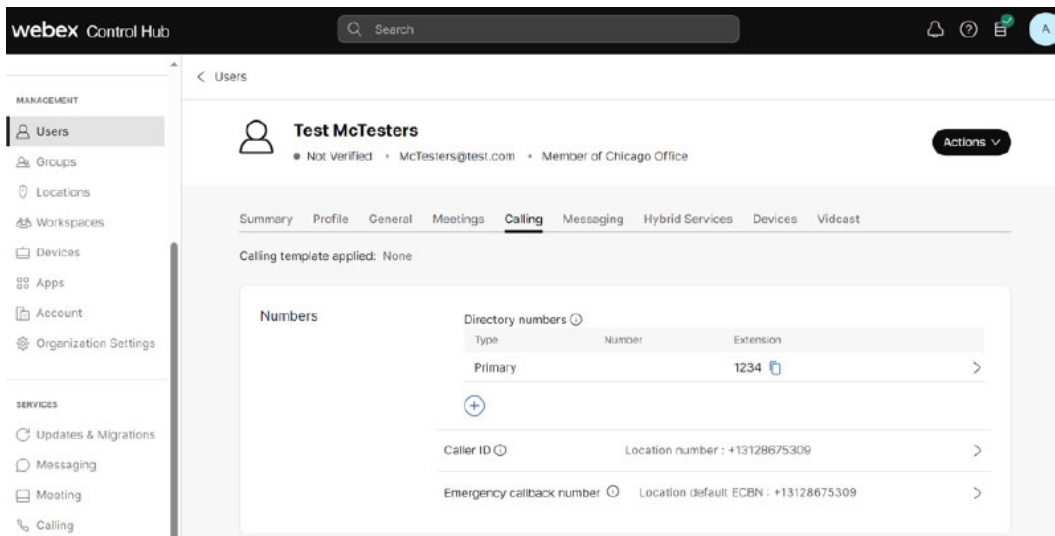


## 4.6 USERS

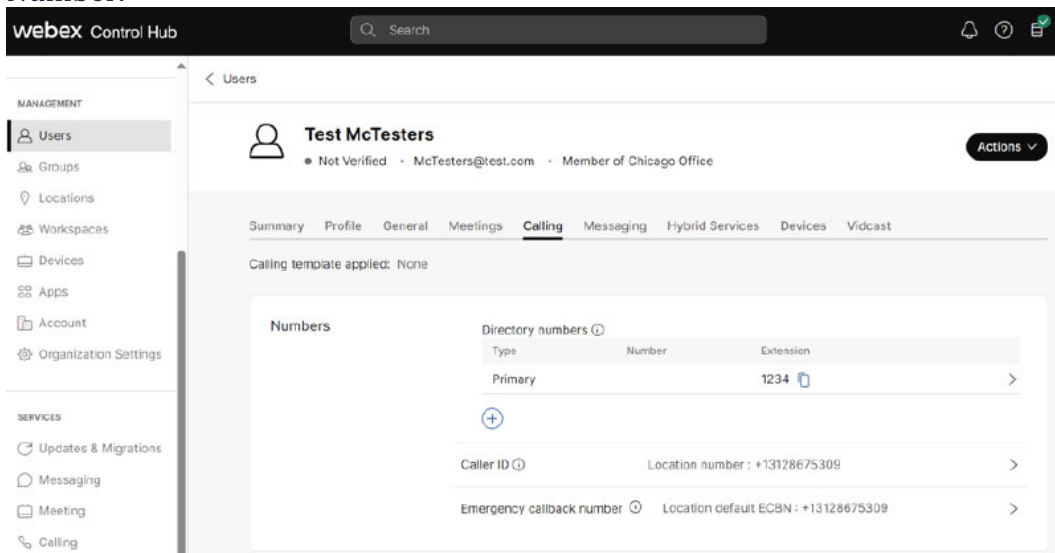
- Navigate to Management->Users



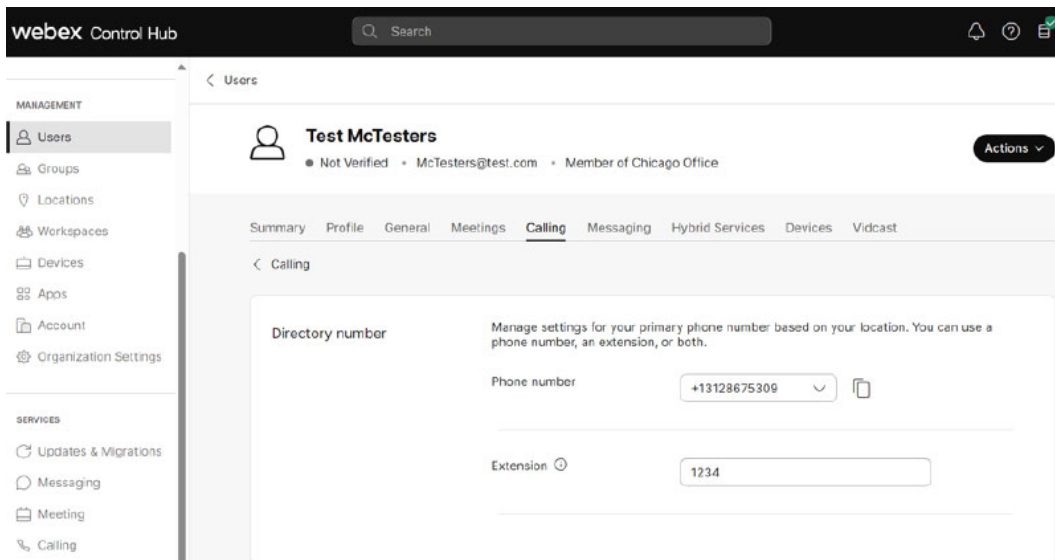
- Create a new user, or select an existing user. In this example we will select an existing user.



- Select the 'Calling' tab to apply Calling changes.
- Select the '>' button under Directory numbers to provision a previously added Number.



- Select a previously added Number from the Phone number dropdown and select 'Save' to continue.



## 4.7 VERIFICATION

- Navigate to Services->Calling->Call Routing->Verify Call Routing
  - Select an appropriate User
  - Enter in an appropriate PSTN number in Select Call Destination field
  - Select 'See Routing Result' to verify it shows it routes via Cloud Connected PSTN using Peerless Network (US)

