CALLING

Transitioning from Unified CM to Webex Calling

Deployment Guide

September 22, 2021

Contents

What's New in This Guide

Table 1 provide a historical list of updated and new topics added to this guide.

Table 1. *Unified CM to Webex Calling Transition Deployment Guide Publication History*

Date	Updated or New Topics	Update Details and Location
Apr. 24, 2020	Initial document publication	*Initial release*
May 25, 2020	Topics throughout document	Minor edits to text and illustrations to correct spelling, grammar, etc. based on feedback made throughout document.
Nov. 18, 2020	Preliminary Transition Considerations	Clarifications regarding single-site with predominately on-net calling, and 150 concurrent calls per LGW limitation (*Introduction > Overview*).
	Webex Calling Bandwidth Calculations	Bandwidth calculation clarifications – IP bandwidth only (*Perform initial readiness assessment of existing deployment > Network Connectivity*).
	PSTN Connectivity with Local Gateway	LGW PSTN and cloud/on-premises interop concurrent calling capacity clarification (*Perform initial readiness assessment of existing deployment > PSTN Connectivity with Local Gateway*).
	Webex Calling Regional Datacenters	Addition of Canadian regional datacenter Webex Calling platform (*Webex Calling region selection*).
	User Migration CSV Settings	Updated CSV setting information for user migration (*User provisioning for Webex Calling*).
	Emergency Calling Enablement	Clarifications related to emergency calling for Webex Calling with RedSky Horizon Mobility service (*Enable emergency calling*).

	Glossary	Added "Glossary" with tables of Unified CM concepts and constructs along with a definition and information on potential alignment with Webex Calling (*Glossary*).
Jan. 27, 2021	Topics throughout document	Product name change: "Webex Teams" to "Webex".
Jun. 17, 2021	Local Gateway concurrent call capacity	Updated per LGW concurrent call capacity to 250 (*Introduction > Overview* and *PSTN Connectivity with Local Gateway*).
September 22, 2021	Topics throughout document	*Product reference name changes from "Cisco Webex" to "Webex" throughout the document along with minor formatting and textual corrections.*

Introduction

Target Audience

This transition deployment guide is intended to be used by teams or individuals with experience configuring and administering Cisco Unified Communications Manager (Unified CM) and Cisco Endpoints including IP desk phones, video devices, and Jabber soft clients. There are links to product and support documentation throughout this document to assist.

Overview

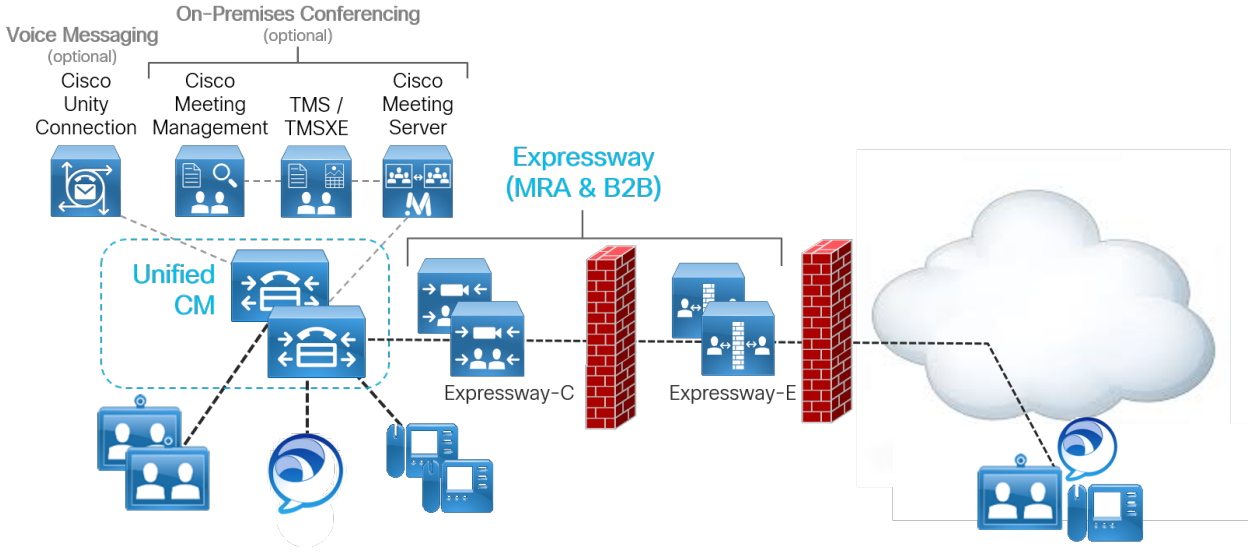
With the growth of cloud-delivered collaboration services, more and more customers are looking to move their existing collaboration workloads to the cloud given the promises of reduced total cost of ownership, simplified management, continuous feature delivery, increased scale, and superior reliability inherent in cloud-based services. As customers look to make the transition from on-premises to cloud collaboration services, it's important for them to understand what the transition entails and the steps required to make the transition.

The purpose of this document is to provide deployment guidance for customers specifically looking to transition from on-premises Cisco Unified CM calling to Webex Calling in the cloud. This deployment guide assumes that the reader has a basic understanding of the calling transition between Unified CM and Webex Calling including what changes when making this transition and what the differences are when moving the calling workload from on-premises to the cloud. Before proceeding ensure you have reviewed and are familiar with the information available in the transition map *Calling: Transition from Unified CM to Webex Calling* available at https://www.cisco.com/c/dam/en/us/td/docs/solutions/PA/mcp/TDM_CALLING_Unified_CM_to_Webex_Calling.pdf. This transition map document provides information about the changes and differences of this transition.

As shown in Figure 1, a typical deployment includes different collaboration infrastructure components on the network, a call control platform, and an edge platform, hardware and software endpoints, and in some cases even conferencing and scheduling platforms. In the Cisco architecture this would include Cisco Unified CM for call control, Cisco Expressway for remote access and business-to-business (B2B) edge services, Cisco Meeting Server / Cisco Meeting Management for on-premises conferencing, Cisco Unity Connection for voice messaging, and user-facing hardware (Cisco IP Phones, Webex DX and Room) and software (Cisco Jabber) IP-based endpoints. These components may vary slightly in some environments, but this is the

starting point for the transition described in the rest of this document.

Figure 1. *On-Premises Collaboration Architecture: Call Control and Remote Access*



Note: The architecture shown in Figure 1 is based on the Preferred Architecture (PA) for Cisco Collaboration Enterprise On-Premises Deployments. For more information on the Enterprise On-Premises PA, refer to https://www.cisco.com/go/pa.

Table 2 lists the key elements of the on-premises architecture prior to transitioning to Webex Calling in the cloud:

Table 2. *Before: On-Premises Calling Infrastructure Components*

Product	Description
Cisco Unified CM	On-premises call control providing device registration and call routing services
Cisco Expressway-C/E	Edge infrastructure providing Mobile and Remote Access (MRA) (business-to-business (B2B)) functionality enabling remote endpoints to connect securely from outside the organization. Expressway is deployed in pairs to provide firewall traversal for external endpoints.

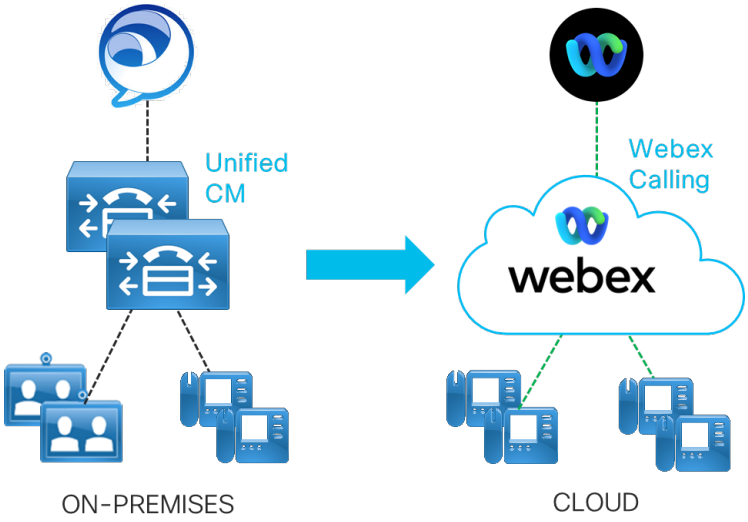
Cisco Meeting Server (CMS), Cisco Meeting Management (CMM), and Cisco Telepresence Management Suite (TMS)	On-premises voice, video, and web conferencing infrastructure providing multipoint meetings, meeting management, and scheduling capabilities. [Optional]
Cisco Unity Connection	On-premises voice messaging platform providing voicemail and unified messaging capabilities. [Optional]
Webex DX, Webex Room / Room Kit, Cisco IP Phones, and Cisco Jabber	IP-based devices registered to Unified CM and providing voice and video calling capabilities

As illustrated in Figure 2, customers who have an on-premises call control with Unified CM and desk and video IP endpoints have a choice of transitioning the architecture toward a Webex Calling cloud architecture.

The decision needs to be made based on customer's functionality requirements. Customers that have the following requirements should consider carefully before making this decision and may ultimately decide to keep call control on-premises:

- Phone models other than Cisco 7800 and 8800 IP phone series.
- Complex or numerous integrations with other on-premises systems / solutions.
- Complex dial plan and/or highly granular classes of service.
- Calling within deployment is predominately on-net within a single site.
- Restrictive, limited, or unreliable Internet access.
- Stringent data privacy and ownership policies.
- Compliance requirement for on-premises or in-country media recording and storage.
- Requirement for more than 250 concurrent PSTN and cloud / on-premises interop calls per Local Gateway (LGW).

Figure 2. *On-Premises Calling Transition Decision Tree*



Customers who wish to start leveraging Cisco cloud calling services should consider Webex Calling. This cloud calling service allows the customer to leverage the Webex global architecture for scale and connectivity. Participants on the corporate network and remote participants outside the corporate network can communicate using IP-based hardware endpoints or desktop or mobile soft client applications.

This document focuses on customers with Cisco Unified CM call control deployments that want to understand the general steps, considerations, and requirements for enabling Webex Calling deployment as depicted in the next section.

Core Components

Roles of the Components Involved

The target architecture for this migration includes several new components. This includes the Webex Calling service for cloud-based calling, Webex App, Cisco Directory Connector for identity integration, and Local Gateway IOS-XE router for PSTN access as well as on-premises to cloud calling integration. Cloud Connected PSTN (CCP) facilitated by a provider partner is another option for PSTN access.

As shown in Figure 3, the new components (Webex Calling, Directory Connector, and Local Gateway) are added to the existing on-premises deployment.

Figure 3. *After: Webex Calling Architecture*

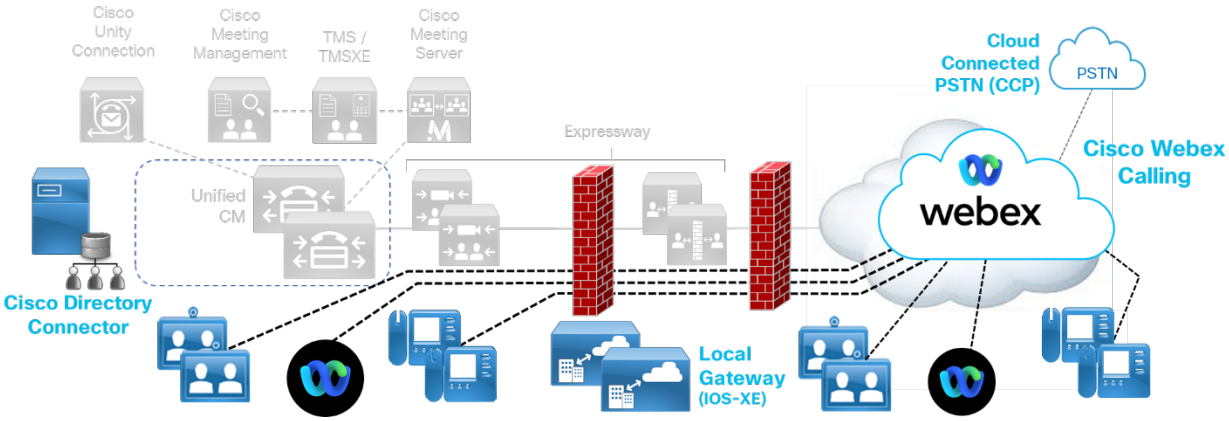


Table 3 lists the new elements of the architecture after transitioning to Webex Calling.

Table 3. *After: Cloud Calling Infrastructure Components*

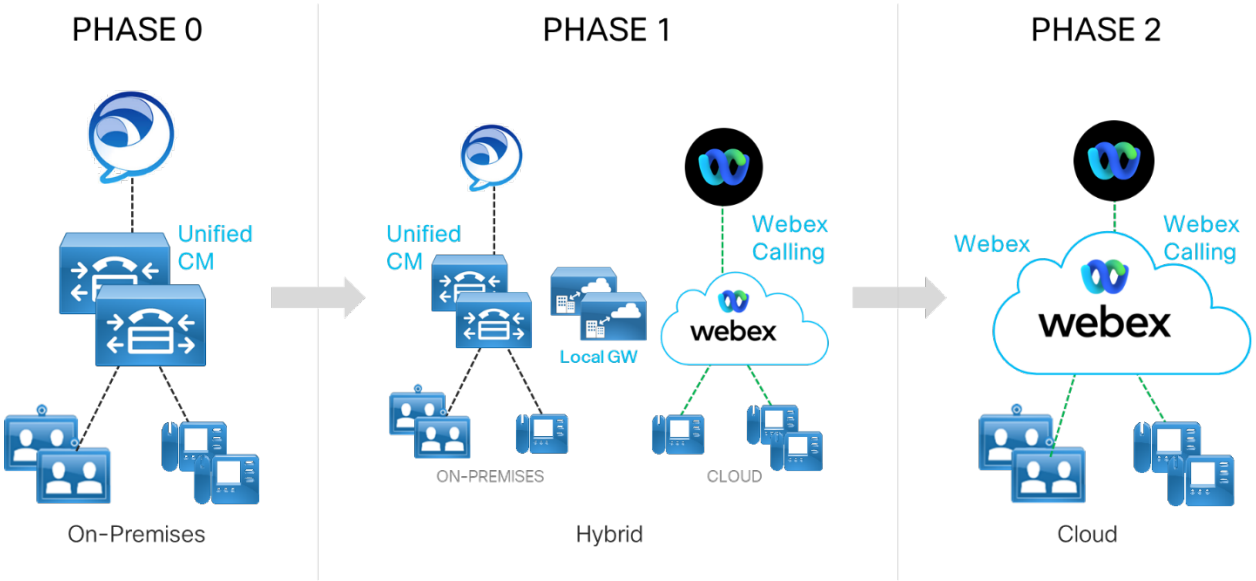
Product	Description
Webex Calling	Cloud-based call service delivered from the Webex platform and providing endpoint registration and call routing
Cisco Directory Connector	Windows application running on a Windows domain machine providing identity synchronization between the enterprise Active Directory and the identity store of the Webex organization.
Cisco IOS-XE Local Gateway	Cisco IOS-XE Integrated Services Router (ISR 1100 and 4000 series) or Cloud Services Router (CSR1000v) deployed on-premises and delivering PSTN access for cloud-registered endpoints as well as calling integration between Unified CM registered and cloud registered endpoints.
Cloud Connected PSTN (CCP)	Cloud Connected PSTN is a cloud-based option for PSTN access by Webex Calling endpoints. PSTN access is facilitated by a cloud PSTN provider and requires no on-premises equipment.
Webex App	Client application running on desktop OS (Windows, Mac) or mobile OS (Android, iOS) and registered directly to Webex Calling platform for calling functionality.

Transition

This section covers the pre-transition preparation steps, the transition implementation steps, and the post-transition steps to be considered for this workflow transition.

This document describes a phased transition in two parts. As shown in Figure 4, the initial transition phase (Phase 1) results in a hybrid deployment with dual call control where some devices are transitioned to cloud calling and other devices maintain on-premises call control for registration and call routing. The final transition phase (Phase 2) results in a pure cloud calling environment where all devices have been fully transitioned to cloud call control. How long an organization takes to transition to cloud calling fully will vary based on the deployment in question. In some cases, organization may make the initial transition and remain in the hybrid dual call control phase (Phase 1) for an extended period (months or even years) while in other cases an organization may fully transition to cloud calling (Phase 2) in a very short period of time (days or weeks). This document is intended to cover both partial (Phase 1) and full transitions (Phase 2).

Figure 4. *Phased Calling Transition: Hybrid and Cloud*



Note: It is possible that some organizations may maintain a hybrid dual call control deployment indefinitely with no plans to ever fully transition to cloud calling.

Pre-Transitions Steps and Considerations

Below is a summary of pre-transition items/steps to consider when performing the transition from Unified CM on-premises calling to Webex Calling.

1. Perform initial readiness assessment of existing deployment.

Prior to transition, to determine the feasibility and potential modifications required, it is important to consider each of the following aspects of your existing deployment. Likewise, you must understand key elements of the Webex Calling offer in comparison with the existing on-premises deployment.

Licensing

Understanding the current licensing structure of an existing deployment is a key consideration when preparing to migrate to Webex Calling. Perform a license assessment of the following areas of your existing Cisco on-premises solution.

- Platform
  The ability to fully articulate what is currently licensed on your core platform will be critical when working with your account team or partner to determine the best path to Flex licensing. Webex Calling is licensed using Flex licensing. For more information on Flex licensing refer to the Cisco Collaboration Flex Plan information available at https://www.cisco.com/c/en/us/products/unified-communications/collaboration-flex-plan/index.html.

- Devices
  Determine what license category your existing and planned new devices will belong to with Webex Calling. Webex Calling licensing station types include knowledge worker, basic user, and non-user assigned devices for common areas. For more information on Webex Calling device licensing refer to the data sheet available at https://www.cisco.com/c/en/us/products/collateral/unified-communications/webex-calling/datasheet-c78-742056.html.

- Local Gateway

  Because Cisco Unified Border Element (CUBE) is required for PSTN access for this transition, CUBE licensing must also be considered. CUBE licensing considerations are covered later in this document.

Deployment Sites

The number and types of sites (large central, regional, branch, and so on) within your existing deployment should be considered when planning this transition. A full understanding of the existing deployment sites will aid in strategically planning for a successful transition particularly when it comes to determining what sites to migrate and in what order. Understanding in detail dial plan requirements (numbering, dialing habits, classes of restriction, and so on), site network connectivity and bandwidth (Internet, WAN, LAN), and PSTN access (local or centralized, IP or TDM) for each site will be critical when making migration decisions. For more information on common deployment models and key considerations, please refer to the collaboration deployment models information available in the Collaboration SRND at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12/models.html.

Another important deployment consideration when transitioning to Webex Calling is location availability. Webex Calling has different capabilities, subscriptions and devices that are available depending on where your deployment is located. For more information on Webex Calling geographic availability, refer to the *Where is Cisco Webex Available* article available at https://help.webex.com/en-us/n6fwepj/Where-is-Cisco-Webex-Available#id_98285.

Finally, it is important to understand the impact the transition to Webex Calling will have on other collaboration services. Based on the objective of this document, the general assumption is that if existing collaboration services outside of the calling workload are to be maintained, then transition to the phase 1 hybrid deployment mentioned above is expected. Examples of collaboration services that may require hybrid deployment include contact center, meetings, paging, call reporting, and so on. For more information about the transition of additional collaboration workloads and services refer to the Collaboration Transitions documentation available at https://cisco.com/go/ct.

Network Connectivity

Consider existing provider data connections (MPLS, SD-WAN, and so on) and generally plan for direct Internet access at each location within your deployment. Because you will be consuming cloud-based services, reliable Internet connectivity with sufficient bandwidth is a base requirement. You should reconsider making this transition if your organization locations' Internet connection(s) are not generally reliable with low latency and adequate up and downstream throughput.

Table 4 shows the call types available with a Webex Calling deployment along with the codec and maximum bandwidth required for each call type. As shown in Table 4 the required audio call bandwidth for each call type can be calculated using the following general formula:

*Number of expected concurrent calls * Bandwidth per call based on codec = Required network throughput.*

Table 4. *Webex Calling Call Type Bandwidth Calculations*

Call Types	Codec - Bandwidth	Bandwidth Calculations
Webex App / MPP1 Phone -> Webex App	OPUS - 70 kbps	Number of concurrent calls * 70 kbps = Required network throughput
Webex App / MPP1 Phone -> MPP1 Phone	OPUS - 70 kbps	Number of concurrent calls * 70 kbps = Required network throughput
Webex App / MPP1 Phone -> PSTN via LGW	G.711 - 80 kbps	Number of concurrent calls * 80 kbps = Required network throughput
Webex App / MPP1 Phone -> PSTN via CCP	G.711 - 80 kbps	Number of concurrent calls * 80 kbps = Required network throughput
Webex App / MPP[1] Phone -> Enterprise via LGW	G.722 - 80 kbps	Number of concurrent calls * 80 kbps = Required network throughput
Webex App / MPP[1] Phone -> Webex Calling Voicemail	OPUS - 70 kbps	Number of concurrent calls * 70 kbps = Required network throughput

[1] Multiplatform Phone (MPP)

By summing the concurrent required network throughput per call type, the total potential bandwidth requirement for a specific site can be determined.

All call legs are always anchored on the Webex Calling access SBCs. To determine the required internet bandwidth for any given Webex Calling location not only the inter-location calls need to be considered, but also intra-location calls and calls to and from a Local Gateway at that location. For example, an intra-site call between two MPPs would need up to 2 x 70 kbps full duplex on the location's internet access.

By summing the concurrent required network throughput per call type, the total potential bandwidth requirement for a specific site can be determined.

Table 5 shows an example of a complete bandwidth calculation exercise assuming that all devices are located in the same site.

Table 5. *Webex Calling Bandwidth Calculation Examples*

Call Types	Number of Concurrent Calls	Total Bandwidth
Webex App / MPP1 Phone -> Webex App	15	2 * 15 * 70 kbps = 2,100 kbps
Webex App / MPP1 Phone -> MPP1 Phone	15	2 * 15 * 70 kbps = 2,100 kbps
Webex App / MPP1 Phone -> PSTN via LGW	50	2 * 50 * 80 kbps = 8,000 kbps
Webex App / MPP1 Phone -> PSTN via CCP	0	0 * 80 Kbps
Webex App / MPP[1] Phone -> Enterprise via LGW	15	2 * 15 * 80 kbps = 2,400 kbps
Webex App / MPP[1] Phone -> Webex Calling Voicemail	5	5 * 70 kbps = 350 kbps
TOTAL CALLS / BANDWIDTH	100 calls	14,950 kbps / 14.95 mbps

[1] Multiplatform Phone (MPP)

Note: All bandwidth values in Table 4 and Table 5 refer to IP bandwidth. Link bandwidth is higher depending on WAN encapsulations.

Note: The bandwidth in Table 4 and Table 5 is for audio calls. For video call bandwidth, Webex App and the MPP 8845/65 phones support H.264 video with maximum resolution of 720p at a maximum bandwidth of 1,500 kbps per call. However, the amount of bandwidth consumed at any point during the will fluctuate based on variable bit rate inherent in video communications.

Webex Calling requires reliable Internet connectivity and offers global reach from all the customer locations thus, eliminating the need for endpoint survivability. If an

SRST like option for endpoints is critical to the existing deployment, then Webex Calling should not be considered as a migration option.

Call Recording

Call Recording integration is between Webex Calling and Dubber (partner offering) data centers and all recorded media is securely kept in the cloud. If compliance and regulation require media be kept on-premises or in your country of deployment, understand this option is not available as part of the call recording architecture.

Voicemail

Voicemail is an integral part of the Webex Calling offer and integration with a premise-based voicemail solution such as Cisco Unity Connection or Cisco Unity Connection Express is not available. Further, there is no ability to migrate existing Unity Connection voicemail messages or greetings to the native voicemail service available with Webex Calling. Likewise, there is no migration of Unity Connection call handlers and auto-attendant functionality to Webex Calling, however, the basic auto-attendant functionality available with Webex Calling may be configured as a possible replacement.

PSTN Connectivity with Local Gateway

Local Gateway is an essential component of the transition strategy and the Local gateway platform must be either a Cisco Integrated Services Router (ISR) 4000 series, Cisco 1100 Integrated Services Router series, or Cloud Services Router (CSR1000v) series.

Currently, Webex Calling allows no more than 250 concurrent sessions from a single Local Gateway, which by default becomes the session count limit for Local Gateway based PSTN calls and Inter-site calls between Unified CM and Webex Calling endpoints. If a Local Gateway deployment requires more than 250 concurrent calls, please contact Cisco Technical Assistance Center (TAC) to request increasing this limit.

Any calls exceeding this limit are rejected with a “403 Forbidden”. The “show call active voice” command can be run on the Local Gateway at any instance to determine the total number of active calls.

Contact your Cisco account team to explore other deployment options if you need a higher number of concurrent sessions.

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. One-way latency between the Local Gateway and the Webex Calling data center should not exceed 100 ms and the jitter should be less than 10 ms.

2. Perform network readiness assessment

Customers need to conduct a network assessment prior to migrating to Webex Calling. It is recommended to confirm network bandwidth availability for expected call volume, ensure quality of service (QoS) requirements are met, and understand the various ports that must be opened in the edge firewall(s).

For more details on network requirements for Webex Calling, refer to the *Cisco Webex Calling Customer Network Minimum Requirements Service Guide* available at: https://callinghelp.cisco.com/wp-content/uploads/2019/05/WC-Customer-Network-Minimum-Requirements-Guide-v2.2_062019.pdf.

Customers can also use cscan.webex.com for network assessment which gives information on customer's network quality, how many calls can be established, latency, and so on. For more information on the cscan tool, refer to the *Use CScan to Test Webex Calling Network Quality* article available at https://help.webex.com/en-us/y27bej/Use-CScan-to-Test-Webex-Calling-Network-Quality.

3. Understand Webex Calling region selection

Webex Calling operates five regional platforms as shown in Figure 5: North America, Canada, EMEAR, APJC (Japan) and APJC (Australia). Each Webex Calling instance provides redundant datacenters within that region.

Figure 5. *Webex Calling Regional Datacenters*

Each Webex Calling customer is provisioned on one of the four Webex Calling instances. All provisioning information of that customer is stored in that Webex Calling instance and the SIP signaling of all endpoints and Local Gateways provisioned for that customer is tied to the Webex Calling instance the customer is provisioned on. Because the initial Webex Calling region selection cannot be changed later it is important to consider all relevant factors as part of the decision process leading to the Webex Calling region selection. To avoid excessive signaling round-trip delay it is important to decide early in the transition process which Webex Calling instance should be used. Cisco recommends selecting the Webex Calling instance which provides the lowest signaling round-trip times for the largest number of users within the deployment.

Another factor to consider in the Webex Calling region selection is the country availability of PSTN services provided by Cloud Connected PSTN (CCP) providers available within that region. While during the transition period PSTN access for Webex Calling devices must be facilitated via a Local Gateway to enable interworking with Unified CM registered devices, after successful completion of the transition, PSTN access for Webex Calling may be switched to CCP. At that point the country availability of the CCP providers available within the Webex Calling region becomes an important factor.

Refer to the Cloud Connected PSTN providers list available at https://community.cisco.com/t5/collaboration-voice-and-video/cloud-connected-pstn-provider-partners-for-cisco-webex-calling/ta-p/3916211. In addition, for

Webex Calling country availability refer to the *Where is Cisco Webex Available* article https://help.webex.com/en-us/n6fwepj/Where-is-Cisco-Webex-Available#id_98285.

4. Analyze deployment dial plan

Each user in Webex Calling is provisioned with an extension. The extension length is a fixed global setting: all extensions in a Webex Calling deployment have the same length. Extension dialing can be used between Webex Calling users both within a location and between locations. Abbreviated inter-site extension dialing (the latter case) only works if the dialed extension is unique.

The dial plan described in the Preferred Architecture for Cisco Collaboration 14 Enterprise On-Premises Deployments, CVD does not support abbreviated inter-site extension dialing. Instead, the *Preferred Architecture for Cisco Collaboration 14 Enterprise On-Premises Deployments, CVD* recommends establishing an enterprise specific numbering plan by prefixing the extensions with a unique steering digit followed by a fixed length site code and to use this number format for abbreviated inter-site dialing.

Table 6 shows an example of three locations where the extension ranges of two locations, NYC and RTP, are identical. Establishing an enterprise numbering scheme with inter-site steering digit “8”, followed by a three-digit site code, and the four-digit extension creates a non-overlapping abbreviated inter-site dialing habit.

Table 6. *Enterprise Numbering Example*

Site	Extension Range	Site Code	Enterprise Range
NYC	2XXX	202	8 202 2XXX
SFO	3XXX	203	8 203 3XXX
RTP	2XXX	204	8 204 2XXX

To allow for a smooth transition the set of dialing habits for users before and after transitioning to Webex Calling ideally should be the same. To prepare for the transition for each location the DID ranges and extension ranges (or abbreviated intra-site dialing habits) need to be documented. Based on this information then the inter-site steering digit needs to be selected.

Table 7 shows an example of three locations and fixed length extension ranges. Because overlapping dialing habits need to be avoided, it is important to make sure that for any extension range the first digit of the range does not match the steering digit for abbreviated inter-site dialing. If for example “8” is selected as the steering digit for inter-site dialing, then no extension range in any site can start with “8”. Typically, the extensions at a given location match the last few digits of the DIDs assigned to that location. To avoid conflicts the first digit of the extension can be changed. If, for example, DIDs in the +1 408 555 8XXX range are used in a given location, then instead of using 8XXX as extension range 7XXX can be used for the extensions in that site.

Table 7. *Fixed Length Webex Calling Extension Ranges*

Site	Extensions (Pre-Transition)	Webex Calling Extensions	Site Code	Enterprise Range
NYC	2XXX	2XXX	202	8 202 2XXX
SFO	8XXX	7XXX	203	8 203 7XXX
RTP	1XX	11XX	204	8 204 11XX

Any seven-digit dial string dialed on a Webex Calling device using the US Webex Calling dial plan always gets transformed to a full 10-digit national number. This behavior makes it impossible to use seven-digit enterprise numbering schemes with Webex Calling. If the existing enterprise numbering schema and the corresponding abbreviated inter-site dialing habit has seven digits, then during the transition to Webex Calling the numbering schema must be changed to a longer or shorter form. The easiest way to achieve this is to add an additional padding digit to the numbering schema. The new longer inter-site dialing schema only needs to be adopted by users already migrated to Webex Calling. Users still on Unified CM can continue to dial seven digits. The enterprise dial plan on Unified CM in this case needs to make sure that abbreviated seven-digit dialing from Unified CM to Webex Calling gets transformed to either +E.164 or to the abbreviated dialing format deployed on Webex Calling. This needs to be done before sending the call to the Local Gateway.

Table 8 shows an example how this renumbering. In this example abbreviated inter-site dialing on Unified CM uses steering digit “8” followed by a two-digit site code and a four-digit extension. To avoid seven-digit abbreviated inter-site dialing

for locations on Webex Calling, the site codes can easily be changed to three digits by prefixing an arbitrary padding digit (“8” in the example) to the two-digit site codes used in Unified CM so that inter-site dialing from Webex Calling phones uses steering digit “8” followed by the padding digit “8”, the old two-digit site code, and the four-digit extension. Users on Webex Calling don’t need to remember new site codes; they only need to remember to use “88” as prefix for inter-site dialing instead of “8” on Unified CM.

Table 8. *Transitioning Seven-Digit Dialing*

		UCM		Webex Calling	
Site	Extensions	Site Code	Enterprise Range	Site Code	Enterprise Range
NYC	2XXX	22	8 22 2XXX	822	8 822 2XXX
SFO	8XXX	23	8 23 7XXX	823	8 823 7XXX
RTP	1XXX	24	8 24 11XX	824	8 824 11XX

In a scenario with different enterprise number formats on Unified CM and Webex Calling if enterprise numbers are presented as calling party information for calls from Unified CM to Webex Calling (for example for calls from devices without a DID), it is important to also implement a mapping between the different number formats for calling party information to ensure callback works. This mapping can be achieved by using calling party transformation pattern on the trunk between Unified CM and the Local Gateway.

5. Inventory existing locations/sites

To prepare for the provisioning of locations on Webex Calling the required information for all migration target locations needs to be collected. Table 9 summarizes the information needed for each location.

Table 9. *Information to Capture for Each Location*

Information	Comment
Extension Range(s)	▪ Each location in Webex Calling can have extensions starting with different digits. One digit must be spared for the inter-site dialing steering digit (for example “8”) and one for the PSTN steering digit (for example “9”). No extension range can start with either of these two digits. ▪ All extension ranges of all locations must be of equal length.
DID Range(s)	
PSTN Steering Digit	
Site Code	All site codes of all locations need to be unique and to have the same length.
Main Number	When creating a location two DIDs need to be provisioned. One as main number (for example to be assigned to an auto attendant service) and one for the voicemail portal.
Voicemail Number	See above
Number of Licenses	
Concurrent Calls in the Busy Hour	Sum of concurrent calls between Webex Calling devices and between Webex Calling devices and the Local Gateway (PSTN and calls to Unified CM devices). Needed to determine the required internet access bandwidth
Country	
Time Zone	
Language	
Contact (Name, Phone, Email)	
Address (Street Address, City, State, Zip Code)	

Emergency Services Physical Dispatchable Location for Endpoints	Device dispatchable location used for emergency calling generally includes the following: building address, building address + floor number, building address + suite number, or building address + floor number + office/cubical number.
Per Device Unique Physical Network Location for Emergency Services	Physical network location for emergency calling generally includes the following: switch / switchport for wired devices, wireless access point (AP) basic service set identifiers (BSSIDs) for wirelessly connected devices, and/or on-premises IP subnet(s) for endpoint devices.

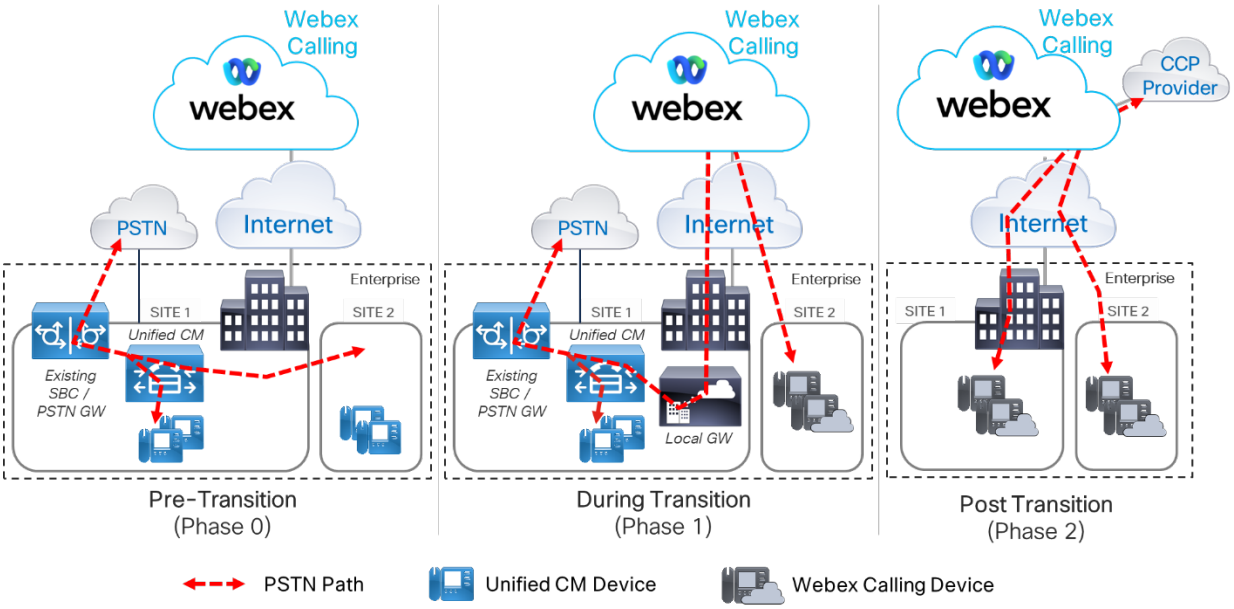
6. Understanding PSTN access options

When it comes to PSTN access for Webex Calling, it is important to understand the following considerations:

- PSTN is required for off-net calling and between enterprises.
- Cisco never supplies the PSTN.
- Only one PSTN option per location is possible:
  - Cloud Connected PSTN (CCP).
  - Cisco Local Gateway for on-premises PSTN and Unified CM integration.
- Emergency call routing and lawful intercept are the responsibility of the PSTN provider.

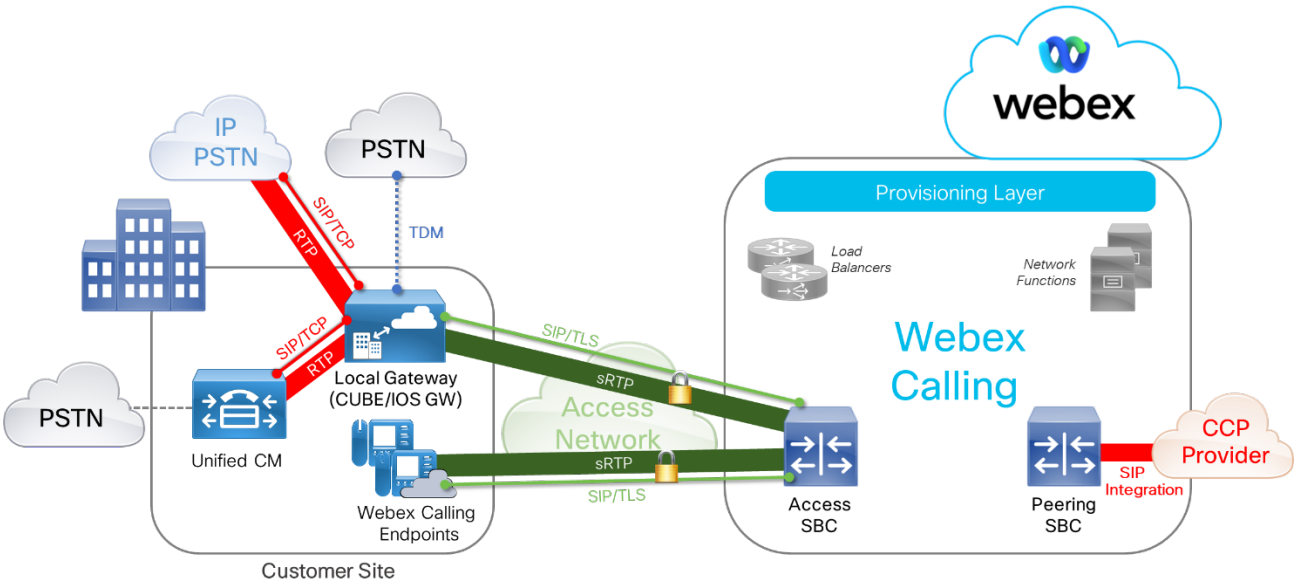
A Local Gateway is required to create a connection between Webex Calling and Unified CM as long as Unified CM and Webex Calling coexist. This connection is not only required to route all calls between Unified CM registered devices and Webex Calling registered devices; it also provides PSTN access for all Webex Calling devices. PSTN access for each location in Webex Calling can either be facilitated by a CCP provider or via a Local Gateway. It is not possible to setup Webex Calling so that inter-enterprise calls originating from within a given Webex Calling location use a Local Gateway while PSTN calls use a CCP provider. This limitation implies that during the transition PSTN access for both, Unified CM registered devices and Webex Calling devices, needs to be through PSTN trunks controlled by Unified CM. Only at the end of the transition, when no Unified CM registered devices remain, PSTN access for all users can be moved to a CCP provider. These steps are shown in Figure 6.

Figure 6. *PSTN Transition*



Before proceeding, it is important to understand the Local Gateway and CCP architectures as shown in Figure 7.

Figure 7. *Local Gateway and Cloud Connected PSTN (CCP) Architecture*



At the on-premises side, the Local Gateway relies on a SIP connection or trunk to Unified CM and in the case of IP-based PSTN connections a SIP trunk is also used to communicate with the IP PSTN provider network. To connect to Webex Calling in the cloud, the Local Gateway communicates via secure SIP TLS to the Webex Calling service. This connection is anchored at a Webex Calling cloud-hosted Access Session Border Controller (SBC) which serves as the gateway to cloud calling services. In all cases, the Local Gateway relies on RTP or sRTP (encrypted RTP) for all media connections.

In the case of CCP, cloud-hosted Peering SBCs serve as the SIP interconnect to CCP provider partner networks. The Peering SBCs are responsible for all integration aspects with the partner network.

The choice to eventually deploy a CCP option or a Local Gateway is up to the customer. With the CCP option, a customer does not need to invest in local gateway hardware and maintenance. Additionally, with a CCP option, there is no media hair-pinning to the Webex Calling cloud.

On the other hand, a Local Gateway allows the customer to re-use an existing UC enabled Cisco ISR or CSR1000v router (assuming the existing ISRs or CSR1000v router are supported for Local Gateway and that the scalability of the existing platforms is sufficient to carry the additional load of the Local Gateway role). This is also preferable if they have a pending service contract with their current PSTN provider. It also allows for PSTN interconnect in locations not supported by the Cloud connected PSTN provider.

A single Local Gateway can be deployed and utilized by multiple Webex Calling locations. However, only a single local gateway can be assigned to a location within Control Hub. Further, if during the transition multiple split sites (some users within the same location still on Unified CM and some already on Webex Calling) with the same extension range exist and extension dialing from Webex Calling to Unified CM is a requirement, then on Unified CM for extension dialed calls from Webex Calling to Unified CM a location specific dialing context needs to be established to enable Unified CM to differentiate between extension dialed calls from different locations. This can be achieved by configuring dedicated Local Gateways for each location requiring split site extension dialing so that the dialing context on Unified CM can be established via calling search spaces inbound on the now location specific trunks from the different Local Gateways. These Local Gateways do not require dedicated CUBEs. All Local Gateways can be configured as “logical”

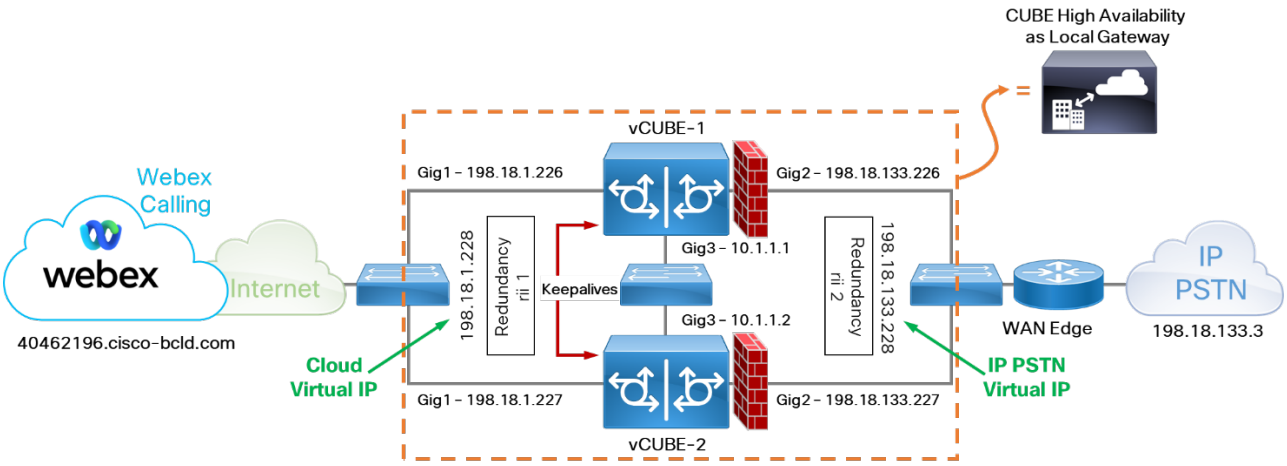
instances on the same CUBE (or CUBE High Availability pair) by overlaying the Local Gateway specific dial-peer sets within the CUBE configuration.

For all IP-based environments, customers have the option to deploy CUBE high availability as Local Gateway for call preservation. CUBE high availability Layer 2 box-to-box redundancy uses the Redundancy Group (RG) Infrastructure protocol to form an active/standby pair of routers. The active/standby pair share the same virtual IP address (VIP) across the respective interfaces and continually exchange status messages. CUBE session information is check-pointed across the active/standby pair of routers enabling the standby router to immediately take over all CUBE call processing responsibilities if the active router should go out of service, resulting in stateful preservation of signaling and media.

Note: Check pointing is limited to connected calls with media packets. Calls in transit are not check pointed, for example, Trying or Ringing state.

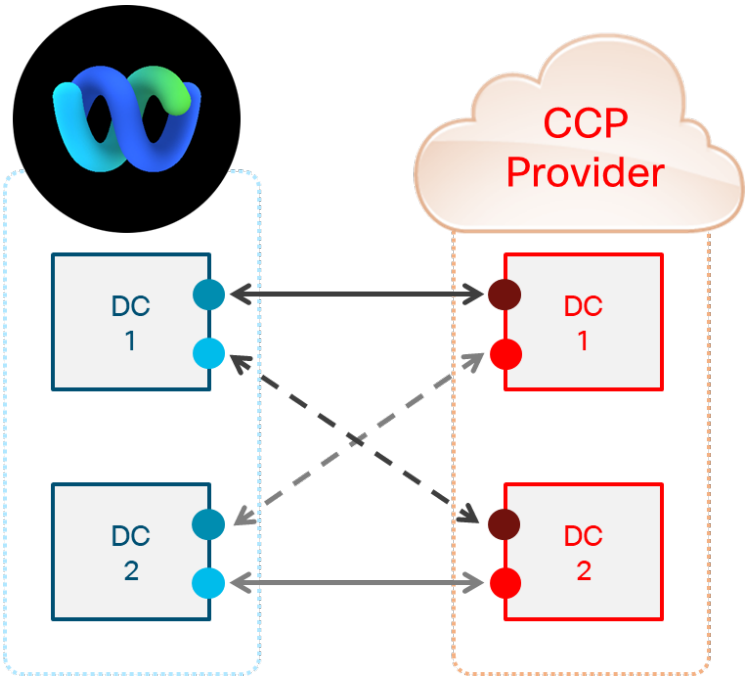
Refer to Figure 8 below which depicts a typical CUBE high availability as Local Gateway setup.

Figure 8. *CUBE High Availability with Local Gateway*



The CCP option entails a static SIP trunk from the Webex Calling data centers to CCP provider data centers. Redundancy is achieved by full mesh connectivity between two Webex Calling data centers and two CCP provider data centers and failover is automatic and transparent to end users. Figure 9 shows this high availability architecture.

Figure 9. *Cloud Connected PSTN High Availability*



When a customer Location is added within Webex Control Hub, the customer administrator may select a CCP provider from the list of integration options. The country of the customer location determines which CCP providers appear in the drop-down list. Only CCP providers that support that country are displayed.

7. Inventory existing endpoints/clients.

Before beginning the transition it's important to inventory your existing hardware and software endpoints. Having a complete list of phone types, models, and quantities will ensure you can adequately plan for transitioning endpoints and mitigating the impact to your deployment for those devices that cannot be migrated to cloud calling. The inventory should be used to determine the endpoints to transition, the endpoints to replace prior to the transition, and the endpoints that may remain managed and registered to on-premises call control.

Desk Phones

For audio and video VoIP desk phones including Cisco IP Phone 7800 and 8800 series, Cisco Unified IP Phone 7900 series, and other personal desk endpoints, only the 7800 and 8800 series endpoints are supported with Webex Calling. Prior to transition these phones must be migrated to Multiplatform Phone (MPP) in order to be transitioned to Webex Calling.

All other desk phones will need to be replaced with 7800 and 8800 series endpoints or must remain registered to Unified CM if you plan to maintain them.

Video Endpoints

Personal and room video endpoints including the Webex series, Webex Room series, Webex DX series, and other hardware video endpoints are not supported with Webex Calling. These devices must remain registered to on-premises Unified CM or transitioned to the Webex platform (registered as Webex Devices) if cloud registration is desired.

Note: When video devices are moved from Unified CM registration to Webex, the URI for these endpoints will change as they are now cloud registered.

The underlying assumption for video endpoints in this deployment is that they are running CE firmware and are shared endpoints used for point-to-point video calling and multi-point on-premises or cloud-based conferencing.

Note: Phone models 8845 and 8865 are personal video endpoints and are supported with Webex Calling.

Soft Clients

Cisco Jabber clients on desktop and mobile platforms are not supported with Webex Calling. Depending on the deployment mode(s) implemented for Jabber (IM-only, phone-only, and/or full UC modes), you may decide to maintain Jabber registration to Unified CM and other on-premises services (Unified CM IM&P, Unity Connection, Cisco Meeting Server). On the other hand, users may be transitioned to the Webex app, the preferred cloud-based software client. Moving users to the Webex app may be done prior to transitioning to Webex Calling to give users time to familiarize themselves with the new application. Alternatively, users may be migrated from Jabber on-premises services in phases after the transition to Webex Calling begins.

To ease the initial transition to Webex app, consider moving users to the Webex app with Unified CM calling, so they can begin using cloud messaging and meeting services while at the same time enjoying the existing on-premises Unified CM calling experience.

8. Inventory and plan for existing users transition to Webex Calling

Determine which users within the existing set of on-premises calling users will be transitioned to Webex Calling. If all users will be transitioned, but the number of users is large, it is a good idea to move users in groups in order to ensure that IT staff and support personnel are able to handle the transition and any issues that may arise. You should also allow some time to provide initial information and training sessions to prepare users for this transition. User transition grouping can be done based on a variety of criteria including the location or site users are assigned to, users' departments, or even user types (knowledge workers, executives, mobile workers, and so on).

As an example, if users in the deployment are divided across three main sites, New York (NYC), San Francisco (SFC), and Research Triangle Park (RTP), a user transition plan may look like the plan outlined in Table 10 below.

Table 10. *User Transition Plan by Site*

User Site / Location	Pre-transition Information and Training Sessions	Transition Period	Post-Transition Support
NYC (1,525 users)	Week of April 1	April 15 – April 27	Week of April 29
SFO (1,600 users)	Week of May 6	May 20 – May 31	Week of June 3
RTP (1,275 users)	Week of June 3	June 17 – June 28	Week of July 1

Transition Steps and Considerations

Below is a summary of transition steps required for the transition from Unified CM on-premises calling to Webex Calling in the cloud. You should only perform these steps during a planned maintenance window for your organization. Before proceeding you should back up all collaboration and infrastructure systems if you must back out or abandon the transition.

Follow these transition steps to move from on-premises calling (Unified CM) to cloud calling (Webex Calling):

1. Order Webex Calling

To begin the transition, a Webex Calling organization with proper licensing is required. For information on ordering Webex Calling and licensing, start with the Webex Calling datasheet available at https://www.cisco.com/c/en/us/products/collateral/unified-communications/webex-calling/datasheet-c78-742056.html.

CUBE Trunk Licenses (CUBE-T-STD or CUBE-T-RED) are included as part of the Webex Calling subscription at a ratio of one license for every two knowledge workers. However, unified communications and security platform licenses for the hardware-based platforms or AX technology package and throughput licenses for CSR1000v platform are not included and must be procured separately before beginning with the transition.

For more details, check the "Local Gateway for Webex Calling Ordering Guide" at https://www.cisco.com/c/en/us/products/collateral/unified-communications/unified-border-element/guide-c07-742037.html.

2. Implement required network and firewall changes.

The first step in transitioning to Webex Calling is ensuring that there is connectivity over the Internet between the on-premises network and the Webex cloud. Most organizations do not connect directly to the Internet, but instead connect through one or more firewalls. For this reason, it is important to understand the traffic flows required between the on-premises network and Webex for Webex Calling. Network and security administrators must understand these flows in terms of direction, protocols, IP addresses, and port numbers so that corporate firewalls and other network components can be configured to accommodate this traffic.

For information on the required flows including IP address, ports, and protocols refer to the *Port Reference Information for Cisco Webex Calling* article available at https://help.webex.com/en-us/b2exve/Port-Reference-Information-for-Cisco-Webex-Calling.

Use this information to properly configure the firewall, proxies, and other network infrastructure in the existing deployment to enable Webex Calling network flows.

3. Prepare Webex Control Hub for directory integration and user provisioning.

Before enabling directory integration between the corporate directory and the Webex cloud identity store, the following set of summarized steps should be taken:

i. Add and verify organization domain(s).
ii. Convert existing users.
iii. Claim organization domain(s).
iv. Set up SSO.
v. Suppress automated user email invitation.
vi. Determine license assignment method.

Each step is explained in detail below.

i. Add and verify organization domain(s).

To add a domain to your Webex organization use the **Add Domain** option under **Settings > Domain** in Webex Control Hub (https://admin.webex.com/). Start by entering the administrator domain and click **Add**. Then find the verification token by selecting **Retrieve verification token** (available by clicking the ellipsis (...) next to the domain name). This verification token must then be added as a DNS TXT record to your DNS host. Once this is done click **Verify** next to the domain. If successfully verified, you will see "Verified" next to your domain. Repeat this process for each domain owned by your organization.

Note: You must add and verify the administrator domain first. Failure to do so will result in administrator lockout.

ii. Convert existing users.

Existing users from other organizations, including in the free consumer organization are not automatically converted to your organization. You will need to convert these users manually. You should convert consumer users to your organization(s) before claiming the domain. That way, these users will not receive notification after domain claim.

iii. Claim organization domain(s).

As a further security measure claim organization domain(s). By claiming your domain(s), you are marking an email domain for use only in your Webex organization. This prevents users with the claimed domain from existing in any other organization, including the free consumer organization.

To claim a domain for your Webex organization under **Settings > Domain** in Control Hub (https://admin.webex.com/), click the ellipsis (...) next to the domain you added and verified previously and select **Claim verified domain**. After claiming the domain, you will see "Claimed" next to your domain. Repeat this process for each domain owned by your organization.

Note: Once a domain is claimed, any administrator outside of your organization that attempts to add a user with this domain will receive an error message.

For more information on adding, verifying, and claiming domains refer to the Add, Verify, and Claim Domains article available at https://help.webex.com/en-us/nxz79m5/Add-Verify-and-Claim-Domains.

iv. Set up SSO.

While optional, the use of single sign-on (SSO) is recommended to provide the best end-user experience. The benefit of SSO is that a user can use a single common set of credentials for authenticating to any Webex services as well as other collaboration applications. With SSO, a user only must provide credentials a single time per session in order to be authorized for any services they are subscribed to.

To enable SSO for the Webex organization, from Control Hub navigate to **Settings** and scroll down to **Authentication**, click **Modify** and then, select the **Integrate a 3rd-party identity provider. (Advanced)**. Next, click the **Download Metadata File** button to download the file for import to your Identity Provider (IdP).

At this point you will need to configure your IdP as appropriate and import the metadata file you downloaded from Control Hub. Then, download or export the metadata file from your IdP.

Return to Control Hub and on the Import IdP Metadata screen, drag and drop the IdP metadata file or navigate to the file using the file browser. Next, under

Signing of Metadata (Advanced) select **Require certificate signed by a certificate authority in Metadata (more secure)** (unless IdP certificate is not signed by a CA, in which case you can select the less secure **Allow self-signed certificate in Metadata** option).

Finally, test the SSO setup by clicking the **Test SSO Connection** button. When prompted enter valid SSO credentials to confirm SSO is working properly. Assuming the test is successful, select **This test was successful. Enable Single Sign-On** and click Next to complete the SSO configuration.

For more information on tested IdPs and enabling SSO for your Webex org refer to the *Single Sign-On Integration in Control Hub* article available at https://help.webex.com/en-us/lfu88u/Single-Sign-On-Integration-in-Control-Hub.

v. Suppress automated user email invitation.

You should prevent automated email invitations to users when assigning licenses to users in your organization in order to activate users without interaction. These email invitations are not necessary and can cause confusion. These automated emails provide an initial password (not required with SSO), requests user validate their activation (not required with verified domain), and requests user provide additional user account details (not required with Directory Connector LDAP integration).

To prevent these automated invitations, from Control Hub navigate to **Settings**, scroll to **Enroll,** and toggle the **Suppress Admin Invite Emails** setting to on and click **Save**.

Note: This setting toggle may only be turned on when SSO is enabled.

vi. Determine license assignment method.

Before proceeding, it is important to decide on the method of license assignment you will use for the deployment. Assigning Webex Calling licenses is a manual process and must be done after users are available in the Webex organizations identity store. However, the simplest method for assigning other licenses (for example, meeting and messaging) is to automatically assign licenses using the Auto-Assign template for all users synced to the organization from Directory Connector.

To configure automatic license assignment for synced users, in Control Hub navigate to **Users** and select the **Manage Users** button. On the Manage Users page click **Set up Auto-Assign Template** link in the Licenses section. Next, select the desired message, meeting, and/or hybrid services licenses you wish to automatically assign to synced users. Review the license summary and then click **Save** to save the template.

Note: One point of consideration when using Auto-Assign template to assign licenses automatically to synced users is the potential for license starvation when importing users from Directory Connector. With this method, all users that are synced from Directory Connector will be automatically consume the licenses assigned by the template, even if they are not actively using Webex services. Alternatively, users can be synced from Directory Connector without using the Auto-Assign template and licenses can be assigned later using the CSV method for bulk user updates.

4. Directory integration

The first step for transitioning users to Webex Calling is importing users from the on-premises directory to the cloud identity store for the Webex Calling organization. In addition to initially importing users to the cloud organization, regular synchronization of user information from the corporate directory to the cloud common identity store is imperative to ensure identity information is consistent in both places.

The preferred method for importing and synchronizing users between the on-premises corporate directory and the identity store in Webex is to use hybrid directory service with Cisco Directory Connector. Directory Connector running on a Windows domain server retrieves user information from the corporate Active Directory and synchronizes this information to the cloud identity store using REST-based APIs. The administrator can determine which users are synchronized and what attributes are mapped between the on-premises and cloud directories based on sync agreement configuration. This synchronization is performed at regular intervals to ensure the cloud common identity store is up to date with any changes to the corporate Active Directory environment.

Deploy and Configure Directory Connector

Begin by downloading the Directory Connector software from Control Hub (https://admin.webex.com/). Navigate to **Users**, click **Manage Users**, click **Enable Directory Synchronization**, and then choose **Next**.

Next, click the **Download and Install** link to save the latest version of the connector installation .zip file to the target Windows domain server. Unzip the file and run the .msi setup file in the ***setup*** folder.

Note: For Windows server and other deployment requirements refer to the *Deployment Guide for Directory Connector* available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cloudCollaboration/spark/hybridservices/directoryconnector/cmgt_b_directory-connector-guide-admins.html

Note: A separate instance of Directory Connector is required for each domain in the organization. Further, for redundancy, two Directory Connectors (per domain) should be installed and configured on two separate Windows domain servers. If the primary Directory Connector server for a domain fails, the secondary Directory Connector can take over and maintain directory synchronization for that domain.

Once installation is complete, launch Cisco Directory Connector and sign into the Webex organization by entering the email address and password of the administrator account for the organization. Note that this is the same email address and password used to log into the Control Hub management portal. Click to confirm the Webex organization and domain.
Next, perform initial configuration of Directory Connector. From the Directory Connector dashboard click the **Configuration** tab. At a minimum configure the following:

- Under *General* specify the Connector Name, for example, DIRSYNC1 (this name will be displayed in Control Hub under Directory Synchronization settings). Add one or more Windows Domain Controllers based on your Windows environment (two domain controllers are recommended for redundancy purposes).
- Under *Object Selection*, at a minimum, tick **Object Type: Users** and configure LDAP filter(s) as required. Specify Base DNs to synchronize by clicking the **Select** button and specifying the container(s) in Active Directory you wish to synchronize.
- Under *User Attribute Mapping* configure any required Active Directory to Webex attribute name mappings by selecting options from the Active Directory attribute drop-down lists. At a minimum, ensure that the Active Directory attribute name **mail** is mapped to the required Webex attribute name

uid. The mail attribute plays a key role in Webex because it uniquely identifies the user.

On the redundant Cisco Directory Connector, configure the same settings outlined above, but use a unique name for the Connector Name setting (for example, DIRSYNC2).

Once the base configuration is complete, click the **Dashboard** tab and then click *Sync Dry Run* and click **OK** to confirm. A dry run validates your configuration and ensures the expected user accounts will be synchronized prior to performing a full synchronization. Review the dry run report to check for user or other sync anomalies (especially mismatched users). Click **Done** to close the dry run report.

If no issues are found, next, perform a full sync by navigating to **Actions > Sync Now > Full**. Click **Yes** to confirm the request.

After the first full directory sync has completed, you should configure the synchronization schedule to ensure any changes to Active Directory (additions, modifications, and/or deletions) are reflected in the Webex organization.
From the Directory Connector dashboard click the Configuration tab and under Schedule specify the following:

- The *Incremental Sync Interval* in minutes (for example, 30 minutes) which determines how often an incremental sync is performed. An incremental sync picks up user account adds, changes, and deletions in Active Directory.
- Tick **Enable Full Sync Schedule** and select the time and day(s) of week to perform a periodic full synchronization (for example, 11:30 PM on F(riday)). A full sync picks up user avatar, attribute mapping, base DN, and LDAP filter addition/changes as well as adds, changes, and deletions of user accounts.
- Specify the *Failover Interval* in minutes (for example, 60 minutes) before the secondary Directory Connector becomes primary and takes over incremental and full synchronization. This setting applies for high availability deployments with more than one Directory Connector (recommended).

For more details on deploying and configuring Directory Connector refer to the *Deployment Guide for Cisco Directory Connector* available at https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cloudCollaboration/spark/hybridservices/directoryconnector/cmgt_b_directory-connector-guide-admins.html.

Alternatively, rather than using Directory Connector, users may be manually added to the Webex organization common identity store individually or in bulk using CSV files. This is a manual process adding administrative overhead but may make sense for small deployments.

5. Setup and verify Local Gateway

Prior to setting up the Local Gateway for Webex Calling, ensure that Webex Calling has been licensed and enabled for the organization, and the Local Gateway has been onboarded within the Control Hub. For more details, refer to the Webex Calling deployment guide.

Baseline Local Gateway platform configuration must be configured according to your organization's policies and procedures and should include:

- Network Time Protocol (NTP) server access for time synchronization.
- Access Control Lists.
- Layer 3 interface(s) with valid and routable IP addresses assigned.
- IOS-XE security changes configuration requirements (master password and AES encryption).
- Enable passwords.
- IP Name Server to enable DNS lookup and ensure it is reachable by pinging it.

Then apply the following Local Gateway specific configuration:

- Enable TLS 1.2 exclusivity and a default dummy trustpoint.
- Update the Local Gateway trustpool by downloading the latest **"Cisco Trusted Core Root Bundle"** from http://www.cisco.com/security/pki/.
- Map parameters obtained from the Control Hub onboarding to IOS-XE CLI including global *voice service voip* configuration.
- Configure appropriate dial-peers for call routing.

Once the Local Gateway registers to the Webex Calling Access SBCs successfully, it will show up as Online within Webex Control Hub. Additionally, registration status can be verified using the **show sip-ua register status** command as shown below with the **reg** (registration) value showing **"yes"**.

```
LocalGateway# show sip-ua register status

Tenant:  200
-------------------- Registrar-Index 1 ---------------------
Line                              peer      expires(sec) reg survival  P-Associ-URI
================================= ========= ============ === ========  ============
LocalGW0135_LGU                   -1         56          yes normal

LocalGateway#
```

6. Configure call routing

During the transition to allow for coexistence of devices registered on Unified CM and on Webex Calling the enterprise dial plan on Unified CM needs to be changed to that at least the following requirements can be met:

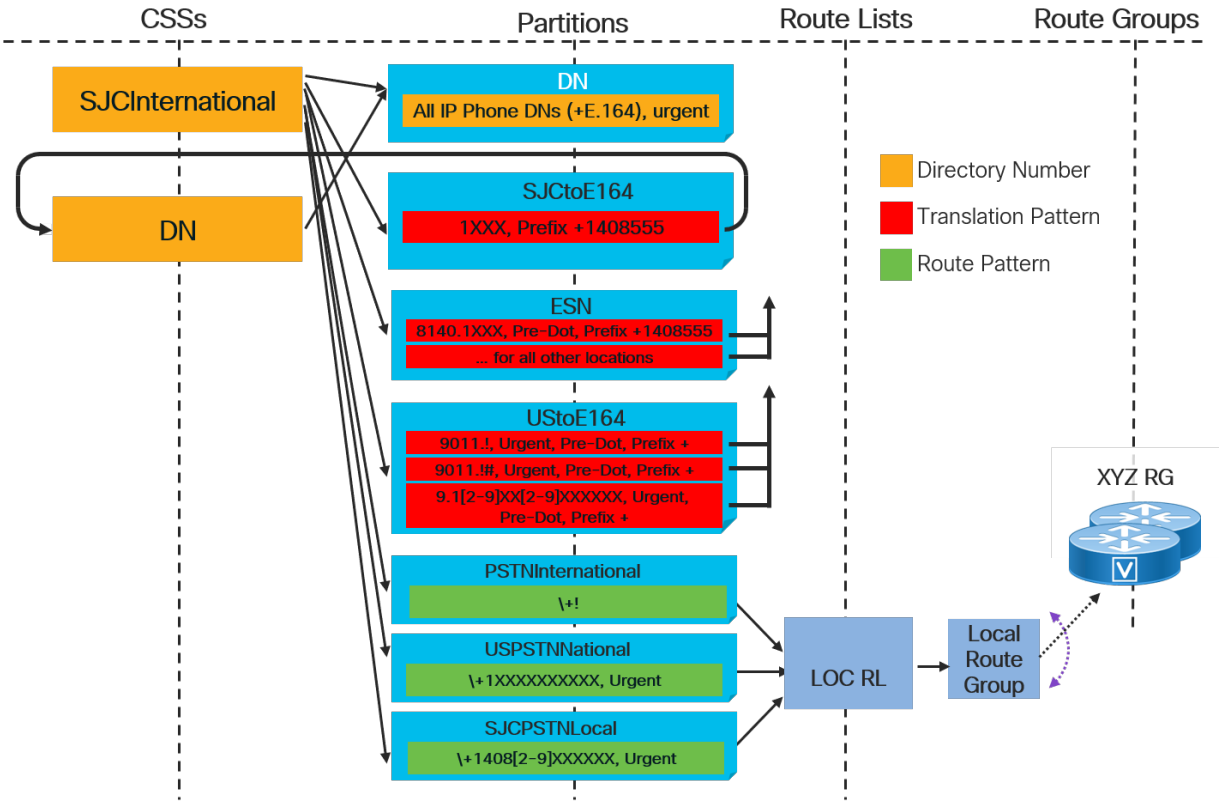
- +E.164 dialing from Unified CM to Webex Calling.
- Extension dialing from Unified CM to Webex Calling (intra-site but also inter-site if the extension ranges are unique).
- Abbreviated inter-site dialing from Unified CM to Webex Calling.
- Forced on-net dialing from Unified CM to Webex Calling.
- Call-back from missed calls directory to destinations on Webex Calling.
- PSTN calls from Webex Calling to PSTN.
- Forced on-net from Webex Calling to Unified CM.
- Extension dialing from Webex Calling to Unified CM (inter-site).

If any of the above are not supported dialing habits prior to the transition, for example no abbreviated inter-site dialing habit exists, then they don't necessarily need to be introduced during the transition.

Figure 10 shows the best practice dial plan approach as described in the "Preferred Architecture for Cisco Collaboration 12.x Enterprise On-Premises Deployments, CVD". Key characteristics of this approach include:

- Single partition for +E.164 directory numbers.
- Core routing based on +E.164 route patterns.
- Normalization of all dialing habits to +E.164 using translation patterns.
- Use of translation pattern calling search space inheritance (option "Use Originator's Calling Search Space" set on translation patterns).

Figure 10. *Best Practices Dial Plan*



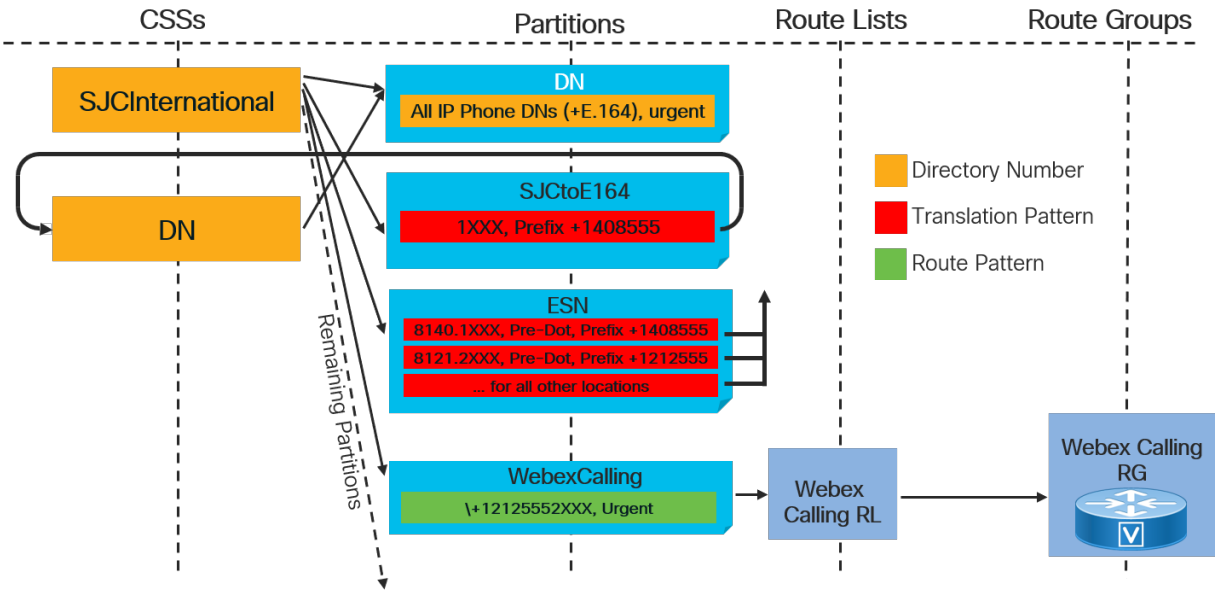
For example, PSTN dialing (9+1+10D) from a device in SJC provisioned with line calling search space “SJCInternational” will first get matched by the “9.1[2-9]XX[2-9]XXXXXX” translation pattern which normalizes the called party number to +E.164. The secondary lookup then uses the same calling search space “SJCInternational” again (calling search space inheritance) and the +E.164-digit string will either get matched by a +E.164 directory number in the “DN” partition or by one of the PSTN route patterns in the “USPSTNNational” or “SJCPSTNLocal” partition. Abbreviated intra-site and inter-site dialing habits are implemented by the translations in the “ESN” and “SJCtoE164” partition. While the “ESN” partition is a global partition (accessible for phones in all locations) the “SJCTOE164” partition is only accessible for users in location SJC. This is assuming overlapping extension ranges.

The first step to enable calling from Unified CM to Webex Calling is to make sure that +E.164 destinations get routed accordingly. This can be achieved by adding a “WebexCalling” partition to the dial plan, adding +E.164 route patterns for all Webex Calling destinations to that partition, and finally adding the “WebexCalling”

partition to all calling search spaces representing classes of service which need to be able to reach Webex Calling. Creating a dedicated “WebexCalling” partition is required to enable creation of a differentiated class of service for calls originating from Webex Calling. To avoid call loops the inbound calling search space on the trunk from the Local Gateway should not have access to the “WebexCalling” partition.

As shown in Figure 11 to enable routing from Unified CM to Webex Calling for a location with +E.164 DID range +1 221 555 2XXX and site code 212 an urgent route pattern matching this +E.164 range needs to be added to the “WebexCalling” partition.

Figure 11. *+E.164 Routing to Webex Calling*



If no site-specific Local Gateway selection is required, then instead of using a route list with a Webex Calling Local Route Group as the destination for route patterns pointing to Webex Calling a single route group can be provisioned with the Local Gateway as the only member and then the Webex Calling route patterns point to a single Webex Calling route list with this one route group as only entry.

To enable inter-site abbreviated dialing to the Webex Calling site the required dialing normalization translation pattern “8121.2XXX” is added to the already existing ESN partition. This is the same dialing normalization translation pattern which also needs to be provisioned for a site with this site code on Unified CM: for

sites to be transitioned to Webex Calling this enterprise abbreviated dialing normalization patterns already exists and does not need to be provisioned to transition that Unified CM site to a Webex Calling location.
With these dial plan changes calls to the Webex Calling location can be placed not only by dialing abbreviated inter-site and +E.164. Also, international and national PSTN dialing are possible because these dialing habits are first normalized to +E.164 via the already existing dialing normalization translation patterns and then get routed to Webex Calling by matching the +E.164 route pattern in the "WebexCalling" partition.

The +E.164 route pattern matching on a the Webex Calling location's DID range can be provisioned while all DIDs are still hosted on Unified CM. The best match pattern matching algorithm of Unified CM makes sure that when a number hosted on Unified CM is dialed then the +E.164 directory number provisioned on Unified CM is a better match than the wildcarded +E.164 route pattern pointing to Webex Calling so that the calls get extended to a line on Unified CM and not sent to Webex Calling.

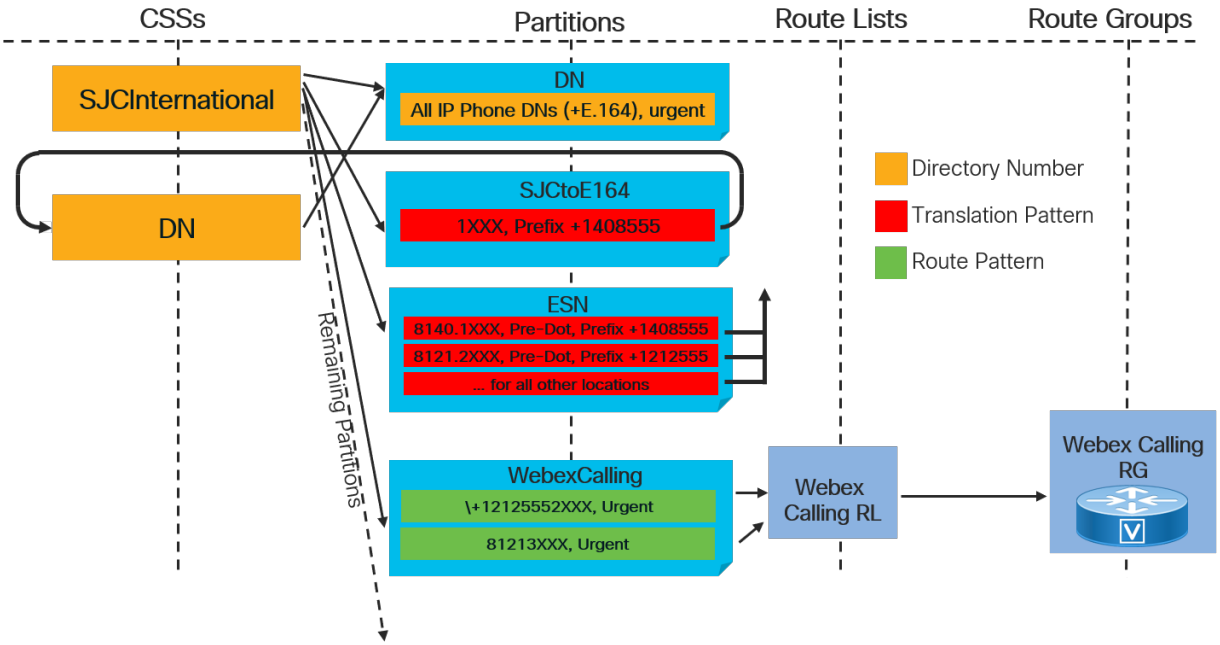
The Preferred Architecture for Cisco Collaboration 12.x Enterprise On-Premises Deployments recommends a dedicated enterprise number range for users and devices without a DID. Directory numbers for these users and phones are provisioned in the regular "DN" partition using the full ESN format (steering digit followed by site code and extension). Table 11 shows an example of three sites with dedicated enterprise number ranges for users without a DID.

Table 11. *ESN Ranges for DIDs and Non-DIDs*

Site	+E.164 Range	Site Code	ESN Range for DIDs	ESN Range for Non-DIDs
SJC	+1 408 555 4XXX	140	8 140 4XXX	8 140 5XXX
RCD	+1 972 555 5XXX	197	8 197 5XXX	8 197 6XXX
NYC	+1 212 555 2XXX	121	8 121 2XXX	8 121 3XXX

To enable abbreviated inter-site dialing to these non-DID destinations on Webex Calling route patterns matching on the non-DID number ranges need to be provisioned in the "WebexCalling" partition. This is shown in Figure 12.

Figure 12. *Abbreviated inter-site route pattern for non-DID destinations*



Equivalent to the +E.164 route pattern again the best match routing logic of Unified CM makes sure that directory numbers using the enterprise number format are a better match than the enterprise number route pattern in the “WebexCalling” partition so that only enterprise numbers which don’t exist on Unified CM get matched and sent over to Webex Calling.

Preparation

Prior to transitioning the first locations and users to Webex Calling the Local Gateway configuration and Unified CM configuration needs to be completed as described in these documents:

- Configure Local Gateway on IOS-XE for Webex Calling:

  https://help.webex.com/article/jr1i3r

- Configure Unified CM for Webex Calling:

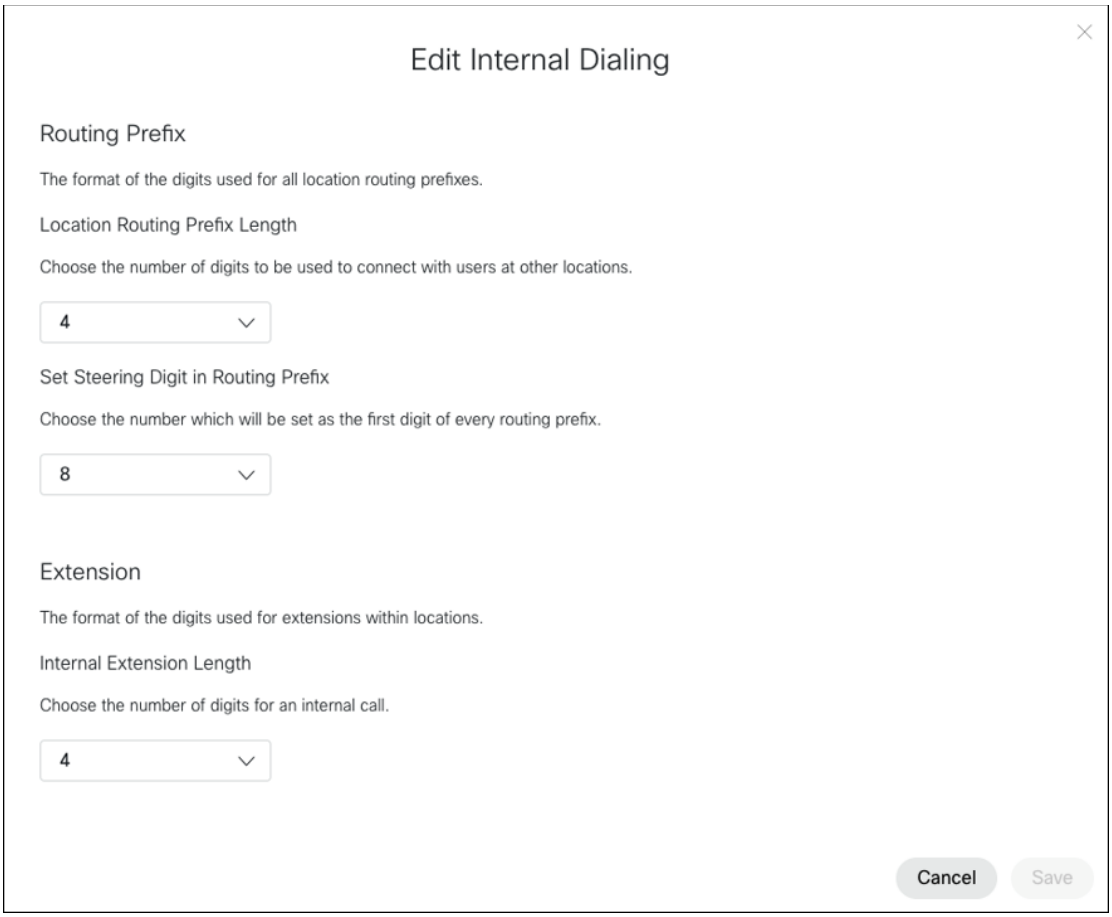
  https://help.webex.com/article/nqqzbk7

Also, in the Webex Calling administration make sure to configure the following internal dialing settings:

- Location Routing Prefix Length.
- Set Steering Digit in Routing Prefix.
- Internal Extension Length.

Figure 13 shows an example with steering digit “8”, three-digit site codes (the steering digit is counted as part of the routing prefix length), and four-digit extensions. The total length of enterprise number in this case is 8.

Figure 13. *Internal Dialing Settings*



Transitioning a Location

To prepare for the transition of users from Unified CM to a Webex Calling location these configuration steps need to be completed:

- Create location in Webex Calling with the setting collected based on the information in Table 6 (see step 5. Inventory existing locations/sites in the Pre-Transition Steps section).
- Define site code and PSTN access code for location.
- Add and activate phone numbers in Control Hub. Adding phone number in Control Hub does not change Webex Calling call routing. Phone numbers become active as soon as the number is assigned to user and a device gets provisioned. All DIDs of users in the location to transition need to be added.
- Provision the site-specific route patterns in the "WebexCalling" partition: +E.164 and (if needed) enterprise numbers.

7. User provisioning for Webex Calling

Because all users already exist in Control Hub (either through LDAP directory integration with Directory Connector or manual individual or bulk provisioning), the next step is to use bulk update (recommended) to enable appropriate users for Webex Calling, assign them to a location, provision their phones and assign phone numbers and extensions. The CSV template for this update can be downloaded from Control Hub by clicking **Manage Users** and selecting the *CSV Add or Modify Users* option. To avoid errors, you can also export all users, filter out the users to modify and then update the settings only for a selected set of users.

Table 12 provides an overview of the CSV file columns relevant for the user migration and the required settings. Quotes are used to indicate literal values; insert the values without the quotes.

Table 12. *User migration CSV settings*

Column	Setting
Extension	extension
Phone Number	DID (if available)
Caller ID Number	Caller ID override, if empty for users with DID the DID is used and for users without DID the locations' main number is used
Caller ID First Name	Caller ID override; can be empty
Caller ID Last Name	Caller ID override; can be empty

Location	Name of location
Jabber with Webex Teams	“FALSE”
Jabber Calling	“FALSE”
Calling Behavior	“NATIVE_WEBEX_TEAMS_CALLING” or “USE_ORG_SETTINGS” if native calling is configured at the organization level
Webex Calling VAR Enterprise	“TRUE”

To enable Webex Calling supplementary user and system features and services, navigate to Control Hub to update users individually or in bulk (Bulk Edit Users) for features like call forwarding, voicemail, and so on. Further, enable system features like auto attendants, hunt groups, call queuing, and so on.

Once users provisioning is complete, the next step is to provision Webex Calling devices and migrate existing phones as described in the next step.

8. Phone migration and provisioning for Webex Calling

Phones that are currently registered to Unified CM will need to be migrated to Webex Calling as part of the cloud transition. To make the migration as simple as possible with minimal chance for failure, Cisco recommends migrating physical sites or departments at the same time.

In order to become Webex Calling cloud registered devices, the migration process will use the cloud-based service to deliver the transitional and MPP firmware loads to complete the migration. Since the transitional firmware loads are phone model specific, the upgrade process will use common device selection criteria to make sure the correct phones are migrated to the Webex Calling platform.

The process to migrate phones to Webex Calling can be summarized in the following 6 steps:

i. Verify phone model support and firmware version.
ii. Upload migration licenses to upgrade.cisco.com.
iii. Provision phones for Webex Calling.
iv. Select phones to migrate.
v. Apply transitional firmware load and MPP firmware load.
vi. Update dial plan to complete migration.

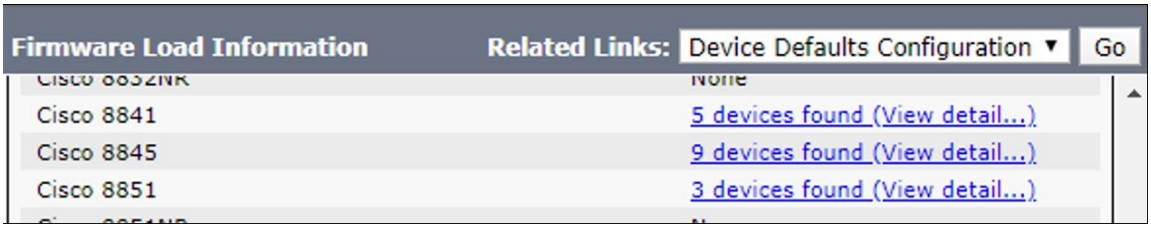
Each step is explained in detail below.

i. Verify phone model support and firmware version.

The phone migration process is only supported on the 7800 and 8800 phones (refer to Convert between Enterprise Firmware and Multiplatform Firmware for Cisco IP Phone 7800 and 8800 Series Guide to verify supported phone models and versions). Any phone device that does not meet the minimum model version will need to be replaced with an MPP phone.
Additionally, any supported phone model must be on firmware version 12.5.1SR2 or later. If any phone is on a previous version, then the phone must be updated to the latest Unified CM firmware prior to applying the transitional firmware to the phone.

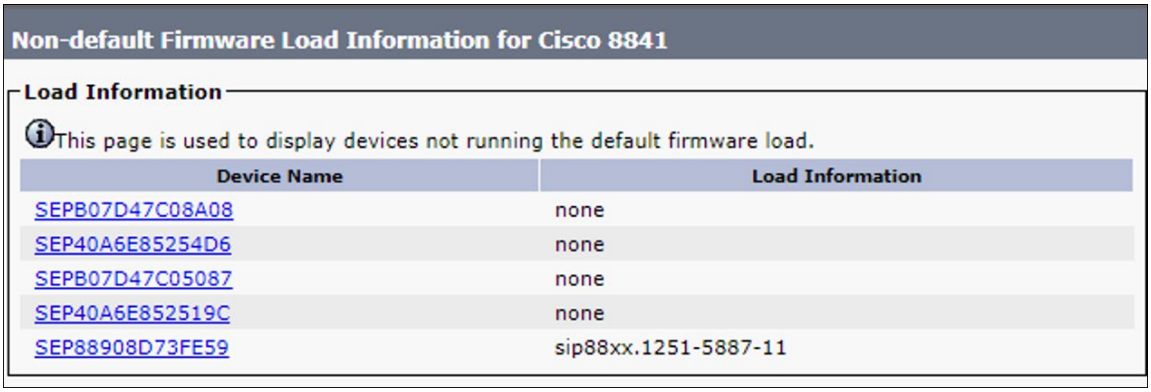
To check to make sure that no phone has a different load than the default firmware load, in Unified CM, navigate to **Device > Device Settings > Firmware Load Information** to load information on firmware loads for the devices configured on the system as shown in Figure 14.

Figure 14. *Firmware Load Information for Existing Devices*



Selecting one of the hyperlinks on the Firmware Load Information page will show you which phones may not be running the current default load. As shown in Figure 15, any phone that has a name in the 'Load Information' field indicates it is not using the default firmware.

Figure 15. *Non-Default Firmware Load Information*



Clicking through on any phone in the list will allow direct access to any device that needs to be upgraded to the model default firmware version.
The migration process should not continue until all phones to be migrated are on version 12.5.1SR2 or later.

Note: This procedure was verified using 12.7.1 as the base firmware.

ii. Upload migration licenses to upgrade.cisco.com.

Refer to Appendix A: Flex plan – Ordering and Appendix B: How to generate licenses for firmware conversion of the *Convert between Enterprise Firmware and Multiplatform Firmware for Cisco IP Phone 7800 and 8800 Series Guide* available at https://www.cisco.com/c/en/us/products/collateral/collaboration-endpoints/unified-ip-phone-7800-series/guide-c07-742786.html.

After ordering and downloading the licenses for migrating firmware, navigate to https://upgrade.cisco.com/e2m_converter to upload your migration licenses to the Cisco Cloud Upgrader service. Cisco Cloud Upgrader is a service that allows customers to easily upgrade/migrate the software on Cisco IP Phones so they can connect to Webex Calling.

Once the licenses have been uploaded, you will need to provision the phones in Webex Calling.

iii. Provision phones for Webex Calling.

The recommended approach for provisioning phones for Webex Calling is to use the bulk operation as described in the *Configure and Manage Webex Calling Devices* article available at: https://help.webex.com/en-us/n9r1aac/Configure-and-Manage-Webex-Calling-Devices#id_118912.

Table 13 provides an overview of the CSV file columns for device provisioning and the required settings. Quotes are used to indicate literal values; insert the values without the quotes.

Table 13. *Bulk device operation settings*

Column	Setting
Username	Email address of user this device is associated with
Type	"USER"
Directory Number	Leave empty; inherited from the linked user
Direct Line	Leave empty; inherited from the linked user
Device Type	"IP"
Model	Phone model. For example, "Cisco 8865"
MAC ADDRESS	MAC address of the phone; 12 characters, no separators. For example, "571432DDDE65"
Location	Leave empty; inherited from the linked user

Bulk provisioning should also be performed as described above for any new phones added as part of this transition. This can be done now or later.

Note: Make sure to complete bulk provisioning for all phones to be migrated before initiating firmware migration in the next step.

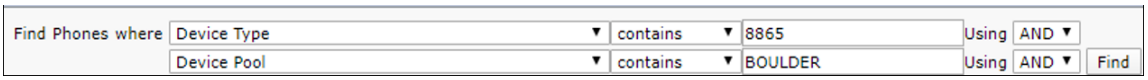
iv. Select phones to migrate.

Cisco recommends migrating groups of phones at the same time. A 'group' as used in this document will be any set of phones that have a common characteristic that can be used to select the phones to be migrated. This may

include configurations related to Device Pool, Physical Location or Description. For smaller installations, the grouping could be phone model type.

In order to select a group of phones to migrate, navigate to **Bulk Administration > Phones > Update Phones > Query**. The first search criteria should be phone model and the second should be the grouping criteria. Device pool is a common setting that can associate a phone to a location. As shown in the example in Figure 16, the search for device type and grouping criteria is for all 8865 phones that are part of the Boulder, CO site (they have a device pool that contains 'BOULDER' in the name).

Figure 16. *Search by Device Type and Grouping Criteria of Device Pool*

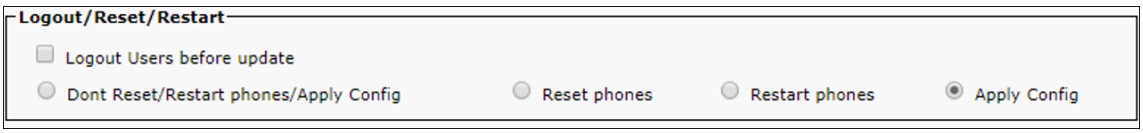


Once a group of phones has been returned, click **Next**.

v. Apply transitional firmware load followed by MPP firmware load.

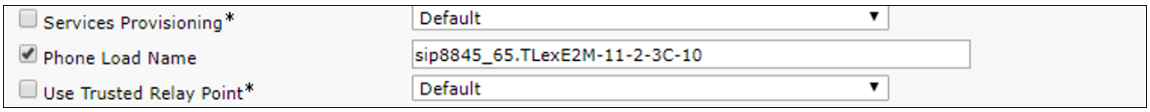
On the Update Phone page, as shown in Figure 17, select the **Apply Config** setting. This instructs the phones to download the new configuration information after executing the update.

Figure 17. *'Apply Config' Setting for BAT Execution*



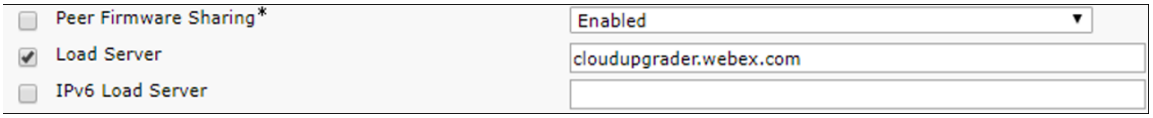
In the **Phone Load Name** field, specify the correct model specific firmware for the phone model that was originally selected. In this example, as shown in Figure 18, the selected 8845/65 phone models will load the configured transitional firmware load: *sip8845_65.TLexE2m-11-2-3C-10*.

Figure 18. *'Phone Load Name' Setting with Configured Transitional Firmware Load*



Next, the phone needs to be configured with the location to find the migration firmware files. The **Load Server** value must be set to use the cloud downloader. As shown in Figure 19, the cloud downloader URL is *cloudupgrader.webex.com*.

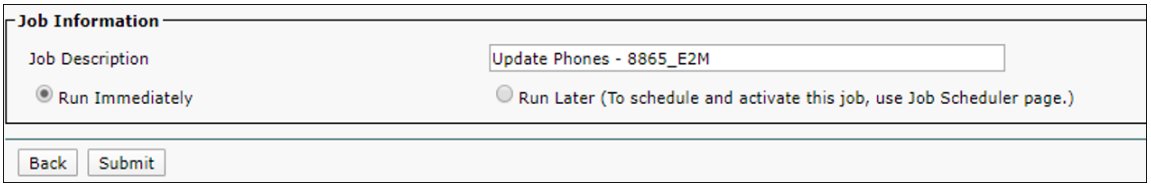
Figure 19. *'Load Server' Setting of cloudupgrader.webex.com*



Note: As shown in Figure 18 and Figure 19, the checkbox on the left side of the Update Phones page is required for the **Phone Load Name** and the **Load Server** entries. The Bulk Administration update job updates only the entries that have a check box selected. In this example, only these two values are getting updated on the selected phones.

At the bottom of the Updates Phones page, change the Job Description name (in this example 'Update Phones – 8865_E2M') and select the **Run Immediately** option in the Job Information section (see Figure 20). After clicking **Submit**, the batch update will occur immediately, and all selected phones will download the migration firmware from the cloud upgrade service.

Figure 20. *BAT Job Description and 'Run Immediately' Setting*



Once the phones complete the loading of the interim load, they will contact the cloud upgrader services, verify the device is entitled to migrate to Webex Calling and download the latest MPP firmware. All of this happens automatically once the interim load is successfully downloaded and the phone reboots.

Each phone model has a different interim load, so the above process must be repeated for each phone model type to be migrated.

Table 14 shows the interim firmware loads for all supported models.

Table 14. *Interim Firmware Loads for Eligible Webex Calling Devices*

Device Type	Phone Load
7832	sip7832.TLexE2M-11-2-3C-10
78xx	sip78xx.TLexE2M-11-2-3C-10
8832	sip8832.TLexE2M-11-2-3C-10
8845/65	sip8845_65.TLexE2M-11-2-3C-10
88xx	sip88xx.TLexE2M-11-2-3C-10

In cases where devices on Unified CM are running firmware versions earlier than 12.5.1SR2, the process above may be used to download the latest enterprise phone firmware needed in order to load the interim code. To use the cloud upgrader to get to the latest enterprise firmware, using the same selection criteria as above, follow the same directions above, but for the Phone Load Name setting specify a newer firmware load (minimum 12.5.1SR2). Table 15 below shows the firmware version used to validate this procedure.

Table 15. *Firmware Version Upgrade to Support Migration to Interim Load*

Device Type	Phone Load
7832	sip7832.12-7-1-0001-393
78xx	sip78xx.12-7-1-0001-393
8832	sip8832.12-7-1-0001-403
8845/65	sip8845_65.12-7-1-0001-393
88xx	sip88xx.12-7-1-0001-393

After the phones download the latest enterprise firmware, repeat the process above to convert the phones to cloud registered phones using the appropriate .TLexE2M-11-2-3C-10 interim build for each phone model transitioned.

vi. Update dial plan to complete migration.

Finally, to make sure that calls get routed from Unified CM to Webex, the existing devices and directory numbers need to be deleted from Unified CM.

To delete the phones from Unified CM follow the steps described in the *Phones Deletions* section of the *Bulk Administration Guide for Cisco Unified Communications Manager* available at https://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html.

If devices provisioned in Unified CM use a site-specific device pool using the "Device Pool" search allows to easily identify all devices provisioned in one site. If all devices are not migrated at the same time, the "Delete Phones Using Custom File" procedure can be used to delete phones based on device names, MAC addresses or directory numbers. A list of directory numbers should be readily available as the same directory numbers have been used above when provisioning the users for Webex Calling.

After deleting the devices from Unified CM, the now unassigned directory numbers need to be deleted using the "Delete Unassigned Directory Numbers" procedure in the *Phones Deletions* section of the *Bulk Administration Guide for Cisco Unified Communications Manager* referenced previously.

Routing from Unified CM to Webex Calling only becomes active after the directory numbers have been deleted because the directory numbers always are a better match than the wildcarded route patterns in the "WebexCalling" partition.

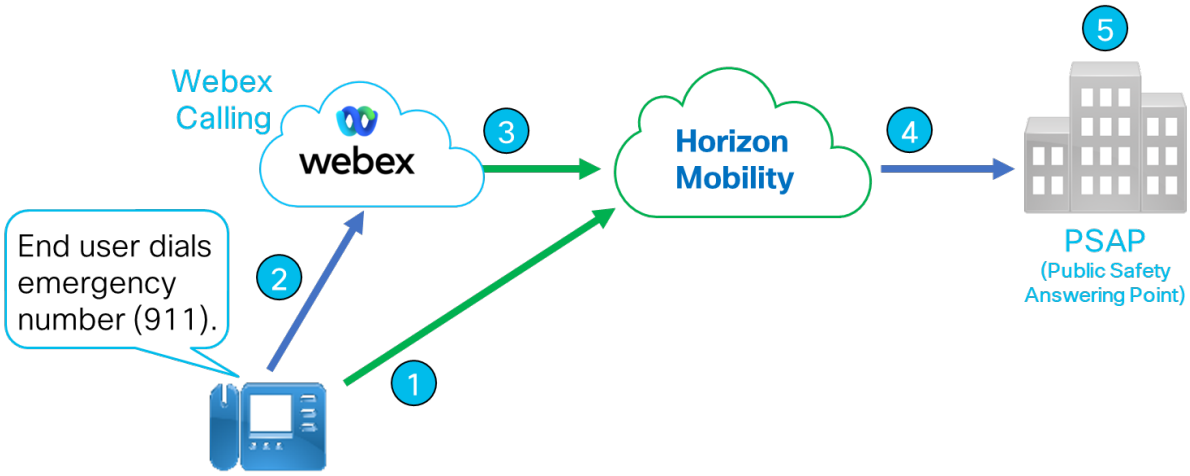
9. Enable emergency calling

Foundational to Webex is the ability to call emergency support numbers. Each country supported by Webex Calling defines the emergency numbers to enable basic emergency calling (for example, 911, 112, 999, and so on). The physical location of an emergency call is presented to the PSAP (Public Safety Access Point) as defined by the PSTN link carrier. It is a requirement of the PSTN carrier to correctly define and deliver the physical address to the PSAP for emergency calls.

For US and Canada based telephony deployments that must provide enhanced emergency calling solutions, Webex Calling may be augmented with Horizon Mobility from RedSky (https://www.redskye911.com/horizon-mobility-for-webex-calling). The Horizon Mobility solution meets all the regulatory requirements under the recently enacted Kari's Law and Ray Baum's Act legislation (compliance requirement anticipated in February 2021). The RedSky Horizon Mobility solution is a subscription model and aligns with the subscription levels of Webex Calling end users. RedSky's Horizon Mobility is a cloud-based solution and has no reliance on an on-premises application server managed by the customer (like Cisco Emergency Responder).

Horizon Mobility service relies on Webex Calling devices to provide network connection information to the cloud at boot time so the device can be associated to a physical location within the customer's deployment. Webex Calling MPP devices will send switch/switchport and/or the wireless access point BSSID along with IP address to the Horizon Mobility service via an "over the top" HTTPS service called HTTP-Enabled Location Delivery (HELD). As shown in Figure 21, step 1, after registering with Webex Calling, the endpoint will receive a URL to access the RedSky Horizon Mobility service as a configuration parameter. The endpoint will send a location update directly to the Horizon Mobility service with the infrastructure information noted above. The endpoint will establish a persistent connection to Webex Calling service through the Access SBC, while the connection Horizon Mobility service will only occur when the endpoint is restarted, moved, or the location information has timed out (every 24 hours).

Figure 21. *Cloud: Webex Calling Enhanced Emergency Calling with Horizon Mobility*



The physical locations within the customers network are configured and managed by the customer / partner in the Horizon Mobility portal. Any time a device sends a location update via the HTTPS service, the device's physical location will be updated to be associated with the new upstream switch, switch port, wireless access point, or IP Subnet.

When a user places an emergency call to 911 from a Webex Calling device (Figure 21, step 2), the call will be sent through a Webex Calling managed link to the RedSky Horizon Mobility service (step 3). Redsky Horizon Mobility service will then identify the location of the calling device and deliver the call to the appropriate PSAP (step4). The PSAP that receives the call will also get the current physical address of the calling device (step 5).

During the transition to Webex Calling the following is required to enable E911 with RedSky Horizon Mobility:

- Configure each physical dispatchable location in the Horizon Mobility Portal.
- Assign each network element (switch, switchport, wireless access point, and IP subnet or endpoint) to the physical dispatchable location.
- Reboot all phones at the physical location to trigger the location registration with Horizon Mobility using HELD.
- Test emergency calling using the 933-test dial string to confirm that the proper physical location of the calling device is processed.

10. Implement call recording

Call recording solutions provide a way to record audio and video calls that traverse various components in a unified communications and collaboration solution. These recordings can then be used by call centers and other enterprise operations for various purposes such as compliance, transcription and speech analysis, or for podcasting and blogging. Existing media forking call recording solutions for an on-premises deployment, for example, CUBE controlled recording, Unified CM network-based recording, and SPAN-based recording solutions are not supported with Webex Calling.

The call recording solution for Webex Calling is provided by the partner Dubber (https:///www.dubber.net). The Dubber solution enables recording of all calls (Internal and PSTN) placed/received on the Webex Calling platform for replay and management. Dubber integrates with the Webex calling platform in the cloud. Prior to transitioning to Webex Calling, ensure your call recording compliance and regulation is met by this architecture. For example, in certain countries, regulation requires the recorded media to stay within the geographic boundary of that country and that requirement may not be satisfied as Webex Calling platform and the Dubber data center integration only exists in certain countries.

The following steps are required to deploy call recording in Webex Calling, which should take place once the end users and devices have been migrated to the cloud:

i. Partner must first establish an agreement and partner account with Dubber.

ii. Partner then sells the service to the Webex Calling customer.

iii. Partner enables call recording for the Webex Calling customer in Webex Control Hub.

iv. The call recording license is assigned to an end user either by partner or customer administrator.

v. The call recording feature for an end user is enabled within the Webex Calling platform.

vi. Lastly, the partner creates the customer administrator and user in Dubber portal using data from the Webex Control Hub.

Dubber call recording feature options for Webex Calling are configurable on a per user basis by the customer administrator or partner and include the following:

- Call recording designation.

- Never: Do not record any calls.
- Always: Record all calls.
- Always with Pause/Resume: All calls are recorded, but the end user has the option to pause recording to protect personally identifiable information (band account numbers, credit cards, PIN, social security number, and so on).

- Call recording announcement.

  Play recording start/stop announcement: If selected, a system message is played to both parties in the language of the customer site.

  - When both parties are connected, the played message is " This call is being recorded."
  - If recording is paused, upon resumption the played message is " Your call recording service has been activated successfully. Thank you."
- Recording Reminder Tone: Option to play message during pause and resume.
- Repeat Tone Every: Option to play a tone periodically (with configurable between tones increments from 10 to 90 seconds).

Post-Transition Steps and Considerations

Once the transition from Unified CM on-premises calling to Webex Calling is complete, there are a few additional steps that should be considered:

1. Transition to Cloud Connected PSTN

Once all endpoints and users are migrated to cloud calling the single purpose of Unified CM is to act as transit between the PSTN gateways and Webex Calling via Local Gateway. Removing PSTN gateways, Unified CM, and Local Gateway from the picture by using Cloud Connected PSTN as PSTN access for all Webex Calling users has several benefits including cost reduction and improved reliability. To transition from Local Gateway based PSTN access to Cloud Connected PSTN follow these steps:

i. Cloud Connected PSTN provider selection.

   Refer to the list of Cloud Connect PSTN providers and select from the available provider(s) available for your organization's location.

ii. Cloud Connected PSTN validation.

Before switching PSTN access for Webex Calling locations to Cloud Connected PSTN, connectivity to PSTN via the selected Cloud Connected PSTN provider should be verified and validated. For this purpose, a test location needs to be provisioned in Webex Calling with some test users provisioned in that test location. PSTN access for this test location is then set to the Cloud Connected PSTN provider before validating PSTN connectivity using the test phones. Upon successful validation the test location can be deprovisioned.

iii. Number porting.

To prepare for the cut-over to Cloud Connected PSTN a port order for all numbers currently assigned to the PSTN trunk terminating on Unified CM needs to be placed. All numbers need to be ported to the Cloud Connected PSTN provider. To maintain inter-location reachability all numbers of all locations need to be ported at the same time.

iv. Switch to Cloud Connected PSTN.

At date of the cut over PSTN access for all locations in Webex Calling needs to be set to the Cloud Connected PSTN provider and inbound and outbound connectivity should be verified.

2. Update on-premises infrastructure

Once all users have been transitioned to Webex Calling and all endpoints have been transitioned to cloud registration (or have been decommissioned), update appropriate on-premises infrastructure now that cloud calling is in use. Updates to the infrastructure include:

- Remove on-premises call control and messaging DNS SRV records from the on-premises DNS server(s) including cisco_uds._tcp.<domain>, cup_login._tcp.<domain>. These SRV records are no longer required for client service discovery.
- Remove edge-related DNS SRV records from the public DNS system including collab_edge._tls.<domain>. These SRV records are no longer required for client service discovery of collaboration edge services.

- Update all relevant DHCP scopes to remove option 66 and option 150 TFTP/boot server addresses. These scopes are no longer required for endpoint call control configuration discovery and download.
- Update/remove appropriate dial-peers in Local Gateway/CUBE that route calls to and from Unified CM. These dial-peers are no longer required for on-premises call routing.
- Delete or remove all Unified CM and Expressway cluster node virtual machines and/or servers. Repurpose compute resources and hardware as needed. These resources are no longer needed for call control and edge services.
- Delete or remove all Unity Connection cluster node virtual machines and/or servers. Repurpose compute resources and hardware as needed. These resources are no longer needed for voicemail and unified messaging services.
- Clean-up: After migrating PSTN access to Cloud Connected PSTN Unified CM, PSTN trunks, PSTN gateways, and Local Gateway can be decommissioned.
- For any existing on-premises e911 solution, delete any locations or numbers that have migrated to Webex Calling and once full transition is complete, remove application virtual machines or servers. Repurpose compute resources and hardware as needed. These resources are no longer needed for emergency calling and location services.
- Update the physical dispatchable location and network element in Horizon Mobility whenever changes occur. Common activities that require updates are:
  - Network switch replacement.
  - Wireless access point replacement.
  - DHCP scope changes.
  - Physical changes inside the building (if resolving to cubical/office).
  - Physical office space expansion or contraction inside a building.

3. Leverage Webex Calling troubleshooting tools

Admin and users can always follow the Webex calling service's status on https://status.broadsoft.com/.

During transition, admin can utilize existing tools to diagnose and resolve issues. These include:

- Unified CM traces.

- IOS-XE debugs.
- Sniffer captures of network traffic.
- Syslog.

Administrators can also open a service request with Cisco Technical Assistance Center (TAC).

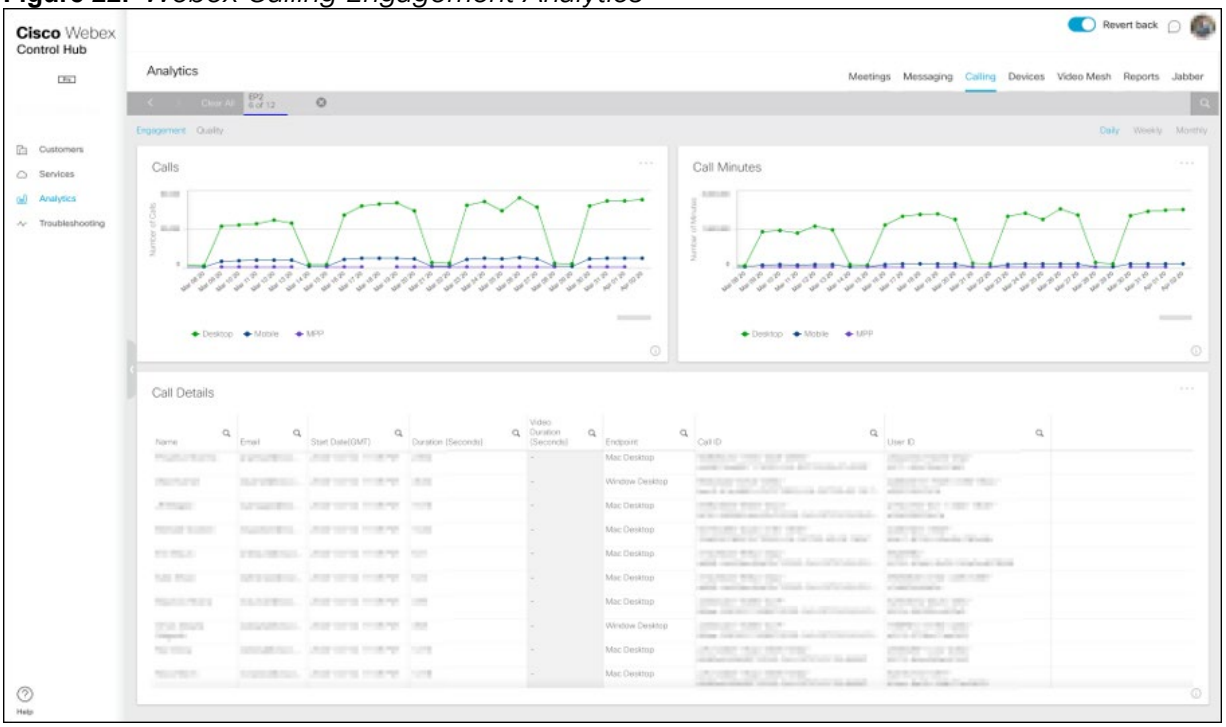
4. Utilize Webex Calling analytics

Webex Calling Analytics in Control Hub brings a new level of insight into your Webex Calling deployments. With the enablement of this feature, historical data of call usage and engagement is available in Control Hub, including media quality records. The Webex Calling analytics are presented in Control Hub under **Analytics > Calling**.

Engagement Analytics

Under **Analytics > Calling > Engagement** are historical engagement analytics for your Webex Calling service (see Figure 22).

Figure 22. *Webex Calling Engagement Analytics*



As shown in Figure 23, the Calls and Call Minutes graphs contain details of all the point-to-point calls made in the organization. The graphs are split into the following categories, depending on the client used for the call:

- Desktop –Webex App for Windows and Mac.
- Browser –Webex App for Web.
- Mobile –Webex App for iPhone, iPad and Android.
- Device –Webex Room Device or Webex Board.
- WXC Desktop –Webex Calling App for Windows and Mac.
- WXC Mobile –Webex Calling App for iPhone, iPad and Android.
- MPP –Webex Calling Multiple Platform Phone.

The Call Details table contains one entry for each call containing the following information:

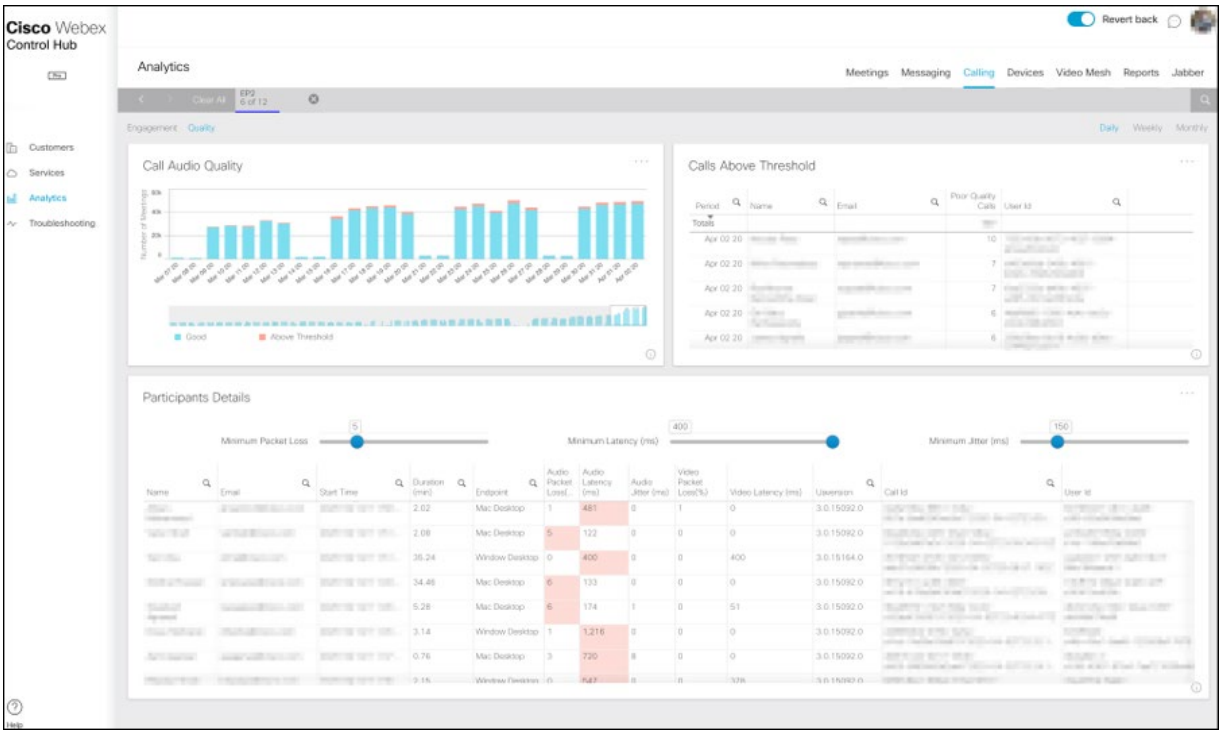
- Name – Username.
- Email – User email.
- Start Date – When the call was made in GMT.
- Duration – Duration of the call-in seconds.
- Endpoint – Type of device or client used. WXC means Webex Calling app, and MPP means Webex Calling Multiple Platform Phone.
- Uaversion – User agent version if reported by the endpoint.
- Call ID – A unique identifier for the call.
- User ID – A unique identifier for the user.

As an example, if Alice calls Bob and both devices are within the same organization, two records will be shown in this view, one for Alice and one for Bob.

Quality Analytics

The Quality Analytics tab allows you to view records for each call and use sliders to filter calls based on quality statistics (see Figure 23).

Figure 23. *Webex Calling Quality Analytics*



The analytic headings available for quality are:

- Name – Username.
- Email – User email.
- Start Date – When the call was made in GMT.
- Duration – Duration of the call-in seconds.
- Endpoint – Type of device or client used. WXC means Webex Calling app, and MPP means Webex Calling Multiple Platform Phone.
- Audio Packet Loss (%) – Packet Loss in percent as reported by the endpoint/soft client.
- Audio Latency (ms) – Audio latency reported by the endpoint.
- Audio Jitter (ms) – Audio jitter reported by the endpoint.
- Video Packet Loss (%) – If applicable, video packet loss reported by the endpoint.
- Uaversion – User agent version if reported by the endpoint.
- Call ID – A unique identifier for the call.

- User ID – A unique identifier for the user on the call.

All statistics are collected from the devices/endpoints directly, and so the statistics (packet loss, jitter, latency) reflect the experience of the call from the perspective of the user's endpoint.

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Glossary

The following tables provide a glossary of Unified CM concepts and constructs along with a definition and information on potential alignment with Webex Calling.

The glossary is divided into the following sections:

Locations and Grouping

Unified CM Concept	Comment
Region	Regions are used to control bandwidth consumption for audio and video calls within a region and between regions. Regions are assigned to devices at the device pool level. In Webex Calling devices are grouped by location and the same codec selection policy is used for calls within a location and between locations.
Call Manager Group	Call Manager Groups are used to control device registration and balance the load over all call processing subscriber nodes of a Unified CM cluster. With a cloud-based call control there is no need for the administrator to define registration policies to manage load and performance.
Device Pool	Device pools allow to group devices of similar characteristics and to apply common configurations like region, location, calling party transformations to devices at the device pool level. In Webex Calling there is no concept of device grouping for administration purposes. All settings need to be applied to the devices individually.

Unified CM Concept	Comment
Common Device Configuration	Using Common Device Configurations in Unified CM allows to create sets of device settings which can then be applied to individual devices together. Settings include softkey template, music on hold, locale, and others. Common Device Configurations get applies to devices at the device level (not at the device pool level). There is no equivalence in Webex Calling. In Webex Calling all device configuration is applied at the device level.
Locations	Locations are used in Unified CM to group together devices residing in the same network location. The location concept in Unified CM is purely used for Call Admission Control (CAC). Devices are assigned to locations by selecting a location at the device pool level. While Webex Calling does not have any CAC capabilities Webex Calling locations group together devices sharing the same geographical location (address), PSTN access, main number, site code, PSTN steering digit etc. Unified CM location assignment among other aspects can be used during a Unified CM to Webex Calling transition to identify devices in the same Webex Calling location.
Device Grouping	Multiple attributes can be used to group devices in Unified CM including: Device Pool, Device Mobility, Locations, DN ranges, Physical Location, Directory Information, Region, Geolocation, SRST reference, Logical Partitioning, MRGL, and others. Webex Calling has no concept of grouping devices other than associating devices with a Webex Calling location.

Unified CM Concept	Comment
Directory Number	Directory numbers represent dialable patterns assigned to endpoints in Unified CM. In Webex Calling endpoints inherit extension and number from the user the device is configured for.
Partition	Dialable patterns are grouped in partitions to form equivalence classes of dialable patterns to be used in calling search spaces to define classes of service and dialing habits in Unified CM. There is no equivalent concept in Webex Calling. Instead in Webex Calling DIDs, extensions, PSTN steering digit, inter-site dialing steering digit, and outbound dial plans are used as parameters to configure dialing habits. From a migration standpoint, partitions, calling search spaces, and patterns can be analyzed to derive this information from an existing Unified CM configuration.
Class of Service (CoS)	Unified CM and Webex Calling use fundamentally different concepts for Class of Service. While CoS in Unified CM is built based on calling search spaces, partitions, and dialable patterns, Webex Calling offers a fixed set of classes of service. Especially there is no (easy) way to build differentiated CoS based on dialed on-net destination.
Calling Search Space (CSS)	Calling search spaces are ordered list of partitions containing dialable patterns. A destination is reachable by a calling device if the dialed destination matches a pattern in any partition contained in the effective calling search of the calling device.

Unified CM Concept	Comment
Local Route Group	Local route groups in Unified CM are used to define site and call type specific egress gateway or trunk selection policy. In Webex Calling a single egress connection can be configured per location. This connection can either be a Cloud Connected PSTN (CCP) trunk or a trunk via a Cisco Unified Border Element (CUBE) acting as Local Gateway (LGW).
Route Group	Route Groups are used in Unified CM to group trunks and gateways into equivalence groups which are used to configure prioritizes egress gateway or trunk selection. In Webex Calling a single egress connection can be configured per location. This connection can either be a Cloud Connected PSTN (CCP) trunk or a trunk via a Cisco Unified Border Element (CUBE) acting as Local Gateway (LGW).
Route List	Route Lists in Unified CM are ordered lists of route groups. They are used to define egress gateway or trunk selection policies. In Webex Calling a single egress connection can be configured per location. This connection can either be a Cloud Connected PSTN (CCP) trunk or a trunk via a Cisco Unified Border Element (CUBE) acting as Local Gateway (LGW).
Route Filter	Route filters are used in combination with national numbering plans to define class of service based on number types defined by national numbering plans. In Webex Calling the class of service concept is different and there is no equivalent to route filters.

Unified CM Concept	Comment
Route Pattern	A route pattern comprises a string of digits (an address) and a set of associated digit manipulations that can be assigned to a route list or a gateway. Route patterns provide flexibility in network design. They work in conjunction with route filters and route lists to direct calls to specific devices and to include, exclude, or modify specific digit patterns. In Webex Calling with a Local Gateway configured as trunk all unknown destinations are sent to the Local Gateway.
Calling/Called Party Transformations	Calling and called party transformations can be applied at various points in Unified CM including implicitly as part of call routing on route and translation patterns, and explicitly using calling and called party transformation patterns in partitions addressed by calling and called party transformation calling search spaces. In Webex Calling called party transformations are limited to normalization of dial strings in line with the respective national numbering plan.

Unified CM Concept	Comment
National Numbering Plan	National Numbering Plans in Unified CM define the characteristics of country specific numbering plans including number types, digit discard instructions, and number formats. When defining the call routing in Unified CM the national numbering plan can be referenced which can simplify the configuration of the enterprise dial plan. In Webex Calling all country specific numbering plans of all supported countries are pre-provisioned and selected based on the county selection of the location of a calling device. There is no need to install country specific numbering plans in Webex Calling. Webex Calling allows to set the caller ID for each user to the direct line, the main number of the user's location, or to a different number from the caller's location.

Users

Unified CM Concept	Comment
Universal Line Template, Universal Device Template, User Profile	User profiles use settings from the following phone and phone line templates to build a profile for the end user: ▪ Universal Line Template—a collection of common phone line settings that are typically assigned to a directory number. Universal line templates allow to quickly configure phone lines for new directory numbers that get assigned to an end user. ▪ Universal Device Template—a collection of common device settings that are typically assigned to a phone or other device. Universal device templates allow to quickly configure new phones that get assigned to an end user. User profiles together with Universal Line and Device templates ease administrative tasks that relate to setting up users and devices and keep a range of device settings on one central, customizable interface. They are also used to determine the settings for users enabled for self-provisioning and are the basis for auto-provisioning of devices for users created via LDAP directory synchronization. Webex Calling does not offer mechanisms to group devices or to prepare templates to be used during user and device provisioning. Each user and device are provisioned individually.

Unified CM Concept	Comment
Feature Group Template	These templates are used to define settings to use when using the quick User/phone add functionality of Unified CM. Furthermore, a feature group template is also used to determine the settings for users and phones added through LDAP synchronization. Webex Calling does not to offer any functionality to pre-define user or device settings. Instead, users and devices are configured individually.
LDAP (System, Filter, Directory, Authentication)	Unified CM can be configured to automatically synchronize user data from an LDAPv3 directory to the Unified CM end user database. Also, password-based authentication can be delegated to LDAP based authentication. Feature Group Templates can be linked to LDAP synchronization agreements to define settings for users and phones automatically created through LDAP synchronization. Directory Connector can be used to link account creation in Webex Common Identity to Microsoft Active Directory. In addition, account creation can be linked to Okta or Azure AD; this does not require any on-premises infrastructure. Authentication can be delegated to enterprise identity management systems using SAML 2.0 compliant Single Sign On (SSO). Webex calling has no option to automatically configure calling related settings like extension, phone number, location, or calling entitlement for users synced from Microsoft Active Directory, Okta, or Azure AD.

Unified CM Concept	Comment
Self-Provisioning	Unified CM supports two ways of self-provisioning: ▪ Users dialing into an IVR from an auto-registered phone and then attributes of the phone are set based on user and device profiles. ▪ Activation code-based onboarding. The latter option of activation code device provisioning also exists in Webex Calling. The main difference is that in Webex Calling the device type is fixed and by entering an activation code a specific device is associated with the actual device.

Phones

Unified CM Concept	Comment
Phone Button Templates	Webex Calling currently does not have any ability to use templates when provisioning phones. All configuration must be applied to each phone individually.
User/Device/Association	In Webex Calling devices get associated to users during the provisioning process and the phone inherits the extension/DID from the user while in Unified CM the configuration is DN centric, DNs get associated to phones via line appearances, and then finally phones are owned by users. A user in Unified CM does not really have a routable number. The information in the user record is mainly used for directory lookups.

Features / Functionality

Unified CM Concept	Comment
Call Admission Control (CAC)	Unified CM offers the option to implement topology aware call admission control based on locations, links interconnecting locations, and available audio and video bandwidth on these links. Calls get rejected if the bandwidth manager identifies lack of available bandwidth on any of the links based on location of calling and called device. Webex Calling is not aware of the network topology the solution is deployed on and thus is not able to offer call admission control. Instead, customers are encouraged to overprovision links and continuously monitor bandwidth utilization.
Automated Alternate Routing (AAR)	Automated Alternate Routing allows to reroute calls to the PSTN or a different network when the call is blocked by Unified CM due to lack of bandwidth as determined by call admission control. The concept of AAR does not apply to Webex Calling because Webex Calling does not offer any bandwidth utilization controls.
Access List	Cisco Unified Mobility allows users to manage business calls using a single phone number and pick up in-progress calls on the desktop phone and cellular phone. Access lists determine the phone numbers that are explicitly allowed or blocked for in-progress call transfers. These access lists are used in the remote destination configuration. With Webex Calling Call Forwarding Selective Users can forward calls at specific times from specific callers.

Unified CM Concept	Comment
Time of day routing	In Unified CM a time schedule can be assigned to a partition which contains dialable patterns. Outside of the schedule, the associated partitions are not considered for call routing. This enables time dependent classes of service. In Webex Calling time schedules can be used in combination with Auto Attendant, call queues, Executive and Executive Assistant, selective call acceptance, and selective call rejection
Dial Rules	Unified CM supports the following types of dial rules: ▪ Application Dial Rules: The administrator uses application dial rules to add and sort the priority of dialing rules for applications such as Cisco web dialer and Cisco Unified Communications Manager Assistant. ▪ Directory Lookup Dial Rules: The administrator uses directory lookup dial rules to transform caller identification numbers and perform a directory search from the assistant console in application such as Cisco Unified Communications Manager Assistant. ▪ SIP Dial Rules: The administrator uses SIP dial rules to perform system digit analysis and routing. The administrator configures SIP dial rules and adds the SIP dial rule to the Cisco Unified IP Phone before the call processing takes place. Webex Calling does not have an equivalent to this. Phones registering to Webex Calling download a core, simple dial map based on the national numbering plan. During digit collection on the phone guard timers are used to terminate digit collection in case of numeric overlaps.

Unified CM Concept	Comment
URI Dialing	In Unified CM each directory number can have multiple associated alphanumeric URIs and both, the alphanumeric URIs and the directory number are dialable destinations. The primary URI also serves as caller ID in conjunction with a numeric identity derived from the directory number value. Webex Calling is a numeric only call control and cannot handle URIs as addresses for users. The only way to dial a URI from a phone is to assign a URI to a speed dial and then dial the URI via a speed dial.
Global Dial Plan Replication (GDPR)	In multi-cluster Unified CM deployments Global Dial Plan Replication (GDPR) allows to replicate dial plan information between clusters. This simplifies the call routing configuration for complex call routing topologies covering large deployments both in terms of number of users and geographic footprint. Webex Calling is a scalable solution with global footprint capable of supporting users at scale in various geographies.
Intercom	Intercom is a type of phone line that combines the functionality of a traditional line and a speed dial. With an intercom line, a user can call the intercom line of another user, which answers automatically to one-way audio whisper. The recipient can then acknowledge the whispered call and initiate a two-way intercom call. In Webex Calling the push-to-talk feature allows users to treat their desktop phones as either a one-way or two-way intercom. When enabled, the administrator can create an allowed or blocked user list.

Unified CM Concept	Comment
Client Matter Codes, Forced Authorization Codes	In Unified CM with client matter codes (CMCs) and forced authorization codes (FACs), you can effectively manage call access and accounting. CMCs assist with call accounting and billing for clients, and FACs regulate the types of calls that certain users can place. CMCs force the user to enter a code; this action specifies that the call relates to a specific client matter. You can assign client matter codes to customers, students, or other populations for call accounting and billing purposes. FACs force the user to enter a valid authorization code that is assigned at a certain access level before the call is completed. Webex Calling has no replacement for CMCs. FAC means Feature Access Code in Webex Calling that is different from Forced Authorization Code in CUCM. Webex Calling supports Authorization Codes to bypass Calling Restrictions.
Presence Subscription	In Unified CM the SUBSCRIBE Calling Search Space determines how Cisco Unified CM routes BLF presence requests that come from the trunk or the phone. The SUBSCRIBE calling search space, which is associated with a watcher, specifies the list of partitions to search for routing information to a presence entity for BLF presence requests. In Webex Calling to prevent someone from monitoring a user's line status selective line status sharing can be configured in the user's advanced calling settings.

Unified CM Concept	Comment
Media Resource Group, Media Resource Group List	A Media Resource Group List (MRGL) provides a prioritized grouping of Media Resource Groups (MRG). Media resource, such as an MOH server, conference bridges and transcoders, from among the available media resources based on the priority order defined in an MRGL. In Webex Calling management of media resources is part of the service provided in the cloud. There is no need for the administrator to configure any selection order
Busy Lamp Field (BLF), BLF presence group	The Busy Lamp Field (BLF) presence feature allows a user who is a watcher to monitor the real-time status of another user from the device of the watcher. A watcher can monitor the status of the user or BLF presence entity (also called presentity) by using the following options: ▪ BLF and Speed Dial buttons. ▪ Missed call, placed call, or received call lists in the directories window. ▪ Shared directories, such as the corporate directory. BLF presence group authorization ensures that only authorized watchers can access the BLF presence status for a destination. Because the administrator ensures that the watcher is authorized to monitor the destination when a BLF or Speed Dial is configured, BLF presence group authorization does not apply to BLF or Speed Dials. In Webex Calling BLF speed dials can be configured in the monitoring section of the Advanced Call Settings in Control Hub. Directories are not presence enabled so that the concept of BLF presence groups is not needed in Webex Calling.

Unified CM Concept	Comment
Extension Mobility	Cisco Extension Mobility allows users to temporarily access their phone settings, such as line appearances, services, and speed dials, from other phones within your system. In Webex Calling Hoteling enables a user's phone profile of phone number, features, and calling plan to be temporarily loaded onto a shared (host) phone. Hoteling can be enabled for users by the administrator in Control hub.
Mobile Remote Access	Expressway Mobile and Remote Access is a core part of the Cisco Collaboration Edge Architecture. It allows endpoints such as Cisco Jabber to have their registration, call control, provisioning, messaging, and presence services that are provided by Unified CM when the endpoint is not within the enterprise network. This concept is not applicable to Webex Calling because all endpoints connect to the Webex Calling cloud service over the top whether they are within or outside of the enterprise network.
Cisco Unified Mobility	Cisco Unified Mobility gives users the ability to redirect incoming IP calls from Unified CM to different designated phones, such as cellular phones. Users can also transition active calls between their Cisco desktop and mobile phone without interruption. Cisco Unified Mobility offers the following mobility-related features: Single Number Reach, Move to Mobile, Mobile Voice Access, Enterprise Feature Access, Intelligent Session Control. In Webex Calling similar functionality can be achieved using Remote Office, Office Anywhere, or Mobility.

Unified CM Concept	Comment
Extend and Connect	Extend and Connect enables users to use basic third-party call control operations from Jabber such as Make Call, Answer, and Disconnect with endpoints not registered with Unified CM. Webex Calling does not have a directly equivalent feature but features such as Remote Office and Office Anywhere address a similar use case.
Hunt Groups	While Webex Calling also offers Hunt Groups the functionality in Webex Calling is not identical to the functionality offered by Unified CM. For example, currently Webex Calling does not offer a hunt groups login/logout option for users.
Call Pickup	In Unified CM the Call Pickup feature allows users to answer calls that come in on a directory number other than their own. There are four flavors of Call Pickup: ▪ Group Call Pickup. ▪ Other Group Pickup. ▪ Directed Call Pickup. ▪ BLF Call Pickup. In Webex Calling the call pickup service enables a user to answer any ringing line within their pickup group. A pickup group is a group administrator-defined set of users within a location, to which the call pickup feature applies. The call pickup feature requires call pickup groups to be added as well as assigning specific users to that pickup group. Call pickup only works within the pickup group

Unified CM Concept	Comment
Call Park, Directed Call Park	Call Park feature on Unified CM allows to place a call on hold so that it can be retrieved from another phone in Unified CM. Directed Call Park allows a user to transfer a call to an available user-selected directed call park number. Directed call park Busy Lamp Field (BLF) on a phone can be used to monitor the busy or idle status of specific directed call park numbers. Users can also use the BLF to speed dial a directed call park number. Call Park monitoring (call reversion) can be configured for directory numbers and hunt pilots. In Webex Calling calls can be parked against members of a call park group and can be retrieved later by dialing the respective call park member. Call Park monitoring (call reversion) can be configured on the location and call park group level.
Ad Hoc Conferencing	Webex Calling supports 3-way and N-way-conferencing (up to 6 participants).
Meet-Me Conferencing	There is no equivalent feature in Webex Calling. Webex Meetings can be used as an alternative.
Conference Now	There is no equivalent feature in Webex Calling. Webex Meetings can be used as an alternative.
Music on Hold (MoH)	No equivalent in Webex Calling: all media resources are cloud based; no need for location specific MRG selection.
Voicemail	Basic VM available in Webex Calling.
Gateways, Trunks	Webex Calling supports Analog Telephony Adapters (ATA), Cisco Unified Border Elements as Local Gateway (Local Gateway), and Cloud Connected PSTN. Local Gateways can be used to interwork with virtually any technology.

Operations

Unified CM Concept	Comment
Credential Policy	Credential policies control the authentication process for resources in Unified CM. A credential policy defines password requirements and account lockout details such as failed login attempts, expiration periods and lockout durations for end user passwords, end user PINs, and application user passwords. In Webex Calling the password policy is determined by the organization level setting in Control Hub.
Access Control Group, Roles, User Rank	An access control group is a list of users and the roles that are assigned to those users. When you assign an end user, application user, or administrator user to an access control group, the user gains the access permissions of the roles that are associated to the group. With Webex Calling users within the organization can be assigned specific administrative roles to determine what they can see and have access to in Control Hub: ▪ Full Administrator ▪ Read-Only Administrator ▪ Support Administrator ▪ User and Device Administrator ▪ Device Administrator ▪ Compliance Office

Unified CM Concept	Comment
Bulk Administration Tool	Unified CM offers various bulk administration options including but not limited to: ▪ Add, update, delete, export phones. ▪ Add, update, delete, export, users. Web Bulk Administration Tools are also available in Webex Calling with some limitations: ▪ Bulk User Import. ▪ Bulk Device Import. ▪ Bulk Device-User Assignment.
Headset management	Headset management is a Webex Calling roadmap item.
APIs	Unified CM has APIs for provisioning (AXL), external call routing (CURRI), Serviceability (SXML), Device Monitoring and Call Control (JTAPI, TAPI, Web Dialer). The Webex Calling administration API is limited to basic provisioning tasks: ▪ Retrieve a Webex Calling license. ▪ Retrieve Locations . ▪ Create a Webex Calling user & assign phone number, extension, location. ▪ Delete a Webex Calling user. ▪ Assign Webex Calling license to existing Webex user & specify phone number, extension, location. ▪ Update phone number, extension for existing Webex Calling user. ▪ Retrieve Webex Calling user with phone number, extension, location. ▪ Query Webex Calling users by Location.

Unified CM Concept	Comment
APIs (cont.)	The Webex Calling call control API allows tasks: ▪ Place a call (dial). ▪ Answer a call (answer). ▪ Reject an incoming call (reject). ▪ Hang up a call (hang-up). ▪ Divert an incoming call (divert). ▪ Hold a call (hold). ▪ Resume a held call (resume). ▪ Transfer an active call (transfer). ▪ Park an active call (park). ▪ Retrieve a parked call (retrieve). ▪ List active calls. ▪ Get the details for a specific active call. ▪ Get the Call History. There are no APIs for device monitoring, serviceability, or external call routing.

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