Cisco Preferred Architecture for Bandwidth Management for Webex

Design Overview

September 2021

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Preface

Cisco Preferred Architectures provide recommended deployment models for specific market segments based on common use cases. They incorporate a subset of products from the Cisco Collaboration portfolio that is best suited for the targeted market segment and defined use cases. These deployment models are prescriptive, out-of-the-box, and built to scale with an organization as its business needs change. This prescriptive approach simplifies the integration of multiple system-level components and enables an organization to select the deployment model that best addresses its business needs.

Documentation for Cisco Preferred Architectures

- Cisco Preferred Architecture (PA) design overview guides help customers and sales teams select the appropriate architecture based on an organization's business requirements; understand the products that are used within the architecture; and obtain general design best practices. These guides support sales processes.
- Cisco Validated Design (CVD) guides provide details for deploying components within the Cisco Preferred Architectures. These guides support planning, deployment, and implementation (PDI).
- Alternative Design Guides — Are post-sales documents that describe optional designs that can be deployed as alternatives to the Preferred Architectures described in the PA overview guides and CVDs. The alternative design may start with the main PA as a foundation and build upon it; therefore, each alternative design guide may be used in conjunction with its corresponding PA overview guide and CVD. In some cases, the Alternative Design Guides can be used as standalone guides.

About This Guide

The *Cisco Preferred Architecture for Bandwidth Management for Cisco Webex Solution Components* is for:

- Sales teams that design and sell collaboration solutions

Customers and sales teams who want to understand the Cisco Webex Edge Connect architecture, its components, and general design best practices.

Readers of this guide should have a general knowledge of Cisco Voice, Video, and Collaboration products and a basic understanding of how to deploy these products.

This guide simplifies the design and sales process by:

- Recommending products in the Cisco Collaboration portfolio that are built for the enterprise and that provide appropriate feature sets for this market
- Detailing a collaboration architecture and identifying general best practices for deploying in enterprise organizations

For detailed information about configuring, deploying, and implementing this architecture, consult the related CVD documents on the Cisco Collaboration Preferred Architectures.

Introduction

Bandwidth management architecture and deployment for the Cisco Collaboration on-premises solution is covered in depth in the latest version of the Preferred Architecture for Cisco Collaboration Enterprise On-Premises Deployments, available at https://www.cisco.com/go/pa. This document attempts to comprise the recommendations for all Webex collaboration solution components in a holistic and modular fashion.

Core Components

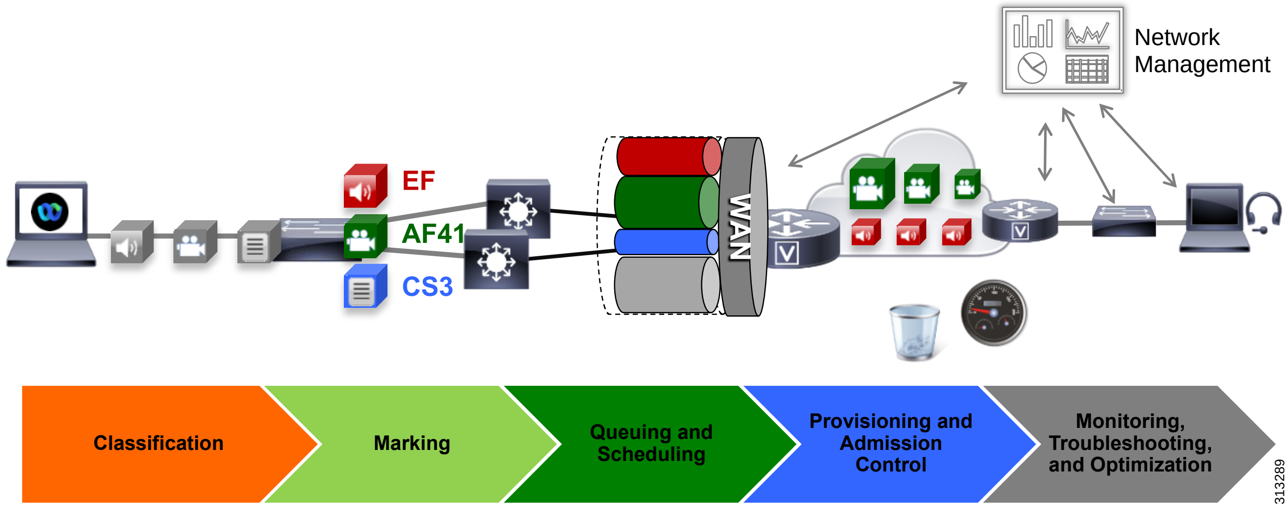
Cisco Webex Services architecture contains these key components:

- Cisco Unified Communications Manager
- Webex App and Webex Devices
- Cisco Expressway
- Cisco Unified Border Element
- Webex Meetings
- Webex Video Mesh Node
- Network infrastructure:
    - Cisco routers
    - Cisco switches

Figure 1 illustrates the design approach to Quality of Service (QoS) used in the Cisco PA for Enterprise Collaboration. This approach consists of the following phases:

- **Classification and Marking** — Refers to concepts of trust and techniques for identifying media and call signaling for endpoints and applications. It also includes the process of mapping the identified traffic to the correct DSCP markings to provide the media and signaling with the correct per-hop behavior end-to-end across the network.
- **Queuing and Scheduling** — Consists of general WAN and Internet queuing and scheduling, the various types of queues, and recommendations for ensuring that collaboration media and signaling are correctly queued on egress to the WAN and Internet.
- **Bandwidth Provisioning** — Refers to provisioning the bandwidth in the network and determining the maximum bit rate that groups of endpoints will utilize.
- **Monitoring, Troubleshooting, and Optimization** — Ensures the proper operation and management of voice and video across the network. Cisco Prime Collaboration offers a suite of tools to perform these functions. Monitoring, troubleshooting, and optimization are not covered in the Preferred Architectures but are part of the overall approach.

Figure 1 Architecture for Bandwidth Management



Recommended Deployment

Modify the existing on-premises QoS switch and WAN and Internet policies to include Webex Services identification, classification, and marking.

- Identify and classify media and signaling traffic from the Webex App and associated workloads as well as Webex Devices.
- Media and signaling marking recommendations:
    1. Mark all audio with Expedited Forwarding class EF (includes all audio of voice-only calls as well as audio for all types of video calls).
    2. Mark all Webex App video with an Assured Forwarding class of AF41 for a prioritized video class of service and AF42 for an opportunistic class of service. The marking of AF41 or AF42 will depend on the choice of whether to deploy opportunistic video during the on-premises deployment phase.
    3. Mark all call signaling with CS3. (All call signaling in HTTPS traffic will be marked based on the enterprise's current policy of traffic marking for HTTP/HTTPS – unless using NBAR which is covered in detail below)
- Configure QoS on all media originating and terminating applications such as the Video Mesh Nodes, Expressway and Cisco Unified Border Element.
- Update the WAN edge ingress re-marking policy.
- Update the WAN edge egress queuing and scheduling policy.

Benefits

This deployment of bandwidth management provides the following benefits:

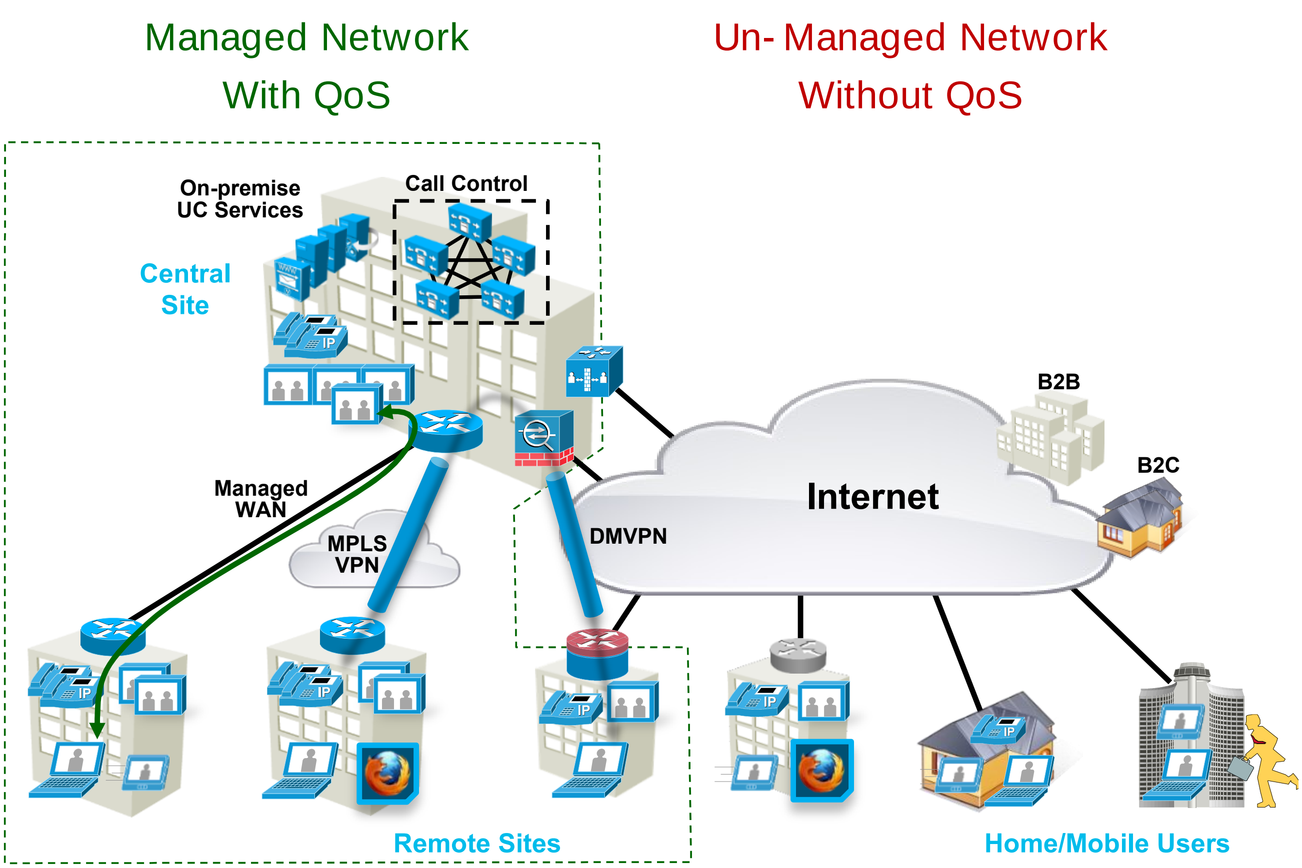
- Ensures the best quality
- Provides prescriptive recommendations to simplify deployment with a simplified QoS architecture that integrates with the Enterprise PA for Collaboration
- Makes more efficient use of network resources
- Supports mobile and multi-media collaboration devices
- Considers unmanaged network segments (Internet)

Architecture

In this Preferred Architecture, usage of the Internet for cloud-based services such as Webex App and Webex Devices is an important aspect of the solution, which means that some of the collaboration infrastructure is located outside of the managed enterprise network and located in the cloud. The enterprise office connectivity options also range from remote sites and mobile users connected over managed leased lines directly connected to MPLS or other technologies, to connectivity over the Internet through technologies such as Dynamic Multipoint VPN (DMVPN), for example. With Webex App an office can be anywhere there is sufficient Internet connectivity.

Figure 2 illustrates the convergence of a traditional on-premises collaboration solution in a managed (capable of QoS) network with cloud services and sites located over an unmanaged (not capable of QoS) network such as the Internet. On-premises remote sites are connected over this managed network, where administrators can prioritize collaboration media and signaling with QoS, while other remote sites and branches connect into the enterprise over the Internet, where collaboration media and signaling cannot be prioritized or are prioritized only outbound from the site. Many different types of mobile users and teleworkers also connect over the Internet into the on-premises solution. So, the incorporation of the Internet as a source for connecting the enterprise with remote sites, home, and mobile users, as well as other businesses and consumers, has an important impact on bandwidth management and user experience.

Figure 2 Managed and Unmanaged Networks



This section presents a strategy for leveraging smart media techniques in Cisco video endpoints, building an end-to-end QoS architecture, and using the latest design and deployment recommendations and best practices for managing bandwidth to achieve the best user experience possible based on the network resources available and the various types of networks that collaboration media traverse.

Media Overview

Background on Audio, Video Packet Delivery and Media Resilience

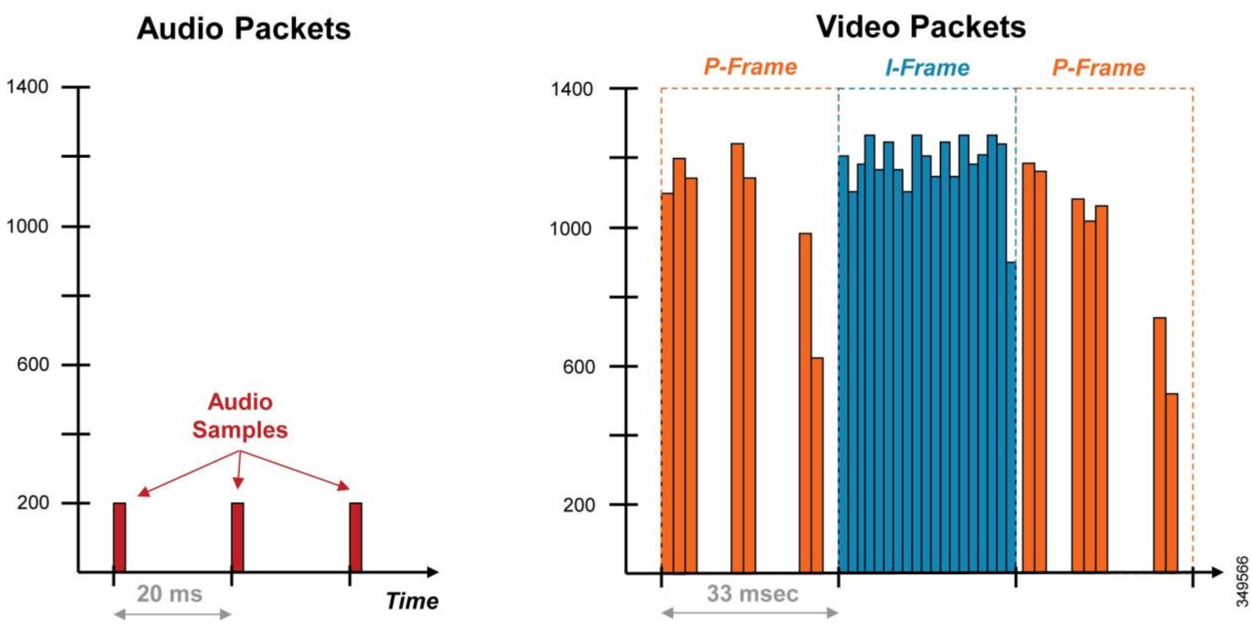
This section covers the characteristics of audio and video streams in real-time media as well as the smart media techniques that Cisco video endpoints employ to ensure high fidelity video in the event of packet loss, delay, and jitter.

Audio versus Video

Voice and video are often thought of as quite similar, and although they are both real-time protocol (RTP) applications, the similarities stop there. Voice is generally considered well behaved because each packet is a fixed size and fixed rate. Video frames are spread over multiple packets that travel as a group. Because one lost packet can ruin a P-frame, and one bad P-frame can cause a persistent artifact, video generally has a tighter loss requirement than audio. Video is asymmetrical. Voice can also be asymmetrical but typically is not. Even on mute, an IP phone will send and receive the same size flow.

Video increases the average real-time packet size and has the capacity to quickly alter the traffic profile of networks. Without planning, this could be detrimental to network performance. Figure 3 shows the difference between a series of audio packets and video packets sent during a specific time interval.

Figure 3 Audio versus Video



As illustrated in Figure 3, the audio packets are the same size, sent at the same time intervals, and represent a very smooth stream. Video, on the other hand, sends a larger group of packets over fixed intervals and can vary greatly from frame to frame. Figure 3 shows the difference in the number of packets and packet sizes for an I-Frame compared to P-frames. This translates to a stream of media that is very bursty in nature when compared to audio. This burstiness is illustrated in Figure 4, which shows the bandwidth profile over time of an HD video stream. Note the large bursts when I-Frames are sent.

Figure 4 Bandwidth Usage: High-Definition Video Call

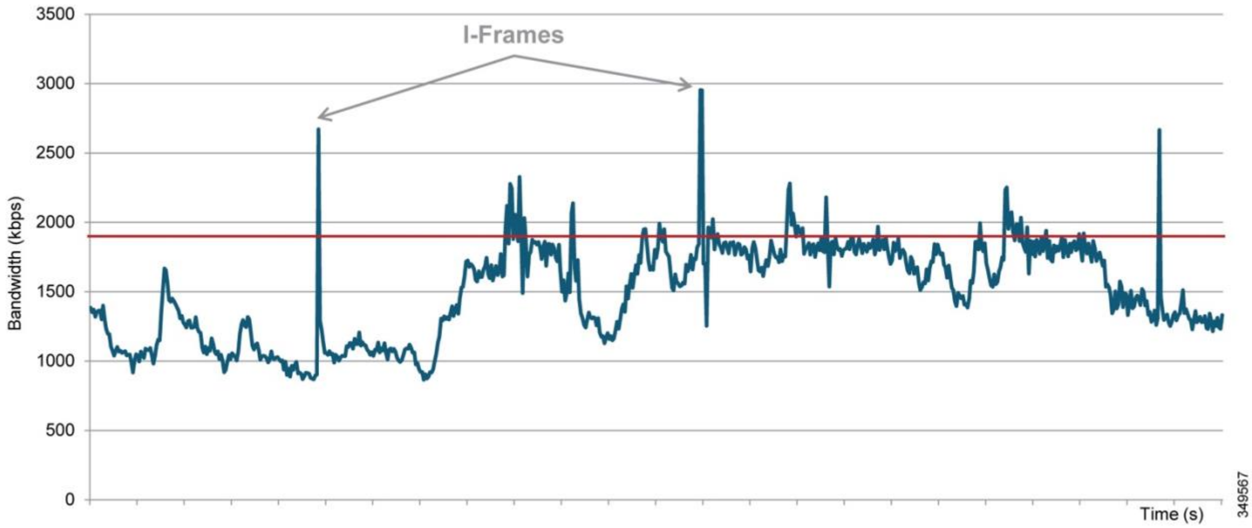
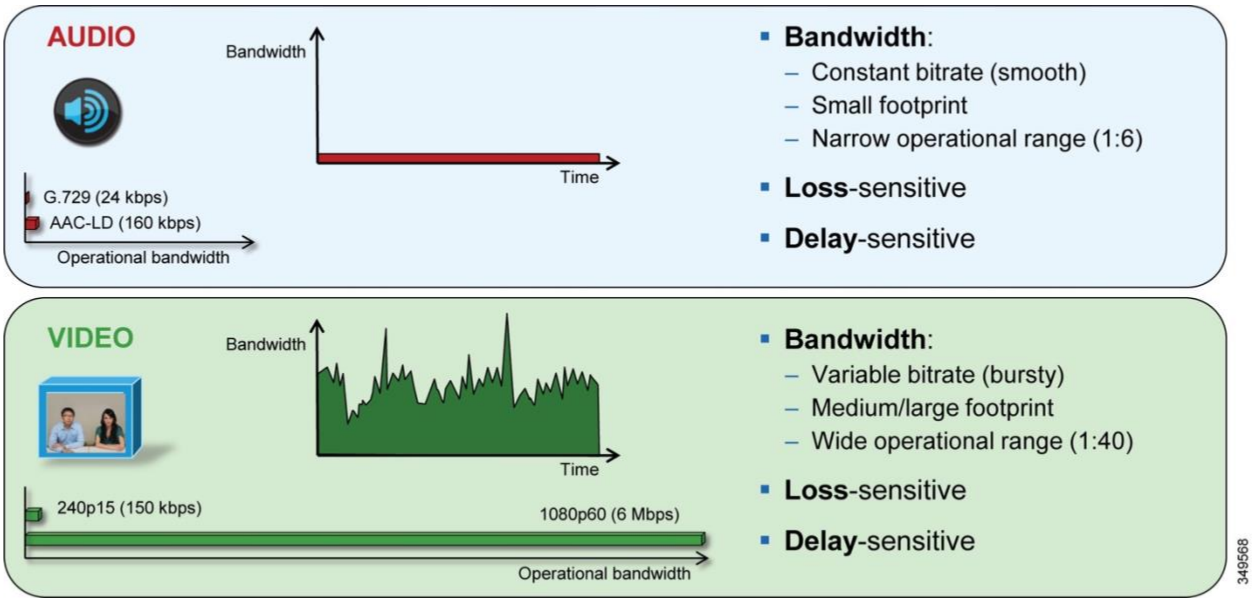


Figure 4 shows an HD video call at 720p at 30 fps and 1920 kbps (1792 kbps video + 128 kbps audio). The red line indicates the average bit rate for the duration of the call.

While audio and video are both transported over UDP and are sensitive to loss and delay, they are quite different in their network requirements and profile. As shown in Figure 5, audio is a constant bit rate, has a smaller footprint compared to video, and has a narrower operational range of 1:6 ratio when comparing the lowest bit rate audio codec to one of the highest bit rate codecs. Video, on the other hand, has a variable bit rate (is bursty), has a medium to large footprint when compared to audio, and has a wide operational range of 1:40 (250p at 15 fps up to 1080p at 60 fps).

Figure 5 Video Traffic Requirements and Profiles



The main point here is that audio and video, while similar in transport and sensitivity to loss and delay, are quite different in the methods employed to manage their bandwidth requirements in the network. Also, while video is pertinent to a full collaboration experience, audio is critical. For example, during a video call, if video is lost or distorted due to a network outage or some other network related event, communication can continue if audio is not lost during that outage. This is a critical concept in bandwidth management in the PA.

"Smart" Media Techniques (Media Resilience and Rate Adaptation)

When deploying video pervasively across an organization, administrators will inevitably encounter insufficient bandwidth to handle the load of video required during the busy hour in some bottleneck areas of the Wide Area Network (WAN). Considering this it is important to prioritize video correctly, to ensure that audio is not affected by any video packet loss that may occur, and to ensure that certain types of video can leverage video rate adaptation to manage the amount of bandwidth used during times of congestion. The media resilience and rate adaptation techniques allow for an optimized video experience in the face of congestion and packet loss over managed and unmanaged networks.

Every Cisco video endpoint employs several smart media techniques to avoid network congestion, recover from packet loss, and optimize network resources. The following smart media techniques are just some of the techniques employed by Cisco Webex App and Devices:

Media resilience techniques

- Encoder pacing
- Forward Error Correction (FEC)
- Rate adaptation

Media Resilience Techniques

The Cisco Webex Apps and Devices in this Preferred Architecture document support some of following media resilience techniques. Not all of these media resilience techniques are found across each Webex App and Device however Cisco Webex strives to provide the best possible experience to every participant regardless of their individual network connection quality or device type. We are on a continuous improvement trajectory and these audio and video resilience techniques and audio and video quality improvements attest to that.

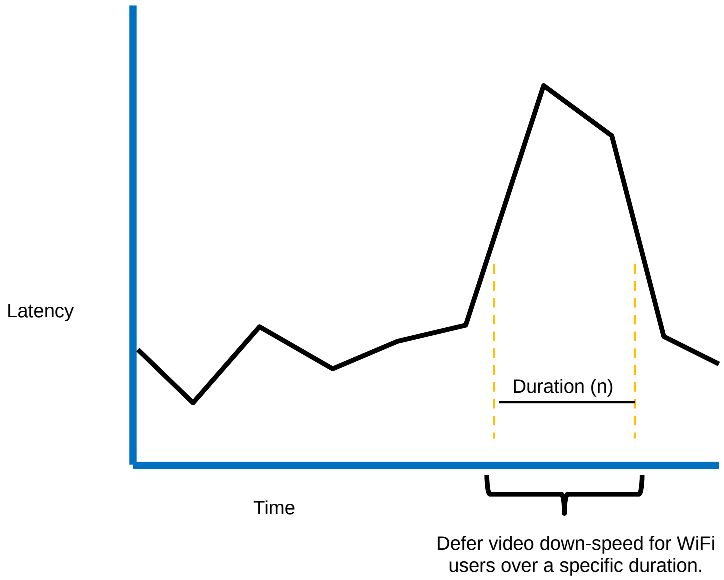
Audio and Video Quality Improvements

Recently we have included adaptive frame rate policy to selectively discard frames with lower information content which improves perceived video quality where lower bitrates are available. We have also rolled out ‘region of interest’ video encoding that spends greater bits on important regions of a video frame such as faces or on actual displayed regions of a meeting based on the user’s behavior. We are also happy to announce that both the audio and video streams are independently resilient for impairment scenarios up to a remarkable 50% packet loss with the combined implementation of several either new or updated media resiliency techniques. The first is via a brilliant UDP retransmit mechanism where we trade a little bit of latency for a whole lot of media (both audio and video) recovery consistent with user tolerances around latency. We have also delivered similar improvements around content sharing to provide even faster, more responsive slide transitions under heavily impaired network conditions. All these improvements improve the Quality of Experience metrics experienced by users when subject to lossy environments of varying degree.

Webex Media Improvements

Improved Video Experience over WIFI (Defer video down-speeding): WIFI environments present a specific set of challenges and through testing we noticed the algorithms were too quick to drop resolution/bandwidth in certain WIFI environments. In Webex version 40.7 the algorithms have been updated to ‘defer the down-speeding” of video for brief WIFI specific network dropouts. This resulted in improved user experience over WIFI. See Figure 6 for an illustration of this. When latency spikes for a specific duration the improved algorithms can detect that this is a WIFI specific event and thus delay the down-speeding of the video resolution until that duration (n) is exceeded in which case would infer a different network event increasing latency.

Figure 6 Deferred Video Down-speeding



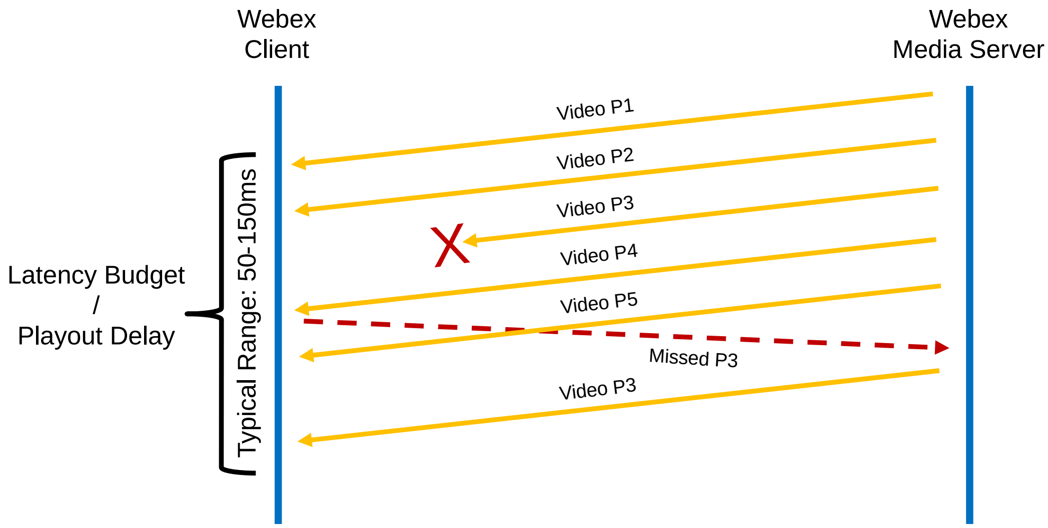
Video Super Scaling is a technique used for users on sub-optimal networks. We provide 720p like experiences to users who are experiencing network impairments resulting in 360p connections to our servers for their meeting. Our updated scaling algorithm provides enhanced video quality when this degradation occurs, making network impairment downgrades less quality impacting. If a 360p image is simply scaled geometrically to a 720p resolution, the quality has not improved, but our technique involves migrating video post-processing tasks to the client for some video flows versus doing it in the cloud, resulting in being able to keep more precious bits of information to reconstruct the video at a given bandwidth.

Music Mode is a feature supporting high quality audio sources. We found it is best to have a 'music mode' setting available to change the audio signal processing in Webex from its default behavior of focusing on speech alone and instead preserving the defined microphone inputs of the original sound for best music quality in a meeting.

Audio background noise improvement: Expanding on our prior experiences with developing ways to determine background audio sources as typing, sirens, dog barks, and more, we've added additional classifications such as office/street background noise attenuation capability resulting in sharper/clearer audio in these more open environments.

Video RTX (Re-Transmit) reduces video pausing/skip scenarios in high packet loss/bursty conditions. We enabled selective video retransmit capability in lossy environments where there is sufficient time to request a retransmission of missing video frames while there is still enough of a video playout buffer to accommodate the request. The advantage here over prior loss recovery methods such as FEC [forward error correction] is that lower bandwidth is required, and objective video quality is improved in these scenarios. Depending on the specific network conditions encountered, video RTX or FEC will be used to improve media quality. Testing has demonstrated that packet loss resilience has been seen in environments of up to 50% packet loss. Figure 7 illustrates a high-level overview of the Video RTX feature. The retransmit mechanism takes advantage of the Webex Meetings App video playout buffer to selectively attempt packet loss recovery via retransmit when video frames are lost in transit and network conditions suggest that retransmission is superior to FEC for video playout recovery.

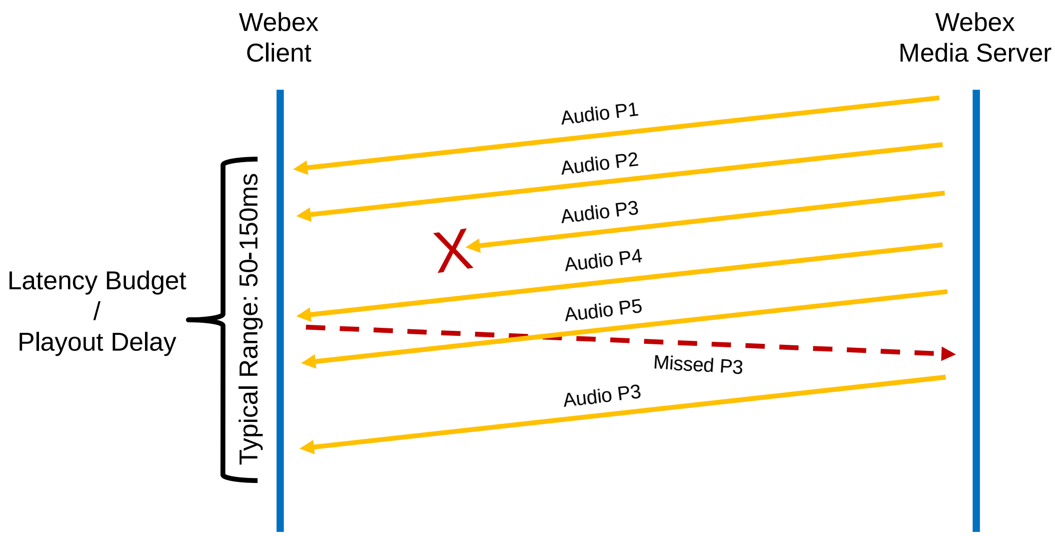
Figure 7 Webex Video RTX high-level overview



Audio RTX (Re-Transmit):

Like Video RTX, Audio RTX attempts to reduce quality deficits due to audio drops in bursty loss environments. Audio retransmit capability is based on currently observed latency where there is still enough time to attempt packet recovery via packet retransmission. Techniques such as FEC [forward error correction] have less desirable results in terms of higher bandwidth utilization. The benefit here is superior audio quality compared to other recovery mechanisms when subject to bursty loss. Testing has demonstrated that packet loss resilience has been seen in environments of up to 50% packet loss. Figure 8 illustrates a high-level overview of the Audio RTX feature. The retransmit mechanism takes advantage of the Webex Meetings App audio playout buffer to selectively attempt packet loss recovery via retransmit when audio packets are lost in transit and network conditions suggest that retransmission is superior to FEC for audio playout recovery.

Figure 8 Webex Audio RTX high-level overview



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Encoder Pacing

Encoder pacing is a simple technique used to spread the packets as evenly as possible to smooth out the peaks of the bursts of bandwidth.

Forward Error Correction (FEC)

Forward error correction (FEC) provides redundancy to the transmitted information by using a predetermined algorithm. The redundancy allows the receiver to detect and correct a limited number of errors occurring anywhere in the message, without the need to ask the sender for additional data. FEC gives the receiver an ability to correct errors without needing a reverse channel (such as RTCP) to request retransmission of data, but this advantage is at the cost of a fixed higher forward channel bandwidth (more packets sent). FEC protects the most important data (typically the repair P-frames) to make sure the receiver is receiving those frames.

Rate Adaptation

Rate adaptation or dynamic bit rate adjustments adapt the call rate to the variable bandwidth available, down-speeding or up-speeding the video bit rate based on the packet loss condition. An endpoint will reduce bit rate when it receives messages from the receiver indicating there is packet loss; and once the packet loss has decreased, up-speeding of the bit rate will occur.

The Self-Regulating Video Network

The self-regulating video network, prioritized audio, and opportunistic video are all QoS concepts as well as a QoS strategy. A self-regulating video network consists of leveraging the smart media and rate adaptation techniques discussed previously, along with proper provisioning and QoS to allow the video endpoints to maximize their video resolution during times when video bandwidth is not fully utilized in the network and to rate adapt or throttle down their bit rate to accommodate more video flows during the busy hour of the day.

Prioritized audio for both audio-only and audio of video calls ensures that all audio is prioritized in the network and is thus not impacted by any loss that can occur in the video queues. Prioritizing voice from all types of collaboration media ensures that even during times of extreme congestion when video is experiencing packet loss and adjusting to that loss, the audio streams are not experiencing packet loss and are allowing the users to carry on an uninterrupted audio experience.

In addition, opportunistic video allows for a group of video endpoints to be strategically marked with a lower class of video, thus allowing them to use available bandwidth when the opportunity arises. This enables endpoints to achieve optimal video resolution during times when the network is less congested, and more bandwidth is available. Conversely, devices can down-speed their video more aggressively than the higher prioritized class of video during times of congestion when the network is in its busiest hour.

This concept of opportunistic video, coupled with prioritized audio, maintains an acceptable video experience while simultaneously ensuring that voice media for these opportunistic video calls is not compromised. This of course applies to the managed network since an unmanaged network such as the Internet is not QoS enabled and thus provides no guarantees regarding packet loss. Nevertheless, the media resiliency and rate adaptation mechanisms also attempt to ensure that media over unmanaged networks have the best possible quality in the face of packet loss, delay, and jitter.

Opportunistic video is an optional deployment choice that adds value to a self-regulating video network with prioritized audio; however, it is not mandatory for a self-regulating video network to function.

QoS Architecture for Collaboration

Quality of Service (QoS) ensures reliable, high-quality voice and video by reducing delay, packet loss, and jitter for media endpoints and applications. QoS provides a foundational network infrastructure technology, which is required to support the transparent convergence of voice, video, and data networks. With the increasing number of interactive applications (particularly voice, video, and immersive applications), real-time services are often required from the network. Because these resources are finite, they must be managed efficiently and effectively. If the number of flows contending for such priority resources were not limited, then as those resources become oversubscribed, the quality of all real-time traffic flows would degrade, eventually to the point of futility. Media resiliency, rate adaptation and QoS ensure that real-time applications and their related media do not oversubscribe the network and the bandwidth provisioned for those applications. These smart media techniques coupled with QoS are a powerful set of tools used to protect real-time media from non-real-time network traffic and to protect the network from over-subscription and the potential loss of quality of experience for end users of voice and video applications.

Cisco Webex Apps and Webex Devices

In this document we categorize the Webex Apps and Devices into the following groupings:

Cisco Webex App for Desktop and Mobile

Cisco Webex Devices (Board, Room and Desk)

Cisco Webex Apps and Devices provide a wide range of features, functionality, and user experiences. Because Cisco Webex endpoints range from low-cost soft clients to presentation, whiteboard, and multi-screen Webex Room Systems, an organization can deploy the right variety of endpoints to meet users' needs. Additionally, these devices enable users to access multiple communication services such as:

- Voice and video calling
- Meetings
- Messaging
- Desktop and content sharing
- Whiteboarding
- On-Premises registration, endpoint and PSTN Calling

Cisco Webex App for Desktop and Mobile

Webex App is a software client for mobile (iOS and Android) and Desktop (Windows and Mac) capable of multiple workloads connecting to different call control services using different media stacks to support the various calling capabilities, features and call flows. Supported calling options in Webex are discussed in this article Webex | Supported Calling Options. The following nomenclature is used in this document going forward to delineate the various calling workloads in Webex App to differentiate the signaling and media stream requirements for identification and QoS classification:

- **Webex App (Teams):** This will refer to the native calling functionality for Webex App. It is used for Webex Space Meetings (meetings started from a Webex space), 1:1 calls to a Webex Account holder ("Call on Webex" calling option in Webex App) as well as SIP URI dialing. This is also the call functionality for Webex scheduled or Personal Room meetings for Webex Orgs enabled with Video Mesh Clusters or on the FedRamp program created prior to May, 15 2021, or if Full-Featured Meetings has not been enabled or has been disabled. This workload was formerly accomplished with Webex Teams.

- **Webex App (Meetings):** Full-Featured Meetings or Meetings Experience is a feature enhancement to Webex App. When you start or join a Webex scheduled or Personal Room meeting from the Webex app, you get access to advanced features such as stage view, breakout sessions, reactions, Webex Assistant for Meetings (where available), and People Insights profiles. This is the default setting for Webex Meetings Orgs enabled after May, 15 2021, unless the customer chose to opt out. All Webex Meetings Orgs enabled prior to May, 15 2021, will have this feature enabled by default unless disabled for Video Mesh Cluster or on the FedRamp program. Alternatively, **Webex Meetings Desktop and Mobile App** is used for this workload.

- **Webex App (Calling):** Integrated calling to Webex Calling, a cloud-based phone system that is optimized for midsized businesses offering enterprise-grade cloud calling, mobility, and PBX features. This solution lets you register your Webex App directly to the Webex Calling Platform (service is offered through a Service Provider (SP) or a Value-Added Reseller (VAR)). This workload was formerly accomplished by **Webex Calling App**.

- **Webex App (Unified CM):** The Calling in Webex (Unified CM) solution lets you register Webex App directly to your Cisco Unified Communications Manager call control environment (on-premises enterprise, Business Edition 6000/7000, Unified CM Cloud or as delivered through an HCS partner solution). This solution enhances the calling experience for end users, allowing them to directly make calls in Webex through your Unified CM environment, use mid-call features, and control their Unified CM registered desk phone from Webex. When dialing from Webex, users can use the same dial strings or prefixes as they do on their desk phones; Webex functions like any other desk phone registered to your Unified CM. Unified CM calls that are established in Webex use the configuration that's in place for your Unified CM deployment (such as location, bandwidth settings, point to point media, and so on). Alternatively, **Jabber** is used for this workload.

It is important to understand the ports and protocols used by interactive audio and video streams generated by one-to-one calls and multipoint meetings in a Webex deployment, so that you can apply the QoS tools in the relevant parts of the network and can provision bandwidth correctly.

As explained Webex App is a software client capable of multiple workloads connecting to different call control services using different media stacks. We'll go through each workload and the associated media and signaling flows where applicable.

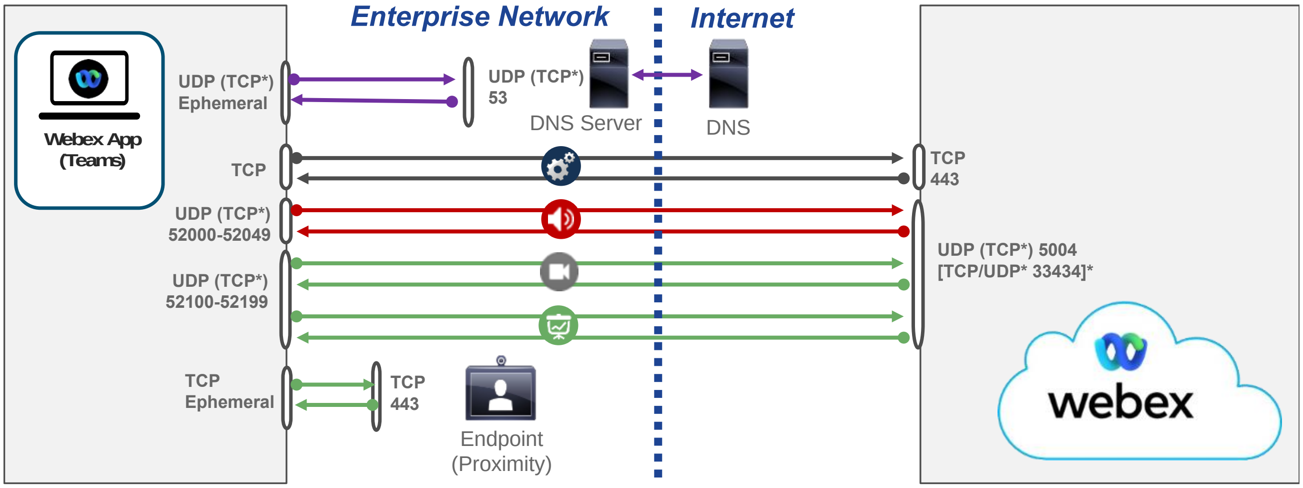
Note: The following sections of this document cover a high-level view of the signaling and media protocols and ports used for communication by Webex App and Webex Devices. It is intended as a primer to understand the ports and protocols for Webex products specifically for QoS prioritization of media within the network and is limited in scope with regards to its coverage of ports, protocols, and communications for this reason. For more information on all of the ports and protocols used by Webex App and Webex Devices please see the Webex help article entitled "Network Requirements for Webex Services" as that is specifically intended for network administrators, particularly firewall and proxy security administrators.

Cisco Webex App (Teams)

This is the functionality for Webex App (Teams) which are meetings started from a space or 1:1 calls to a Webex Account holder ("Call on Webex" calling option in Webex App – see Webex | Supported Calling Options for more details).

Figure 9 illustrates a high-level overview of Webex App and some of the various signaling and media sessions that are set up by the application joining Webex Spaces and 1:1 meetings.

Figure 9 Webex App (Teams)



* Backup connections

Special Considerations for Cisco Webex App (Teams) for Microsoft Windows

Webex App (Teams) for Windows currently uses ephemeral source ports for media provided by the Windows OS. There is a backend configuration to allow the ability to set Webex App for Windows to use the same source port range as other Webex App (Teams) clients. This is ONLY for Webex App (Teams) and does not apply to Webex App (Meetings) joined via an invite or from a “Call on Webex” calling option. The Webex feature toggle is called ***desktop-qos-enabled***. Please contact your account team if you would like this enabled for your Webex Org.

Caveats

There are a few caveats to this feature functionality, which is also why it is not enabled by default or a standard configuration at this time.

- Caveat #1 Microsoft Defender Firewall:

  Webex App (Teams) has special functionality for the Microsoft Windows platform that allows it to bind itself to specific source port ranges. These are the same port ranges that all other Webex App platforms (MacOS, Apple iOS, or Android) use natively. To allocate source ports from the specific ranges. Webex App for Windows application must make a request to the underlying operating system when it is first installed on a device. This results in the following behavior on devices running Microsoft Windows with Windows Firewall enabled:

  > Because of a limitation in the Microsoft Windows APIs, whenever an application requests a specific source port from the operating system, it also gains permission to listen for unsolicited incoming traffic on that port. The Webex App (Teams) application does not need these privileges because it receives packets only on a given port after transmitting from that same port, but it has no way of communicating this to the operating system.

Therefore, in Microsoft Windows system configurations where Microsoft Windows Firewall is enabled, a security alert might be displayed to the end user when Webex App (Teams) is first run, informing them that Windows Firewall has blocked some features of the application, and prompting them to allow access (which requires administrator privileges) or to cancel.

It is important to note that, regardless of the action chosen by the end user for this alert, the Webex App will operate correctly using the appropriate source ports, and no other alert will be displayed after the initial session.

- Caveat #2 Cisco Anyconnect Secure Mobility Client:

  When source ports are used in conjunction with Cisco Anyconnect Secure Mobility Client the Webex traffic is bound to the Anyconnect VPN adapter and not the local interface. This results in all Webex traffic being forwarded over the VPN adapter, which over-rides any split-tunnel traffic configuration that may have been implemented in the VPN design.

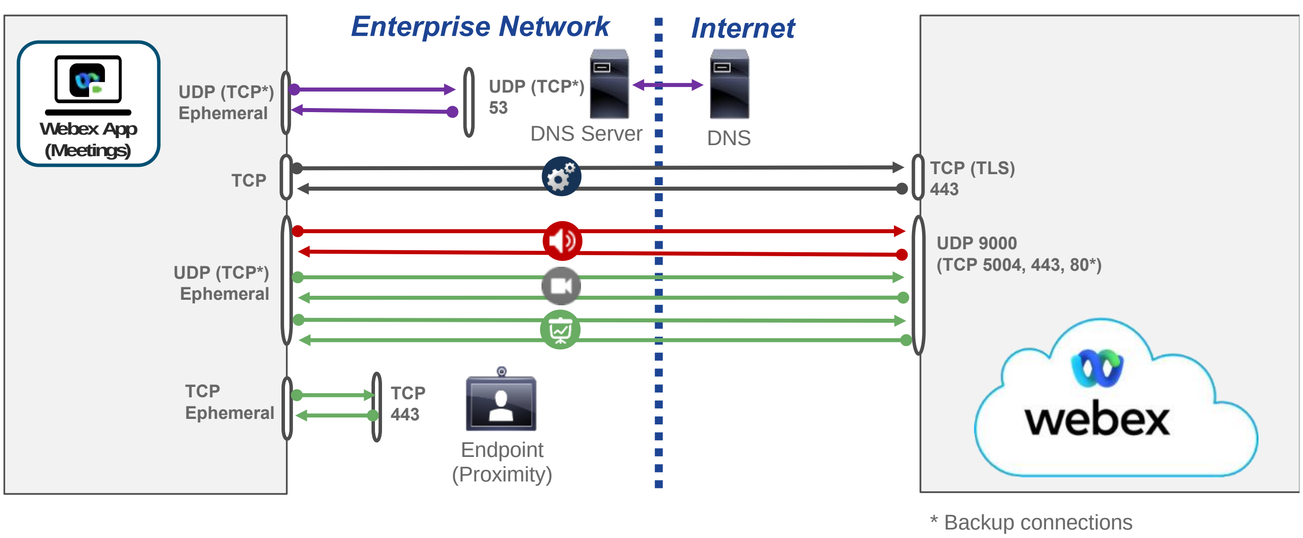
Note: This feature support applies only to Microsoft Windows, and it does not affect the Webex App on other platforms such as MacOS, Apple iOS, or Android.

Cisco Webex App (Meetings)

This is the functionality for Webex App (Meetings) which are Webex Meetings joined via an invite or from a "Call on Webex" calling option in Webex App – see Webex | Supported Calling Options for more details). As explained earlier in the **Webex Meetings Desktop and Mobile App** was used for this workload.

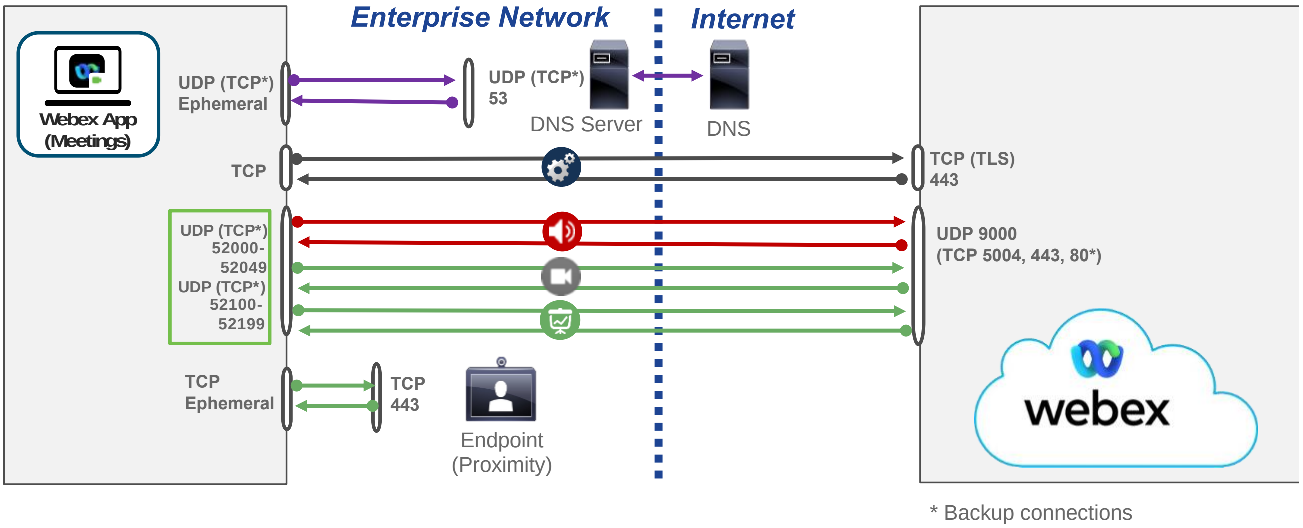
Figure 10 illustrates a high-level overview of Webex App and some of the various signaling and media sessions that are set up by the application joining Webex Meetings.

Figure 10 Webex App (Meetings)



One major difference between Webex App (Teams) and Webex App (Meetings) is that Webex App (Meetings) only recently supports source port enablement to differentiate audio and video/share media, as of version 42.7. In this new release Webex App (Meetings) all clients get source port enablement for differentiated audio and video media traffic as illustrated in Figure 11.

Figure 11 Webex App (Meetings) Version 42.7 and above



This is enabled by default for all client types, however there is a specific enablement for Windows OS clients that require the installation of the Webex Meetings Desktop App with a command line switch prior to running the Webex App installation. This switch enables the source port feature and Windows Defender Firewall ACLs as well as the required Windows registry settings for enablement. Once enabled the installer can install Webex App in order to get the full Webex App suite.

Both Apps are available at https://www.webex.com/downloads.html. The Webex Meetings Desktop app is available under *Other download options* > *Our previous app, Meetings*. From this drop down the installer can select the Windows platform to download the installation program. This command line switch is only needed once during the first installation and thereafter any upgrades will still get source port enablement. This includes any upgrades simply using the Webex App.

1. Step 1 install Webex Meetings Desktop App with a command line switch during installation:

   32bit MSI: msiexec /i webexapp.msi LOCALPORTRANGE="1“

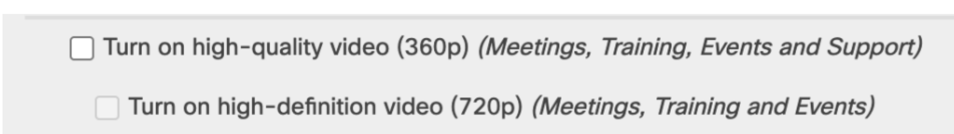
2. Step 2 install the Webex App

Note: Installing Webex Meetings Desktop app and running the installation switch is the easiest way of enabling media source ports for Windows clients for Webex Meetings workloads however if running the installation switch during installation is not an option for your environment there is another option that allows an administrator to enable this feature post-installation. Please follow the *“Enable the Feature (Post-Installation)“* section of the document Enable Dedicated Media Source Port Ranges for Webex App (Full Featured Meetings) and Webex Meetings Desktop App.

Webex App (Meetings) Bandwidth Controls

Webex administrators have 2 key controls to help control bandwidth as used by clients that connect to Webex meetings should they choose to. Namely, you can cap the meeting layouts at either 360p as the max available resolution, or to enable 720p layers. Whether your site is administered on Webex Control Hub or Webex Site Administrator, the following controls are available in Configuration > Common Site Settings > Options:

Figure 12 Webex App (Meetings) Bandwidth Controls



See table 1 for bandwidth values for each resolution above.

Table 1 Webex Meetings Desktop App Bandwidth per Resolution Table

Layer	Bandwidth Range
90p active thumbnail [each]	~60-100 kb/s
180p main video	125-200 kb/s
360p main video	470-640 kb/s
720p main video	900k-1.5 mb/s
Content sharing [sharpness, 1080p/5]	120k – 1.3 mb/s
Content sharing [motion, 720p/30]	900k – 2.5 mb/s

Webex App for Mobile

Video features for mobile clients

Webex App multipoint video is supported on Android and Apple iPads and iPhones. Webex App on mobile offers customizable video and share layouts. The initial layout will be 1:1 video, as shown in Figure X.

Bandwidth for the mobile phone grid-view option with two panes

Number of participants	Grid view mobile with two panes - no content transmit	Grid view mobile with two panes - no content receive
3	920 kbp/s	1.5 mb/s

Table 2 Bandwidth for a mobile phone grid-view option with four panes

Number of participants	Grid view mobile with four panes plus content transmit	Grid view mobile with four panes plus content receive
4	920 kbp/s	2.1 mb/s

Table 3 Bandwidth for a tablet with four active thumbnails

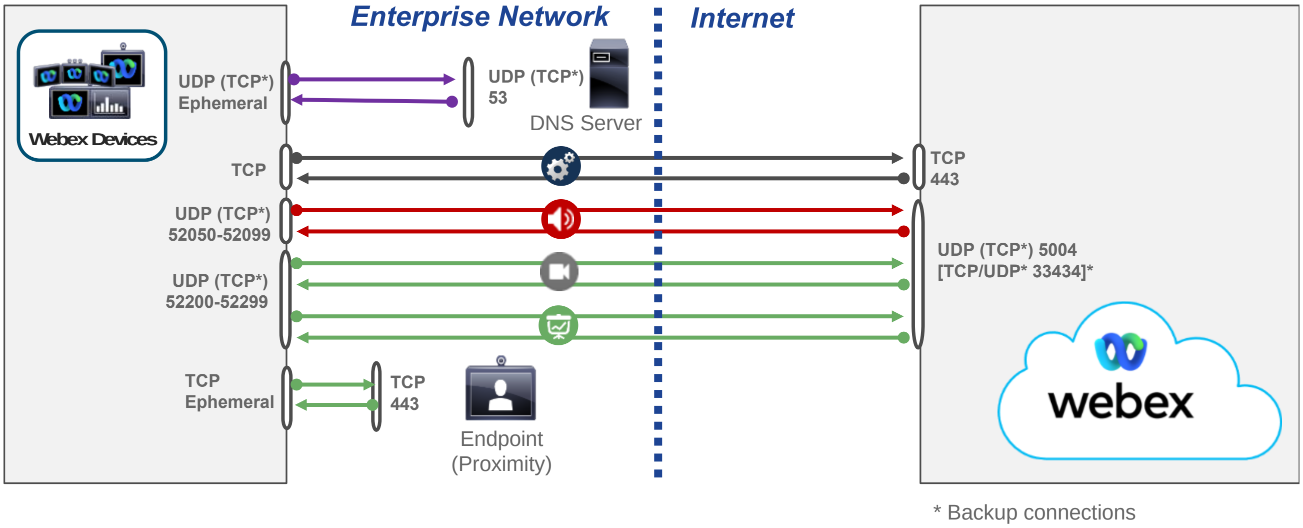
Number of participants	Share view with four thumbnails - transmit	Share view with four thumbnails - receive
5	~560 kb/s (participant)	1.4 mb/s

Note: Mobile devices are efficient at conserving bandwidth when the meeting is idle, when speaker video is mostly talking heads, and when content transitions do not occur frequently. Phones and tablets typically consume around 80 kb/s in an idle meeting such as these and use around 100 kbps with slide transitions limited to every 30 s or so. The figures in the tables in this document are for dynamic meetings with considerable speaker motion (common on mobile participants), and frequent content scene changes.

Cisco Webex Devices (Board, Room and Desk)

Figure 8 illustrates a high-level overview of Webex Devices and some of the various signaling and media sessions that are set up by the devices joining Webex Meetings.

Figure 13 Webex Devices (Board, Room and Desk)



Cisco Webex Devices come in 3 classifications based on usage, Webex Boards, Webex Room Devices and Webex Desk Devices.

The **Cisco Webex Board** is an all-in-one whiteboard, wireless presentation screen, and video conferencing system for smarter team collaboration.

The **Cisco Webex Room Devices** are Intelligent video conferencing devices for meeting rooms of all sizes.

The **Cisco Webex Desk Devices** are simple-to-use and compact video conferencing devices designed for desktops.

For bandwidth discussions it is more appropriate to categorize these types of endpoints based on their maximum bit-rate support. There are 2 categories of these devices, those which support up to 3 mbps bit rate and those which support up to 6 mbps. There are also devices which have dual screens and thus have a different level of bandwidth consumption in aggregate.

3 mbps Bit-rate Support

SX10, DX Series,

6 mbps Bit-rate Support

All Webex Room Series

SX20, SX80, MX Series

Video Stream Layouts for Webex Devices

Cisco Webex Video Conferencing Devices are purpose built to help teams collaborate and create together in real time, no matter how far the distance. When enabled with video conferencing, they allow users to see every video stream from the meeting in a variety of layouts. As seen in previous sections bandwidth usage can vary based on the type of layout selected. Below we'll show the types of layouts that are available for Webex Devices.

Figure 14 Video Stream Layouts for Webex Video Conferencing Devices

You can choose how video streams appear during a meeting by selecting one of the following layout options depending on your device and configuration.

- **Overlay/Active Speaker view**— This layout shows the speaker in the primary video and provides thumbnail videos of the other participants.
- **Equal view**— This grid layout shows an equal view of all the participants. It can be either a 2x2, 3x3, or a 4x4 grid view of meeting participant's video and shared content streams.
- **Single view**— This layout shows a full-screen video of the speaker.
- **Prominent view**— This layout for single screen Webex video conferencing devices display thumbnails of meeting participants above the active speaker.
    - **Prominent view** layouts on dual screen Webex video conferencing devices display the main speaker on the left screen, while meeting participants appear in a 3x3, or a 4x4 grid view on the right screen, depending on how it's registered.
- **Activity view**— This default layout for Webex Board devices enabled with digital whiteboarding allows you to draw, share and modify visual content with others in real time.
- **Minimize content view**— This layout shows the main speaker with any shared content displayed in a nearby thumbnail.
    - The **Minimize content view** layout for dual and triple screen Webex video conferencing devices shows the main speaker and shared content minimized across all screens.

For more information on different layouts for Webex Board and Webex Room and Desk Devices as well as examples of the layouts see Webex Rooms | Video Stream Layouts.

For all intents and purposed the bandwidth utilization is comparable across all layouts except for Equal (Grid) View. As such Active Speaker will be used as it is comparable to most other layouts apart from Equal (Grid). We'll also show Equal (Grid) view as this has a larger impact on bandwidth utilization compared to the other layouts.

Cisco Video Device Bandwidth Utilization Table. Any of the Cisco Webex Devices Cloud registered to Webex will have the following bandwidth utilization, broken into common scenarios and showing values for Equal view.

Table 4 Cisco Device Bandwidth Utilization

Meeting Scenario	Configured Device Calling Rate
Cisco Device Point-to-Point call (3mbps)	**Depending on available network resources, Video endpoints will use approx 70-80% of configured call rate based on lighting/motion, and up to 100% of configured call rate during moments of high motion, and less during low activity points.**
Cisco Device Point-to-Point call (6mbps)	**As above**
Cisco Device Webex Meeting (3mbps)	**As above**
Cisco Device Webex Meeting (6mbps)	**As above**
Cisco Device Webex Meeting - Dual Screen (3mbps)	**As above**
Cisco Device Webex Meeting - Dual Screen (6mbps)	**As above**

Grid View for Devices

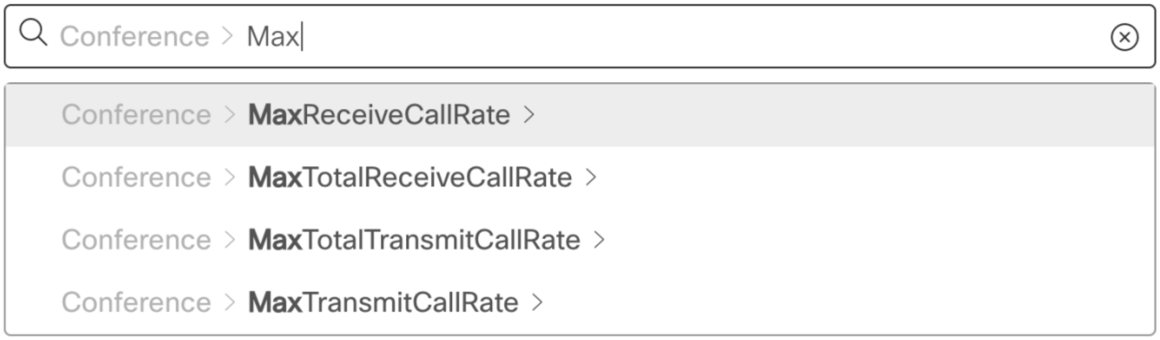
Table 5 Bandwidth vs Resolution, Cisco Devices

Bitrate	**Main Video Resolution**
Up to 1 mb/s	720p in 1:1 448p in meetings
1-2 mb/s	720p in 1:1 448p in meetings
2-4 mb/s	1080p in 1:1 / 2-way meetings 720p in meetings
4-6 mb/s	1080p in1:1 and meeting
6 – 20 mb/s	Main: 1080p60 / Content: 4Kp15

Control Hub Device Level Bandwidth Controls

Control Hub (admin.webex.com) offers the ability to set a minimum and/or maximum transmit and/or receive bit rate for calls for Webex Devices in Webex Meetings. This allows the administrator to reduce the transmit or receive bit rate for specific devices if they are deployed in bandwidth constrained environments. Figure 9 illustrates the Control Hub bit rate settings for Webex Devices. See this document for more information on accessing **Advanced Configurations for Room and Desk Devices and Webex Boards**.

Figure 15 Control Hub bit rate controls for Webex Devices



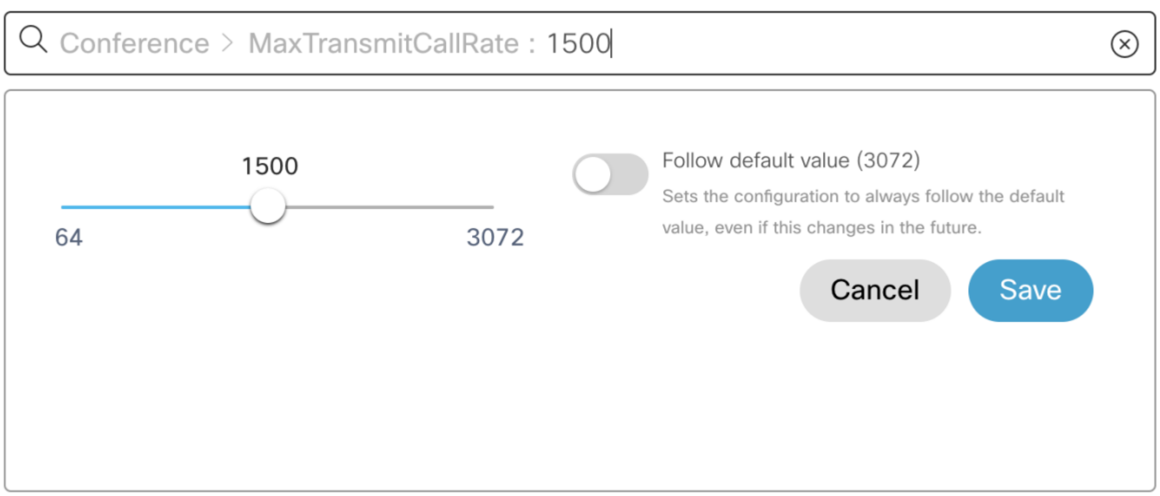
The following are definitions for each bit rate setting:

MaxReceiveCallRate: Define the maximum receive bit rate to be used when placing or receiving calls. Note that this is the maximum bit rate for each individual call and is inclusive of all media bit rate: audio, video and presentation sharing.

MaxTransmitCallRate: Define the maximum transmit bit rate to be used when placing or receiving calls. Note that this is the maximum bit rate for each individual call and is inclusive of all media bit rate: audio, video and presentation sharing.

One example of where it might be applicable to modify the bit rate settings is in environments where the transmit bandwidth is much lower than the receive bandwidth, such as home environments with limited upload speeds. These environments are prime candidates for this type of bandwidth control as the administrator can set the MaxTransmitCallRate to be lower than the device capability while leaving the MaxReceiveCallRate to the default or the maximum bit rate that the device can receive. So, as an example a Cisco Webex DX 80 can support up to 3 mbps of transmit or receive. If the site where the device is located only has a maximum of 2 mbps upload speed then it might make sense to reduce the transmit for this device to something like 1.5 mbps so that during a meeting it doesn't attempt to transmit more leaving 500k for other traffic during the meeting. See figure 10 for an example of this configuration in Control Hub.

Figure 16 Example: Changing MaxTransmitCallRate for a Device

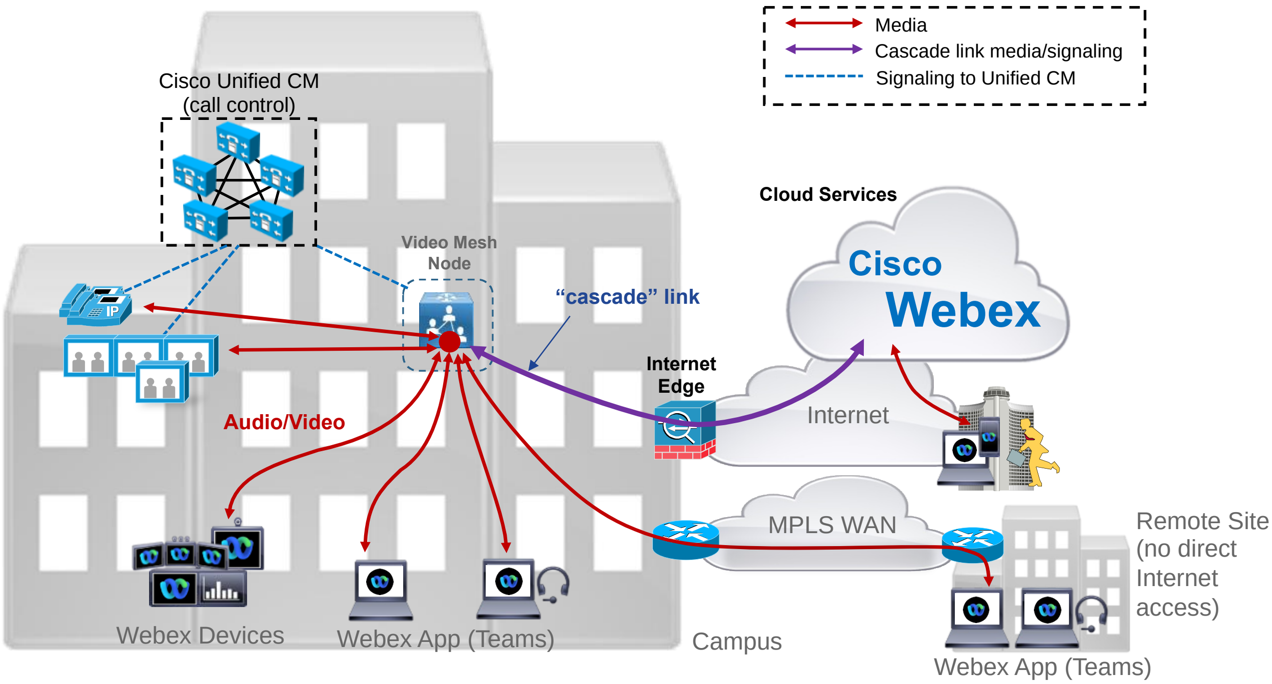


The **MaxTotalTransmitCallRate and MaxTotalReceiveCallRate** settings apply to a video system's built-in MultiSite feature (optional), which is not applicable to Webex Meetings and can be disregarded for the purposes of this document:\

Video Mesh

A unique aspect of a Webex deployment is that it allows enterprise customers to deploy Webex Video Mesh Nodes on their corporate network to optimize media flows. Figure 17 shows a Webex App (Teams) deployment with a Video Mesh Node located in the main site.

Figure 17 Webex Video Mesh Node Forms a Cascade Link to Webex



When a Video Mesh Node is present, Webex App (Teams) endpoints and applications located inside the corporate network automatically detect it and send their audio/video streams to it. If any participants to a multipoint meeting are located on the Internet (for example, the mobile user in Figure 17), they will send their audio/video flows to Webex and a "cascade" link will automatically be set up between Webex and the Video Mesh Node, so that all meeting participants can have the same experience. For more information on the Video Mesh Node, See the Cisco Preferred Architecture for Webex Edge Video Mesh.

The exception to this is for on-premises endpoints connecting to meetings hosted by Webex. It is possible to configure a SIP trunk to an on-premises Video Mesh Node for Webex Meetings so that on-premises SIP endpoints can then connect to a Webex meeting by leveraging the on-premises Video Mesh Node. In this case the media and signaling from the Video Mesh Node for the on-premises endpoints will follow the same path and use the same destination ports as for the Webex App (Teams) clients and endpoints. (Source ports for Unified CM endpoints are configured in the Unified CM SIP profiles.) See the Cisco Preferred Architecture for Webex Edge Video Mesh for more information on integrating the Video Mesh Node for on-premises endpoints to connect to Webex Meetings.

Multistream Capabilities and Bandwidth Management

Webex App (Teams) Webex Devices support multistreaming of video. This allows endpoints to render their own video experience instead of relying on the Cloud Video Services or Video Mesh Node to provide them with a transcoded composite experience. Webex App (Teams) can typically send up to 4 video streams simultaneously. Streams are sent at different resolutions depending on what the other participants in the meeting are requesting based on their layout selection (for example, active speaker, equal layout, and so forth).

Typically, resolutions of 1080p, 720p, 360p, and 180p can be sent. The exact resolutions may be affected by factors such as endpoint type, media assurance, and so forth. The resolutions are usually based on the following layouts:

Active Speaker — 720p for the active speaker and 180p for the other participants in picture-in-picture (PIP) mode

Equal Layout — 360p for all participants

1080p for Video Mesh meetings only, and enablement requires specific configuration on the Video Mesh cluster (see the Cisco Preferred Architecture for Webex Edge Video Mesh for more details.)

Multistream Cascading

The multistream technology applied to Webex App (Teams) also applies to the cascade link of the Video Mesh Nodes. For example, a Video Mesh Node can send multiple video streams per endpoint inside the cascade link to Webex. For more information on Video Mesh cascade links, see the section Cascades in the Cisco Preferred Architecture for Webex Edge Video Mesh.

Classification and Marking

When you deploy Webex App and Devices on an enterprise network across multiple sites, you must classify real-time media flows correctly (that is, identify them as audio, video, or other application traffic) and mark them as close as possible to the media source or whenever they enter the enterprise network domain.

Webex App, Webex Devices, and the Video Mesh Node always attempt to set DSCP for the traffic they originate, as indicated in Table 6. The table also shows the corresponding 802.11 User Priority (UP) values used when the connection is to an enterprise wireless network. While 802.11 User Priority is attempted there are specific actions that are required to ensure that it is functioning on devices across the portfolio. More about that in the section Wireless Considerations.

Table 6 DSCP Values Used by Webex App, Webex Devices, and Video Mesh Nodes

Traffic Type	DSCP (PHB; decimal value)	802.11 User Priority (UP)	Notes
Audio	EF; 46	6	Includes audio streams of voice-only calls, audio streams of video calls, and related RTCP packets
Prioritized video	AF41; 34	5	Includes video streams (main video and presentations or content) and related RTCP packets
Opportunistic video	AF42; 36	5	Includes video streams (main video and presentations or content) and related RTCP packets
Other traffic	Best Effort; 0	0	Includes messaging, file transfer, configuration, call and meeting setup

The DSCP values for media traffic are aligned with the RFC 4594 recommendations and with Cisco's design guidance for on-premises Collaboration deployments. (For more details, refer to the latest version of the *Preferred Architecture for Cisco Collaboration Enterprise On-Premises Deployments*, available at https://www.cisco.com/go/pa.)

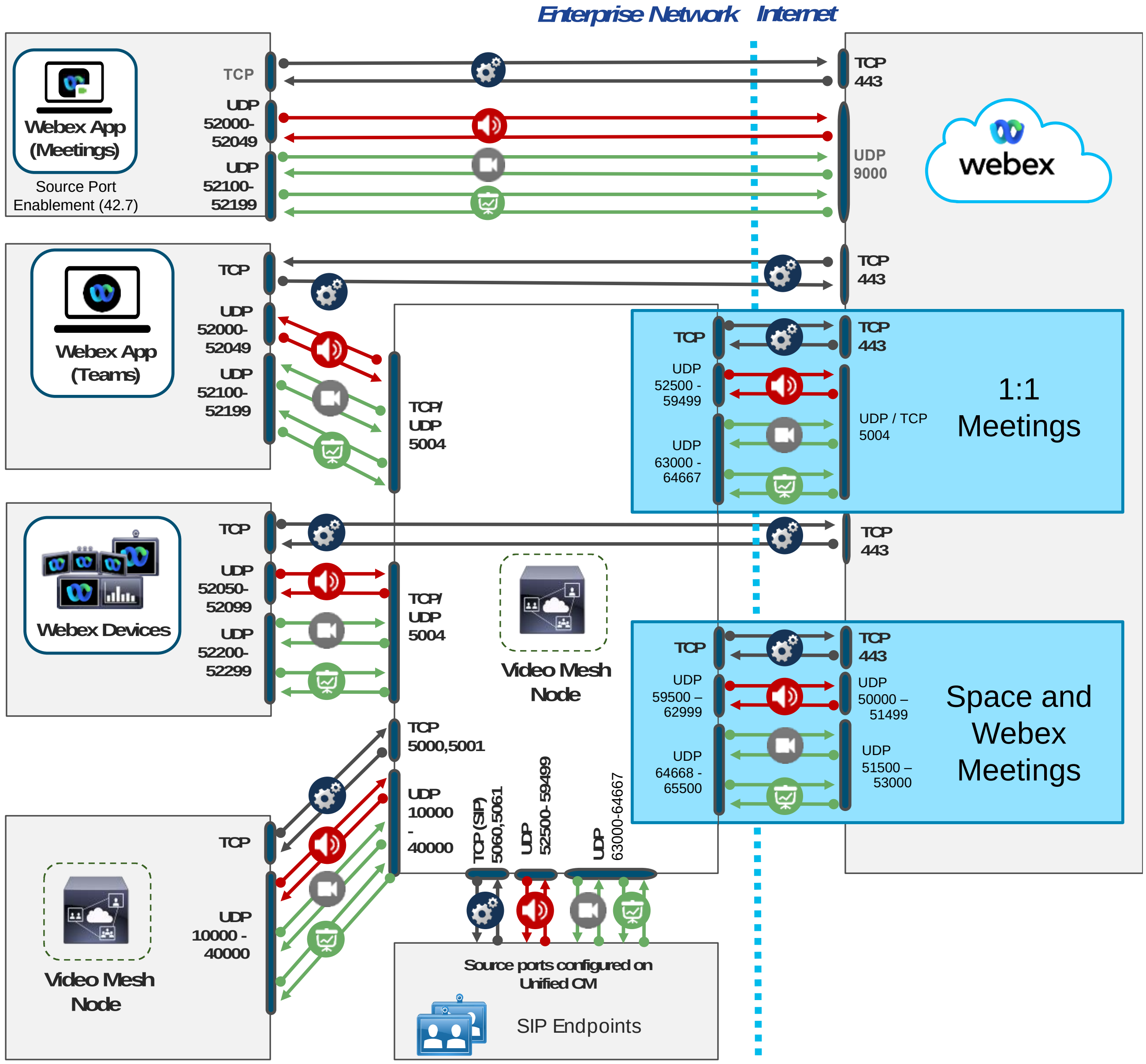
While it is possible to configure your network to pass through the DSCP values natively set by the endpoints, we recommend classifying traffic at the campus access layer to simplify ingress policy configuration. It is also worth noting that DSCP values are not preserved over the Internet, so network-based classification is necessary for media flows that originate from the cloud and are directed to endpoints on the enterprise network.

The ability of the network to identify Webex App media flows relies on a consistent usage of specific UDP port ranges for each media type, which essentially provide identifiable traffic "signatures." These traffic signatures can then be used to create access control lists (ACLs) to reclassify the flows in the network according to the implemented QoS policy. The traffic signatures are also leveraged by Cisco's Next Generation Network-Based Application Recognition (NBAR2) libraries and EasyQoS for easy creation of QoS policies.

When a Video Mesh Node is deployed, the Webex App endpoints and applications use local ports from the same ranges to communicate with it, but media streams are terminated on different port ranges at the Video Mesh Node. A "cascade link" may also need to be created between the Video Mesh Node and the cloud if the meeting has external participants.

The Video Mesh Node requires larger port ranges than endpoints and applications, given the number of media streams that may be terminated by a single node. However, the IP addresses of Video Mesh Nodes are well known to the enterprise network administrator, so specific access control lists can be used to reclassify the traffic pertaining to these nodes if necessary. Figure 18 depicts the port usage for media flows between Webex App endpoints and applications, Video Mesh Node, and Webex Cloud media services.

Figure 18 Port Ranges for Webex Video Mesh Nodes



In summary, Table 7 shows all the traffic signatures for Webex App media flows and the corresponding recommended DSCP settings. In this table, flows are listed in the egress direction, which is from the endpoints toward the cloud, but the same port ranges apply to the flows in the ingress direction, which is from the cloud toward the endpoints. You can simply swap source and destination IP addresses and ports to obtain the traffic signatures for the ingress direction.

Table 7 **Traffic Signatures for Webex App Real-Time Media (Symmetric[1])**

Source IP Address	Destination IP Address	Source UDP Ports	Destination UDP Ports	Recommended DSCP [2]	Media Type [3]
Webex App [4]	Webex cloud media services or Video Mesh Node	52000 to 52049	5004	EF	Audio
Webex App [4]	Webex cloud media services or Video Mesh Node	52100 to 52199	5004	AF41 or AF42	Video
Webex Devices	Webex cloud media services or Video Mesh Node	52050 to 52099	5004	EF	Audio
Webex Devices	Webex cloud media services or Video Mesh Node	52200 to 52299	5004	AF41 or AF42	Video
Video Mesh Node	Webex cloud media services or Video Mesh Node	56000 to 59499	50000 to 51499	EF	Audio
Video Mesh Node	Webex cloud media services or Video Mesh Node	63835 to 64667	51500 to 53000	AF41 or AF42	Video
Unified CM SIP endpoints	Video Mesh Node	Unified CM SIP Profile	52500 to 62999	EF	Audio
Unified CM SIP endpoints	Video Mesh Node	Unified CM SIP Profile	63000 to 65500	AF41 or AF42	Video

[1.] Symmetric in this case means that the same ports are used in the reverse direction where the source port becomes the destination port, and the destination port becomes the source port, for the return media path. For example, if the media source port from a Webex App application is 52010 and the destination port to the cloud is 5004, then the media return path from the cloud will have a source port of 5004 and a destination port to the Webex App application of 52010.

[2.] These values are the recommended values for DSCP marking based on UDP port ranges and not necessarily the "native marking" of the flows.

[3.] As elsewhere in this document, Audio in this table refers to audio streams of voice-only calls, audio streams of video calls, and related RTCP packets, while Video refers to video streams (main video and presentation or content sharing) and related RTCP packets.

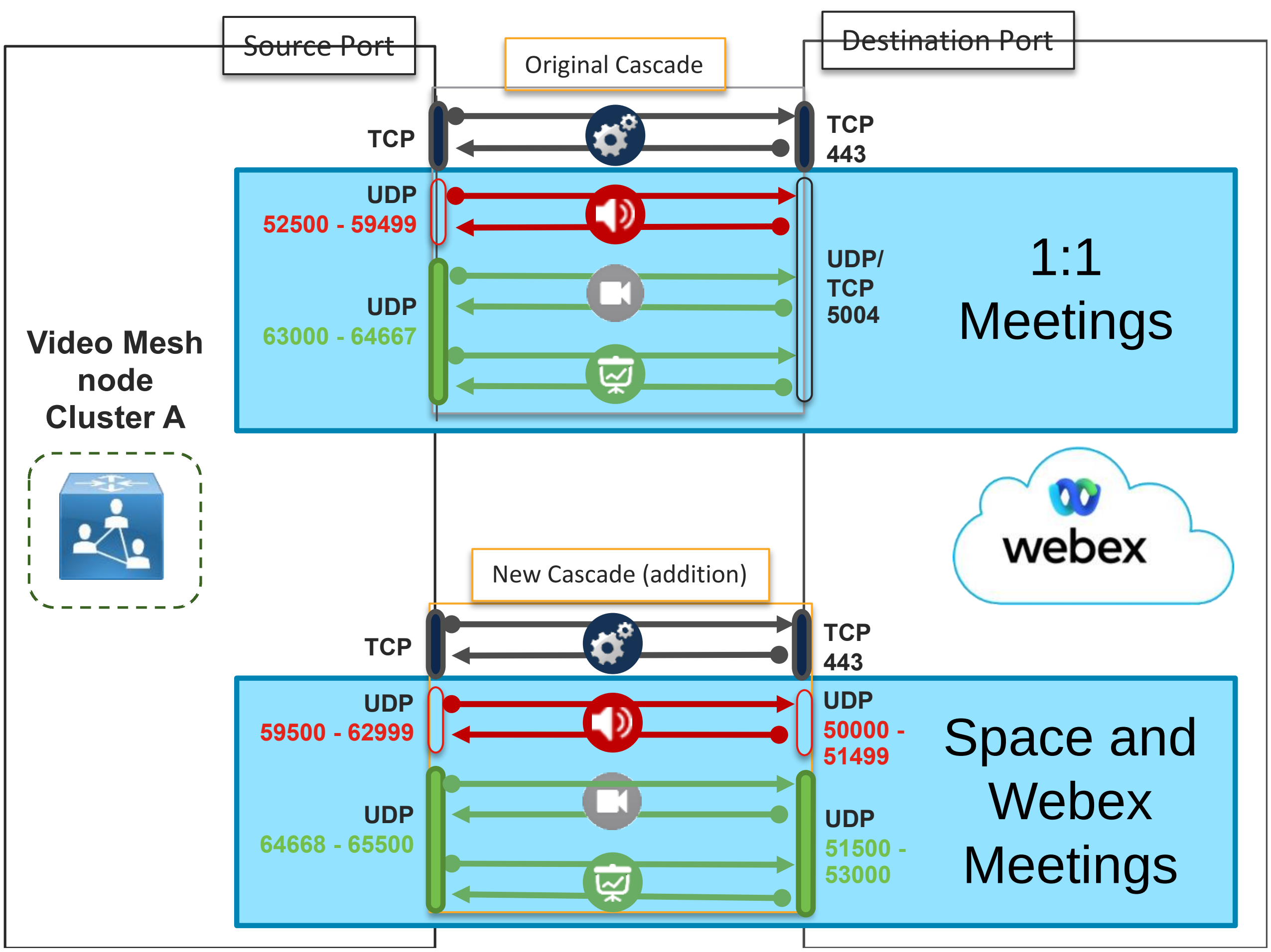
[4.] This table does not currently apply to Webex App for Windows. Webex App for Windows currently uses ephemeral source ports for media provided by the windows OS unless source port enablement is configured. Please see the section on *Cisco Webex App for Desktop and Mobile* for more information.

With the traffic signatures listed in Table 7, you can classify Webex App real-time media using a common access control list (ACL) as close as possible to the network edge – that is, at the campus access layer and at the Internet edge.

If your deployment includes Video Mesh Nodes, you can also classify the audio and video traffic related to the cascade link between the Video Mesh Nodes and the Webex cloud media services, based on the UDP ports shown in Table 7 and the individual IP addresses of the Video Mesh Nodes.

Note: As of February 2022, Video Mesh updated its media cascade port range as part of the continuous improvements in the Webex platform. Video Mesh added new media cascade ports using UDP 50,000 – 53,000 destination ports. 1:1 meetings will still use UDP port 5004 while Space and Webex meetings will use the new UDP 50,000 – 53,000 destination ports. Figure 19 illustrates the changes.

Figure 19 Video Mesh Cascade Port Updates



As seen in Figure 19 the updated source and destination ports are seen in Green and Red. The top set of ports are the original cascade port ranges while the bottom set of ports are the new cascade port ranges. The port ranges shown in Figure 18 and Table 7 document the new cascade media port ranges.

All media traffic for calls to/from on-premises endpoints registered with Cisco Unified CM to/from Webex App meetings, endpoints, or applications is routed through the Expressway pair that is used for interconnecting the on-premises endpoints with the Webex App applications and meetings. The media and signaling DSCP values for the streams from the on-premises endpoints are set by Unified CM. This is covered in detail in the latest version of the *Preferred Architecture for Cisco Collaboration Enterprise On-Premises Deployments*, available at https://www.cisco.com/go/pa.

Media and signaling DSCP values for the streams from Webex cloud media services to the on-premises endpoints are set by Expressway-C on ingress into the enterprise. The media and signaling are marked with the same DSCP settings as all other incoming Expressway media and signaling. Therefore, if the deployed Expressway edge equipment has been installed and configured as part of the *Preferred Architecture for Cisco Collaboration Enterprise On-Premises Deployments*, then nothing more is required on Cisco Expressway.

For deployments where the Video Mesh Node sits in the DMZ, there is a Video Mesh Node configuration setting in the Webex Control Hub that allows the administrator to optimize the port ranges used by the Video Mesh Node. This **Quality-of-Service** setting, when disabled (enabled by default), changes the source ports that are used for audio, video, and content sharing from the Video Mesh Node to the range of 34000 to 34999. The impact of this, however, is that the Video Mesh Node will natively mark all audio, video, and content sharing to a single DSCP of AF41; and since the source ports are the same for all media regardless of destination, it is not possible to differentiate the audio from video or content sharing based on port range with this setting disabled.

If your deployment requires the Video Mesh Node to be deployed in the DMZ, this setting may be helpful to reduce the firewall port openings. For more information on this setting and the impacts, refer to the *Deployment Guide for Cisco Webex Video Mesh*, available at

https://www.cisco.com/c/en/us/support/unified-communications/spark/products-installation-guides-list.html

Queuing and Scheduling

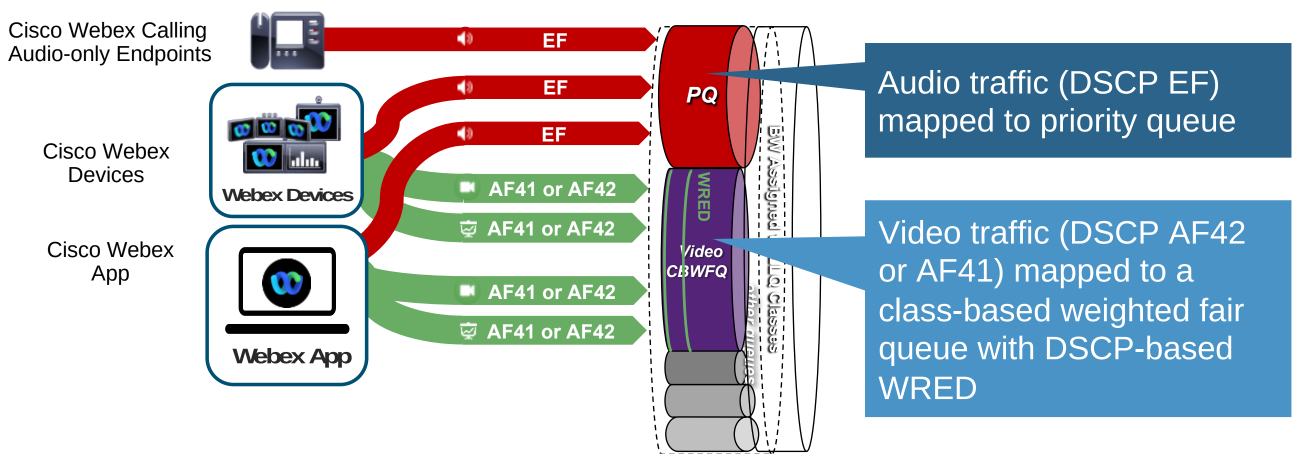
Once real-time media traffic has been correctly identified and classified with DSCP, it can be assigned to the appropriate queues in the network devices it traverses. Because WAN and Internet links are the most common bandwidth bottleneck points in an enterprise deployment, this section shows an example based on the Low-Latency Queuing (LLQ) features found in Cisco IOS routers, but the same considerations can be applied to other parts of the network such as the campus or data center.

In alignment with existing recommendations for on-premises Cisco Collaboration deployments, the WAN queuing and scheduling model adopted here is based on two separate queues for interactive media traffic, and the queue assignment is based on DSCP settings:

- A Priority Queue for all audio traffic marked with DSCP EF
- A Class-Based Weighted Fair Queue for all video traffic marked with DSCP AF41 for a prioritized class of video, or AF42 if an opportunistic class of video is configured

Figure 20 illustrates how media streams from Webex App endpoints and applications are assigned to queues in a Cisco IOS router.

Figure 20 Assigning Webex App Audio and Video Traffic to Queues



In Figure 20 the audio streams of voice-only calls and video calls are marked as EF and placed into a Priority Queue (PQ). Priority queues are generally associated with a policer that limits how much bandwidth can be allocated to the queue, to avoid starving other traffic types.

Video streams (main video and content or presentation sharing) are marked as AF41 or AF42 and placed into a Class-Based Weighted Fair Queue (CBWFQ) with Weighted Random Early Detect (WRED). AF42 marking is used if opportunistic video has been deployed and Webex App endpoints are used as the opportunistic video endpoints. These queues guarantee that the matching traffic will receive at least the configured bandwidth, but they can also take advantage of any unused bandwidth from other queues. WRED is a congestion avoidance mechanism that was originally developed for TCP applications, but it can also be effective with UDP applications that support loss-based dynamic rate adaptation, such as Cisco Collaboration endpoints that implement Media Assure. In a nutshell, when the queue length reaches a certain threshold, WRED preemptively starts to drop an increasing percentage of packets in the queue, thus triggering the loss-sensing rate adaptation algorithm before the tail of the queue is reached.

Bandwidth Provisioning and Capacity Planning

Figure 21 lists audio and video bandwidth recommendations for various types of Webex components. There are several variables that go into determining send and receive bandwidth values. These bandwidth values are simplified for easy calculation as this value will be augmented for both growth and extra overhead to ensure a lower utilization rate of the circuit so that it's not running at or near capacity. It's also worth noting that Webex Apps and Webex Devices have both main video and shared content video. While there are differences in sending and receiving video bandwidth based on devices sharing content vs sending video only, these differences are marginal and not relevant when doing larger scale capacity planning. In general, the larger the number of users used in the bandwidth calculation the more the averages will flatten out with regards to the variability of video bandwidth utilization.

Figure 21 Bandwidth Requirements for Webex Components (Including Layer-3 Overhead)

Webex Component	Audio Bandwidth	Video Bandwidth Average (Max)	High FPS Content Share	Grid 5x5
Webex App (Meetings)	100 kbps	2 mbps (3 mbps)	1mb - 2.5 mbps	5 mbps
Webex App (Teams)	170 kbps	2 mbps (4 mbps)	1mb - 2.5 mbps	NA
Video Mesh Deployments (Webex App (teams), Unified CM Integration)	600 kbps	12 mbps (20 Mbps)	NA	NA
DX Series, SX10, MX Series, SX20, SX80, Webex App Room Kit, Webex Board *	100 kbps	2 mbps (4 mbps)	NA	NA
Expressway Edge Deployments (Webex Edge Audio / Webex Video Meetings / Webex Hybrid)	80 kbps	2 mbps (Customer Configured**)	NA	NA

* For dual screened systems double the video bandwidth utilization

** Expressway video flows maximum bitrate is configured on the Unified CM SIP Profile