Troubleshooting Audio and Video Quality with Webex Control Hub

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Introduction

At the core of Webex meetings is the audio and video communication that occurs between participants. While this communication is usually a great experience, on rare occasions the audio and/or video quality can be degraded. This degradation can be caused by issues anywhere along the audio or video path, including laptop resource issues, network issues, or work-from-home bandwidth issues. These issues can lead to a poor meeting experience for attendees.
With a focus on end user environments, this document is targeted at Webex administrators, partners, and other Control Hub users who have a working knowledge of voice and video in the context of Webex meetings. Various features and functions for audio and video troubleshooting are available in Webex and knowing how and when to apply these improves the resolution time for these issues.

Divided into four main sections, this white paper teaches voice and video fundamental concepts and then applies these to troubleshooting real world use cases using the Webex administrative platform, Control Hub. The first section is Audio and Video Fundamentals. In this section, foundational elements covering topics, such as the digitization of audio and video, Quality of Service (QoS) impairments, and UDP versus TCP transport are discussed. Understanding these topics is critical for establishing a proper foundation in how audio and video communication streams work in today's networks.

In the next two sections, Audio in Webex Meetings and Video in Webex Meetings, an introductory look is taken at how audio and video work in general, but also in the context of

Webex meetings. After completing these sections, you will have the proper foundation for learning how to effectively troubleshoot audio and video quality issues in Webex meetings.

The section, Control Hub Audio and Video Troubleshooting Analytics, guides you through the various functions and capabilities in Control Hub. At the conclusion of this section, you should feel comfortable moving through the various Control Hub screens and knowing where to go to get more detailed information at the meeting and individual user level.

The last section, Use Cases and Troubleshooting Scenarios, shows you real world examples of audio and video quality issues. With each example, you are guided step by step through the various screens in Control Hub that lead to the successful resolution or definitive next steps to be taken for that particular problem.

By the end of this document, you should feel confident handling audio and video troubleshooting issues from your users or your customers. This document can also serve as a handy reference for future consultation.

> ***Note****: This white paper is focused exclusively on troubleshooting Webex Meetings audio and video* <u>*quality*</u> *issues, not meeting join or other issues that may occur prior to the establishment of the audio and video connections. Quality issues typically deal with audio and video streams that present with impairments of some sort that in turn affect the viewing or hearing of the media.*

Audio and Video Fundamentals

In the not-too-distant past, telephony and data networks were separate. Data traffic traveled over common Local Area Network (LAN) and Wide Area Network (WAN) technologies like ethernet and MPLS using routers and switches. Meanwhile, telephony traffic was carried over FXO/FXS connections and T1/E1 circuits using Private Branch eXchanges (PBXs) and telephony carrier switches connected to the Public Switched Telephone Network (PSTN). Starting around the turn of the century, the Internet Protocol (IP) telephony revolution, however, completely changed this paradigm.

IP Telephony merged voice and video onto the existing data network infrastructure. IP at this point was becoming the ubiquitous data protocol of choice and so it made sense to use this for transporting telephony information as well. The analog waveforms of traditional voice and video were digitized and transferred across data networks in packets. The ability to take sound and light, which is passed using analog waveforms in the real world, and turn that into a digital representation of 0s and 1s is the foundation that allows the transmission of voice and video over today's packet networks.

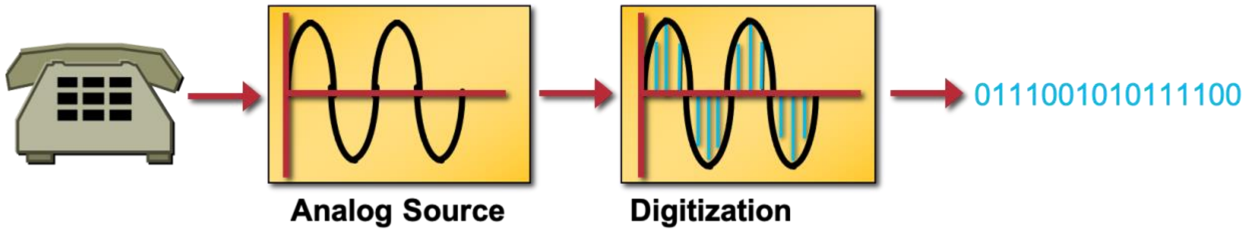


Figure 1: *Digitizing an Analog Signal*

In Figure 1, a high-level overview of the digitization of an analog signal is shown. In this case the source would be speech from a phone, but it just as easily could be your voice going into a headset on an IP phone, a teleconferencing system, or even your PC. No matter what you are talking into, this same process must occur. The analog speech coming out of your mouth must be digitized to go over an IP network.

> ***Note****: Both audio and video must be digitized. While Figure 1 is representative of audio, the light captured in video is also an analog waveform initially. This must also be captured by the camera on your PC, video conferencing system, etc. and digitized before it is transmitted.*

Both audio and video can be digitized or encoded using a number of different algorithms. These algorithms are usually referred to as codecs, which is short for coder-decoder. Depending on the codec being used, the audio and video stream that is encoded will typically vary in quality, bandwidth, and amount of compression. Naturally, the best audio and video quality is desired, but this usually requires more bandwidth and often more processing power as well. Therefore, balancing several factors are necessary when audio and video codecs are selected by a device.

For audio specifically, the digitized stream produced by a codec is divided into segments and packetized. The amount of audio information stuffed into each packet is determined by the packetization period of the codec. This is also referred to as the packetization time. For example, if a codec has a packetization period of 20ms, then 20ms worth of the audio conversation is placed into a packet and sent. Then 20ms later, another packet is sent with the next 20ms of information for that conversation. What this means from a troubleshooting perspective is that for each audio packet you lose, the packetization period determines how much of the conversation is lost. Losing one packet is rarely a problem but groups of consecutive packets are much more noticeable.

Webex Meetings supports multiple video and audio codecs to be compatible with many different endpoints. Webex negotiates with each endpoint or system connected to it to ensure ideal codec selection. If a codec match is not possible then codecs can be converted or transcoded to allow for interoperability. Transcoding can occur at the endpoint locations or elsewhere in the network, including the Webex cloud in some cases.

Note*: With Webex Meetings, the audio and video codecs are selected automatically, so end users are rarely ever aware of the codec that was chosen or if transcoding is occurring. For audio connections, Webex Meetings prefers the Opus codec. For video streams, Webex typically uses H.264 and more recently, AV1.*

Impairments

The convergence of voice and video traffic with traditional data traffic, like web and email, has posed some challenges. One of the biggest has been that voice and video are real-time traffic. This means that packets of voice and video must arrive at their destination quickly, with little delay. While it may be acceptable for an email to take a few seconds or even minutes to make it to its recipient or for a web page to take a few seconds to load, no one is ok with seconds of delay in their voice and video communications.

There are three main factors that impact the quality of an audio or video call. These factors are **packet loss**, **latency**, and **jitter**. As shown in Figure 2, **packet loss is simply losing one or more packets within a stream of packets**. In this example, a Voice over IP (VoIP) packet is lost going from the Webex client to the Webex server. Loss can occur for a number of reasons in a network but most commonly it is caused by congestion or resource contention in the network or on the endpoints. This leads to routers, switches, or the endpoints themselves dropping or delaying packets. In other cases, packets encounter such a high delay that they are received too late to be played out. These packets will also be counted as lost.

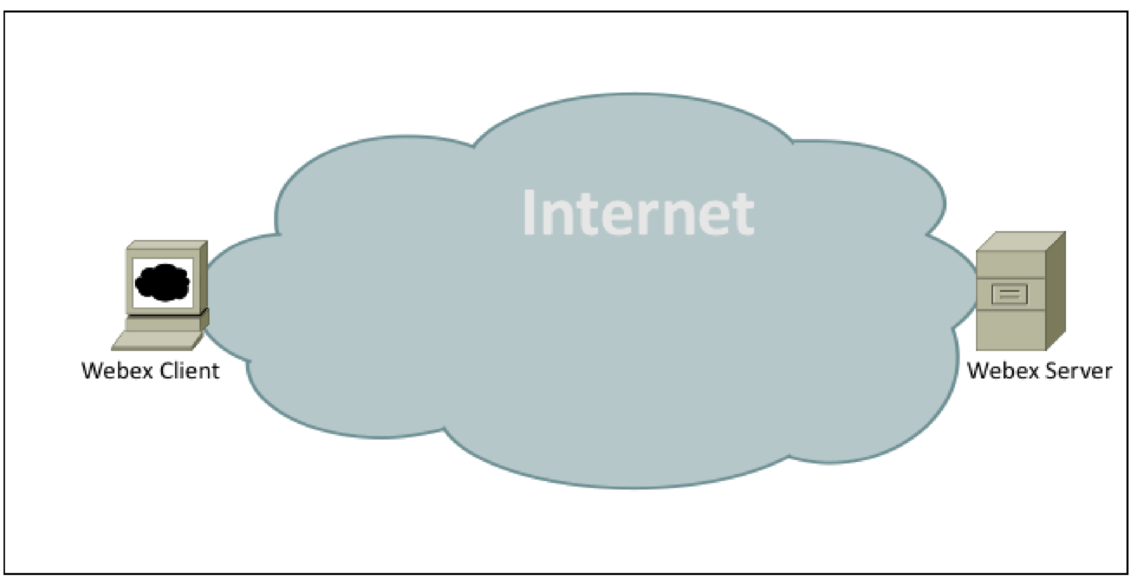


Figure 2: *Packet Loss Example*

Latency is the one way delay between when a packet is sent from one device and received on the other. Latency is part of Round Trip Time (RTT), which is the time it takes for a packet to be sent and for a response to be received. Figure 3 illustrates delay for a VoIP packet.

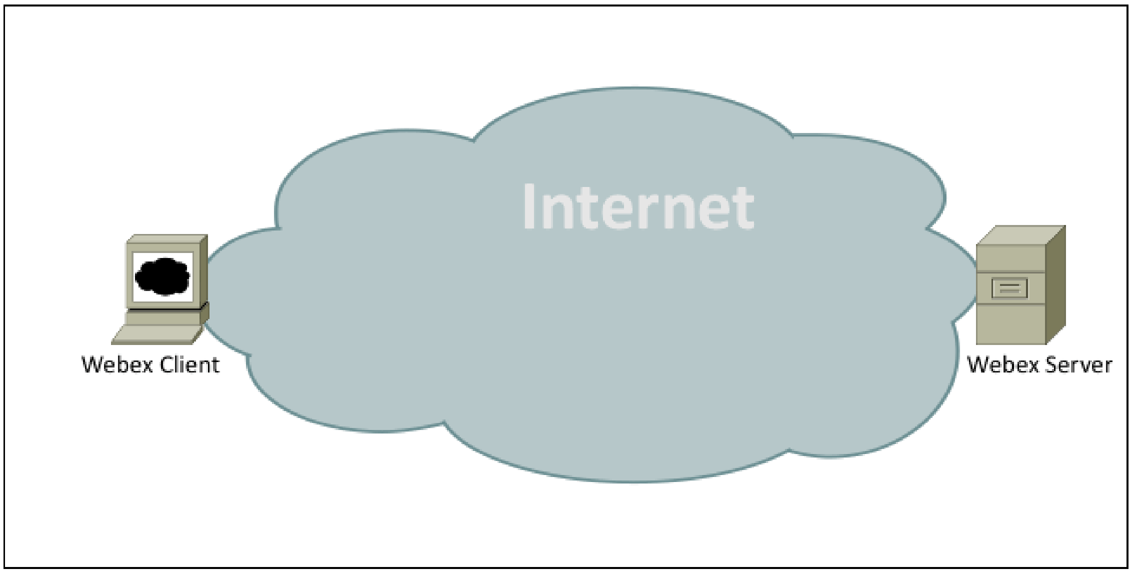


Figure 3: *Latency Example*

Jitter is a little more complicated than loss or latency. **Jitter refers to the variance in the latency between packets**. For example, if two packets arrive 5ms apart and then the next packet arrives 100ms later, the difference in the interarrival times or the jitter between the 3 packets is 95ms. Figure 4 provides a graphical example of jitter. You can think about jitter as the difference in sizes between the two arrows (representing time) between the VoIP packets.

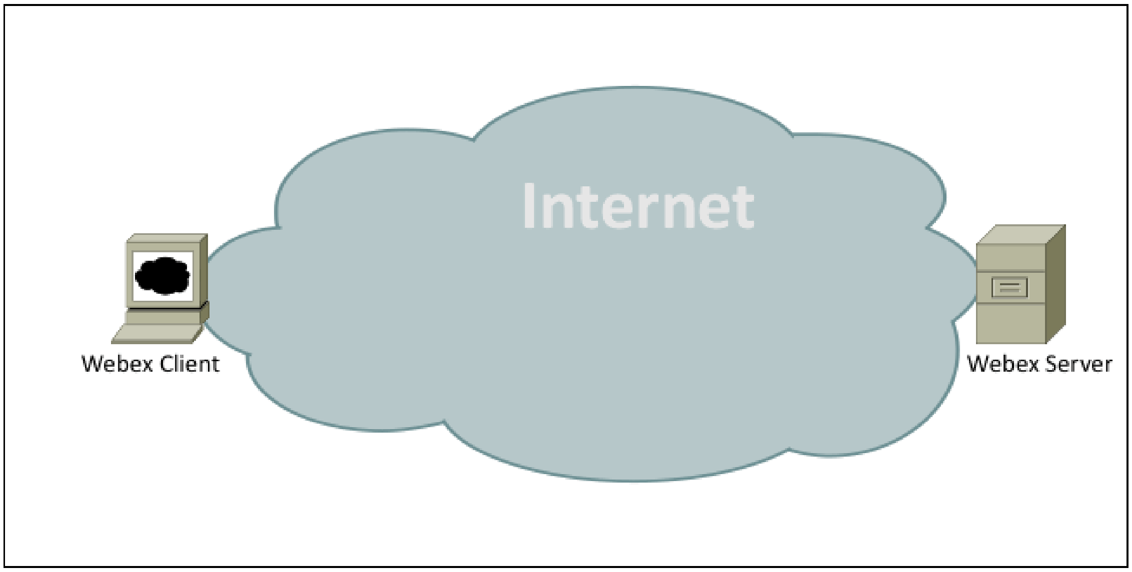


Figure 4: *Jitter Example*

> ***Note****: Impairments can occur anywhere along the path of the audio or video stream. While Control Hub will usually show that an audio or video stream has been affected by packet loss, jitter, or delay, it is usually difficult to pinpoint the exact location. When possible, it is recommended to start at the endpoint and then work towards the Webex Cloud to isolate the issue.*

Effectively handling the real-time nature of voice and video communications over IP networks requires prioritization of these packets throughout the network infrastructure. This is commonly referred to as QoS or Quality of Service. In addition to prioritization, QoS also focuses on handling impairments that may cause the loss or delay of the voice and video packets.

Because of the real-time nature of voice and video communications, QoS is critical to ensuring optimal quality. While there are a number of mechanisms that Webex can employ to compensate for issues related to QoS, it is still recommended that packet loss, jitter, and delay be kept to a minimum whenever possible.

> ***Note****: Control Hub Diagnostics highlights potential problems with packet loss and delay by coloring voice and video streams as Good, Fair, or Poor. It is important to understand that just because Control Hub Diagnostics flags part of a conversation as Fair or Poor, it does not necessarily mean that the user had a bad experience. Voice and video compensation mechanisms and algorithms in combination with other factors can mitigate the impact of loss and delay to the point where it is not noticeable by the user.*

TCP Versus UDP

In the world of IP, one of two transport layer protocols is typically used by an application. These two options are the Transmission Control Protocol (TCP) and the User Datagram Protocol (UDP) as shown in Figure 5. The purpose of these transport layer protocols is to provide end-to-end delivery of IP packets.

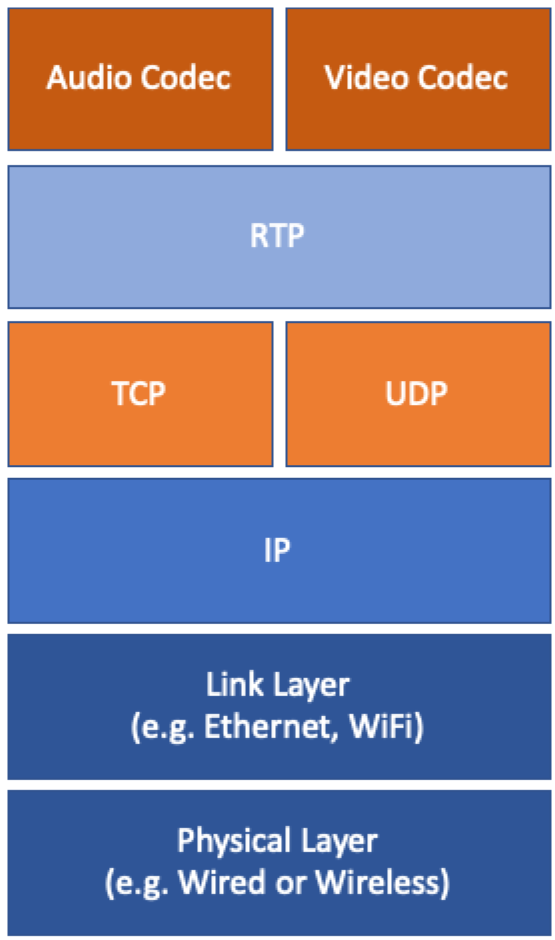


Figure 5: *Common VoIP Protocol Stack*

Both TCP and UDP as shown in Figure 5 commonly use the Real Time Transport protocol (RTP) to carry payloads from an audio or video codec. Designed specifically for streaming media, RTP was constructed to help detect issues, such as jitter, packet loss, and out of order packets.

TCP and UDP take two completely different approaches in providing for the end-to-end delivery of IP packets. TCP takes a connection-oriented approach. Before any IP data packets are sent, each side of the TCP transaction agrees to a session using a handshake. Then, TCP guarantees that every IP packet makes it to the other side by using sequence numbers, acknowledgements, and retransmissions when required. This makes TCP “slower” than UDP but ensures that all data transmitted is received reliably.

On the other hand, UDP handles delivery in a connectionless manner. It is not necessary to establish a session at all with UDP and IP packets can be sent immediately. UDP also does not guarantee a packet’s arrival at the destination because it does not retransmit packets that are lost. While UDP may initially sound like it is not useful in its functions, for applications involving real-time data, like voice and video, UDP is critical.

The reason that UDP is used when transporting voice and video codec payloads, is that it is lightweight and retransmitting lost data is not necessary. If an audio packet is lost during a conversation occurring using VoIP, by the time the origination is notified that a packet has been lost and it is resent, it is too late. That part of the conversation has already been played out at the other end. In most cases, real-time audio and video sessions do not have time to resend lost or even late packets. This is why UDP is recommended, and almost always used, for the audio and video traffic.

On occasion, UDP is not used for audio and video IP packets. This is most often caused by UDP ports 9000 and 5004 being blocked by a firewall or similar device. When a Webex client is unable to communicate with the Webex cloud using UDP, it will attempt to use TCP as a fallback mechanism. However, for the reasons mentioned previously, TCP is not ideal, and audio and video quality may be degraded more severely as compared to UDP if impairments are encountered.

With Control Hub the usage of either UDP or TCP for each user is shown. In Figure 6, you can see a screenshot of the Webex Meeting Participant Details that can found by looking at a specific meeting after clicking on Troubleshooting from the selections on the left side of the screen. Circled in red in Figure 6 are the audio and video transport layer protocols being used for the audio and video media. In this case, it is UDP for both. You should also note that below this information are the specific audio and video codecs being transported by UDP.

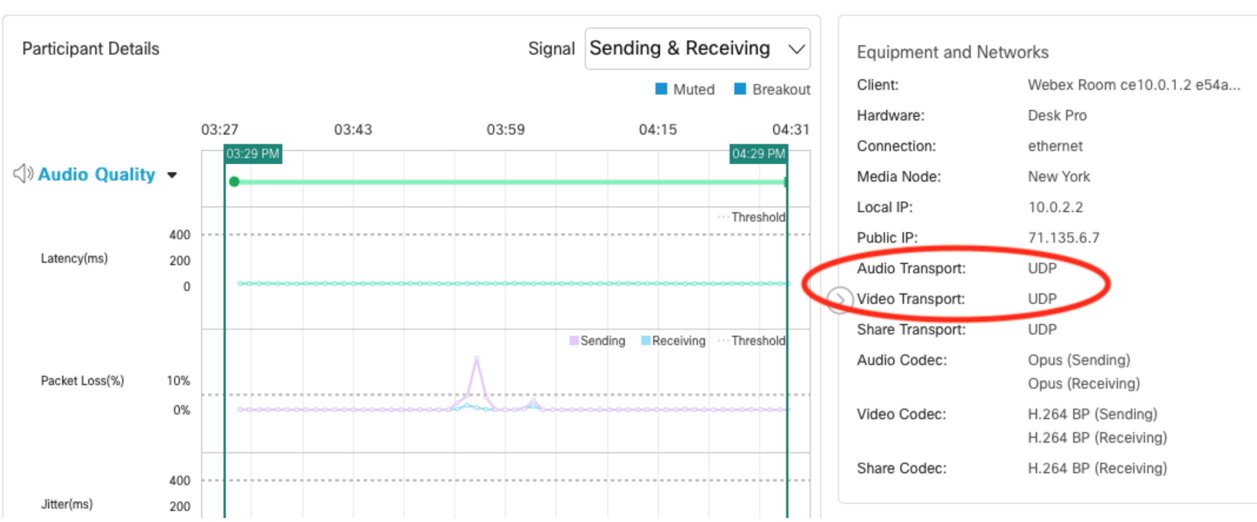


***Figure 6**: Screenshot of Control Hub Participant Details Highlighting the Transport Layer Protocol Being Used by a Participant*

***TIP**: UDP offers the best performance for audio and video media streams and is the preferred option from a Webex perspective. Devices performing firewall or NAT functions may restrict UDP traffic. When this occurs, Webex will fall back to TCP as the transport protocol, but this can result in reduced audio and video quality when impairments are encountered compared to UDP.*

Audio in Webex Meetings

Audio is an important part of the meeting experience. Therefore, audio quality is critical for ensuring a productive and successful meeting. A number of factors can impact a participant's audio quality, including their join method and connection type, codec selection, endpoint device, and the participant's surrounding environment. In this section, you will learn more about these factors and how they can impact audio quality.

Webex offers different join methods for connecting the audio path to a user. Depending on licensed options, the user can select the option they want to use when they join the Webex conference and can even change their selection during the meeting, if desired. Each of the join methods will utilize either VoIP or the PSTN for connecting the user to Webex. Table 2 overviews the Webex methods for joining a conference.

Webex Join Method	**Connection Type**
Device Audio – Utilize the speaker and microphone and the Internet connection for the device joining the meeting for audio.	VoIP
Call Back – Allows the participant to enter a phone number for the Webex service to call them at.	PSTN
Call In – Participant calls in to Webex from a phone and can connect to the conference.	PSTN

***Table 2**: Overview of Webex Meeting Join Methods*

When Call Back or Call In are used to join a Webex meeting, the connection between the participant's device and Webex occurs through a third party PSTN provider in most cases. This provider often uses VoIP internally but there is little to no visibility into this PSTN provider's network and the audio connection that is traversing it. Naturally, this makes any audio quality issues that occur for Call back or Call In a little more difficult to troubleshoot.

For example, let's take the following scenario shown in Figure 7. Two participants, David and Arun, join a Webex meeting. Arun joins using Device Audio and David prefers a call back to his desk phone.

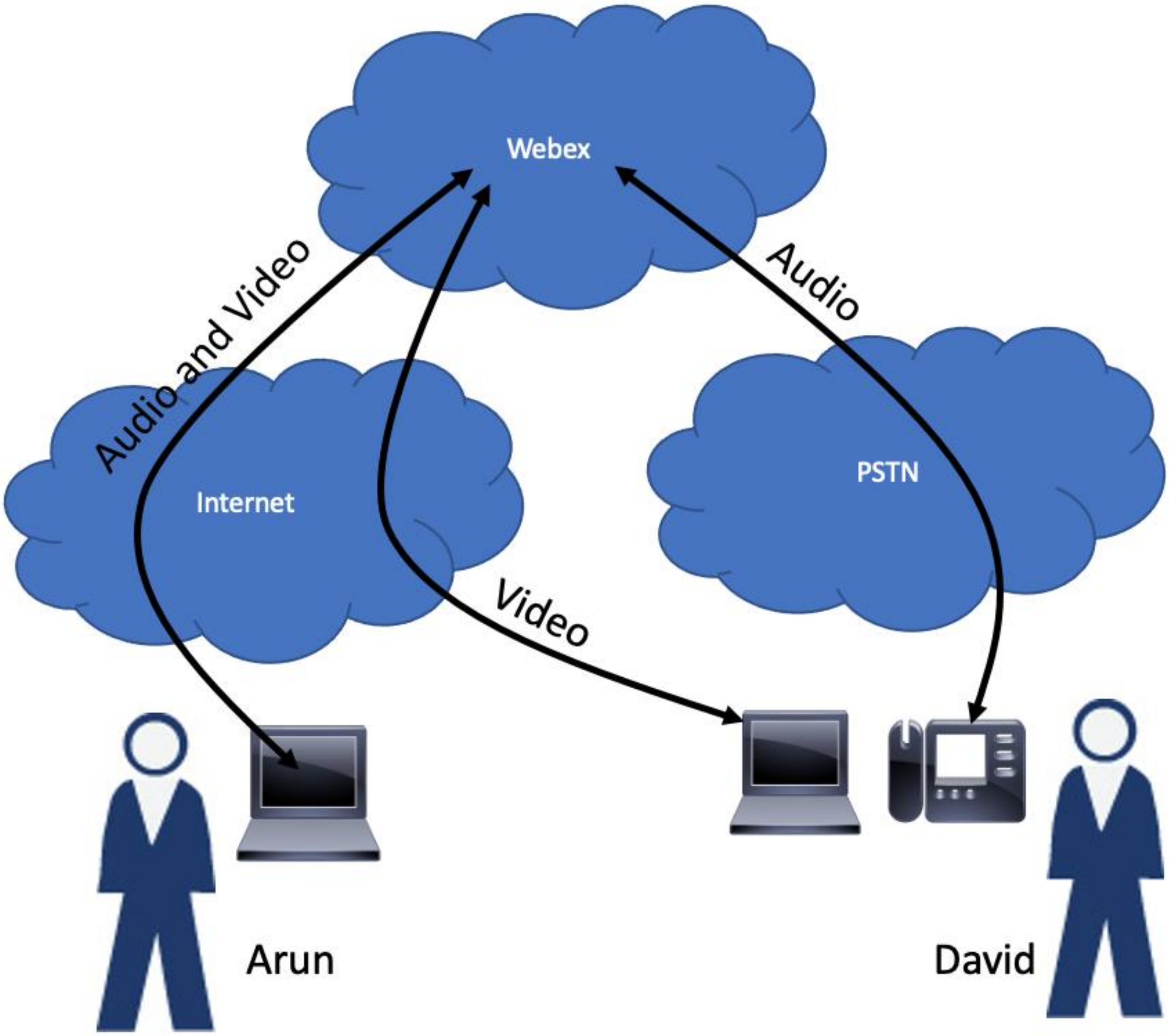


Figure 7*: Webex Meeting Showing a Call Back Participant and Device Audio Participant*

In Control Hub you can view this meeting and naturally David and Arun are shown as the participants. With the Audio tab selected in Figure 8, you will notice that David is listed twice as a participant though.

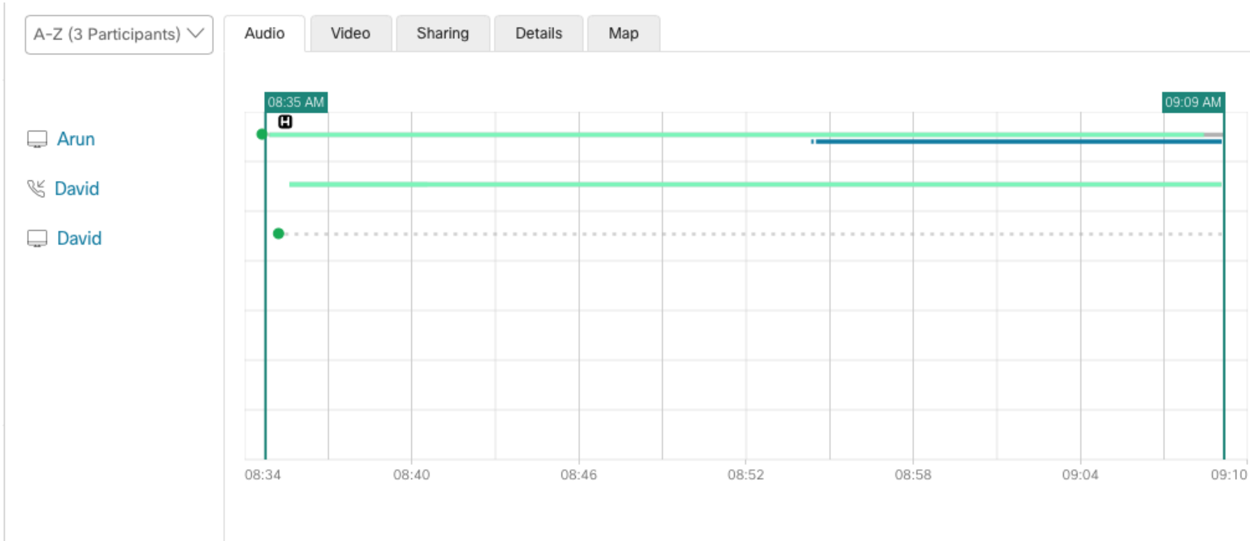


***Figure 8**: Webex Control Hub View of a Meeting with a Call Back Participant and Device Audio Participant*

The participant David in Figure 8 joined via the Call back method. This is why he is listed twice as a participant. The first listing is the audio path between David and Webex using Call Back as notated by the phone icon beside his name. The second listing is the IP connection between the device David is using and Webex. This connection is used for management and device communications that can optionally enable other services like Video or screen sharing. If video is enabled for a PSTN user, it will flow over the device's Internet connection. Clicking on the Call Back connection for the participant David shows more details about the PSTN connection as detailed in Figure 9.

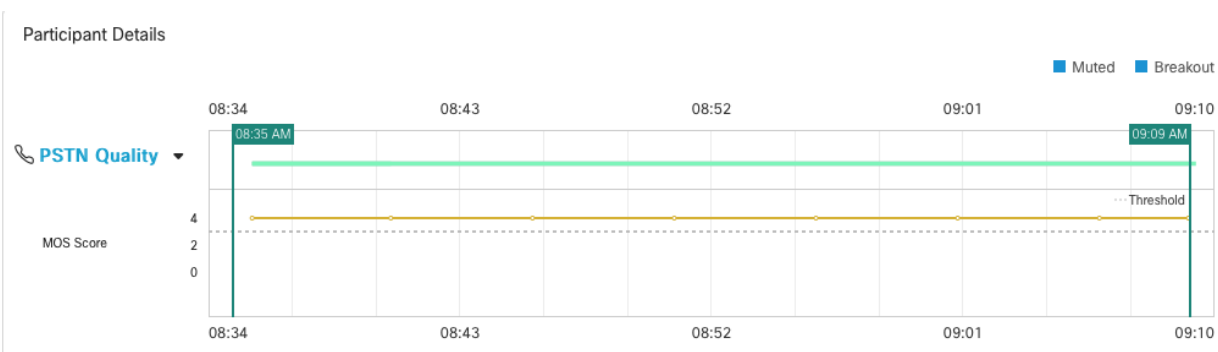


***Figure 9**: Control Hub Audio Quality Details for a Call Back Participant*

As you can see in Figure 9, the troubleshooting information available for this Call Back join method using a PSTN provider is quite limited. You can get a general sense of the audio quality being received by Webex from the PSTN connection through the Mean Opinion Score or MOS. However, the MOS value provides no details about what may be causing a low score.

The other meeting participant for the conference in Figure 7 is Arun and he joined using Device Audio as his meeting join method. In this case, a VoIP stream is created between Arun's device and Webex. A negotiation takes place using the SIP protocol where a number of different parameters are determined, including the audio codec. Once an audio codec has been agreed upon, a bidirectional stream of packets that handles the incoming and outgoing audio for Arun is constructed. Because of this direct negotiation and connection to Webex, Control Hub offers more insight into the audio stream. In Figure 10, you can view the audio details for Arun.

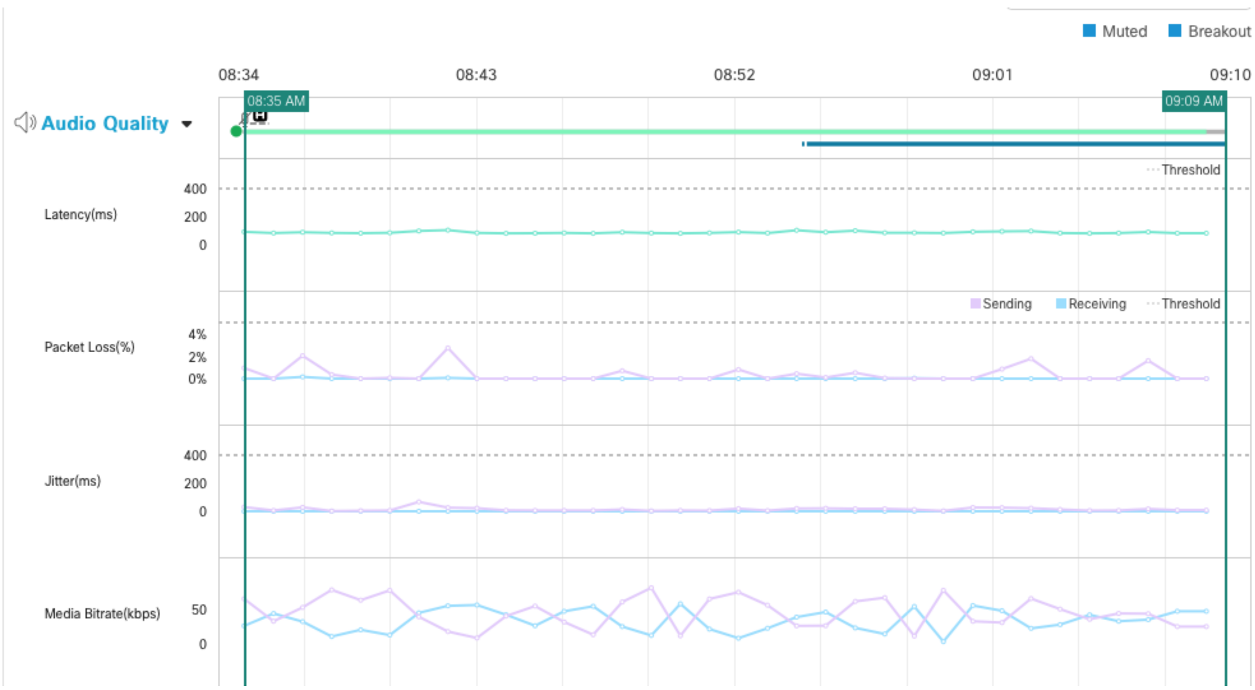


***Figure 10**: Control Hub Audio Quality Details for a Device Audio Participant Using VoIP*

As mentioned earlier in this document, Webex supports various audio codecs so that it can effectively support just any type of device. However, the preferred codec of Webex is Opus and you will often see this listed as the audio connection codec when troubleshooting issues in Control Hub. The Opus codec is a wideband codec optimized for audio in meeting environments while still being highly resilient to impairments, like packet loss. More information on the Opus codec can be found at https://opus-codec.org

Opus and just about all other audio codecs are able to utilize algorithms that are either run on dedicated Digital Signal Processors (DSPs) or CPU resources to deal with impairments, like packet loss, delay, and jitter. In fact, some of these algorithms are even specified and built into the codecs themselves. These algorithms can "fill-in" for missing or late VoIP packets. The algorithms can be quite simple where just silence is played out to more complicated schemes that interpolate and conceal for missing audio samples. The amount of compensation for loss packets that can be achieved with these types of algorithms is impressive and gives audio streams the ability to deal with packet loss in a manner that is usually unnoticed by the meeting participant.

> ***TIP***: *In Control Hub, any audio quality issue detected is highlighted for transparency, but these issues often go unnoticed by the end user, unless the issue is extreme or happens in a short time interval. As mentioned in this section, algorithms and other methods are able to compensate for most impairments that impact a VoIP stream. Exceptions often occur when a burst of errors occur over a short time frame. For example, losing 100 VoIP packets over the course of a 1 hour call is not noticeable but losing that many over the course of a few seconds is much more impacting to the participant.*

> ***NOTE***: *Video streams are unable to use the same algorithms and mechanisms to the same extent as audio streams when it comes to handling packet loss and other issues. Therefore, meeting participants will sometimes notice problems with their video and not their audio, depending on the severity of the impairment that is being encountered. Video compensation for packet loss and other impairments is discussed in detail in the upcoming section, "Factors That Influence Video Quality".*

While having more visibility to the audio stream in Control Hub is a decided advantage when participants use Device Audio for a VoIP connection, the device's audio speaker and microphone along with the device's surrounding environment can sometimes be a disadvantage. PCs, laptops, tablets, and so on are not necessarily optimized for VoIP. The quality of the speakers and microphones can be poor and usually they just act as a speakerphone of sorts. Additionally, these devices can have resource constraints as the CPU and memory must be shared with other applications. This can result in audio packets not being processed in a timely manner, which can cause poor audio quality.

The environment where these devices are located can also be a problem. Often, the background can be noisy and microphones on these devices are designed to be omnidirectional to pick up all surrounding sounds. This can add additional noise to the audio path, which negatively affects the audio quality. For these reasons, participants sometimes choose the Call Back or Call in Webex join method. This enables them to use a dedicated phone device optimized for human speech that usually allows for a headset or handset to be used as well to cut down on background noise.

However, with the appropriate equipment and environment, VoIP audio is superior to PSTN audio. Even with some background noise present Webex implements specialized background noise removal algorithms that have excellent performance in most cases. Additionally, VoIP audio utilizes wideband codecs instead of the narrowband codecs used by PSTN audio. Wideband codecs are of much higher quality and fidelity than narrowband codecs.

In most cases, the decision to use Device Audio, Call Back, or Call In is the choice of the participant and they pick the method that is best for them and their situation. At the same time, it is important to understand the advantages and disadvantages of these join methods, especially when it comes to resolving issues with them in Control Hub.

***TIP**: Having participants mute themselves when they are not talking is more important than is often realized. Poor audio quality (and even other issues like meeting echo) can sometimes be related to end devices and their external environment. Muting resolves these issues in a simple manner. The option to Mute is available to all participants and should be heavily encouraged, especially on meetings with large numbers of participants.*

Video in Webex Meetings

The number of participants who turn on video in virtual meetings has increased dramatically in the recent years. Participants feel more intimate and involved as video enables them to get connected at a human level and naturally convey their thoughts and feelings through facial expressions. Video also allows participants to visualize the impact of their words and conversation. For example, it empowers teachers to get insights into whether the remote student in virtual learning classrooms is paying attention or needs more help to understand the concept being taught.

Video quality plays a critical role in the overall satisfaction of an end user's Webex meeting experience. It is essential for Webex administrators to quickly analyze video quality issues reported by users and take swift action to resolve them. Often, media quality issues don't get reported and hence it is highly recommended to proactively monitor Webex meeting analytics to determine trends that would impact end users. Having a solid foundation of what is video, how it works, common artifacts observed in a meeting, and factors that influence video quality will prepare administrators to isolate the source of the quality issue quickly and engage the right person to address the issue.

Video is a sequence of images played at a specific rate. The images are captured by the video camera in the end user's device (e.g., computer, smartphone, or video endpoint) and is made available to the video encoder as raw video frames. The encoder uses a codec such as H.264 or AV1 to convert raw video frames into IP video streams that are then transmitted to the Webex cloud. The encoder can also use H.264 Scalable Video Coding (SVC) method to transmit the same video in different video resolution formats. This enables Webex Cloud to minimize the need to use transcoders in order to send the video stream of correct resolution based on the meeting client needs and capabilities.

The meeting client application or the endpoint requests one or more video streams from Webex cloud depending on the meeting layout selected by the end user. The Webex cloud determines the active speaker and sends the requested number of participant video streams to the meeting client in different video resolutions as shown in Figure 11. (for example, high resolution for main video showing the active speaker, standard resolution for videos shown in participant panel and low resolution for video thumbnails).

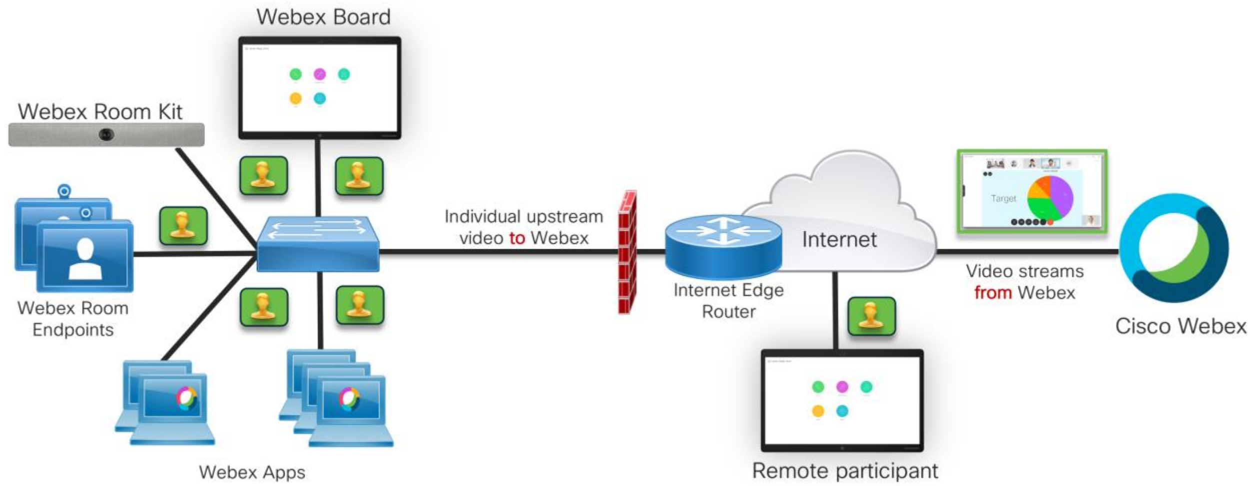


Figure 11*: Video Streams in a Webex Meeting*

The IP video streams used for video conferencing consists of two primary frame types:

1. I-Frame – Intra frame. This frame is larger in size and contains information about <u>all</u> the pixels within a video frame.
2. P-Frame – Prediction frame. This frame is much smaller than the I-Frame and contains information only about pixel changes since the last I-frame. It is generated by using the I-Frame as the reference frame and is sent more frequently as shown in Figure 12.

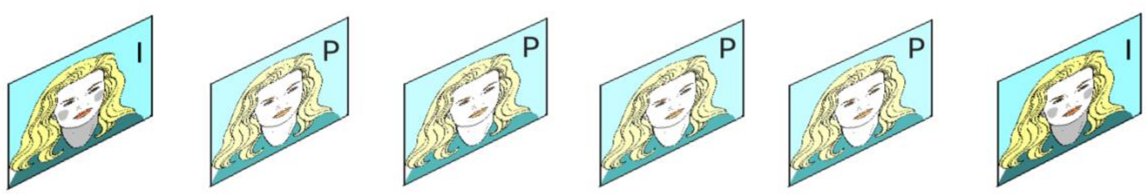


Figure 12*: Video Frame Types in a Conference Call*

Video resolution and frame rate are two key parameters that define the video quality at the source device. Video resolution specifies the number of pixels in the horizontal direction (width) by the number of pixels in the vertical direction (height) captured in each video frame. The higher the video resolution, the better the video quality and higher the bandwidth requirements.

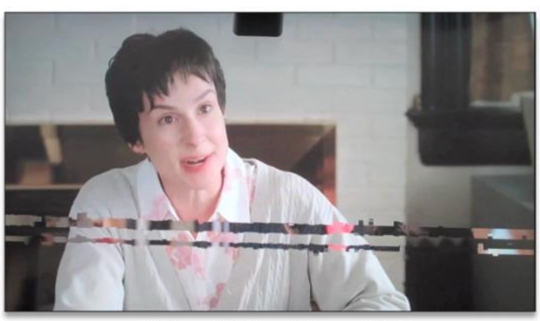
Frame rate specifies the number of video frames per second. Higher frame rates are particularly useful when there is frequent motion in the given scene. Typically, the primary video stream in a Webex meeting that contains the participant's face is sent at a higher frame rate whereas the secondary video stream used for content sharing is sent at lower frame rates. Table 3 shows the video resolution, frame rate, and bandwidth for video stream types used in Webex meetings. The aggregate bandwidth utilization for a Webex meetings depends on a variety of factors, such as meeting layout, type of client device, and the number of screens used at the client device. The Bandwidth Provisioning and Capacity Planning section of Preferred Architecture for Cisco Webex Hybrid Services, Cisco Validated Design document provides typical and maximum video bandwidth requirements for Webex endpoints and applications.

Video stream type	Video resolutions	Frame Rate (fps)	Video bandwidth (bits per second)
High-definition primary video (720p)	1280x720	30	900 Kbps to 1.8 Mbps
High quality primary video (360p)	640x360	30	320 Kbps to 512 Kbps
Standard quality primary video (180p)	320x180	30	128 Kbps to 256 Kbps
Content sharing (motion and video)	1280x720	30	0.25 Mbps
Content sharing (text and images)	1280x720	3	0.13 Mbps

Table 3: *Video Parameters for Common Video Stream Types*

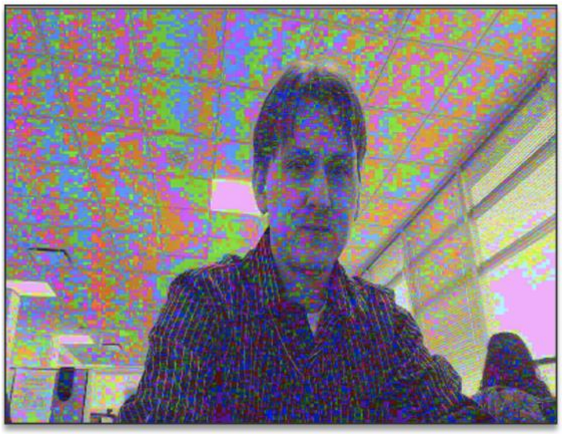
Video Quality Artifacts

The first step in troubleshooting video quality is to clearly understand the symptom experienced by the end user. Getting a screenshot or a picture of the display screen usually gives more information than a verbal description of the problem. The type of video artifacts observed in a screenshot or picture can be used to determine the troubleshooting path. For example, in Figure 13, pictures (a) and (b) are artifacts that are caused when video streams experience packet loss. Just by seeing the picture, you can immediately start troubleshooting to identify the source of the packet loss.

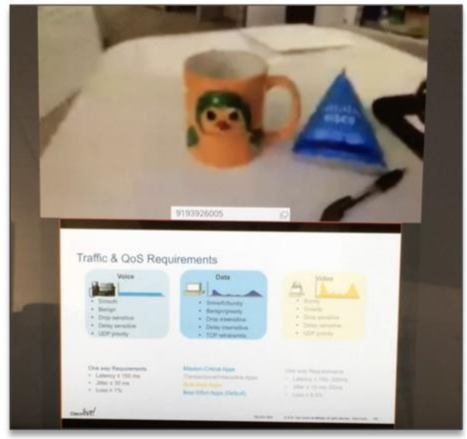


a. Video with line stripe artifacts

b. Frozen video with block artifacts



c. Pixelated video



d. Blurry primary video

Figure 13: Video Quality Artifacts

Picture (c) in Figure 13 is an example of an artifact caused when an incorrect camera driver is installed in the end user's device computer, which causes a strange color pixelization. Picture (d) shows a blurry video in the primary video stream while the secondary video stream containing the presentation content has good quality. The blurriness is visibly noticeable when you try to read the letters on the character shown in the primary video. This scenario is typically observed when the resolution of the primary video stream is downgraded to accommodate the secondary video stream during low bandwidth conditions.

Factors That Influence Video Quality

There are three main factors that influence media quality:

1. **Input and Output Components**
2. **Negotiated Media Capability**
3. **Network Impairments**

Input and Output Components (or I/O Components) refers to the microphone, camera, speaker, and display screen components of the device and headsets used by the end user to join a Webex Meeting. The quality of the video generated by the source device depends on the brightness, contrast, and saturation settings of the video camera. Fortunately, the local self-camera view is available across most devices today and end users are adept at testing their local video before submitting a video quality issue ticket. To ensure the best experience, administrators should provide a recommended set of headsets, cameras, and PC requirements as part of the user onboarding process and validate whether the operating system (OS) software versions, and 3rd party camera driver version requirements are met. Performing these checks is critical for troubleshooting and preventing issues like the pixelated local video as shown in picture (c) of Figure 13. The video quality of the camera output can be improved by adjusting the brightness, contrast, saturation, sharpness levels available in the camera settings and also by using better lighting in the end user environment.

Webex Control Hub makes it quite easy to find the device type and operating system version used by the end user to join a Webex meeting. All you need is the end user's email address and meeting time to find the right meeting and to get end user's device details. Figure 14 shows the list of meetings attended by a participant with the email address, rtpmsuser1@gmail.com.

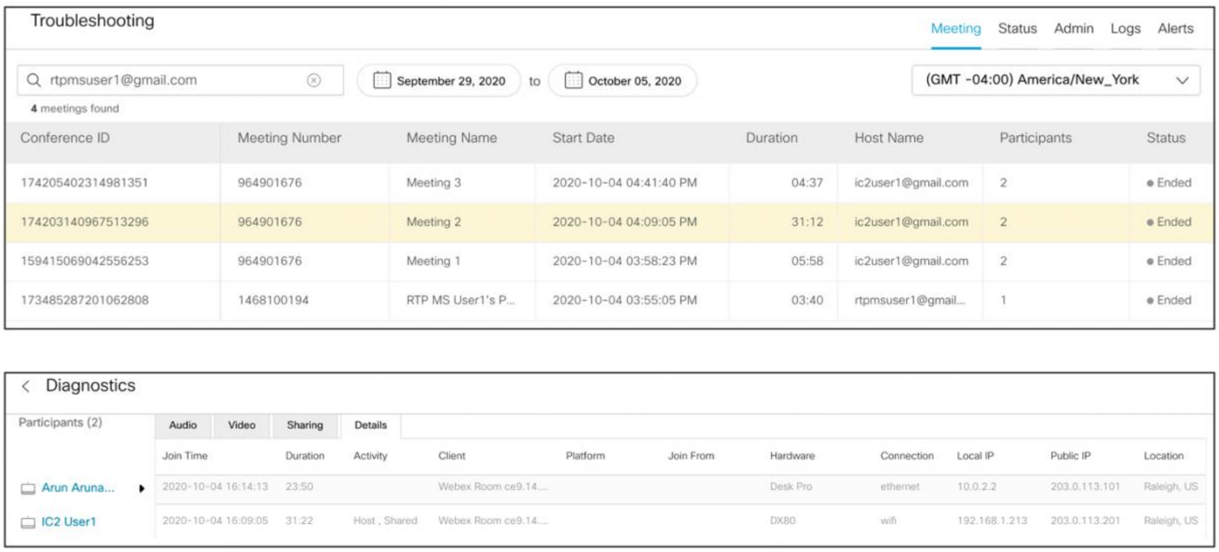


***Figure 14**: Participant Device Details*

Clicking the meeting of interest provides audio, video, and content sharing statistics along with other information for all participants in the meeting. The **Details** tab displays key device level information such as hardware type and software version of the Webex client that was used to join the meeting. It can be inferred from Figure 14 that the user Arun joined the meeting using Cisco Webex Desk Pro device running CE 9.14 software version.

Negotiated Media Capability refers to the audio codec, video codec, resolution, frame rates and maximum media bitrate (bandwidth) that are negotiated and currently in use for the primary video and secondary video (content sharing). Clicking the participant's name in the participant list on the left side of Diagnostics page (Figure 14) provides insights into the codec used for audio, video, and content share along with video resolution. Figure 15 shows a participant that is using the Opus audio codec (which was discussed in more detail in the previous section "Audio in Webex Meetings") and the H.264 video codec configured for the BP (Baseline Profile) capability. Additionally, this H.264 stream is currently receiving video at a resolution of 288p (512x288), which is in fact a low video resolution value. This information about the video stream indicates that the video may be blurry when viewed on a larger screen. It is important to note that the frame rate and video resolution can **dynamically change** during the call depending on network conditions or occurrence of events such as sharing a presentation.

> ***NOTE**: The next generation AV1 codec is slowly replacing the aging H.264 video codec in Webex. The AV1 codec provides much better quality but is not as broadly supported by endpoints and does currently require more computing resources for the encoding. More information can be found here -* https://blog.webex.com/engineering/the-av1-video-codec-comes-to-webex/

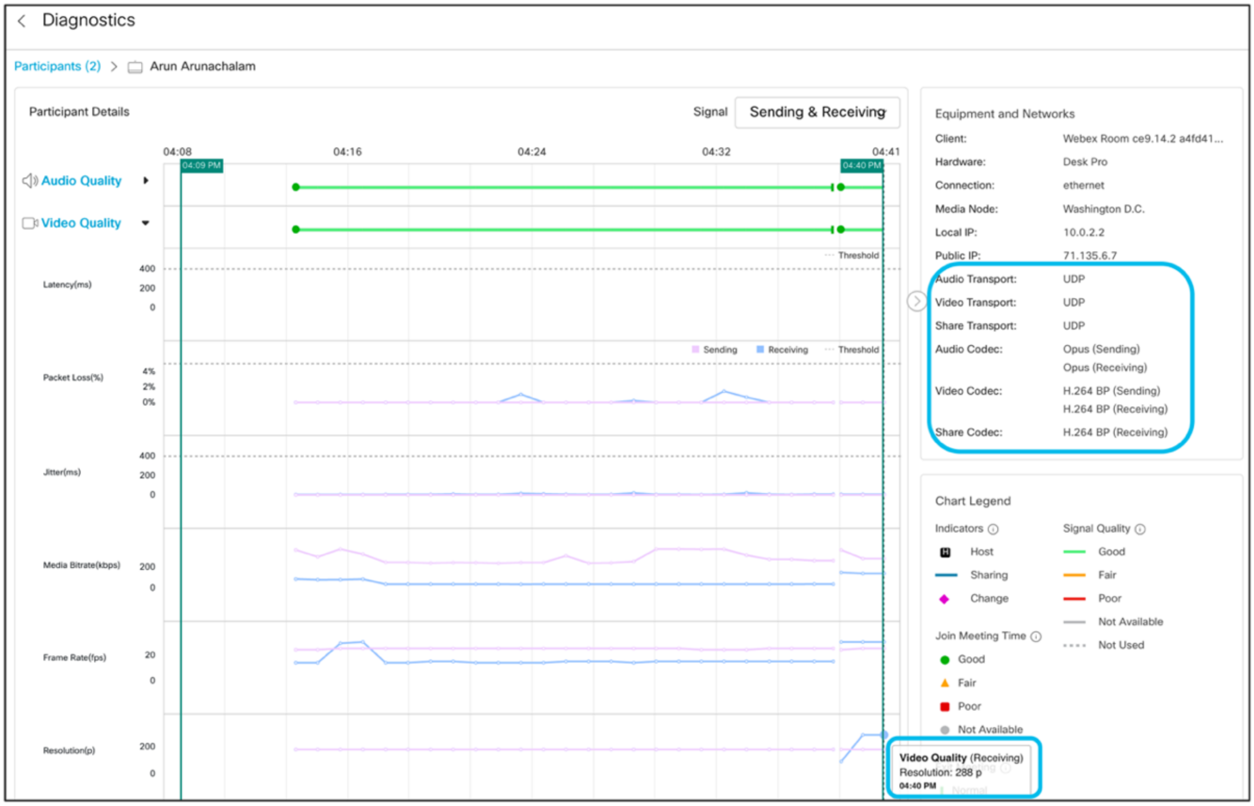


***Figure 15**: Negotiated Media Capabilities*

> ***NOTE:** The device type and software version of each participant's device are determined based on information received from the client. Currently, the Diagnostics details tab shows the audio and video codecs negotiated initially at meeting setup time and is <u>not</u> updated if the codecs get re-negotiated during the meeting.*

Network impairments such as packet loss, delay, and jitter can also impact video quality. The video frame data is highly compressed through the encoding process to minimize bandwidth utilization. This **high compression makes video streams more sensitive to packet loss than audio streams.** Even a 1% packet loss may cause noticeable artifacts if the client does not use advanced techniques to handle packet loss. Cisco collaboration endpoints and applications make use of several adaptive mechanisms to preserve high video quality even during packet loss situations. The animation in Figure 16 shows how missing I and P frames can affect video quality.

The artifact observed by end users depends on which video frame type is missing due to packet loss. If an IP packet containing an I-frame is lost, the receiving endpoint would not be able to decode the video correctly even if it receives all the subsequent packets that contain the P-frames. This scenario is highlighted in the first animation of Figure 16. This is because the video decoder will not be able to decode the P-frames without the reference I-frame and hence need to discard all the P-frames until the next I-frame is successfully received. End users observe frozen video artifacts in this scenario. On the other hand, if the I-frame is received correctly but only some of the P-frames are lost during transit, then the video decoder will continue to process the received P-frames and the video gets rendered. Some of the motion changes captured in the missing P-frames cannot be rendered resulting in line stripe like artifacts shown to end users as indicated in the second animation of Figure 16.

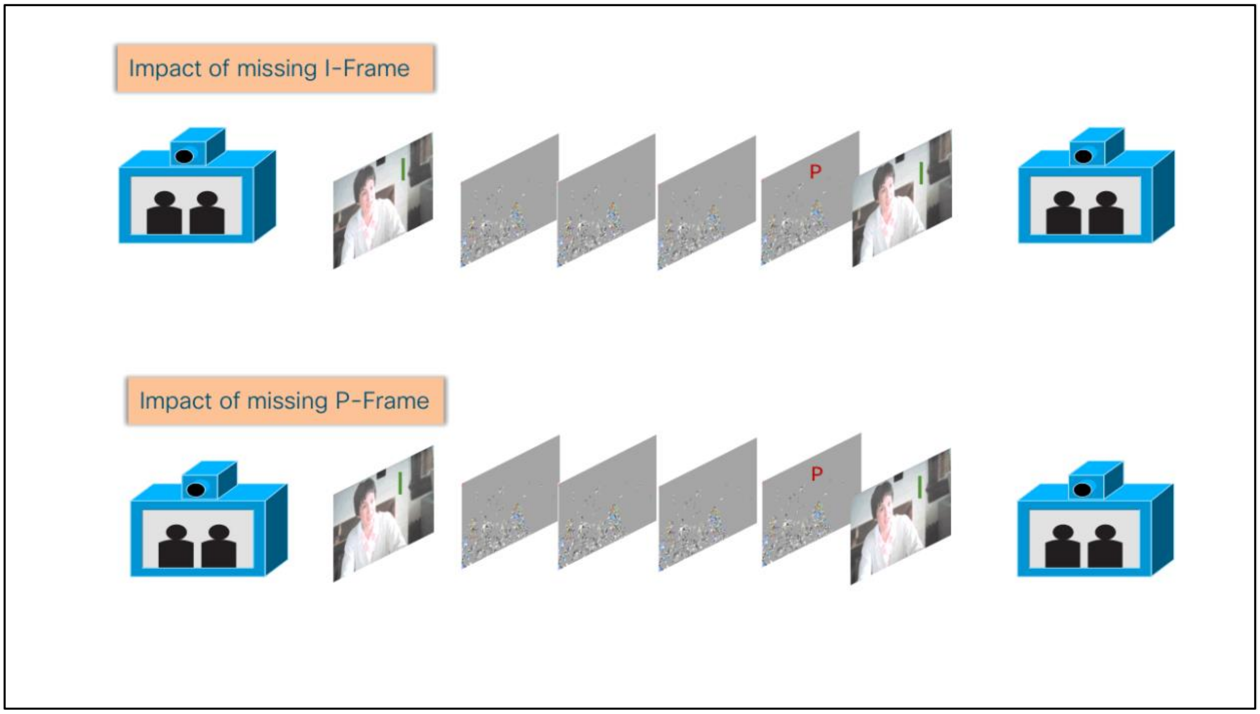


Figure 16*: Video Quality Artifacts During Packet Loss*

Cisco video endpoints use several intelligent mechanisms to handle packet loss in media streams. The first mechanism, called RTCP based dynamic bit rate adjustment, uses Real-time Control Packet (RTCP) packets to become aware that the remote endpoint is experiencing packet loss consistently. Then, it dynamically reduces the upstream video bit rate to minimize the possibility of packet loss.

The second technique uses Repair-P frames. Typically, endpoints experiencing packet loss request the remote peer to resend the I-Frame to recover from packet loss. As I-Frames are larger in size, resending them in a lossy network scenario usually makes the situation worse. To avoid resending I-Frames, the endpoint stores the initial I-Frame as a Long Term Reference Frame (LTRF) and uses small Repair-P frames to resync with the peer after experiencing packet loss. The size of a Repair-P frame is only 10% of the I-Frame and helps in faster video data recovery.

The third mechanism is Forward Error Correction (FEC), which works by sending redundant video data within RTP packets. This enables the endpoint experiencing packet loss to automatically recover video data without asking the other end to resend frames. In the FEC scenario shown in Figure 17, even if the packet P2 is lost, the destination endpoint will be able to recover it by using the data available in P1. This avoids the needs to resync with the originator right after the packet loss.

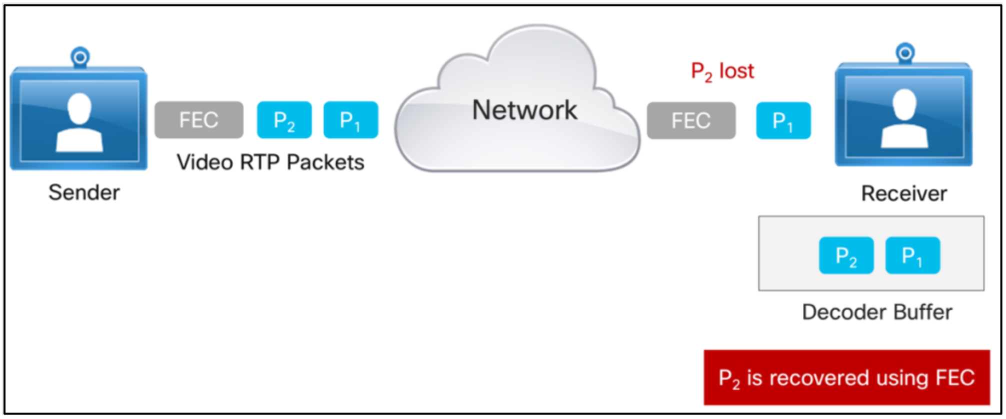


Figure 17*: Packet Loss Recovery Using Forward Error Correction*

The fourth mechanism called Media Adaptation and Resilience Implementation (MARI) is a combination of several techniques. These techniques include video packet pacing which minimizes packet loss due to burstiness, RTP retransmission (RTX) to recover lost packets in delay tolerant low frame rate content sharing video streams and FEC to recover multiple lost packets in main video and high frame rate content sharing video streams to ensure high video quality experience even in lossy networks.

Collaboration endpoints dynamically choose RTX or FEC depending on negotiated bandwidth and delay tolerance for a given media stream. FEC results in higher bandwidth utilization due to redundant video data but it doesn't introduce additional delay to recover lost packets. Whereas RTX doesn't contribute to higher bandwidth utilization because the RTP packets are retransmitted only when the receiver indicates packet loss in RTCP feedback channel. RTX introduces packet recovery delay due to the time it takes for the RTCP packet to reach the receiver from the sender and for the retransmitted packet to reach the receiver from the sender.

Reducing the bandwidth consumption for Webex meetings reduces the possibility of packet loss due to any rate-limiting configuration within the Service provider as the Webex traffic traverses the Internet on its way to the Webex cloud. Video Mesh Node can be deployed in enterprise networks to optimize up-stream video bandwidth in Webex meetings, specifically when several on-premise video endpoints join the same meeting. Figure 18 shows an example scenario in which a virtual meeting is attended by students in class rooms using their laptops, local school teachers using Cisco DX80 in their offices, Webex room kit from Media lab and Webex board from Library. Without Video Mesh Node, each endpoint sends audio and video streams directly to the Webex Cloud resulting in about 23.7 Mbps for just one Webex meeting. Whereas with Video Mesh Node (VMN) deployment, Webex cloud instructs the endpoints to send their media stream to the local VMN and the local VMN sends only the media streams of the last six active speakers resulting in only 8.54 Mbps of upstream bandwidth.

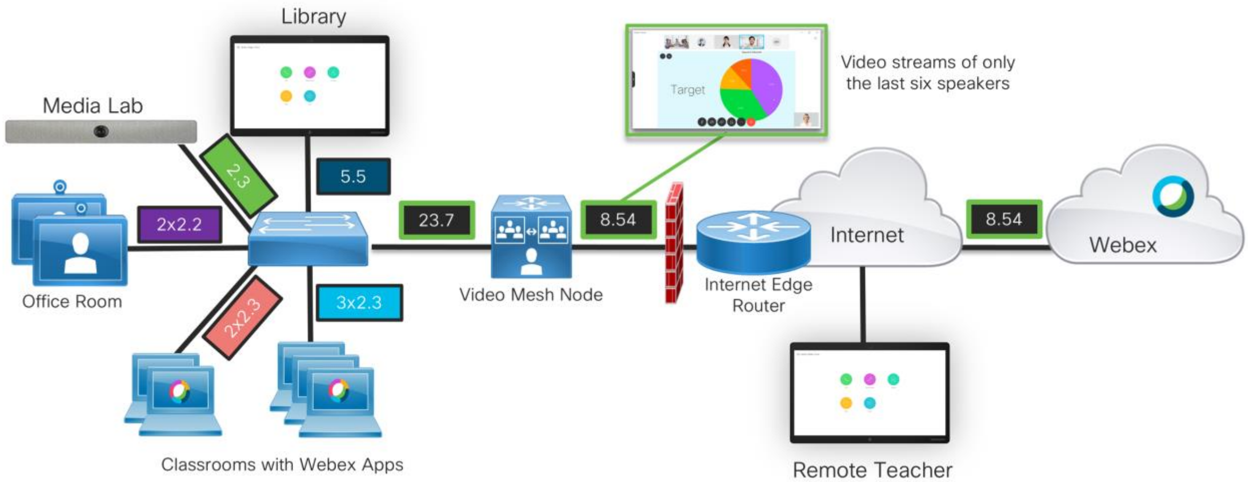


Figure 18: *Reduced Up-stream Bandwidth using Video Mesh Node*

The audio and video RTP streams from the Webex Meetings client, Webex App, Webex Room Device, and Webex Mesh Nodes to the Webex Cloud are sent across a single UDP connection. UDP is the preferred transport method and hence is referred as Primary transport type. When UDP connection is not available, for example when UDP ports are blocked by firewall, clients use alternate transport methods such as TCP to send audio and video RTP streams. Table 4 lists the primary and alternate transport types and port numbers used by Webex clients. The port number 33434 is used only if the port numbers 5004, 9000 are blocked.

Webex client type	Primary transport type	Alternate transport type	UDP port numbers	TCP port numbers
Webex Meetings client	UDP	TCP	9000	80 / 443
Webex App	UDP	TCP	9000, 33434	5004, 33434
Webex Room Devices	UDP	TCP	5004, 33434	5004, 33434
Webex Mesh Node	UDP	TCP	5004	5004
Cisco Video Collaboration Devices through Expressway-E	UDP	N/A	49152 - 59999	N/A

Table 4: *Port Numbers Used by Webex Meetings Media Streams*

The RTP packets that belong to the audio and video streams have separate Differentiated Service Code Point (DSCP) values which enables the Network to provide higher preference to audio packets than the video packets. DSCP utilizes 6 bits in the IP header field to assign values to an IP packet so network devices, like routers, can ensure their priority.

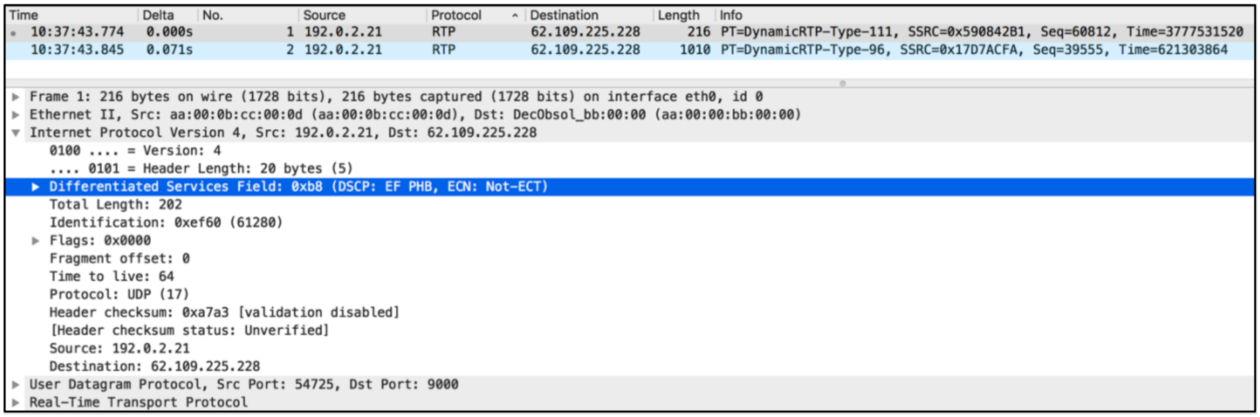


Figure 19*: Audio RTP Packet Marked with EF DSCP*

Audio RTP packets are marked with Expedited Forwarding (EF, DSCP=46) and Video RTP packets are marked with Assured Forwarding (AF41, DSCP=34) as shown in Figure 19 and 20.

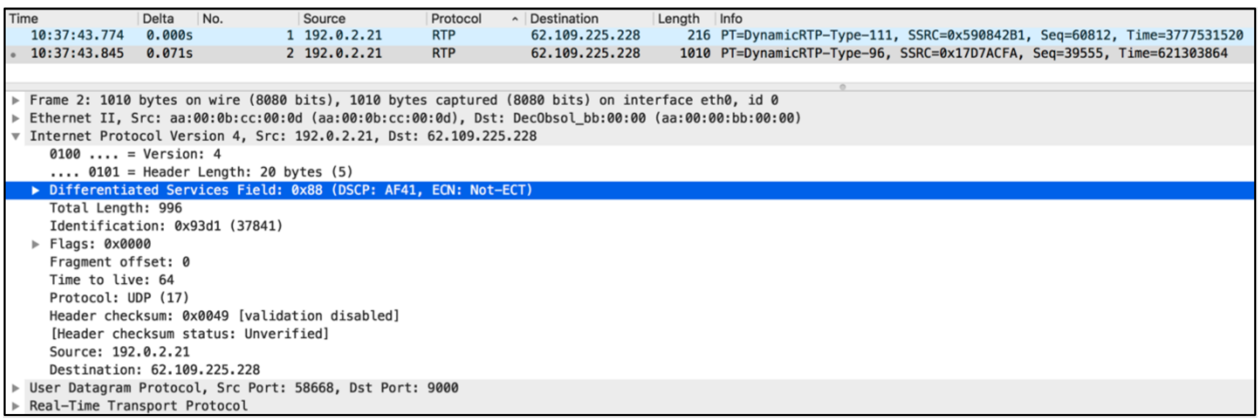


Figure 20*: Video RTP Packet Marked with AF41 DSCP*

> ***NOTE:*** *Windows OS 7 and higher require applications to have administrator privileges in order to be able to mark DSCP value of media streams. Webex app installed in Windows devices don't have admin privileges in most deployments and hence the DSCP value of audio and video packets generated from such devices will be zero. To overcome this limitation, it is recommended to mark the Webex meeting audio and video RTP streams with the correct DSCP value by using an ingress QoS policy at the WAN edge router. The class-map defined within the QoS policy can use the source and destination port numbers or Network Based Application Recognition (NBAR) to match the Webex traffic. Please reference the session [Webex Bandwidth Management - BRKCOL-2777]() in the Cisco Live On-Demand Library for additional design and configuration details.*

Since the network is providing differential treatment to Audio and Video RTP streams, it is possible for these streams to experience different one-way network delay from the client to Webex Cloud. Transcoding is another reason for the network delay difference between audio and video streams. For example, when an endpoint that supports only H.263 video codec and G.711 audio codec joins a Webex meeting (Webex CMR cloud), Webex performs transcoding of H.263 → H.264 video stream while the audio stream is processed without any transcoding. The receiving endpoints use a technique called RTCP based media synchronization to align audio and video streams together and play out the media in a synchronized fashion. If there is significant **network delay** difference between these two streams, for example audio packets consistently arrive earlier than the video packets during network congestion, there is a possibility of end users experiencing lip synchronization issues. During a lip-sync issue, the lip movement of the active speaker doesn't align with speaker's voice and the voice either lags or leads the lip movements. It is harder for the meeting participants to listen, focus and participate in the conversation in such situations.

The variance in delay between packets of the same stream is referred to as **network jitter.** The receiving endpoints use a jitter buffer to store the audio and video data and media is played out in a periodic time interval. This allows packets that arrive early to be stored in the jitter buffer and played out in a synchronized fashion. Packets that arrive late are considered as lost packets and are dropped resulting in video frame loss. Table 5 provides a summary of commonly observed video-related issues and their most likely root causes.

Root cause	**Issue symptoms**
Low packet loss	Line stripes, Blocky video
High packet loss	Frozen video
Network delay	Lip synchronization
Low network bandwidth	Blurry video, Low video resolution
Camera driver	Grainy video, Video pixelization

Table 5*: Common Video Quality Symptoms and Their Causes*

Cisco validated design (CVD) guides are good sources of best practice recommendations that help you select the right architecture for on-prem, hybrid and cloud collaboration deployments. The following CVD guides provide detailed information about Quality of Service and Quality of Experience from Collaboration applications and end users perspective:

1. Bandwidth Management chapter of Preferred Architecture for Cisco Webex Hybrid Services, CVD.
2. Implementing Quality of Experience for Webex in Work from Home Environments.

Control Hub Audio and Video Troubleshooting Analytics

Often, users who experience audio and video quality issues may not take the time to report the issue to their internal helpdesk. Therefore, depending solely on the user feedback isn't sufficient to guarantee the best experience to end users. The Webex Control Hub Analytics enables you to be proactive by making it easier to visualize the user experience and performance of all meetings within their organization in an intuitive way.

Figure 21 shows the first level view of meetings in a 30-day timeframe from a media quality perspective. You are informed of the location that has the most participants with poor audio or video quality. This functionality saves troubleshooting time by eliminating the need to manually analyze the media statistics to isolate the source of the media quality issues. You can also identify the percentage of the meeting participants who had good, fair, and poor meeting quality and their geographical region and location distribution. In the example illustrated by Figure 21, India has the most participants experiencing voice and video quality issues.

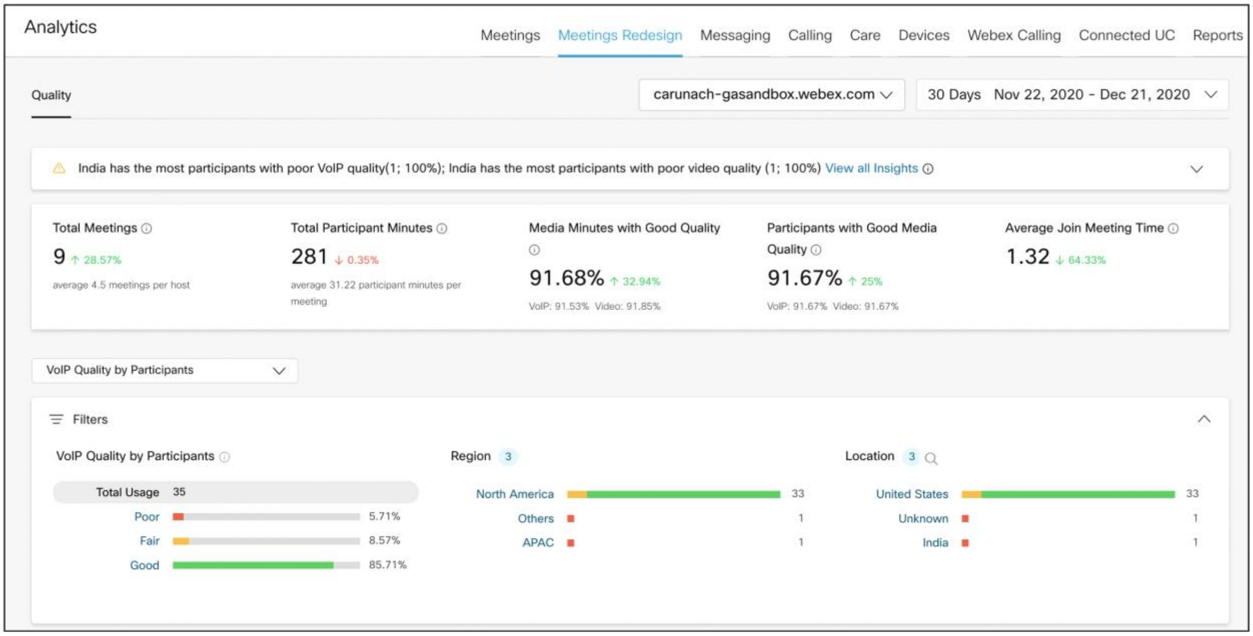


***Figure 21**: Meeting Quality Insights*

Clicking the **View all Insights** provides additional granular insights, such as device types and connection types used by participants experiencing problems. For the example scenario being discussed, all the participants who experienced quality issues are using a Mac computer and are connected to Webex across a Wireless connection as shown in Figure 22.

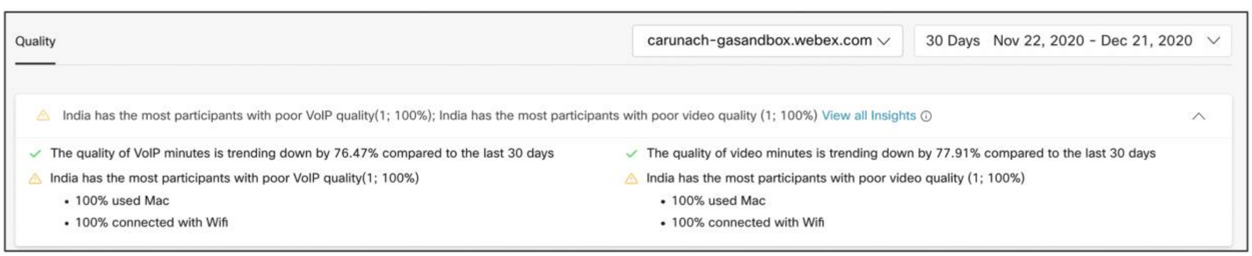


***Figure 22**: VoIP Quality Insights*

The **VoIP Quality by Participants**, **VoIP Quality by Connection**, and **VoIP Quality by Platform** graphs in Figures 23-25 provide information about when the poor media quality was observed, the type of network connection used by the client devices (Ethernet, Wi-Fi, Cellular), and the client device types (participant joined from web browser in a device running Windows, MAC, Android, or iOS operating system or participant joined from the Webex application). The Other category in Video Quality by Platform graph includes participants joining from collaboration endpoints, such as Cisco DX 80 or Cisco Desk Pro.

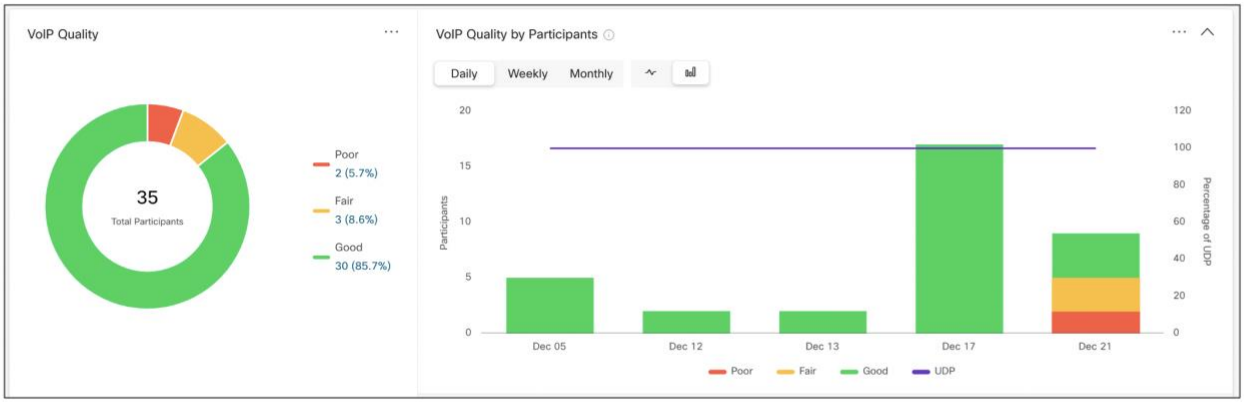


***Figure 23**: Media Quality Level Distribution*

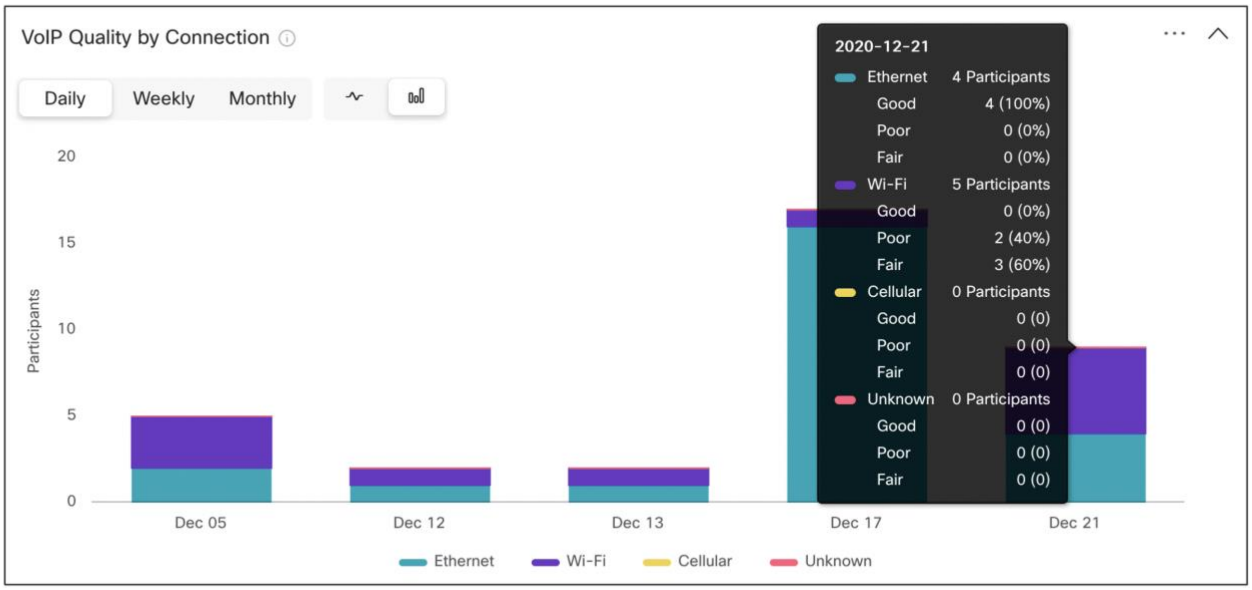


***Figure 24**: Meeting Participant Connection Types*

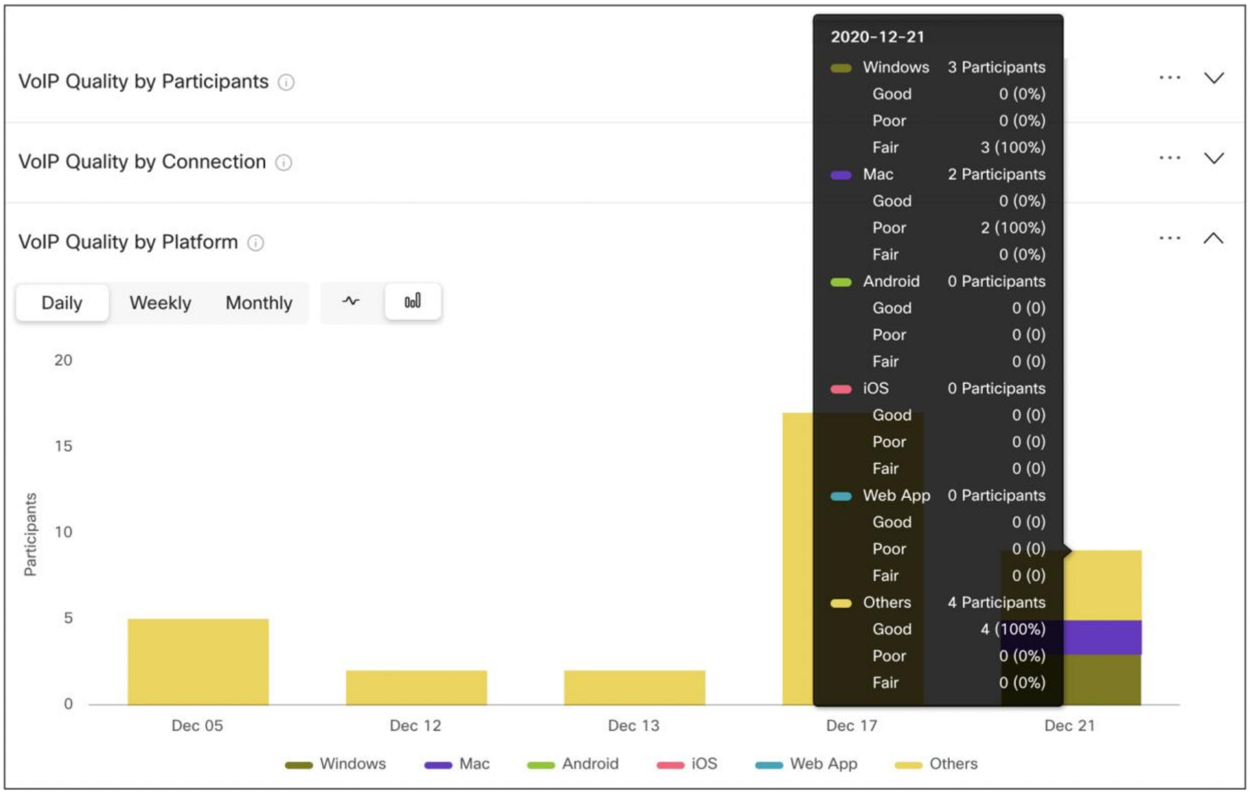
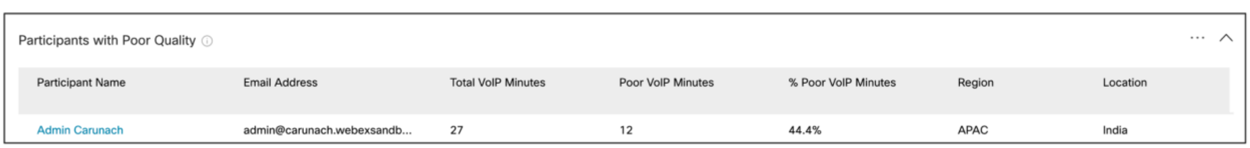


***Figure 25**: Meeting Participant Device Types*

The **Participants with Poor Quality** section in the Quality insights page displays the list of participants who had an average end-to-end packet loss greater than 5% and average network latency greater than 400 milliseconds during the duration of one or more Webex meetings that occurred in the specified data range (e.g. 30 days). It is important to note that end-to-end packet loss refers to packet loss experienced by a client device after it has applied error recovery techniques such as Forward Error Correction. Clicking the participant name link in Figure 26 will take you to the **Troubleshooting** section of Webex Control Hub auto-populated with the list of meetings attended by the participant in the date range set at the top of Quality insights page. You can reference the Troubleshooting scenarios in the next section of this document for further steps to isolate the root cause of the poor quality problem.

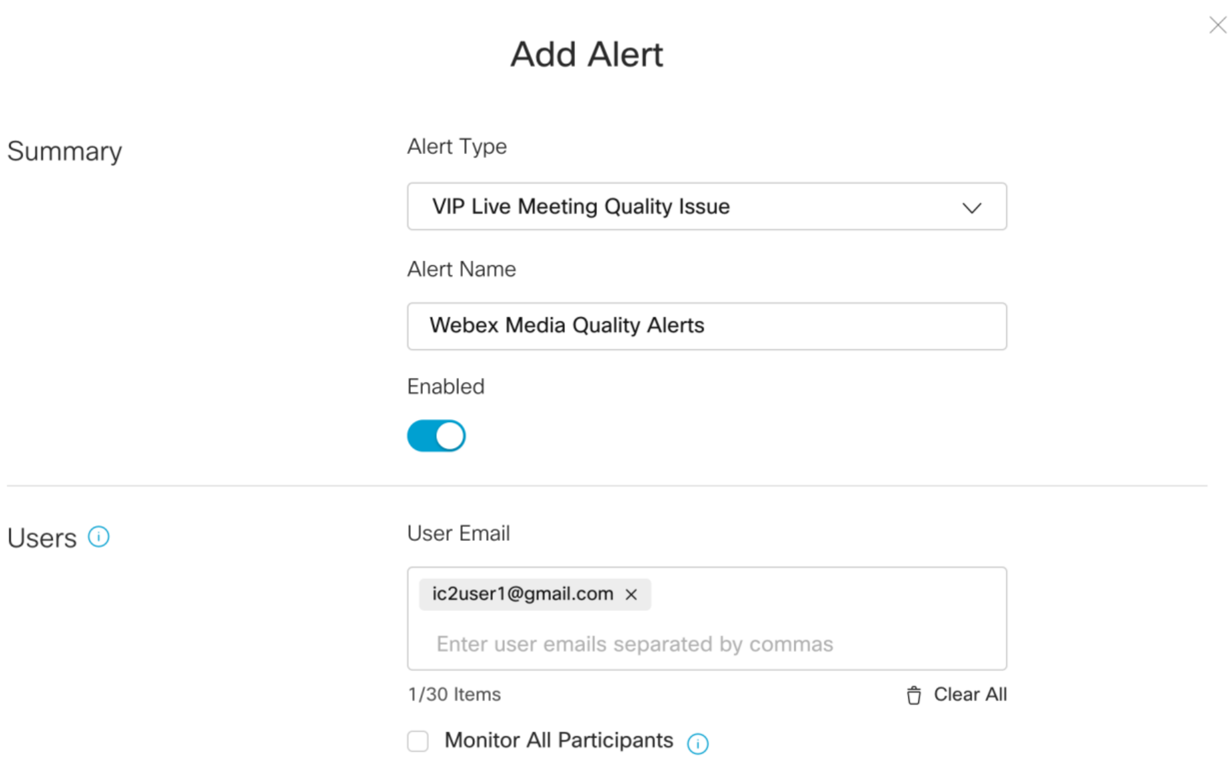

Participants with Poor Quality

Participant Name	Email Address	Total VoIP Minutes	Poor VoIP Minutes	% Poor VoIP Minutes	Region	Location
Admin Carunach	admin@carunach.webexsandb...	27	12	44.4%	APAC	India

***Figure 26**: Participant with Poor Quality*

Proactive Notifications

It is useful to get proactively notified when specific users such as executives within your organization experience media quality issues so that your team can take action to resolve the issue right when it happens instead of working on an escalation later. You can setup Webex to notify your helpdesk team via email or Webex app when packet loss, jitter and latency reaches a specific threshold for a specific duration during the meeting of specific users. The alert details are defined in Control Hub (Troubleshooting > Alerts) section. Figure 27 shows an example of an alert that notifies **helpdesk@example.com** when the user ic2user1@gmail.com experiences 5% video packet loss or more than 400ms latency for one minute (accumulation time), one or more times in a three minute window (consecutive time). The alert is also generated when 10% audio packet loss or more than 400ms latency is observed during the same accumulation and consecutive times.



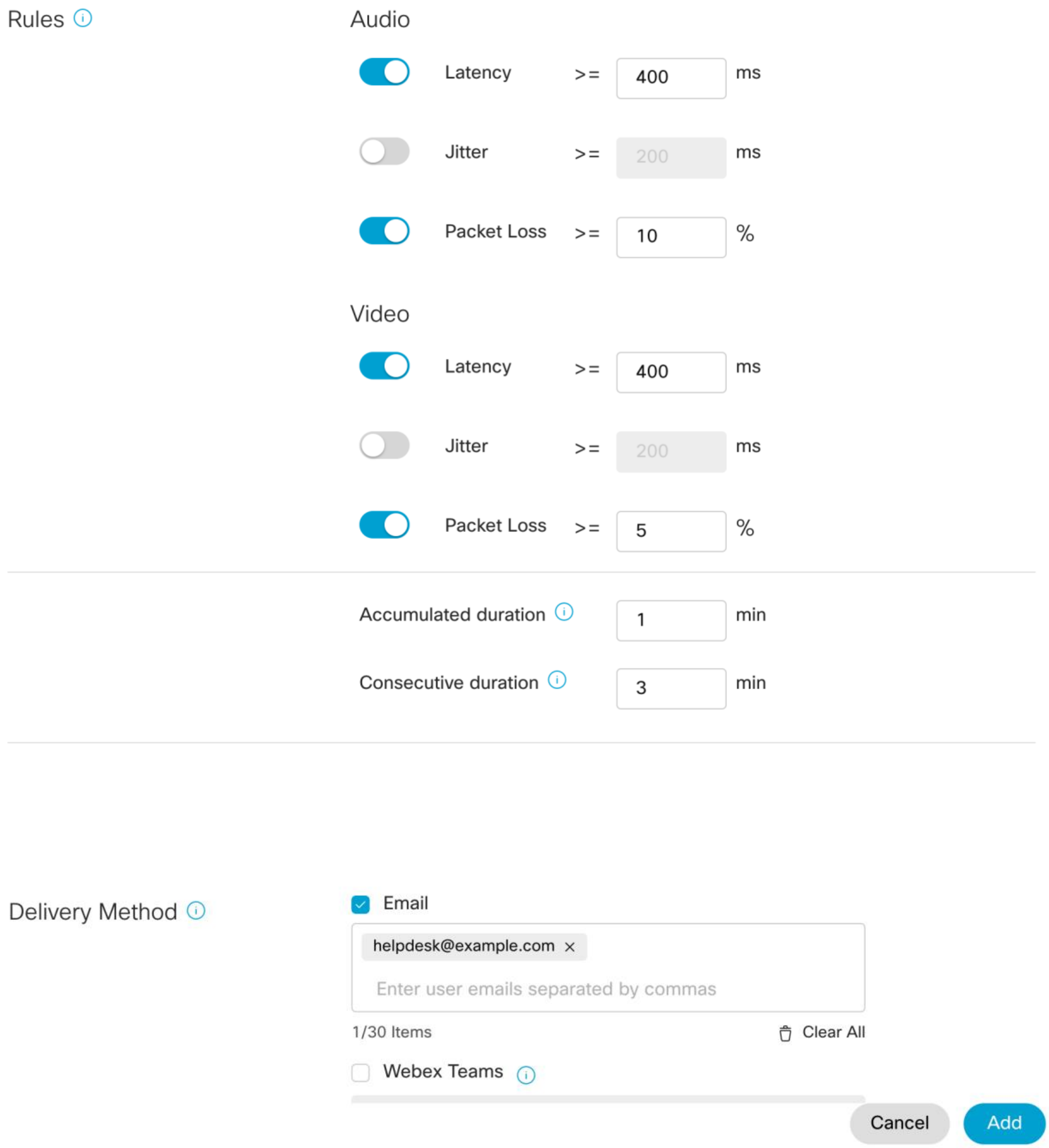


***Figure 27**: Proactive Network Impairment Notification Alert Configuration*

Use Cases and Troubleshooting Scenarios

Garbled Voice and Line Stripe Artifacts in Video

Problem description
You receive a report from User 1 (rtpmsuser1@gmail.com), one of the participants of a recent meeting, stating that the audio quality was intermittently poor. The described symptoms are audio drops, garbled audio, or choppy audio during the meeting # 1463195095 at around 2:50 PM EST on 8th Oct. 2020. The users also report that at times they see line stripe artifacts in the video stream as shown in picture (a) of Figure 13.

Problem Analysis using Control Hub
Control Hub provides a wealth of information for every meeting that you can use to quickly isolate whether the media quality issue is caused by user's device, user's network, or the Webex cloud. The **first step** is to find the meeting of interest by using the **Monitoring > Troubleshooting** menu in the Control Hub Navigation bar as shown in Figure 28. Selecting the meeting leads you to a web page that shows the quality of each participant's experience in a graphical format. The Audio and Video tabs in this webpage provides a graphical representation of the audio and video stream statistics from the time each participant joins the meeting.

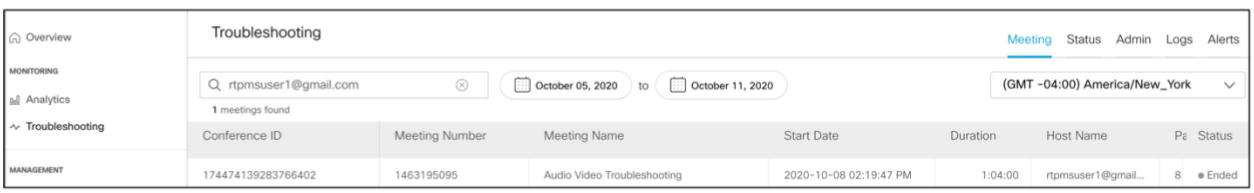


***Figure 28**: Finding the Meeting of Interest*

Figure 29: *User 1's Media Statistics Graph*

The media statistics of the meeting participant at a specific point in time can be retrieved as shown in Figure 29. You can quickly observe that the audio stream received by User 1 at around 2:50 PM is marked as Red (Poor) and that User 1's endpoint is experiencing a 10% packet loss. It is a good practice to check whether other participants experienced the same issue at the same time. User 3, User 5 and User 6 experience similar packet loss in the **Receiving** direction (i.e. from Webex to the user's device) as shown in Figure 30. A key point to note is that the **Receiving packet loss** value displayed in the graph is the **end-to-end packet loss** experienced by the Webex client after forward error correction is applied.

__Figure 30__: Meeting Participant Media Statistics Graph

Interestingly, User 4's statistics gives different information. The packet loss is observed in **Sending** direction instead of Receiving direction. The **Sending packet loss** value displayed in the graph is the **hop-by-hop packet loss** experienced by the Webex cloud in the RTP media stream received from the client.

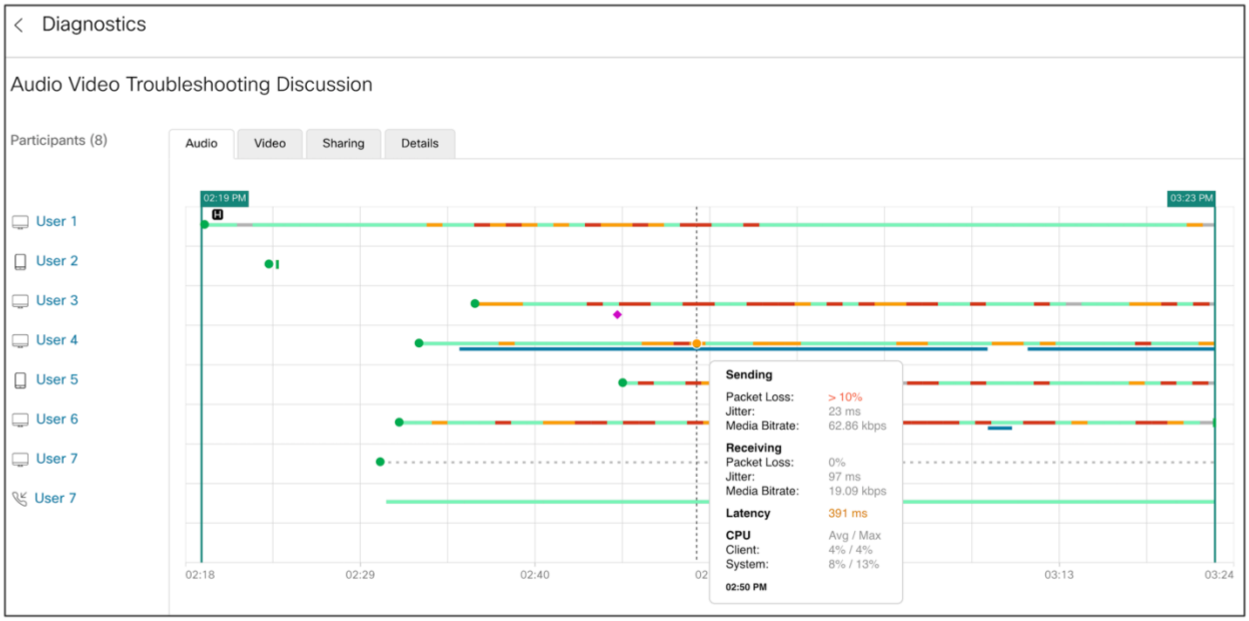


__Figure 31__: Meeting Participant Media Statistics Graph

Packet loss in the sending direction indicates that Webex is not getting all the audio RTP (Real time protocol) packets from User 4's endpoint or is getting the audio packets in a delayed fashion so that they end up being discarded. If there is packet loss in the media stream received from the active speaker, it impacts all the recipients. In this case, the media statistics timeline will show a RED (Poor quality) or ORANGE (Fair quality) color for all other participants if the packet loss can't be recovered by the participant's client device as shown in Figure 31. If an RTP packet is not received in a timely manner, the endpoint tries to predict the audio content based on recent past audio frames to conceal the missing audio gaps. This results in noticeable garbled audio when the time duration for audio concealment is long. At this point, you have successfully isolated the problem to one participant.

***NOTE:** It is important to note that even though the media statistics graph is showing Red and Orange colors indicating network issues, it is possible for the Webex client to compensate the packet loss and thereby resulting in no audio or video interruptions to the end user. It is always a good practice to leverage the media statistics in the context of an issue reported by an end user.*

The next step in the troubleshooting process is to identify the root cause for User 4's audio stream packet loss. Is it due to delayed transmission of packets at the source device, is it due to packets getting dropped at the user's network edge, or is it due to packets getting dropped while traversing the Internet on the way to the Webex cloud? Clicking a participant's name in the media quality graph leads to the **Participant Details** graph which provides a consolidated view of audio, video, content share, CPU and memory usage, and the Webex media node to which the user's device is connected as shown in Figure 32.

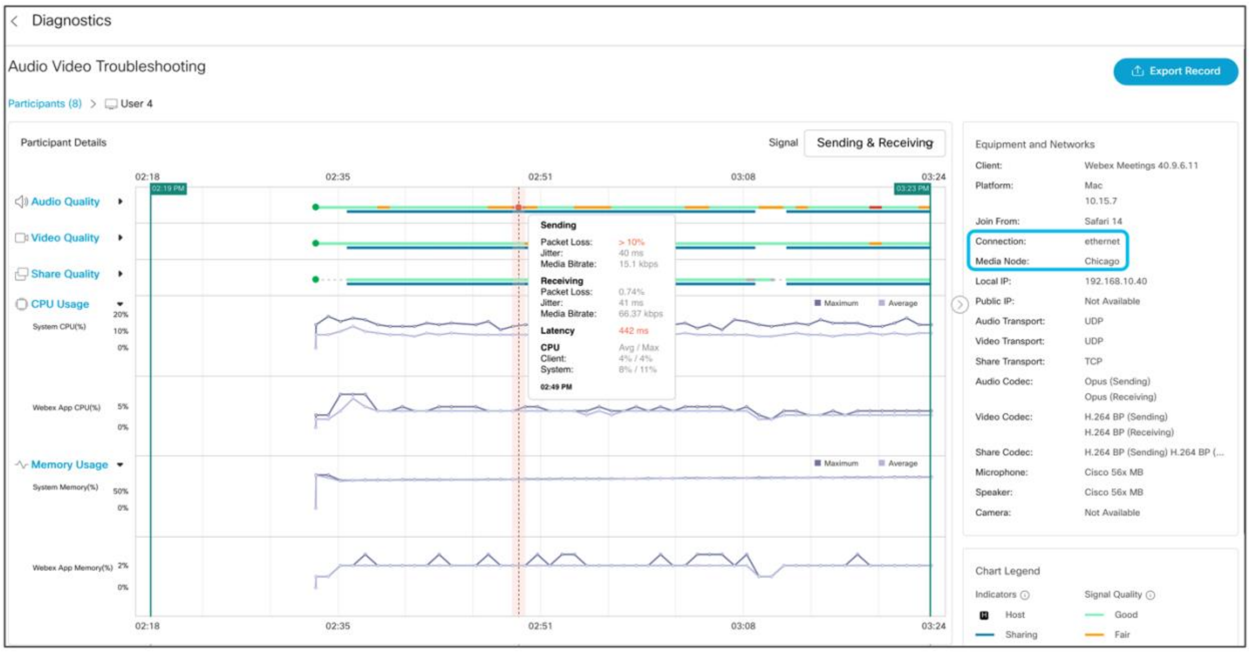


***Figure 32**: Participant Details*

The Participant details graph of User 4 shows that System CPU (13%), Webex App CPU (4%), System Memory (85%) and Webex App Memory (2%) utilizations are at normal usage levels. Therefore, we can rule out the user's device contributing to the media quality problem. Given that the packet loss (10%) and latency (442 milliseconds) of the audio stream is high while other system parameters are normal, we can conclude that the impairments are caused by the network connection between end user's device and Webex media node in Chicago.

The troubleshooting techniques used to identify the source of packet loss between User 4 and Webex media node differs based on the type of network to which User 4 is connected. If User 4 is joining the Webex meeting from an enterprise network, we can use [Performance monitor] feature in the enterprise network edge to isolate whether the packet loss is introduced within the enterprise network or within the Internet. If the User 4 is connected to a home network, we can use the troubleshooting best practices discussed in the "Tips to Troubleshoot Network Issues in Home Networks" section. Fortunately, the upcoming Webex – ThousandEyes integration will bring more visibility into the end-to-end network performance in such scenarios. This whitepaper will be updated once this integration is generally available for customers.

Tips to Troubleshoot Network Issues in Home Networks

As a Webex administrator, you can provide the following check list to an end user experiencing packet loss or network delays in their home network to enable them to self-solve many audio and video quality issues.

1. **Avoid VPN Connection for Webex meetings**
   Webex meeting service can be accessed by end users directly through the Internet. Hence it is recommended to bypass VPN connection in order to avoid routing Webex meeting traffic to the VPN headend and then to the Webex cloud. VPN clients can be configured with Split Tunneling feature to identify Webex Traffic and not send the corresponding IP packets through the VPN connection. The article titled "Optimize AnyConnect Split Tunnel for Microsoft Office 365 and Cisco Webex" provides an example Split Tunneling configuration for Cisco Adaptive Security Appliance (ASA) with AnyConnect VPN clients.

2. **Check the network connection of the device**
   Most end users connect to the home network across a Wi-Fi connection. The quality of the network connection greatly depends on the wireless signal strength and lack of interference. There are a number of free and inexpensive Wi-Fi network analyzers available for desktop and mobile devices. These analyzers provide insights into WiFi parameters, settings, and the radio signal itself. The following WiFi information is usually included:
   - Mode (802.11 a/b/g/n/ac/ax)
   - Band (5Ghz, 2.4 Ghz)
   - Channel
   - Signal strength
   - Signal to Noise Ratio (SNR)
   - Vendor

   Typically, the Wi-Fi network analyzer displays intuitive visual indicators to highlight good, average, poor wireless network connection based on the SNR level. SNR values stronger than 40 dB indicate a good wireless connection. Figure 36 shows information collected on a per channel level using NetSpot WiFi Analyzer.

BSSID	Channel	Band	Security	Mode	Level (SNR)	SNR	Signal	Signal %	Avg	Max	Min	Noise	Nois...	Last seen
F4:17...	149	5GHz	WPA2 Personal	ac		19	-77	23%	-76	-75	-78	-96	4%	now
42:2...	6	2.4GHz	WPA2 Personal	b/g/n		46	-50	50%	-48	-43	-52	-96	4%	now
40:2...	6	2.4GHz	WPA2 Personal	b/g/n		45	-51	49%	-49	-47	-53	-96	4%	now
48:00...	6	2.4GHz	WPA2 Personal	b/g/n		-	-	0%	-93	-93	-94	-	0%	1min 11s ago
FE:01...	3	2.4GHz	WPA2 Personal	g/n		13	-83	17%	-81	-79	-84	-96	4%	now
A8:9...	1	2.4GHz	WPA2 Enterpri...	b/g/n		52	-44	56%	-45	-42	-45	-96	4%	now
A8:9...	48	5GHz	WPA2 Enterpri...	a/n		43	-53	47%	-55	-53	-55	-96	4%	now
C6:8...	149	5GHz	WPA2 Personal	ac		17	-79	21%	-79	-78	-82	-96	4%	now
46:2...	149	5GHz	WPA2 Personal	ac		35	-61	39%	-61	-58	-63	-96	4%	now

***Figure 36**: Wireless Connection Details*

Some home routers with pre-built WiFi access points provide native WiFi congestion testing tools and WiFi connection statistics. Figure 37 is an example screenshot collected from such a router that uses a congestion score from 1-10 to represent the congestion level in each WiFi channel. The “Transmit Discard Packets” indicate the number of packets that were dropped by the device. It is a best practice to clear the connection statistics and monitor whether the drop packet counter is incrementing, especially during a Webex call.

Channel	AP Count	Congestion Score
36	3	3
40	1	3
44	0	3
48	3	3
52	1	3
56	0	3
60	0	3
64	0	3
100	1	1
104	0	1
108	0	1
112	0	1
132	0	5
136	0	5
149	6	10
153	0	10
157	0	10
161	0	10

	2.4 GHz	5 GHz
Transmit Bytes	3652052259	2946879275
Receive Bytes	1542599478	740354274
Transmit Packets	15781253	44838800
Receive Packets	4876340	11903925
Transmit Error Packets	4606	37
Receive Error Packets	0	6288
Transmit Discard Packets	0	503896
Receive Discard Packets	2	0

Figure 37*: Example of WiFi Access Point Congestion Information*

To remedy a poor WiFi connection at home, end users can use WiFi extenders, change the location of the device (i.e. move closer to the access point) to get a better signal, switch to the 5Ghz band to minimize interference, and/or select a wireless access point that supports faster speeds (e.g. 802.11ac).

3. **Change or upgrade the network connection**
   Another approach is to choose a more reliable network connection. For example, the end user can connect their device directly to an ethernet port and eliminate wireless interferences.

   Observing the time of the day in which audio and video quality issues happen also helps to isolate the problem. Does the problem always occur when multiple users on the home network are attending online meetings at the same time? A common scenario is kids and parents attending school and business meetings at the same time. Irrespective of whether you are connected to a wireless or wired port, you will experience packet loss or delay if your home network experiences network congestion. Performing a network test right from the cable modem will give you accurate internet upload, download speeds, and latency. End users should ensure that their Internet service and home network can handle the amount of traffic being placed on it during peak usage periods.

   End users can upgrade their internet services to have higher upload speeds if their bandwidth consumption exceeds their upload speed. As a last resort, end users can also choose to use a dedicated Hotspot connection from a cellular device to bypass the local home network.

4. **Engage the Internet service provider**
   After ensuring that the local wired or wireless connections are good, engaging the service provider can often be helpful. Sometimes, the problem observed by end users may be caused by software issues in the cable or DSL modems, a poor connection between the modem and the service provider network, or by issues within the service provider network itself.