Integrated VoIP Audio

General Questions

- What are the features of Integrated VoIP audio?
- What is the attendee capacity for Integrated VoIP audio?
- How does WebEx VoIP audio help me know if attendees cannot hear me?

What are the features of Integrated VoIP audio?

The following are brief descriptions of the features of Integrated VoIP audio:

- **Services Support**: Integrated VoIP is supported in the services and platforms listed in the following table:

Center	Windows	Solaris	Linux	Macintosh
Meeting Center	Yes	Yes	Yes	Yes
Training Center	Yes	Yes	Yes	Yes
Event Center	Yes	Yes	Yes	No
Support Center	Yes	Yes	Yes	Yes

- **Hardware requirements**: No special hardware is required to use integrated VoIP. A full duplex sound card and speakers or headset are all that is required. A headset is recommended for users wishing to speak. USB devices are also supported on Windows and Mac as long as the device has drivers for the operating system.

- **Number of attendees**: Integrated VoIP supports up to 500 attendees (1,000 for Training Center).
- **TCP/UDP support**: Integrated VoIP can use UDP or TCP as a transport method. UDP transport allows offers lower latency for VoIP sessions. TCP offers optional SSL security with slightly higher latency. When VoIP is started as a service, each client attempts to connect via UDP first and then reverts to TCP as a transport protocol. Meetings can support a mix of UDP and TCP attendees.
- **SSL Support**: Integrated VoIP can (if the site is SSL enabled) use SSL as a transport method. SSL provides highly secure transport for VoIP traffic. SSL may introduce additional latency for VoIP connections.
- **Echo Cancellation**: Integrated VoIP features built in echo cancellation offering improved performance under most conditions.
- **Automatic Gain Control**: Integrated VoIP features automatic gain control provides level equalization for meeting attendees.
- **Cross-platform support**: Cross-platform support for Support Center

What is the attendee capacity for Integrated VoIP audio?

You can invite up to 500 attendees to a session (1,000 for Training Center).

How does WebEx VoIP audio help me know if attendees cannot hear me?

Integrated VoIP displays a network indicator in the **Volume** window (available from the Audio menu) that shows how your network is performing and the overall quality of the audio your attendees hear. The indicator displays one of the following colors:

- Green, when more than 85% of your attendees are experiencing good-quality audio
- Yellow, when fewer than 50 to 85% of your attendees are experiencing good-quality audio
- Red, when fewer than 50% of your attendees are experiencing good-quality audio

System Requirements and Technical Information

- Is VoIP a full or half duplex transmission?

What are the system requirements?

Your system must meet the requirements shown in Cross-platform Features (WBS30).

Which WebEx service sites offer VoIP audio?

You can use Integrated VoIP with the WebEx services and computers listed in the following table:

Center	Windows	Solaris	Linux	Macintosh
Meeting Center	Yes	Yes	Yes	Yes
Training Center	Yes	Yes	Yes	Yes
Event Center	Yes	Yes	Yes	No
Support Center	Yes	Yes	Yes	Yes

What do I need to use WebEx Integrated VoIP?

To use WebEx Integrated VoIP, you will need a full duplex sound card and speakers or headset. To speak, you should have a microphone that is connected to your computer. For best results, we recommend that you use a headset.

Can I use TCP, UDP, or PSTN with WebEx Integrated VoIP audio?

You can use the UDP or TCP protocols with WebEx VoIP audio. With UDP, you may experience lower latency rates (delays) than with TCP, but with TCP, you can use the SSL security protocol (and the latency rate will probably be greater). When VoIP starts, WebEx tries to connect using UDP and then switches to TCP. You can conduct sessions where some attendees use UDP while others use TCP.

UDP is only supported for non-SSL sites. In order to use UDP, the IP ports 9000 and 9001 must be opened for outbound communication using UDP on the corporate firewall. UDP is selected automatically if the ports are open.

Can I use WebEx Integrated VoIP if my site is SSL-enabled?

Yes. You can use SSL if you also use the TCP transport protocol.

Can I use VoIP over dial-up connections?

Integrated VoIP is not recommended for dial-up connections. UCF-based PowerPoint sharing should work satisfactorily as long as video is not enabled and only one active microphone is in use. Application and desktop sharing in concert with Integrated VoIP is not supported on connections of less than 56Kb/s.

Can I provision WebEx VoIP from an EMX node.

Integrated VoIP can be provisioned from a WebEx™ Extended MediaTone eXchange (EMX) node on a case-by-case basis. Please contact Product Management for further information.

Is VoIP a full or half duplex transmission?

Integrated VoIP is full duplex, meaning multiple attendees can speak at the same time. This is similar to a traditional teleconference using PSTN. Half duplex is a VoIP conference where only one attendee can speak at a given time, similar to a CB radio.

Troubleshooting

- Why is there a delay in the audio during my VoIP conferences? Why does the quality seem to be not as good as traditional telephony?
- Why do I get good quality on some VoIP calls, but not on others?
- What if the customer experiences any technical issues with Integrated VoIP?

Why is there a delay in the audio during my VoIP conferences? Why does the quality seem to be not as good as traditional telephony?

Traditional PSTN-based teleconferencing is circuit-based, which gives each participant a dedicated channel to the teleconference bridge; the delay is virtually unnoticeable. Typically, the only delay one encounters in a circuit-switched voice environment is due to the distance the voice must travel). A good VoIP solution will have delay of about 0.25 - 0.5 seconds, depending the following factors:

- **Network congestion**: VoIP solutions send the voice information over an IP network (such as the Internet), which is a shared medium on which the packets are routed on a first-in/first-out basis. Congestion on any of the routers between the meeting participants will delay and/or degrade the audio.
- **Encoding process**: When you speak into the microphone, the sound card in your PC captures and digitizes the sound. This information is then broken up into data packets that are sent over the network to the conference server(s). The conference server(s) sends these packets to the other attendees' PCs, where the encoding process is reversed. The encoding process for integrated VoIP relies on audio components (microphone, speakers, and sound card), and these can vary greatly from PC to PC. Lower quality components will produce lower quality audio.

Such delay and audio quality issues are common to the VoIP solutions offered by all the vendors—not just WebEx. VoIP solutions offered by vendors such as Centra, et al., suffer from the same shortcomings when compared to PSTN. Based on our testing, the delay and audio quality of WebEx VoIP is at least on par with that of Centra's.

Why do I get good quality on some VoIP calls, but not on others?

It is hard to have a straight answer to this question due to the number of possibilities. You can have a perfect VoIP conference with a 28-Kbps connection to a country halfway around the world, followed by a scratchy mess for a call to the next state with a 56-Kbps or a 300+-Kbps connection. The quality is almost entirely determined by the sample rate (number of "slices" per second it uses to reproduce your voice) of the VoIP software, plus the throughput of your internet connection. A 56-Kbps (or a 300+-Kbps LAN, for that matter) connection does not ensure that you can move data across the Internet at that speed; the actual speed is determined by traffic levels on all networks between the source and end point, and the equipment capabilities at the source and end point. In general, poor-quality transmissions are a result of traffic and cannot be avoided completely in VoIP that uses Internet for all or part of the voice-data traffic.

What if the customer experiences any technical issues with Integrated VoIP?

Follow the standard Technical Support escalation process.