Two way VoIP over wireless nets requires low latency

By Jan Linden, Global IP Sound Inc., San Francisco, Ca.
iApplianceWeb
December 1, 2003
URL: http://www.iapplianceweb.com/story/OEG20031201S0089

Many factors affect the perceived quality in two way communication. An extremely important parameter is latency. In traditional telephony, long delays are basically experienced only for long distance calls and calls to mobile phones. This is not necessarily true for voice over Internet Protocol (VoIP), where the effects of excessive delay have often been overlooked, resulting in significant quality degradation even in short distance calls. Wireless VoIP, typically over a wireless local area network, is becoming increasingly popular, but even further elevates the challenges of delay management.

The impact of latency on communication quality varies significantly with the use. For example, long delays are not perceived as annoying in a cell phone environment as in a regular wired phone because of the added value of mobility. The presence of echo also has a significant impact on sensitivity to delay. Echo occurs either because of impedance mismatch in the PSTN network or because of acoustic feedback in a speakerphone or similar device. As long as the latency is not too high, echo cancellation algorithms can remove most of the effects.

For very long delays (greater than 200 milliseconds), even if echo cancellation is used, it is hard for users to maintain a two way conversation without interrupting each other. This effect is often amplified by shortcomings of the echo canceler design. The International Telecommunication Union's Standardization Sector, or ITU T, recommends in standard G.114 that the one way delay should be kept lower than 150 ms for acceptable conversation quality.

There are many sources of delay in a VoIP system that add up to the total latency. Among them are the following.

Causes for delay in VoIP

Algorithmic delay. This source of delay is related to the speech codec used, and occurs because of framing for block processing, including lookahead. Also, part of the algorithmic delay is the delay incurred by pre and postprocessing; for example, echo cancellation, noise suppression and filtering.

Processing delay. This is related to the signal processing performed and depends on the available CPU power, but is limited by the duration of one frame for real time operation.

Network delay. This is caused by physical delay in the transmission lines, buffers in routers, jitter buffers and so on, and is time varying. The transmission delay is split into two parts: a constant or slowly varying network delay, and rapid variations referred to as jitter. Included in the network delay is also the effect of sending several frames in one packet, a technique used to reduce the protocol overhead in IP communications. There are other issues related to implementing speech processing functions in the endpoint devices that can have a significant impact on quality and delay. Examples are the analog to digital converter, voice activity detection, echo cancellation and general speech processing issues.

Hardware delay: One major potential contributor to the hardware delay is clock drift, which occurs when the two devices talking to each other have slightly different clock frequencies. If the clock drift is not detected accurately, delay will build up during a call.

WLAN characteristics

The recent boom in wireless LAN deployment has generated a lot of interest in making VoIP available over such networks. Applications include wireless phones in the enterprise, dual mode cell phones that can connect to a WLAN for access to higher bandwidth when available and VoIP enabled personal digital assistants (PDAs).

The challenges in deploying VoIP over WLAN stem mainly from issues related to access point congestion and to those that affect the link quality. The result is that WLANs experience significantly higher delay, with more network jitter and packet loss, than wired LANs. When several users are connected to the same access point, congestion easily occurs. The result is jitter that can be very significant, especially if large data packets are present. Also, the efficiency of the system quickly deteriorates when the number of users increases.

An informal test has been carried out that showed a bandwidth utilization factor of less than 4 percent for VoIP due to access point congestion. Interestingly enough, using a higher compression speech codec did not increase the number of channels that could be handled.

Degraded link quality and, consequently, reduced available bandwidth occurs because of limited radio coverage and interference from other devices. Poor link quality also leads to an increased number of retransmissions, a situation that directly affects the latency and jitter.

Wireless VoIP

In the previous section it was pointed out that the level of jitter is much higher in a wireless than in a wired LAN. In fact, in an 802.11b network, jitter with standard deviation of up to 50 to 100 ms is not unusual, while in an enterprise LAN, rarely more than a couple of milliseconds of jitter is experienced. Hence, when designing VoIP devices for WLANs, it is of utmost importance to take the characteristics of the WLAN into account and minimize overall delay.

There are two main approaches available to reduce the delay: One is to make the VoIP devices less sensitive to the WLAN characteristic, and the other is to make the WLAN itself cope better with voice applications.

No other endpoint voice processing functionality has a bigger impact on the latency in a WLAN environment than the jitter buffer. Jitter buffer design is a trade off between low latency and good speech quality. To keep the delay as short as possible, the jitter buffer algorithm must adapt rapidly to changing network conditions. However, the traditional packet buffer approach is limited in its adaptation granularity by the packet size, which significantly limits the achievable performance.

Global IP Sound has developed an advanced adaptive jitter buffer control combined with error concealment. This unique approach allows the jitter buffer to quickly and with high resolution adapt to changing network conditions, and to ensure high speech quality with minimal buffer latency. Experiments show that, with this type of approach, one way delay savings of 30 ms to 80 ms are achievable in a typical 802.11b environment while sustaining the same speech quality. Another major advantage is that the algorithm automatically takes care of clock drift without introducing any audible artifacts or extra delay.

Powerful approach

Implementing quality of service (QoS) mechanisms is another powerful approach to improving VoIP over WLAN performance. The goal is to improve current standards with additions that take into account the different requirements of regular data traffic and time sensitive voice traffic.

However, even if QoS mechanisms are implemented, it is clear that WLANs will continue to offer much more challenging conditions than the typical wired LAN. Hence, QoS has to be combined with efficient jitter buffer implementations and careful latency management in the system design.

Another major challenge in wireless VoIP is to achieve proper handling of the audio in the wireless device. These devices are often small and may have been originally designed for other applications. PDAs, for instance, impose challenges in achieving low delay.

Factors that influence the latency include buffering in the audio interface, effects of using a non real time operating system, proper task scheduling and the use of low delay speech processing components. If such considerations are taken into account and audio processing expertise is applied in the design process, significant latency savings can be achieved.

High latency has a significant impact on the perceived quality of a telephone conversation, which imposes additional requirements on wireless VoIP due to the characteristics of wireless networks. However, proper design, including an efficient jitter buffer, can mitigate most of the negative effects.

Jan Linden (jan.linden@globalipsound.com) is the vice president of engineering at Global IP Sound Inc. (San Francisco).