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Finding Harmony Between VoIP and WLANs

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Delivering voice services over IP links has been a challenges members of the communication design community have been tasked with solving for many years. Now, with deployments on the rise, developers must embark on their next challenge—delivering VoIP capabilities to lossy wireless LAN environments.

Combining VoIP and WLAN is an obvious match that receives significant interest as users learn that connecting to WLANs with their portable devices provides the environment necessary to implement "anywhere, anytime" access. However, as VoIP moves into the wireless world, performance issues arise because the characteristics of wireline and wireless networks differ.

In this article, we'll examine how traditional VoIP issues, such as delay, are magnified in an 802.11 WLAN environment. We'll then show some techniques designers can implement to bring harmony between WLAN and VoIP technologies.

Traditional VoIP Challenges
Before looking at the factors that impact VoIP performance over a WLAN link, let's first examine some of the key factors that impact the quality of VoIP performance in traditional IP networks.

Many factors affect the voice quality for a VoIP call. Many of them are similar to what is experienced in the PSTN or regular wireless phone systems such as the impact of equipment design, echo, and the speech codec used. However, there are three major factors specifically associated with IP networks that have significant impact on speech quality for VoIP: delay, jitter, and packet loss.

Delay causes several problems in a VoIP connection. The biggest is talker overlap, i.e. it is hard to maintain a two-way conversation without persons starting to talk at the same time. The presence of echo also has a significant impact on sensitivity to delay. The International Telecommunication Union (ITU) recommends in standard G.114 that the one-way delay should be kept lower than 150 ms for acceptable conversation quality.

In a VoIP design, the major delay contributions are algorithmic delay, processing delay, network delay, and delay stemming from hardware interfaces. Algorithmic delay is related to the speech codec used, and occurs because of framing for block processing, including look-ahead. Also part of the algorithmic delay is the delay incurred by pre- and post-processing; for example, echo cancellation, noise suppression, and filtering.

Processing delay 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. It is worth noting that low bit-rate speech codecs typically have longer algorithmic delay and require much more processing power than a high bit-rate codec which results in significantly higher delay.

The delay in the IP network is a time varying delay that is caused by propagation delay in the transmission lines, buffers in routers, and jitter buffers. Transmission delay is split into two parts: a constant or slowly varying network delay and rapid variations referred to as jitter. Because of the nature of the IP network, the amount of delay is different in each direction.

The jitter present in packet networks complicates the decoding process in the receiver device because the decoder needs to have packets of data readily available at the right time instants. A jitter buffer is normally used to make sure that packets are available when needed, resulting in additional delay that increases with the magnitude of the jitter.

Packet loss occurs either if a packet is lost in the network or if a packet arrives too late to be handled by the decoder. By allowing for a long delay in the jitter buffer, the latter type of packet loss can be almost completely removed, but at the price of increased system delay.

Applying it to Wireless LANs

Because of the lossy nature of the 802.11 medium, the VoIP effects detailed above are even further accentuated. The challenges in deploying VoIP over WLAN stem mainly from issues related to access point congestion and various issues that affect the link quality. The resulting effect is that WLANs experience significantly higher delay, 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 sent in the same network. The efficiency of the system quickly deteriorates when the number of users increases.

When roaming around in a wireless network the mobile device will at some point have to switch which access point it is associated with. In a traditional WLAN it is common that such a hand-off will introduce 500 ms of delay, which will have a very audible impact on the call quality. However, many solutions are now available that cut that delay number to about 20 to 50 ms, if the user is not switching between two IP subnets. In the case of subnet roaming the hand-over is more complicated and no really good solutions exist currently. Therefore it is common to plan the network in such a way that likelihood of subnet roaming is minimized.

An informal test has been carried out. The impact on packet loss and jitter of the number of simultaneous calls over an access point with perfect coverage for all users is depicted in **Figures 1** and **2** . Each call was using G.711 with a packet size of 20 ms which, including IP headers, results in a payload bandwidth of 80 kb/s. These results suggest that only five calls can be allowed for acceptable quality, resulting in a bandwidth utilization factor of much less than ten percent.

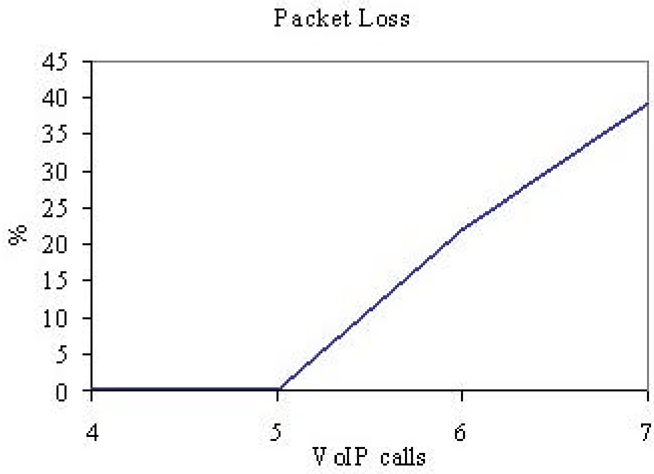

Figure 1: Effect of access point congestion on the amount of packet loss as a function of the number of simultaneous VoIP calls through one access point.

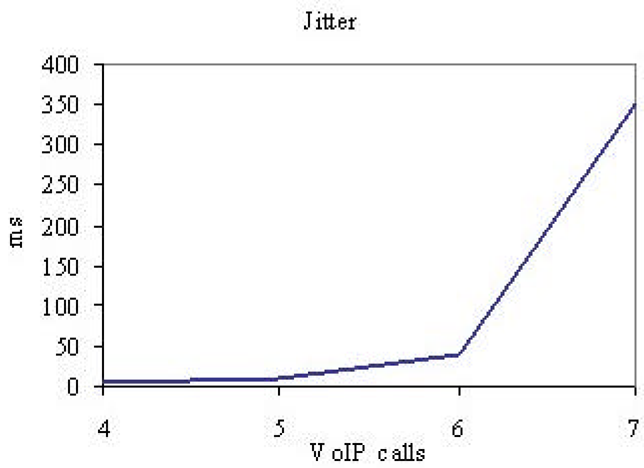

Figure 2: Effect of access point congestion on network jitter as a function of the number of simultaneous VoIP calls through one access point.

Interestingly enough, using a higher compression speech codec did not increase the number of channels that could be handled. The reason is that access point congestion depends much more on the number of packets the access point has to process than on the actual bandwidth. Voice packets are small and sent very frequently which explains the low throughput for voice packets. Because of this limitation it is common to put several voice frames into the same packet, which reduces the number of packets and hence increases the throughput. However, as a result the delay will increase.

Link Quality Issues

Sensitivity to congestion is but one of the limitations of 802.11 networks. Degraded link quality, and consequently reduced available bandwidth, occurs due to a number of reasons. 802.11 systems typically operate in the unlicensed 2.4 GHz frequency range and share this spectrum with other wireless technologies, such as Bluetooth and cordless phones which causes interference with potentially severe performance degradation since a lower connection speed than the maximum is chosen.

Poor link quality also leads to an increased number of retransmissions, which directly affects the delay and jitter. The link quality varies rapidly when moving around in a coverage area. This is a severe drawback, since a WLAN is introduced to add mobility and a wireless VoIP user can be expected to roam around the coverage area. Hence, the introduction of VoIP into a WLAN environment puts higher requirements on network planning than for an all-data WLAN.

The result of the very high delays that occur due to access point congestion and bad link quality is that often the packets arrive too late to be useful. Therefore, the effective packet loss rate after the jitter buffer is typically significantly higher for WLANs than for wired LANs.

Solving the WLAN / VOIP Challenge

As has been pointed out here, WLAN systems offer a much tougher environment than a wired LAN. Hence, when designing portable VoIP devices for WLAN, it is of utmost importance to take the special characteristics of the WLAN into account. Both software and hardware issues need proper attention to achieve the best overall quality.

The use of high compression codecs results in higher delay and since the access point congestion is mainly affected by the number of packets rather than the bandwidth there are basically no good reasons not to deploy a high quality, high bit-rate, and low complexity codec such as G.711.

A poor jitter buffer can have a disastrous effect on the delay and quality for a wireless device. In order to keep the delay as short as possible, it is important that the jitter buffer algorithm adapts rapidly to changing network conditions. Therefore, jitter buffers with dynamic size allocation, so-called adaptive jitter buffers, are now most common. In these buffering schemes, adaptation is achieved by inserting packets in the buffer to increase the delay and discarding packets to lower the delay.

Packet insertion is typically accomplished by repeating the previous packet, thus causing audible distortion. Therefore, adaptive jitter buffer algorithms are very cautious when it comes to changing the delay. This traditional packet buffer approach is limited in its adaptation granularity by the packet size.

A new algorithm has recently been introduced that combines an advanced adaptive jitter-buffer control with error concealment. Combining adaptive jitter control and packet loss concealment into one unit makes this algorithm capable of adapting the buffer size on a millisecond basis. The approach allows it to quickly adapt to changing network conditions, and to ensure high speech quality with minimal buffer latency.

Experiments show that with the combined adaptive jitter/error concealment approach one-way delay savings of 30 to 80 ms are achievable in a typical 802.11b environment. Another major advantage is that this type of algorithm is capable of automatically taking care of clock drift between devices without introducing any audible artifacts or extra delay.

To seamlessly handle a hand-off between two access points, it is important that the jitter buffer is capable of very rapidly increase the delay without audible distortion. When the hand-off is completed the delay should be quickly reduced again for best performance.

Managing Packet Loss

As was mentioned earlier, the amount of packet loss is typically much higher for WLANs than for wired LANs, which put high requirements on the capability to cope with packet loss. When a packet is lost, some mechanism for filling in the missing speech signal must be incorporated. Simple methods, like repeating the previous packet, do not provide sufficient quality for wireless applications. A sophisticated algorithm, on the other hand, can handle 10 percent of packet loss without noticeable degradation.

Another approach to handle packet loss is to deploy a speech coding technique that has been specifically designed to handle packet loss. None of the current speech coding standards (e.g. ITU codecs) have been designed in such a way and hence are all sensitive to packet loss. However, new robust codecs are being adopted outside of the traditional standards bodies for speech coding. For example, the Internet Engineering Task Force (IETF) is currently standardizing the iLBC speech codec.

Implementing quality of service (QoS) mechanisms is another powerful approach to improve VoIP over WLAN performance. The goal is to introduce additions to the current 802.11 media access control (MAC) layer that take into account the different requirements of regular data traffic and time-sensitive voice traffic.

Prioritization of voice (and video) packets is the first level of QoS that is typically introduced. However, the result of the recently conducted case study suggests that prioritization is not enough since only a few VoIP calls could be carried by one access point even without any data traffic present. Therefore, to be successful, the WLAN sector must compliment its QoS work with changes to the 802.11 standards that will reduce the jitter and delay by changing the actual transmission scheme.

The development of WLAN QoS standards has been quite slow, and it will take significant time before significant deployment will be seen. Also, QoS mechanisms are necessary but not sufficient to guarantee good quality. 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 other techniques to achieve acceptable voice quality.

Many factors in the design of mobile devices affect quality. Obvious examples are microphones, speakers, and analog-to-digital converters. Several signal processing components such as, choice of speech codec, filtering, and echo cancellation also significantly impact the quality and delay. These issues are all very similar to challenges well known from designing devices for wireless telephony.

Another major challenge is to achieve proper handling of the audio in the hardware interfaces of a wireless device. These devices are often small and/or originally designed for other applications, e.g., PDAs, which imposes challenges in achieving good voice quality and low delay. It is important that the latency be minimized in every component of the implementation.

Overcoming the Challenge

It is clear that many challenges of enabling VoIP in WLANs do exist. However, with proper hardware design, and utilization of advanced sound processing most of the difficulties can be overcome, helping make the spread of the technology easier and faster.

About the Author

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