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Challenges for VoIP Deployment in the Enterprise

VoIP is coming of age but it still faces a myriad of issues, especially in the wireless enterprise. Understanding the technical issues of VoIP in wireless environments makes development and deployment a bit less inundating. |By Jan Linden

Telephony solutions based on voice over Internet protocol are quickly overtaking traditional telephony system installations in the Enterprise space. Many factors in the past have slowed down the anticipated growth of VoIP. Today, VoIP solutions that achieve levels of quality and reliability that customers are used to receiving from the public switched telephone network are being observed as the market grows quickly. However, reasons for concern about quality remain. Both the Internet protocol network and the actual VoIP equipment have to be carefully designed and managed to achieve good quality. Major challenges face the designers and integrators of VoIP enterprise solutions.

Enterprise IP Networks

Introducing VoIP into an enterprise network places serious requirements on the IP network. The network then has to be capable of handling time-critical voice packets together with a significant load of data packets generated by applications, such as email and web browsing.

There are three major factors associated with packet networks, that have a significant impact on perceived speech quality. They are:

Delay – the transmission time for the voice packets through the network, plus related signal processing algorithmic and processing delays. The goal is to keep the one-way delay less than 100 ms. At this rate, users do not notice the delay. If, however, delay exceeds 150 ms, the effect is clearly noticeable. These delay guidelines assume that there is no echo, or that echo cancellation is deployed — otherwise, delays less than 50 ms are annoying.

Jitter – the variation in transmission time that occurs due to congestion and queuing. To guarantee the availability of a constant stream of voice data for loudspeaker playout, a jitter buffer must be used on the receiving endpoint device. Jitter is one of the most challenging problems to be resolved when deploying VoIP enterprise networks.

Packet loss – when a packet is discarded or delayed so long that it is no longer useful to the receiver when it finally arrives. In a well-managed enterprise network, packet loss rates are low (a common goal is to keep the packet loss less than 1%). Even so, this problem cannot be overlooked because every lost packet results in audible distortion if a proper concealment technique is not deployed.

All three factors stem from the nature of a packet network, which does not guarantee that a packet of speech data will arrive at the receiving end in time, or even that it will arrive at all. In contrast, traditional telephony networks rarely, or never, lose packets, and often the transmission delay is a fixed parameter. These network effects are the most important factors distinguishing speech processing for VoIP from traditional solutions.

The effect of delay on communication quality varies with the type of use. For example, long delays are not perceived as being as annoying in a cell phone conversation as they are in a wired phone conversation because of the added value of mobility.

Even though enterprise networks typically are already, or can be modified to be, suitable for VoIP, connections to other networks such as the Internet and wireless networks raise some concerns. Delay, jitter, and packet loss must be taken into account for a VoIP solution to be successful. Two approaches have been

GLOSSARY OF ACRONYMS

802.11 - a family of specifications developed by the IEEE for wireless LAN technology

802.1p/q - an IEEE standard for quality of service at the medium access control level

Bluetooth - a standardized, short-range, wireless interface at 2.4GHz used to connect desktop and notebook computers, personal digital assistants, mobile phones, camera phones, printers, digital cameras, headsets, keyboards, a computer mouse and some household electronic devices

Diffserve - differentiated services, an architecture for providing different types or levels of service for network traffic

G.711 - the international standard for encoding telephone audio on an 64 kbps channel using pulse-code modulation an 8 kHz sample rate, with 8 bits per sample

G.723.1 - an audio codec for voice that compresses voice audio in chunks of 30 milliseconds, mostly used in VoIP applications for its low bandwidth requirement

G.729A - a low bit rate speech coder standard for compressing toll quality speech (8,000 samples/second)

GIPS - a family of codecs and related software from Global IP Sound that is intended to maintain voice quality with as much as a 30% packet loss

IETF - Internet Engineering Task Force

ILBC - Internet Low Bit-Rate Codec

IP - Internet Protocol

LAN - Local Area Network

PR - Packet Repetition

PSTN - Public Switched Telephone Network

QoS - Quality of Service

RSVP - a resource reservation protocol that defines several data objects that carry resource reservation information

USTA - United States Telephone Association

VoIP - Voice over Internet Protocol

VPN - Virtual Private Network, a network using public communications media (such as the Internet) to connect nodes. With encryption and security mechanisms, only authorized users can access the network, and its data cannot be intercepted, making it "virtually private."

WLAN - Wireless Local Area Network

ZS - Zero Insertion

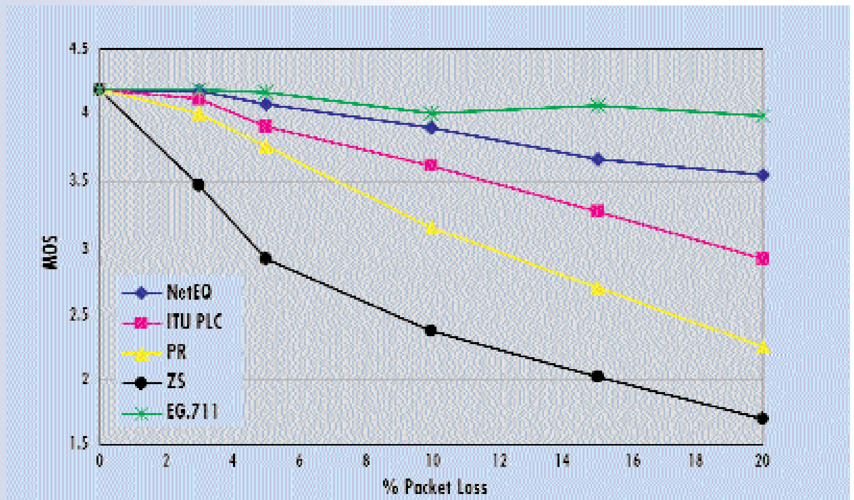

Figure 1. Subjective test results for different approaches to handling packet loss concealment. Source: Lockheed Martin Global Telecommunication (COMSAT).

brought to this challenge.

Network QoS

Because network imperfections have a direct effect on voice quality, it is of utmost importance that the network is properly designed and managed. Measures taken to ensure network performance are usually referred to as *quality of service* techniques. High quality of service can be achieved by applying the following types of techniques:

Capacity management – Inside the enterprise, the amount of bandwidth available rarely is a problem. Most bandwidth problems are related to the connection to a wide area network or in other access links. If a major upgrade of the internal network capacity is necessary, it can be costly.

Prioritization techniques – By prioritizing voice over less time-critical data in routers and switches, delay and jitter can be significantly reduced without any noticeable effect on the data traffic. Many standards are available for implementing differentiated services, including 802.1p/q and DiffServ. In addition, RSVP can be used to reserve end-to-end bandwidth for the voice connection.

Network monitoring – As the needs and requirements of networks continuously change, it is important to implement ongoing monitoring and management of the network.

QoS techniques are covered extensively in literature available on the topic. These types of techniques can be successfully integrated in an isolated enterprise network. Challenges typically arise when the traffic is routed through an unmanaged network. One particularly interesting and challenging example is a telecommuter connecting to the enterprise network through a virtual private network.

Wireless LAN Challenges

Traditionally, enterprise networks have been limited to wired Ethernet solutions that are fairly straightforward to manage. The recent boom of wireless local area network solutions, however, is quickly changing the enterprise network landscape. WLAN, in particular the 802.11 family of standards, offers mobility for computer access but also the flexibility of wireless IP phones and are hence of great interest for VoIP systems.

The development of QoS standards for WLAN has been quite slow, and it will take time before large-scale deployment will be seen. In fact, due to the nature of the wireless network (interference and congestion), even if QoS mechanisms are implemented, it is clear that WLANs will continue to offer much more challenging conditions than the typical wired LAN.

Jitter and effective packet loss rates are significantly higher in WLAN than in a wired network. Furthermore, the network characteristics are rapidly changing. In addition, as the user moves around in the network, levels of coverage vary and interference levels from other radio source change. Interference sources include cordless phones, Bluetooth devices and microwave ovens. The result is that high-level voice quality is much harder to guarantee in a wireless network than in a typical wired LAN.

WLANs are advertised as having high throughput (11 Mb/s for 802.11b and 54 Mb/s for 802.11a and 802.11g). However, field studies show that actual throughput is often only half of the advertised figures, even when the client is close to the access point. It has been shown that, because of the high packet rate of VoIP, these numbers are even worse; with typical throughput values of 5% to 10%[1].

Although security issues once were a serious concern, the advent of authentication and encryption standards for the 802.11 family of wireless systems has resolved those problems for most users.

QoS techniques, as powerful as they are, cannot overcome all VoIP problems in practical enterprise networks, especially not in WLANs. Therefore, to achieve the best possible VoIP quality, other methods are necessary to complement the network QoS solutions.

Endpoint QoS

Previous sections covered network issues related to VoIP in the enterprise and concluded that a perfect network does not exist. If a VoIP device cannot cope with at least some level of network degradation, the quality can never be acceptable. Therefore, the characteristics of the IP network must be taken into account in the design and implementation of VoIP products as well as in the choice of speech processing components, such as the speech codec.

Network QoS offers a powerful way of achieving high network performance, which in turn facilitates high-quality voice. A properly managed enterprise IP network is well behaved in terms of delay, network jitter, and packet loss — factors that directly affect VoIP quality. When network issues have been the reason for sub-par VoIP performance, the traditional approach to solving the problems has been to try to fix the network, either through increasing the capacity or introducing/improving network management. Although capacity and network management improvements work, they do not necessarily represent the most scalable and cost-efficient solution.

For example, if a significant portion of the total IP traffic is voice packets, QoS methods based on prioritization will fail. Furthermore, for a certain QoS technique to work properly it has to be implemented for every single piece of equipment in the network, otherwise the whole benefit could be lost. A typical example would be an initial installation where all switches were properly configured for voice prioritization. But when the need arises to split an Ethernet endpoint for several users, a switch is installed that either does not support QoS at all or is improperly configured. Also, portions of the network cannot be managed in the same manner as the basic enterprise LAN, such as WLANs and VPN connections, and sometimes a call has to be routed over an unmanaged network.

VoIP endpoints can never be implemented under the assumption that the network is perfect; therefore, every VoIP device has to be designed to cope with network degradation. The concept that the endpoints are made robust against network degradation to compensate for lack of network QoS is referred to as "endpoint QoS."

Endpoint voice processing software can be improved in many ways to cope with higher levels of network impairments as an alternative, or complementary, solution to traditional QoS techniques. This allows for higher flexibility in coping with the network load variations typically seen during the day. Moreover, having more robust VoIP endpoints allows for more flexibility when equipment failures and network upgrades are handled.

No other end-point voice processing functionality has a bigger effect on the latency in a VoIP environment than the jitter buffer. Jitter buffer design is a trade-off between low latency and good speech quality, and 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 limits the achievable performance.

A new technique combines advanced adaptive jitter-buffer control with error concealment. This 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.

The choice of speech codec sets the upper limit for the achievable quality in VoIP, disregarding the network issues discussed here. Unfortunately, often in the interest of saving bandwidth, a codec that does not provide high basic quality is deployed (e.g. G.723.1 and G.729A). Another drawback is that these codecs do not provide high robustness against packet loss. For G.711, which does not have an internal packet loss concealment solution, an external packet loss technique has to be deployed.

The figure below depicts subjective listening test results for a number of approaches to handle packet loss for the G.711 codec. The purpose is to show what remarkable improvements can be achieved by deploying endpoint QoS solutions. Clearly, simple zero insertion and packet repetition are not providing quality that can be classified as acceptable. The packet loss concealment method described in G.711 Appendix I (marked ITU PLC in the figure) offers reasonable quality. As an example of what recent developments can provide in terms of robustness, results are included for the method described in[2] (NetEQ). This method is clearly outperforming the traditional methods. The next step is to deploy a speech codec specifically designed for packet networks such as Enhanced G.711 (EG.711 in the figure). These two latter methods are examples of new wave of techniques that have been specifically designed for IP networks and therefore provide better robustness than traditional solutions. Another example is the iLBC speech codec[3] that is currently being standardized by the Internet Engineering Task Force.

The levels of jitter and packet loss experienced in a wireless LAN are many times higher and vary much quicker than what is observed in a well behaved wired LAN. The requirements in terms of network robustness on wireless VoIP devices are thus quite tough, but no less important than other endpoints, such as IP phones and gateways. These devices have to be able to cope with the characteristics of wireless networks for participants on both sides of a conversation to receive good quality voice reproduction.

Summary

Network QoS techniques are necessary for successful deployment of enterprise VoIP. However, deploying such methods are not sufficient to guarantee good quality. Also the actual VoIP endpoints have to be specifically designed to cope with network degradations. In addition to choosing an appropriate speech codec, the most important signal-processing component is the jitter buffer. When properly designed, the endpoint should be able to handle even tough cases such as WLANs, VPNs and unmanaged networks. The end result is a system that provides graceful quality degradation and great scalability. The good news is that the software components necessary to provide endpoint QoS are available, placing hence high-quality enterprise VoIP within reach.

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