GLOBAL IP SOUND

Software company demonstrates edge device architecture for QoS

The challenges inherent in using packet networks for interactive voice communications arise from the real-time characteristics of speech. The three most important factors that affect speech quality are packet loss, delay, and jitter. The level of packet loss and the amount of delay and jitter vary greatly with the network, location, and time. VoIP calls between Global IP Sound offices in Europe, the US and Asia have shown substantial differences in the sound quality during interoffice calls. All international calls were affected by latency, packet loss, and jitter. Within Asia, however - specifically between Hong Kong and Mainland China - audio was most delayed and distorted, sometimes becoming unintelligible.

To better understand the nature of these Quality of Service (QoS) problems, we monitored and analyzed the transmission characteristics of the Internet between several endpoints over a four-week period. The data show that, compared to Europe and the US, degradation in the quality of communications was more severe with calls to, from, and within Asia.

The QoS problems inherent to voice-over-packet are magnified in Mainland China , most likely due to the light IP infrastructure and the surge of demand for data and voice traffic in this region. In addition, the bandwidth of the pipes that connect China to the rest of Asia and the world is very restricted. Monitoring also confirmed that the number of hops does not give a good indication of the quality of a connection.

Solving QoS problems at the network core is costly and inflexible. Spikes in Internet demand can be addressed by adding more bandwidth and more sophisticated network equipment, such as intelligent routers; other common approaches are to create separate networks for voice and data and/or to prioritize voice packets. But these solutions undermine the "common" nature of the global public Internet and are not as cost effective and scalable as other solutions. And no matter how much infrastructure is improved, packets will be lost and rerouted; it is the nature of the Internet.

Fortunately, there is another way to improve sound quality. Sound processing software running on edge devices can improve sound quality by providing robustness to packet loss, delay, and jitter.

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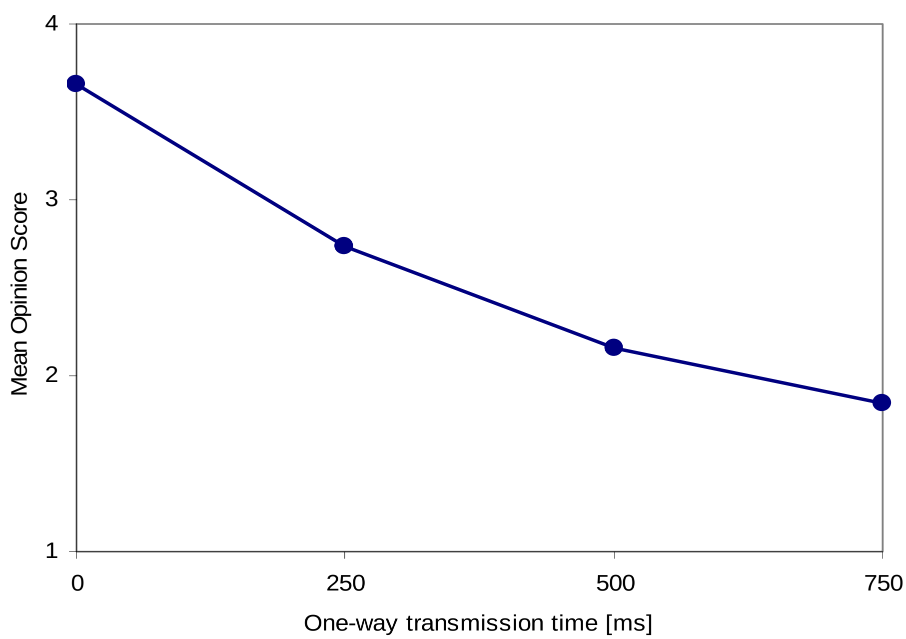
Sound information sent through the Internet always originates at an edge device, such as a voice gateway, PC, PDA, or IP phone. The analog sound signal is digitized, packetized with headers containing needed information such as destination and source IP addresses, and then is sent through the "Internet cloud" to its destination. There, it is unpacked and converted back into an analog sound signal. It is inevitable that between the sender and the destination, as packets progress from router to router, some will be lost, some will arrive out of sequence, and there will be variations in the arrival times of the packets.

For telephony applications, lost packets create gaps in voice communication. This may be heard as clicks and muted or unintelligible speech. Packet loss is caused by several sources. For example, a router may intentionally discard a packet because it was damaged during transmission or timed out of a queue due to congestion problems.
Congestion can also contribute to latency, which can make two-way voice conversation difficult. Latency is the sum of the time it takes to encode analog sound into a digital signal, transmit the digital information through the network, and decode back into sound at the receiving end. The acceptable delay for telephony, as set forth in the ITU-T G.114 Recommendation, is a maximum of 150 milliseconds (ms).



Source: ITU-T Standards Document G.114

Transmission delay is split into two parts, one being the constant or slowly varying network delay and the other being the rapid variations on top of the basic network delay, usually referred to as jitter. The jitter is defined as a smoothed function of the delay differences between consecutive packets over time.

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 instant. If not, the decoder will not be able to produce smooth, continuous speech. Thus, on top of its contribution to delay, it also causes a timing problem for the receiver. A jitter buffer (which introduces additional delay) is required to make sure that packets are available when needed for play-out.

On the Internet, these interrelated problems are all unpredictable. Calling over the Internet can be like making a call using a wireless phone. A caller can experience few to no interruptions in service one minute, and the next, the sound can be extremely delayed or unintelligible. It all depends on the connection – infrastructure, equipment, and route.

Our Hong Kong office experienced deterioration in quality more often than our U.S. and European offices. Calls were particularly difficult between Hong Kong and Mainland China.

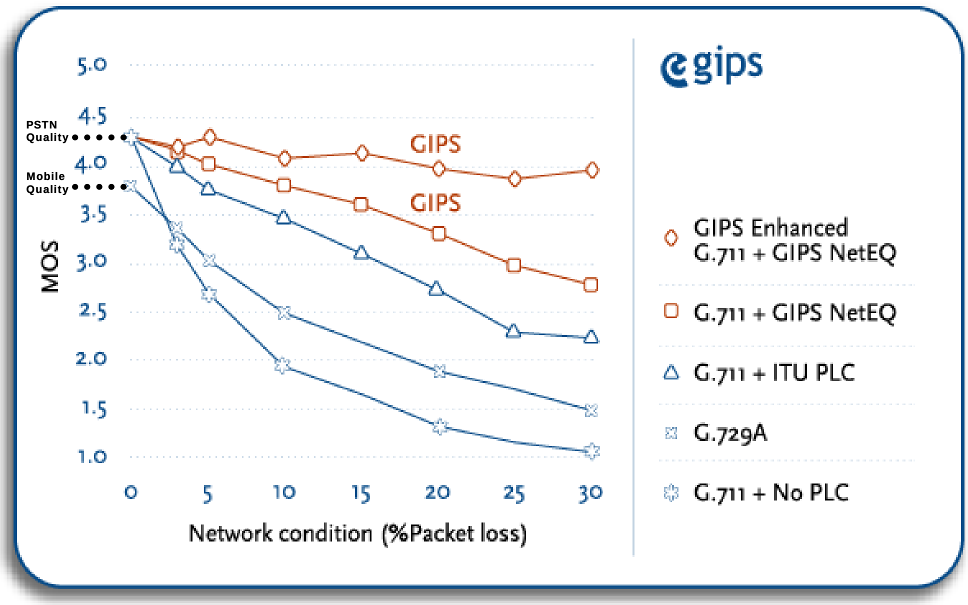
We observed that between Hong Kong and Urumgi, Xinjiang (in northwest China), Hong Kong and Beijing, and Hong Kong and Tianjin, delays could reach up to 650 and 900 milliseconds and packet loss peaked between 45 and 50 percent. Interestingly, the number of hops was roughly the same for calls in all tested connections, including calls from Hong Kong to San Francisco and even Stockholm. Furthermore, calls between Stockholm and San Francisco, Stockholm and Hong Kong, and San Francisco and Hong Kong all experienced packet loss but to a lesser degree. Average packet loss for typical calls during the “testing period” was about three percent, with average delays about half that of the Hong Kong-to-China calls.

Between	Packet Loss	Delay	Hops
Hong Kong < -- > Urumgi, Xinjiang	Up to 50%	Up to 900 ms	20
Hong Kong < -- > Beijing	Up to 45%	Up to 900 ms	18
Hong Kong < -- > Tianjin	Up to 45 %	Up to 650 ms	18 – 20
San Francisco < -- > Stockholm	Up to 3 %	Up to 250 ms	16 – 18
San Francisco < -- > Hong Kong	Up to 3 %	Up to 400 ms	17
Hong Kong < -- > Stockholm	Up to 3 %	Up to 400 ms	~ 16-18

The ratio of infrastructure to traffic demand determines the degree of resource contention, which in turn affects packet loss, delay, and jitter. With larger, built-out networks, packets travel between routers, finding the fastest route and avoiding bottlenecks to arrive within a reasonable time. As illustrated by the calls from Hong Kong to mainland China, when network conditions are less than ideal, communication quality deteriorates rapidly.

In the hundred-plus year history of the public switched telephone networks (PSTN) and the recent development of digital wireless telephony, the tradeoffs between voice quality and bandwidth requirements were addressed by standard codecs (G.711, G.729, G.723.1, etc.) that are used in all network equipment today. Unfortunately, in the design of these codecs, the unique requirements of IP telephony networks were not considered. Although they are being used for packet networks, they were designed only for circuit-switched networks.

Global IP Sound has developed a suite of sound processing software, including codecs, an adaptive jitter buffer, and error concealment, which when deployed on edge devices can sustain high voice quality even under severe network degradation conditions. An example of listening tests for these improved GIPS codecs is shown in these MOS (Mean Opinion Score) test results:



Source: Lockheed Martin Global Telecommunication (COMSAT)

Alternative solutions to GIPS sound processing software include hardware-based solutions for the network core, which tend to be inflexible, costly, and inefficient. Options such as rebuilding the entire network from the core and replacing all existing equipment with better, faster, "optimized" systems is costly and sometimes impractical. Another approach pursued by some carriers and equipment manufacturers is to support packet prioritization, thus ensuring that the effect of delay and packets on voice packets is minimized. This option might not be as costly as rebuilding, but it does introduce more fixed costs and is often inefficient.

Prioritization of voice packets can be an effective way of getting around congestion problems, by penalizing data traffic. However, in “voice-heavy” networks, voice traffic will still experience congestion. This solution does not scale as needed, and sooner or later more bandwidth will need to be deployed.

Building QoS solutions into edge or gateway devices is more cost-effective, scalable, and simpler, thus enabling high-fidelity sound even with poor networks, such as that between Hong Kong and mainland China. QoS becomes a function of the edge, and is not left up to the network operators.

About Global IP Sound

Global IP Sound develops voice-processing technologies for real-time communications on packet networks that result in better than cellular/PCS and PSTN calls. The company's codecs and algorithms are embedded into soft phones, IP phones, and gateways to overcome delay, jitter, packet loss, and acoustic and network echo. The increased quality and robustness enable the confident deployment of "traditional" VoIP and today's newer applications, including voice over WiFi and multi-user conferencing.

For additional information, please contact:
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