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VoIP Quality Issues -- Are they Really a Thing of the Past?

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The VoIP market is undergoing rapid expansion, with the improvement in voice quality making the platform more attractive to telecom carriers and enterprises aiming to cut costs and utilize greater bandwidth.

As more and more standards are being ratified for VoIP coders and signaling protocols, some analysts predict that the media gateway market will eventually standardize. But there is one issue that still keeps all VoIP media gateways from becoming equal – voice quality.

When VoIP technology was launched in the 1990s, voice quality was a big issue. The quality was lower than what could be achieved via PSTN phone lines. The technology was used primarily to save money on phone calls. Not only was voice quality poor compared with regular PSTN, but there were also problems with the network. At that time, the Internet was less reliable and the technology could not cope with problems such as packet loss, delays and cutoffs.

Since then, VoIP technology has evolved. Voice quality has improved and the gap between PSTN and VoIP has narrowed. In particular, the introduction of broadband Internet has enabled even more improvement. Now, media gateways can achieve voice quality through VoIP technology that is equivalent or even superior in some cases, to PSTN quality. In the near future, VoIP is expected to deliver a consistently better performance than PSTN.

VoIP can overcome the limitations of the PSTN bandwidth, which is limited to 3.3 KHz. VoIP technology is moving toward wideband technology, which enables IP telephony users to hear at up to 8 KHz.

For the user, wideband technology for speech transmission improves the perception of the voice of the other party. Voice transmitted over VoIP is becoming more natural, pleasant and clearer, making it easier, for example, to quickly and correctly identify the other caller. On regular phone lines, it can sometimes be hard to initially recognize a voice. Regular phone lines have a bandwidth that limits the transmission of information to only a portion that relates to the intelligibility of the speech. But this small missing portion is important for recognition.

The technological advantages of VoIP are therefore clear. The start of the VoIP takeover has already begun.

Media gateway capabilities not uniform

The foundation of the VoIP revolution is the media gateway, the unit that links telecommunications networks into VoIP.

Many media gateway elements have become standard, including the G.723 voice compression technology that enables networks to transmit voice signals, RTP protocols for packetization, and signaling protocols. But there is no strict standardization for the technologies of voice quality enhancement.

Speech sound quality and the capability of the media gateway to handle different problems that occur in the network are what make one vendor different from the other. As telecom operators race to bring the competing VoIP services into their portfolio, the importance of speech quality has never been greater.

Indeed, in the world of VoIP the media gateway is not just a compilation of parts, but a system that must be carefully designed with the capability to handle the changing variables that arise during communications and affect the quality of communications.

The system must handle variables such as echo, packet loss, double talk and transmission quality in the presence of background noise, and jitter or delay. Algorithms in hardware to manage these problems are not standardized. Therefore no two VoIP gateways will ever be equal.

VoIP is voice on top of an IP network that changes every second. Quality control depends on mechanisms in the gateway. Only the media gateway maker vendor can determine this quality.

Most customers looking to purchase a media gateway think they are all the same. They believe that the technology has become standardized based on SIP, even though it is still being updated.

However, the main parameter that will never become standard is the preservation of voice quality when moving from TDM to IP. TDM networks are based on G.711 coding, which makes all voice equal or with the most negligible differences. An IP network is never the same because of the number of factors influencing the ability of a network to move a packet from one destination to another.

Even the same IP network will not provide the same quality of service at any given point of time. The environment is constantly changing. The ability to maintain premium quality while moving voice on top of changing networks depends on the way the media gateway’s mechanisms are designed to work to overcome all of these factors. None of this work is standardized. Years of work must be invested in designing algorithms to manage voice transfer as best as possible.

While a call is in progress, the challenge is to compress without losing voice quality. The mechanism for compression differs from one media gateway to another, with the compression algorithms the variable. One technology enables the transfer of several calls simultaneously in a thru packet that uses less bandwidth on the network.

One challenge for media gateway makers is to create buffers to handle jitter, or delay variation. If delay were at a constant pace, the other party would not notice. But in VoIP delay is a changing factor. Buffering streamlines the packets. If there is no buffering the sound gets to the other side with different time variations and won’t sound right. The buffer streamlines the packet transfer and arrival. There can be different delay variations during a conversation.

Designing an optimal buffer is a challenge for a media gateway provider. Small buffers are cheaper to make, but won’t provide the tool to ensure quality. A buffer that is too large can be the source of delay. Therefore the challenge is to adjust the size of the buffer to solve delay without creating delay. Larger buffers will make a product more expensive but can manage changing delay situations.

Another expertise is packet loss concealment, or the technology for covering up for lost packets. Algorithms can be created in the media gateway to fill up the lost packets with something to cover for the cutoffs, which are usually only a fraction of a second. The goal is to predict the lost packet and fill it in.

Double talk is another challenge to maintain when two parties are talking together to ensure that there is no speech clipping and discontinuity when two parties talk at the same time. The echo canceller suppresses residual echo to the background noise level so it will be perceived by the user as regular background noise rather than disturbing echo.

Achieving satisfying VoIP quality is not easy, and will always require ongoing, and significant R&D investment, especially in proprietary algorithms. In fact, maintaining cutting-edge voice quality in the media gateway is all about R&D.

All media gateways are therefore not born equal and as yet, have not eliminated voice quality issues throughout their evolution. Much investment must be made in creating features to preserve state-of-the-art voice quality, the most important factor in a voice network. Only when this happens, will voice quality become an issue of the past.

About the Author

Lior Aldema is Vice President of Marketing at AudioCodes. He began his career at AudioCodes in 1998 where he was a team leader and then headed up the System Software Group in the research and development department. Prior to 1998, Mr. Aldema served as a Major in the Technical Unit of the Intelligence Corps of the Israeli Defense Forces (IDF), where he was in charge of both operational units and large development groups relating to various technologies. He was responsible for one of the largest projects conducted by the IDF and executed by the Israeli Defense industry corporation. Mr. Aldema holds an M.B.A. (Finance and Marketing) with distinction from Tel Aviv University and a B.Sc. with distinction from the Technion in Haifa, Israel.

About AudioCodes

AudioCodes., a key originator of the G.723 VoIP compression standard, enables the new voice infrastructure by providing innovative, reliable and cost-effective Voice over Packet technology and Voice Network products to OEMs, network equipment providers and system integrators. AudioCodes provides its customers and partners with a diverse range of flexible, comprehensive media gateway and media processing technologies, based on VoIPerfect, its best-of-breed, core media gateway architecture.

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Its customers include the leading global telecom and data network equipment providers. AudioCodes voice network products feature media gateway and media server platforms for packed-based applications in the converged, wireline, wireless, broadband access, and enhanced voice services markets.
AudioCodes' headquarters and R&D facilities are located in Israel with an R&D extension in the US. The company maintains offices in Europe, the Far East and Latin America. For more information please visit www.audiocodes.com