WHITE PAPER

IN JEANS

Enhancing QoE in VoIP Networks with AudioCodes SBCs

July 2014

AudioCodes

Barry Spielman
Marketing Manager, AudioCodes

End-to-End QoE

While non-real time data applications are less susceptible to delay (latency), jitter and packet loss in IP networks, it is well known that these three network impairments are the archenemies of VoIP performance. Network administrators invest great efforts in fine-tuning their networks so as to minimize the negative effects of these factors.

Legacy networks were designed to take into account these factors, and therefore connecting between wireline networks was relatively straight forward. However, many networks were fitted with devices with low performance CPUs resulting in built-in tools with limited ability to overcome these negative effects from non-wireline networks. As networks developed and new types of devices introduced new types of traffic behavior to the network, the challenge in handling jitter, delay and packet loss increased.

That challenge increases further with Voice over IP. VoIP may traverse either wireline, WAN, cellular or wireless networks (such as LTE or 3G or Wi-Fi using Smart Phones or Tablets, or other devices). A call might start on one provider's network and end on another and go over additional networks along the way. All the networks involved need to ensure that the quality of the call meet the expectations of the users. This is critical in today's highly competitive communications market.

Different Networks - Different Behavior

The behavior of delay, jitter and packet loss varies between these different networks and hence its effects on VoIP calls traversing these networks differs as well. When designed, legacy networks didn't take into account traffic coming in from networks such as Wi-Fi with its frequent retransmissions and 3G with its bandwidth restrictions that can change from call to call and even within calls, in which the jitter levels are higher and behave differently than those in wireline networks. And wireless networks were designed first and foremost, for data applications.

Inconsistent Wireless Traffic Behavior

Due to variables such as distance, speed, and potential impeding obstacles, the behavior of wireless traffic is inherently inconsistent. Add mobile devices to picture and the potential for erratic traffic behavior only increases. In the data focused scenario, the most important thing is for the payload to arrive complete. It matters less how long it will take to arrive. Thus, compensating measures can be taken to ensure the complete end-to-end delivery of the data over the network. To accomplish this, wireless networks were designed with capabilities such as retransmission and dynamic bit rate change. While these capabilities ensured the arrival of the packets, they also increased the delay and the jitter. But while the delay for data applications is tolerable, in the case of VoIP it may render the call unfeasible.

The Need to Handle the Effects of Wireless Traffic on VoIP Networks

As long as the right level of packet loss is maintained on a call, a sufficient level of voice quality can be maintained as well. But delay on a call is different story. Here the call participants have little tolerance for delay and if too long, the call will be considered to be of very poor quality and in the worst case scenario, people will simply drop the call. These two voice-related behavior patterns – the ability to deal with partially lost data on the one hand, but not being able to handle long delays on the other, are totally opposite the way the wireless networks were designed for data applications, thus creating a problem of VoIP traffic on wireless networks that must be resolved to allow for quality calls.

As most VoIP entities in the network (SBC, GW, ATA, IP Phone, Mobile Client, etc.) were designed to handle wireline and not wireless impairments, they have a hard time handling (without help) traffic emanating from wireless networks.

There is a way around this problem. AudioCodes' SBC technology includes special tools that provide that help, and handle the impairments emanating from transition between wireless and wireline networks. By placing AudioCodes Mediant Session Border Controllers with their wide array of built-in tools between the wireless network and any other network (wireless, wireline, cellular, etc.), the impairments from the wireless network traffic can be managed and reduced dramatically to allow for the end-to-end quality call.

AudioCodes Session Border Controller (SBC) Tools to Enhance Voice Quality

AudioCodes SBCs are uniquely able to combat and neutralize the negative effects of jitter, delay and packet loss and enhance QoE on VoIP calls which traverse between different networks. This is done through an arsenal of flexible built-in tools, rendering the AudioCodes offering quite unique in the marketplace.
These tools include:

1. **Adaptive Jitter Buffer** - Almost all devices today (Smartphones, IP phones, gateways, etc.) have built in jitter buffers. Legacy networks (which were LAN focused when designed) usually have older devices with less sophisticated jitter buffers. When designed, they didn't take into account traffic coming in from networks such as Wi-Fi and cellular, in which the jitter levels are higher than those in wireline networks. And even if the network is designed to handle jitter, today's OTT applications via Smart Phones add yet another variable to the equation. There are hundreds of such devices out there, and the audio interfaces of these devices create jitter that is passed into the network. For these situations, the AJB is necessary. AudioCodes has developed a highly advanced Adaptive Jitter Buffer (AJB) built into the SBC that neutralizes the incoming jitter so that it is handled without problem on the other side. The AJB can handle variable jitter rates as high as 1 or 2 seconds (there is no need to go beyond this point as calls with delays of more than 2 seconds would not be feasible). Additionally, the AJB can operate with or without the need for transcoding.

2. **Transcoding** Beyond being able to mediate between different codecs on the different networks on either end of the SBC, the SBC can transcode an incoming codec that is less resilient to packet loss (such as narrowband G.729 or wideband G.722) to a more resilient codec (such as Opus). By transcoding to a more resilient codec, the SBC can lower the effects of packet loss. Transcoding can also lower the bandwidth on the network. Additionally, the SBC can transcode from narrowband (8Khz) to wideband (16Khz) (and vice versa) as well as wideband transcoding, where both endpoints support wideband codecs but are not using the same ones. AudioCodes SBCs support all the standard codecs used in wireline and cellular networks. In addition, more complex codecs such as RTA, Silk, OPUS, and SPEEX are also supported.

3. **Tools for Channels Characterized by High Packet Loss Burst** - The SBC has several tools that can be used to balance networks characterized by high packet loss burst. One tool is multiple RTP redundancy in which several copies of the voice packets are sent to ensure they are received. The norm in the industry is generally up to 2 packets. Additional tools supported by the AudioCodes SBC include Forward Error Correction (FEC) and Interleaving. The SBC can choose between these tools based on the different characteristics of the network.

4. **Trans rating –** Trans rating can be used not only to mediate between two end devices using different Ptimes, but also as a means of balancing the network by reducing bandwidth and reducing CPU pressure during traffic peaks. The AudioCodes SBC has the ability to handle trans rating from 10 msec up to 100+ msec (and even higher upon demand). AudioCodes SBCs are also unique in that they don't require DSP hardware to perform trans rating. This improves the cost-effectiveness of the SBCs.

5. **Quality-based Routing –** Another tool used by the SBC is Quality-based routing. The SBC, which is monitoring all the calls on the network all the time, can decide (based on pre-defined thresholds and parameters) to reroute calls over different links that have better quality.

The above built-in SBC capabilities provide a powerful arsenal for combatting the ill effects of Jitter, delay and packet loss. Each tool on its own can play a major role in significantly enhancing QoE. However, when taken together, and given the ability to fine tune the balance between the variables to reach the ideal equation by using an AJB, Transcoding, Trans rating, Redundancy, routing and bandwidth management, the network administrator has at his fingertips a formidable set of tools to ensure end-to-end QoE on the VoIP network.

Triggering the Quality Enhancement Tools

As we have seen, all the tools mentioned above serve important functions, but their overall effectiveness relies on a mechanism that can automatically trigger these tools upon encountering certain network conditions. On the receiver end, AudioCodes devices are very flexible and they receive and handle the incoming bit stream that may have the different variables (Ptime, redundancy level, bit rate, etc.) affected on the fly by all sorts of external factors. On the transmitting end, AudioCodes SBCs can be pre-configured with thresholds based on a combination of voice quality parameters. These parameters are closely correlated to a subjective MOS (Mean Opinion Score) metric.

The SBCs run a real-time algorithm that measures these parameters on every call. If one or more thresholds are crossed, the SBC can activate any of the above-mentioned quality enhancement tools. The SBC knows how each change in each parameter can effect voice quality and it knows how much needs to be changed in each parameter to maximize the voice quality.

About AudioCodes

AudioCodes Ltd. (NasdaqGS: AUDC) designs, develops and sells advanced Voice over IP (VoIP) and converged VoIP and Data networking products and applications to Service Providers and Enterprises. AudioCodes is a VoIP technology market leader focused on converged VoIP & data communications and its products are deployed globally in Broadband, Mobile, Cable, and Enterprise networks. The company provides a range of innovative, cost-effective products including Media Gateways, Multi-Service Business Routers, Session Border Controllers (SBC), Residential Gateways, IP Phones, Media Servers and Value Added Applications. AudioCodes' underlying technology, VoIPerfectHD™, relies on AudioCodes' leadership in DSP, voice coding and voice processing technologies. AudioCodes High Definition (HD) VoIP technologies and products provide enhanced intelligibility and a better end user communication experience in Voice communications.

International Headquarters
1 Hayarden Street,
Airport City
Lod 7019900, Israel
Tel: +972-3-976-4000
Fax: +972-3-976-4040

AudioCodes Inc.
27 World's Fair Drive,
Somerset, NJ 08873
Tel:+1-732-469-0880
Fax:+1-732-469-2298

Contact us: www.audiocodes.com/contact
Website: www.AudioCodes.com

Ref # LTRM-80036 07/14 V.1

AudioCodes