Your new VoIP Network is working great… Right? How to Know

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WHITE PAPER

Executive Summary

This paper discusses the importance of measuring and monitoring the voice quality of VoIP calls traversing the data links into a corporate network. Data collected must be viewed in the context of the network topology, and should provide enough insight and detail to enable network engineering and troubleshooting. The Session Experience Manager from AudioCodes is introduced as a way to achieve this goal.

> *You've just installed the new Unified Communications infrastructure in your corporate network. It was a pretty high profile assignment. The executive team had been involved, and everyone was excited about the new integrated IM, Presence, and Collaboration features. Everything seemed to be working great, until you started to get complaints about voice quality.*
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> *Pretty quickly it became obvious that all the cool new features in the world don't count for much if basic voice isn't working well. Whatever tolerance users may have for cell phone quality issues doesn't extend in any way to business systems.*

Unfortunately there are lots of things in corporate networks that are not very voice (or video) friendly. And ironically, one of the most unfriendly of all is the very voice/data convergence that helped make Unified Communications so attractive in the first place. Because convergence also means competition for bandwidth on the links connecting the various corporate sites to each other and to the Service Provider networks. Even if the switches are configured to give priority to real time services like voice, load spikes from data servers can result in jitter, delay, and packet loss.

To resolve voice quality complaints, and even better make sure they don't happen in the first place, tools to monitor quality and provide sufficient information to troubleshoot any problems detected are required. Ideally, the tools should go beyond passive monitoring, and actively intervene to head off problems before they even start.

So how to do <u>that</u>?

How to KNOW about the state of your Voice Quality

To start with, UCC systems typically provide a means to monitor quality as perceived by the individual users it manages. This gives a perspective on their experience, but doesn't provide much information regarding what in the network might be causing any issues reported. What is also needed is a <u>network</u> view of voice quality.

Session Border Controllers and Media Gateways are excellent places to gain this network view as they are positioned at the entry points to the corporate network from Service Providers and between corporate sites, and are aware of all the media flows going over those links.

And for a network wide perspective, the information about voice quality on all the links in the network needs to be aggregated, analyzed, and presented in a useful way. And again ideally, automatically acted upon in an intelligent way.

So how to do that?

Collecting Voice Quality Information

The first thing to do is to ensure that voice (and video) packets are "tagged" for real time treatment, and that the routers and switches in the LAN are set to give prioritized treatment to packets tagged in this manner. Specifically by using Differentiated Services (DiffServ) and the 6-bit Differentiated Services Code Point (DSCP) field in the IP header for packet classification. Most UCC systems and switches do this by default, but not all networks are set up in this manner. For example, Wi-Fi networks use specialized protocols in the Wi-Fi client and Access Point for QoS treatment, and not all clients and Access Points support them.

For data collection, there are a couple of widely used standards that provide for capture of voice quality information. RTCP (Real Time Control Protocol – RFC3550), and RTCP-XR (RTCP Extended – RFC3611) provide for ongoing and per call collection and reporting of key metrics such as latency, jitter, packet loss, echo, and even Mean Opinion Score (MOS - a derived metric defined by ITU-T PESQ P.862 intended to provide an overall measurement of voice quality). These metrics must then be integrated with call control information to allow analysis tools to associate quality data with the calls it came from. Media Gateways and Session Border Controllers are an excellent place to gather this data as they have access to both the signaling and media, and are located at the points where voice enters and exits the corporate network.

The voice data generated at each device must first be aggregated, and then analyzed. Not only must the media flows be associated to the calls they came from, but calls traversing the same links must also be grouped in order to provide the per link network perspective that the UCC monitoring system can't. Finally the information needs to be presented in multiple ways. There should be alarms and other alerting mechanisms for when quality thresholds are crossed as well as a way to view the overall state of voice traffic in the network in real time. In addition there should be a reporting mechanism to track volume and quality metrics over time.

Finally a mechanism to generate calls to/from any user is required to troubleshoot and isolate any issues reported. Detailed information about these calls, including signaling and media traces needs to be available for analysis.

Beyond Monitoring - Active Intervention

Active intervention on the part of the network in order to preserve voice quality is not new, for example echo cancellers monitor, measure and remove echo, and wireless vocoders adapt their rate based on radio link quality. In UCC systems, Enterprise Session Border Controllers use Call Admission Control to limit the number of voice calls over a specific link, but it's often a rudimentary control, where the number of calls is partly used to control bandwidth consumption, a proxy that High Definition voice and video has made less reliable.

However, these quality metrics combined with policy controls in Session Border Controllers and Media Gateways can provide for some very sophisticated ways to intervene both gracefully and effectively. For example, 4 Mb/s on a 10 Mb/s link might be allocated for voice, and a Layer 3 rule implemented in the router to enforce that. By itself, such a limiting mechanism protects data capacity, but is pretty destructive to voice - as excess voice packets will be discarded across all calls, with no block for additional calls.

Standard Call Admission Control helps with this by blocking calls that would cause voice capacity to exceed the 4Mb/s limit, but unless the Admission Control is bandwidth aware it could allow the limit to be exceeded if the existing calls have a higher average bandwidth consumption that was assumed. In devices with hybrid capability, rather than blocking the limit topping call it could be placed via the PSTN or some other route instead.

But even bandwidth aware Admission Control is to a significant extent reactionary. The bandwidth is consumed and calls blocked by the time it intervenes. A more sophisticated approach would be to intervene earlier - before the bandwidth capacity was so fully consumed. For example, once a capacity threshold is reached (e.g. 75% of capacity) policies in the SBC or MGW could be invoked to conserve bandwidth, for example by forcing the use of a compression codec, or increasing the packetization time (i.e. reducing the amount of RTP header overhead). Alternatively, calls can be routed based on link quality, with alternate routes selected when the primary route is experiencing quality problems.

The Session Experience Manager from AudioCodes

AudioCodes has introduced the Session Experience Manager (SEM) to serve this need. It provides network operators with a comprehensive overview of the network status, both quality and volume. Subsets of the network can be examined in isolation, and the analysis can extend down to not only individual calls, but how the quality metrics vary over a call.

Information gathered from RTCP and RTCP-XR packets by the Media Gateways and Session Border Controllers in a corporate network are combined with signaling information about the call and sent to the SEM, including mid-call if SEM defined alarm thresholds are exceeded. The SEM sorts the data by link for display and reporting. The aggregate metrics for each link are calculated and compared against alarm threshold values, and stored for display and report generation. When thresholds are exceeded, alarming and alerting actions are taken.

One traditional hurdle of Voice Quality analysis is encryption of the media with Secure-RTP (SRTP), in that the SRTP, RTCP, and RTCP-XR data can't be accessed without being de-crypted. The Media Gateways and Session Border Controllers perform this decryption as a matter of course and are able to send the associated Voice Quality to the SEM where more passive solutions cannot.

The information gathered can be displayed in a variety of ways:

- A Network view by link, either in tabular or graphical form provides an overview of call quality per link or device and in aggregate, along with any current alarms (Figures 1 & 2).
- A Statistics view provides key metrics over a settable time scale, including call success/fail, overall call quality, and individual jitter, latency, packet loss, and MOS statistics (Figure 3).
- A Call View lists key metrics per call. Individual calls can be selected and examined over time, with key information summarized (Figure 4).
- An alarm view provides a list of both active alarms as well as alarm history. Alarms can be sorted, searched, and filtered.
- A wide range of pre-defined reports are available, or can be defined by the user to run ad hoc or on a user defined schedule. Reports can be sent in a variety of common formats (e.g. CSV, XML, PDF, email).

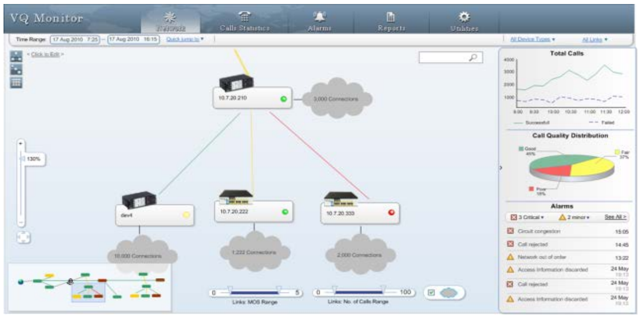


Figure 1 Graphical Network View

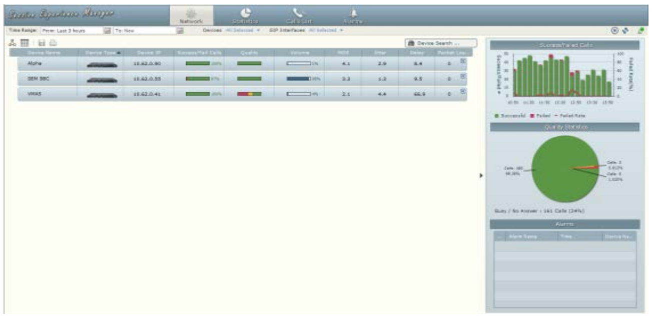


Figure 2 Tabular Network View

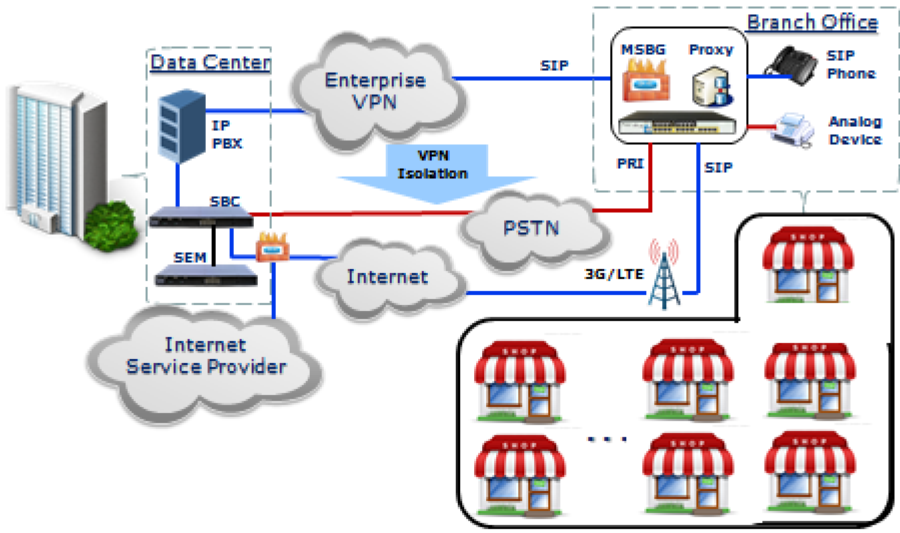
Figure 4 Statistics View

Figure 3 Call Detail View

For additional help with trouble shooting, a test agent in the media gateway or session border control can be invoked from the SEM. In this fashion test calls can be made to and from various users in the network, and subsequently have the data on those calls presented at the SEM. And if additional detail is required, signaling and media traces can be extracted and examined.

Typical Customer Deployment

One recent customer needed to connect a state-wide network of remote sites into a Data Center, which in turn provided SIP Trunking to an Internet Telephony Service Provider ITSP). They wanted a multiservice device in the branches providing local PSTN connectivity as well as Voice and Data connectivity to the Data Center. Being able to monitor voice quality between sites, as well as the Service Level Agreement (SLA) with the Service Provider was just as important as the providing the service itself.



Example Customer Deployment

The customer deployed a Mediant 4000 SBC in the Data Center, and Mediant 800 Multi-Service Business Gateways in each of their branches. The Mediant 4000 SBC provided needed adaptation between their UCC System and the Service Provider, and provided Perimeter Security for voice traffic at the access point to their Data Center. The Mediant 800s provided data connectivity at each of the remotes, as well as local PSTN access and Standalone Survivability, ensuring that the branch users maintained voice capability even if the data connection to the UCC system in the Data Center was lost.

Mediant 4000

With the Mediant 4000 and Mediant 800 built from the same VoIPerfect software platform, the customer appreciated having common voice technology for both Data Center and Branch, and SIP Trunking and PRI. Additionally, the ability to add SIP Trunking capability to any of the branch Mediant 800s provided future flexibility for connectivity to their ITSP.

Mediant 800

And finally, the Session Experience Manager gave them the ability to monitor, report, alarm, and protect the voice communications throughout the entire network, allowing them to know that the voice quality their users were receiving met their expectations.

Summary

If voice quality is important to you, make sure you select Mediant Media Gateways, Session Border Controllers, and Multi-Service Business Gateways for the voice connectivity for your next network upgrade, and take advantage of the insight that the Session Experience Manager can bring to your organization.

You're new VoIP Network is working great… Right? Are you sure?

To find out, schedule a consultation on the Session Experience Manager by visiting www.audiocodes.com/SEM and completing the Sales Inquiry form. For information on AudioCodes Mediant Media Gateways, Session Border Controllers and Multi-Service Business Gateways with voice quality monitoring features and SEM compatibility, visit www.audiocodes.com/products-lobby.

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, Enterprise networks and Cable. The company provides a range of innovative, cost-effective products including Media Gateways, Multi-Service Business Gateways, 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.

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