WHITE PAPER

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Enabling VoIP quality of service from the design platform

Introduction

Increased attention to voice quality in IP networks is directing focus on the evolving standards needed by quality initiatives. The use of quality monitoring and advanced real-time algorithms requires a standardized architecture in which quality metrics can be exchanged between end-points and communicated to a quality monitoring agent for more comprehensive analysis.

Many architectural and technological options are currently available to implement this functionality. For core carrier networks, standard quality architecture is necessary before deploying any large-scale, next-generation quality assurance system. In this paper, we'll describe several of the most pertinent quality architecture issues as they relate to IP networks and to existing circuit-switched and cellular networks.

The need for a "quality overlay"

In the past, the goal of IP quality of service (QoS) initiatives was to guarantee a certain level of service for a particular service. QoS schemes attempted to guarantee a minimum throughput speed or maximum delay time, and provided a framework for preferential treatment to certain customers or types of traffic. They did not guarantee a fault-free environment, nor protect against network packet loss or jitter.

QoS initiatives typically do not guarantee a quality user experience for real-time applications such as voice and video. Also, they do not enable remedial actions in the event of service degradation.

Another critical piece absent from these approaches is a view of the signaling layer used throughout the network to establish and maintain user connections. In today's complex IP network topology, it is not uncommon for a connection to traverse 10 or more different "networks" to establish a complete channel. These various networks may involve IP or Asynchronous Transfer Mode (ATM) technologies, the public switched telephone network (PSTN) or a variety of cellular radio frequency (RF) air interfaces.

Comprehensive quality architecture would more effectively address these and various hand-off issues among the various networks. Existing mechanisms, which are typically limited to allocating bandwidth and minimizing delay, are simply inadequate. Establishing complete quality architecture requires distributed intelligence throughout the network and across service provider boundaries. Placing intelligence in network endpoints is the initial step toward realizing this goal.

Endpoints

The most common endpoint instrument in any telephone call is the handset. It is easy to visualize two IP phones as the endpoints in a conversation. In addition to the call itself, these IP phones can also transmit statistics on call quality.

The definition of an endpoint should be extended, however, to include any point in the total communications infrastructure involving media or technology. Possibilities include cable, PSTN, wireless and others where a transition in protocol or coding occurs. Under this definition, all gateways and transcoding nodes qualify as endpoints. Figure 1 illustrates endpoints for the purposes of this discussion.

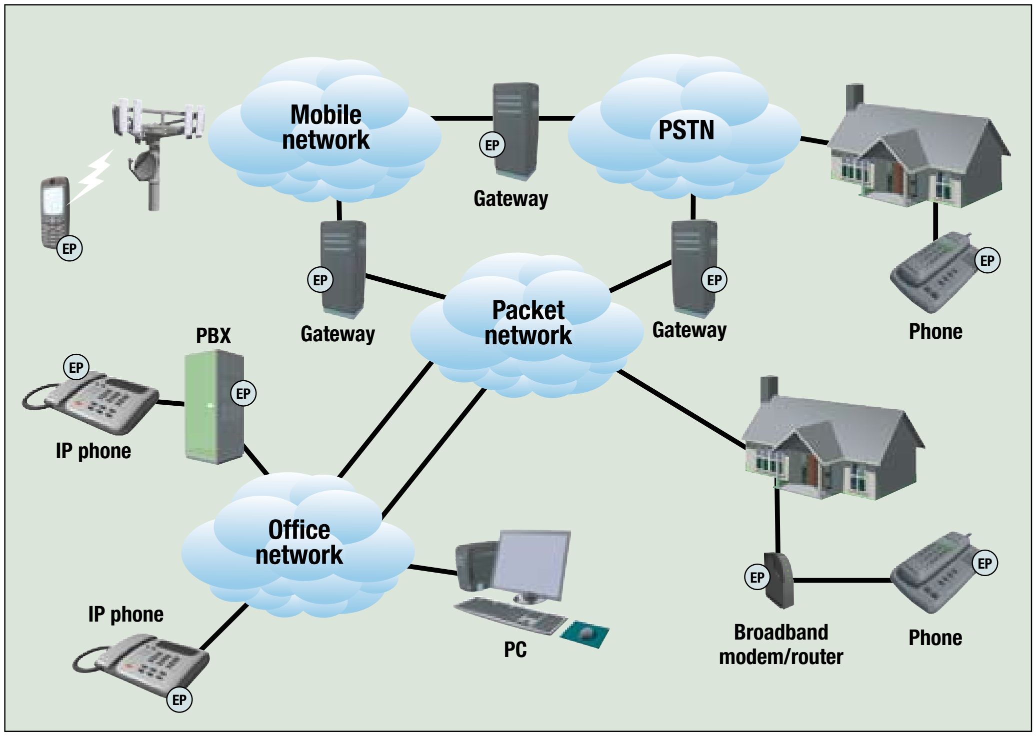

Figure 1– IP network endpoints

Intelligent IP phones and wireless handsets provide the most reliable and objective indicators of the user experience as the closest possible monitoring point to the user where both packet and content impairment data is accessible.

For IP phones, statistics are easily accessible through the IP network. A variety of protocols are available for this purpose. (See sidebar, "Protocols for compiling user experience statistics on IP networks.") For mobile phones, quality metrics can be sent as data and would require minimal bandwidth over today's cellular networks.

The design of the PSTN precludes quality reporting from a plain old telephone service (POTS) telephone because no data channel is associated with the instrument. For the most part, quality reporting is not necessary because the 4-kHz POTS voice channel has dedicated bandwidth with a fixed delay. Problems do arise in this environment, but no mechanisms currently exist for gathering user quality experience metrics. A POTS device is an "on/off" technology: it either works or it doesn't.

Placing a call from one phone to another requires multiple gateways within today's communications landscape. For example, calls from an IP phone in a modern digital home can be connected to remote and even primitive villages in faraway lands. Multiple types of gateways are involved in all of these complex connections.

Sometimes these gateways perform the complex task of converting from traditional circuit switched technology, as found in the PSTN, to IP technology. In this case, PSTN and IP connections are effectively terminated and the gateway is in an ideal position to report on voice quality from both directions.

The evolution of mobile networks parallels the development of several voice coding techniques, with many new codecs under development. When subscribers to a particular mobile network want to talk with subscribers on different wireless, broadband or even PSTN networks, transcoding is likely necessary. Transcoding deconstructs and reassembles the voice packet and is yet another point in the network where quality monitoring can discreetly occur.

This user quality monitoring functionality requires unobtrusive technologies that will not unduly affect the user experience itself or the processing load on the endpoint device. Scalable processing functionality is most often provided by a digital signal processor (DSP). Because of their programmability and flexibility, DSPs are well suited to the changing landscape of today's communications infrastructure. As new codecs are developed and features and functionality evolve to meet the demands of new applications, the flexibility of DSPs will be highly beneficial.

Quality metrics reporting

In addition to the quality data gathered at the endpoints, probes deployed in a network can collect and analyze a variety of quality metrics. Historically, active or passive probes strategically placed in the network acquired the necessary data for troubleshooting.

Today, these probes can be embedded as software into endpoints, allowing for comprehensive and real-time fault isolation and providing an economic and scalable way to monitor performance on the edges of the network. In addition, specialized agent probes can be downloaded to endpoints when certain faults are detected.

Whether from an endpoint or from a network probe, equipment vendors and network operators can choose from any one of several protocols for reporting quality metrics. New standards for this purpose are currently

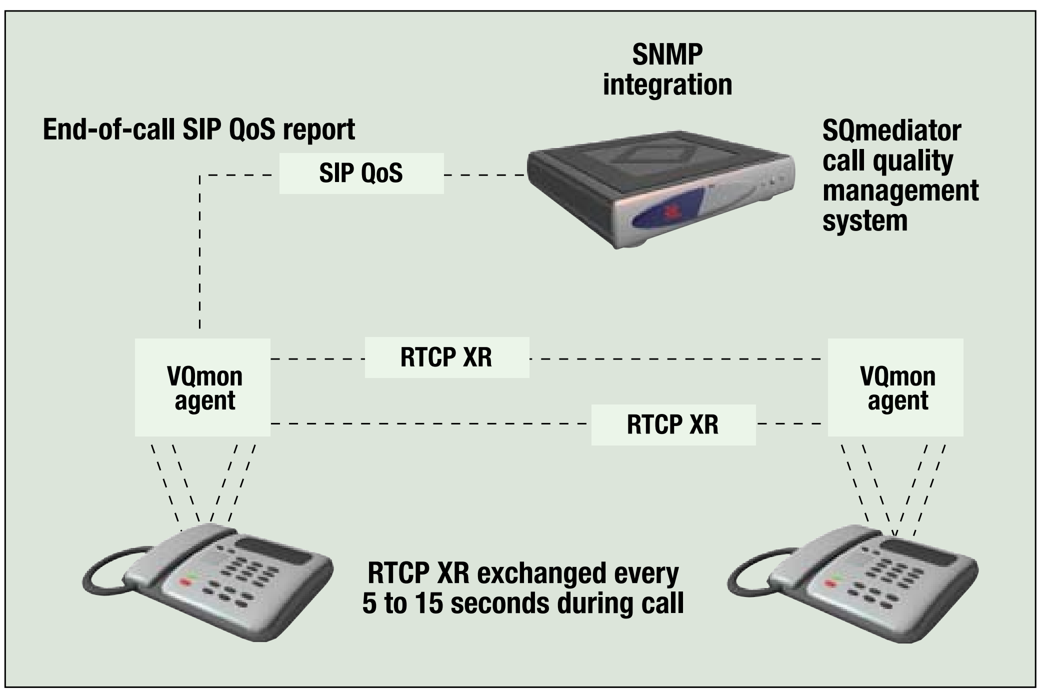

Figure 2– *Embedded software agents monitoring quality*

being developed and existing standards are evolving. (See sidebar, “Protocols for compiling user experience statistics on IP networks.”)

New quality management overlay systems for the IP network can collect information from intelligent endpoints on the quality of real-time voice or video services. These quality management overlay systems are capable of expert analysis of call-quality data and are designed to integrate into existing Simple Network Management Protocol (SNMP)-based management architectures. These overlay quality systems also allow service providers to proactively identify degradations in the quality of the user experience and to take remedial action.

Figure 2 illustrates the use of industry standards to overlay a quality management system on an IP network. Embedded software probes, such as industry-standard VQMon agents in IP phones, monitor call quality in real time.

These agents create RTCP XR messages that are exchanged every five to 15 seconds to provide feedback and allow for more accurate quality estimates, since each endpoint can incorporate information on the QoS provided by the other endpoint, such as echo level.

At the end of a call, a Session Initiation Protocol (SIP) QoS report is sent to a call quality management system. (Telchemy’s SQMediator is shown in Figure 2.) The voice quality manager can then provide comprehensive summaries and analysis via SNMP to network managers.

Ensure quality now

Many of the capabilities of a quality architecture that could be overlaid onto IP networks already exist today. As previously mentioned, industry standards have been developed and approved that enable endpoints to monitor and accurately report quality data on a regular basis. Being based on open standards enhances the scalability of overlaid quality architecture, which is a critical feature for rapidly expanding IP networks.

Moreover, quality architecture must be able to rapidly respond to ongoing conditions. Interrogating endpoints in real time to determine the root cause of a fault or degraded service could initiate corrective action to mitigate the degraded conditions immediately. Mid-level points in the network would collect and manage the quality data generated at the endpoints. Of course, the deployment of such an architecture must be cost-effective, and its impact on the performance of endpoints negligible.

Moving the pieces into place

Even though all of the pieces for quality architecture are available, establishing architecture to address voice quality will not happen overnight. Regulatory and privacy issues, as well as intercarrier issues like the unwillingness to share quality metrics, will slow the process. Exactly what parameters reported in RTCP-XR, for example, would be considered private and not shared?

Will Carrier A allow the transmission of quality metrics from its subscribers' IP phones to the IP phones of Carrier B? Or will Carrier B transmit quality metrics from its session border controllers or gateway systems to the IP phones of Carrier C? These issues delve into uncharted waters and will certainly affect how quickly any quality architecture is deployed.

Nevertheless, the nature of IP and the desire for a higher quality user experience is a driving force in moving quality standards forward. This trend is likely to gather momentum. Once in place, this same infrastructure can support video in addition to voice communications.

Protocols for compiling user-experience statistics on IP networks

RTCP voice quality reports

- RTCP-XR (VoIP metrics) is defined in RFC-3611 and includes such parameters as packet loss rate, packet discard rate, burst density and duration, gap density and duration, round trip delay, end system delay, signal level, noise level, residual echo return loss (RERL), MOS -listening quality, MOS conversational quality, and information on jitter. TI's PIQUA™ software uses VQmon from Telchemy, which provides the MOS score values. VQmon provides a proven method for accurately measuring MOS scores.
- RTCP HR (high resolution) is currently under development in the Internet Engineering Task Force (IETF) and is intended to augment the existing RTCP XR by adding new block types for reporting higher resolution metrics as needed by some applications in carrier networks.

Session Initiation Protocol (SIP)

SIP is one of many protocols used to provide end-of-call quality reports. With SIP, voice quality information can be pushed to a call server at the end of each call for offline analysis and reporting.

Real-Time Streaming Protocol (RTSP)

RTSP is a diagnostic protocol that allows remote clients to set up and configure one or more diagnostic flows that can include selective collection of all available system statistics and diagnostics. Once the diagnostic flows are set up, the diagnostic streams are pushed to the client, where it is collected, processed and forwarded as required.

Standard RTSP methods such as SETUP, PLAY, PAUSE, TEARDOWN, GET_PARAMETER and SET_PARAMETER are used to set up and configure diagnostic streams.

Other relevant standards

- Signaling-based reporting
  – H.248.30 – Megaco extensions
  – H.460.9 Annex B – H.323 extensions
- Standards that mandate RTCP XR
  – G.799.1 – VoIP Trunking Gateway Spec
  – Packet Cable 1.5 U.S. Cable Industry Spec