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REVIEW OF VOIP TECHNIQUES

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Voice transmission on net has increased significantly in last three years. Voice quality can be degraded due to number of reason such as unpredictable short term loads, lack of guarantees on network performance, lack of control over the end systems. Enormous advances have been made in computer technology to send packet voice data. This research paper reviews the concerns and technological advancement in this area.

1. Introduction

The quality of packet audio is largely determined by the mouth-to-ear delay and the packet loss. The contribution of this paper is to study and provide techniques to improve the packet audio quality. To achieve our goal of good quality audio communication, some changes might be needed to the ubiquitous Internet. The degradation of voice quality which can occur when using a multi-user packet switched network is the fundamental problem. Unpredictable short term loads, lack of guarantees on network performance, lack of control over the end systems and stringent requirements on the voice quality make VoIP a challenging application to realize successfully on the Internet. Our Quality of Service (QoS) research is orthogonal to the investigations being carried out by the network community. These investigations focus on changing the packet switching techniques to be more reliable, more timely and more fair. This is especially the case for time sensitive traffic such as voice. Protocols have been developed to signal routers and end systems that certain data types need to be treated differently, again in the case of voice traffic often at higher priority. We look at allocating resources given the current conditions of the network or adapting to it, also by measuring the current state so that we can make decisions based on these measurements rather than assuming the certain functionality will be available. A media gateway plays a critical role of interoperability between packet networks and the existing telephone networks. Inevitably, it involves many complex processes, such as packetization/depacketization of voice frames, jitter smoothing, and error concealment, etc. It has been found that three factors can profoundly affect VoIP quality. They are delay, jitter, and packet loss.

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2. Fundamental Concepts of VOIP

Voice over Internet Protocol (VoIP) is often confused with Internet telephony. VoIP is voice transmitted as packets over a data network, whereas Internet telephony refers to voice transmitted as packets over the public Internet—a special case of VoIP. [1-3] Quality of Service (QoS) typically comes to mind. In [4] Qos is defines as “the set of technologies that enables network administrators to manage the effects of congestion on application traffic by using network resources optimally, rather than by continually adding capacity”.

Three fundamental concepts affecting real-time data transmission must be considered while designing the IP network for audio and video data. These are network provisioning, queuing, and classifying.

Provisioning–provisioning the network simply means installing more network bandwidth or capacity than is actually needed for all of the audio, video, and regular data applications that will run over the network.

Queuing–Buffering issues may be overcome by enabling separate voice and video data queues in the network switches and routers. Separate queues allow time critical data such is audio and video to be transmitted in a priority fashion.

Classifying–Several different schemes currently exist for providing priority to network packets. These include Resource Reservation Protocol (RSVP), IP precedence, differentiated services (DiffServ), and Multi-Protocol Label Switching (MPLS).

3. Protocols and Societies

- H.248 is an ITU Recommendation that defines “Gateway Control Protocol.” H.248 is the result of a joint collaboration between the ITU and the Internet Engineering Task Force (IETF).
- H.323 is an ITU Recommendation that defines “packet-based multimedia communications systems.” In other

words, H.323 defines a distributed architecture for creating multimedia applications, including VoIP.

- IETF refers to the Internet Engineering Task Force (http://www.ietf.org/), a community of engineers that seeks to determine how the Internet and Internet protocols work, as well as to define the prominent standards.
- ITU, is the International Telecommunication Union (http://www.itu.int/home/index.html), an international organization within the United Nations System (http://www.unsystem.org/) where governments and the private sector coordinate global telecom networks and services.
- Megaco, also known as IETF RFC 2885 and ITU Recommendation H.248, defines a centralized architecture for creating multim Gateway Control Protocol (MGCP), also known as IETF RFC 2705, defines a centralized architecture for creating multimedia applications, including VoIP.
- Real-Time Transport Protocol (RTP), also known as IETF RFC 1889, defines a transport protocol for real-time applications. Specifically, RTP provides the transport to carry the audio/media portion of VoIP communication.
- Session Initiation Protocol (SIP), also known as IETF RFC 2543, defines a distributed architecture for creating multimedia applications, including VoIP.

4. Voice Quality and Impairments

The traditional measurement for voice quality measurement in telecommunications is the Mean Opinion Score (MOS). The MOS test is also called the Absolute Category Rating (ACR) test. The ACR is described in detail in ITU Recommendation P. 80. Using the MOS method, listeners are asked to rate speech and classify it into categories.

Using the MOS method, listeners are asked to rate speech and classify it into categories. These categories are shown in Table 1.

Table 1
Absolute Category Rating (ACR) System

Quality	*Score*
Excellent	5
Good	4
Fair	3
Poor	2
Bad	1

The MOS level 4.0 which is considered to be "good" quality of speech has traditionally been considered "Toll Quality". This was the quality that could be expected for a connection in the United States public switched telephone network (PSTN). Many other tests are also used to gauge quality for voice connections. These include the Percent Good or Better (%GOB), Percent Poor or Worse (%POW), Degradation Category Rating (DCR), and the E-model.

5. E-Model

The E-model tries to address in a qualitative manner several of the quality issues that will affect voice over packet systems. One of the driving forces behind the E-model is that the actual quality of the speech is not always as crucial as the perceived quality. The European Telecommunications Standards Institute (ETSI) developed the E-model to address the needs of network planners [5-11]. The E-model is based on the premise that "Psychological factors on the psychological scale are additive" [12]. The E-Model defines the "*R*" value as the measure of voice quality.

Comparing the MOS scale and E-model provides a reference as to what is considered acceptable. Table 2 shows a comparison of the two scales.

Table 2
E-model vs. MOS values

User Satisfaction	*E-model - R*	*MOS*
Very Satisfied	90	4.3
Satisfied	80	4.0
Some Users Dissatisfied	70	3.6
Many Users Dissatisfied	60	3.1
Nearly All Users Dissatisfied	50	2.6
Not Recommended	0	1.0

6. PESQ

In Perceptual Evaluation of Speech Quality (PESQ) the original and degraded signals are mapped onto an internal representation using a perceptual model. The difference in this representation is used by a cognitive model to predict the perceived speech quality of the degraded signal. This perceived listening quality is expressed in terms of Mean Opinion Score, an average quality score over a large set of subjects. PESQ is able to predict subjective quality with good correlation in a very wide range of conditions, that may include coding distortions, errors, noise, filtering, delay and variable delay. Other related evaluations are-Perceptual Speech Quality Measure (PSQM), Measuring Normalizing Blocks (MNB), and Perceptual Analysis Measurement System (PAMS).

7. E-Model Optimization

Speech quality is judged by human listeners and hence it is inherently subjective. The Mean Opinion Score (MOS) test, defined by ITU-T P.800 [13-14], is widely accepted as a norm for speech quality assessment. However, such subjective test is expensive and time-consuming.

The E-Model is very important in our study for two reasons. First, it quantifies the MOS degradation due to delay and loss impairments. In addition, the E-Model models the effect of noise and other speech related impairments, thus allowing us to take them into account without going into details. Second and most important, the E-Model combines all the impairments into a single rating using additive in the appropriate scale *R*.

The E-model has not been fully verified by the researcher or laboratory tests for the very large number of possible combinations of input parameters. For many combinations of high importance to transmission planners, the E-model can be used with confidence, but for other parameter combinations, E-model predictions have been questioned and are currently under study. As per ITU-T recommendations, the algorithm for the E-Model is considered as the common Transmission Rating Model. This computational model is useful to transmission planners to help ensure that users will be satisfied with end-to-end transmission performance. The primary output of the model is a scalar rating of transmission quality. A major advantage of this model is the use of transmission impairment factors that reflect the effects of modern signal processing devices.

8. CONCLUSIONS

The primary conclusion from the research is that the voice over IP network is feasible given that certain stringent requirements are met. First, an E-Model optimization was completed to determine delay, coder type, and other crucial parameters. Second, on the edge of the network (where link bit-rates are small), packet size and the number of voice sources must be controlled. Third, the core of the network must be tightly controlled with respect to voice load and scheduling mechanisms.

REFERENCES

[1] Fowler, Thomas B., "So You Want to Deploy VoIP? A Decision Maker's Guide," *The Telecommunications Review,* **14**, Mitretek Systems, (2003).

[2] Surinder K. Juneja, "Voice Over Internet Protocol: Technology, Political, and Regulatory Considerations", *The Telecommunication Review,* (2003).

[3] Cory C. Beard, Victor S. Frost, "Dynamic Agent-Based Prioritized Connection Admission for Stressed Networks", *IEEE International Conference on Communications,* Vancouver, Canada, (1999).

[4] Olivier Hersent, David Gurle, Jean-Pierre Petit, "IP Telephony, Packet-Based *Multimedia Communications Systems"*, Addison Wesley, (2000).

[5] Grenville Armitage, "MPLS, The Magic Behind the Myths", *IEEE Communications Magazine*, (2000).

[6] Y. Bernet, P. Ford, R. Yavatkar, F. Baker, L. Zhang, M. Speer, R. Braden, B. Davie, J. Wroclawski, E. Felstaine, "A Framework for Integrated Services Operation over Diffserv Networks", RFC 2998, (2000).

[7] Francois Le Faucheur, Liwen Wu, Bruce Davie, Shahram Davari, Pasi Vaananen, Pierrick Cheval, Juha Heinanen, Telia Finland, "MPLS Support of Differentiated Services", *IETF Internet Draft,* (2001), *draft-ietf-mpls-diff-ext-09.txt,*

[8] "Understanding Packet Voice Protocols," Cisco Systems, *www.cisco.com,* (2002).

[9] ITU-T Recommendation P.80, "Methods for Subjective Determination of Transmission Quality", (1993).

[10] T. Porter, Practical VoIP Security, Syngress Publishing, (2006).

[11] Lehtinen, R., and G. T. Gangemi, Sr., Computer Security Basics, O'Reilly, (2006).

[12] Endler, D., and M. Collier., Hacking Exposed VoIP: Voice over IP Security Secrets and Solutions, McGraw-Hill Osborne, (2007).

[13] Anis, M. I, Khan, M. Z, Inam, J. Ajmatullah, R., VOIP using the CBR and VBR Voice Schemes, (2008).

[14] B. Ngamwongwattanam, Effect of Packetization on VOIP Performance, (2008).