

QoS Evaluation Based on Extend E-Model in VoIP

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Abstract—Voice over Internet Protocol (VoIP) is a technology that transports voice data packets across packet switched networks using the Internet Protocol (IP). The key research is to develop a new method for QoS determination and dynamic control of the network that is applicable to VoIP industry. The paper investigates the effects of packet loss, delay and jitter on speech quality for VoIP. It extends the coverage of the standard E-Model. This paper also compares the performances of the original and the modified E-models by applying them to an evaluation of conversational quality in practical VoIP systems (Skype, windows live messenger, Google Talk) for various delay, packet loss and jitter. The results show that the enhanced E-Model achieves good accuracy.

Keywords—VoIP, QoS, E-Model

I. INTRODUCTION

While the evaluation of speech communication systems has been an important field for both academia and industry for decades, the introduction of VoIP systems has created a new set of issues that requires new evaluation methods of QoS. To date, there have been only a small number of comprehensive evaluations of commonly used in VoIP systems.

The Quality-of-Service can be measured using different techniques. Mean Opinion Score (MOS) is one of the widely chosen methods of measuring subjective quality of voice. Other methods include: Perceptual Speech Quality Measurement (PSQM), Perceptual Evaluation of Speech Quality (PESQ) and Perceptual Analysis/Masurement System (PAMS).

Packet Delay, jitter and packet loss dramatically degrades the quality of VoIP calls. Lijing Ding investigates the effects of packet loss on speech quality for random loss and different packet size [1]. Christian Hoene study the impairment of individual packet loss [2]. Furthermore, Alexander Raake classify packet loss into short- and long- term, the short-term behavior relates to the effect of packet loss on the coder and packet loss concealment performance, the long-term loss behavior is defined so that it reflects the loss behavior that ultimately leads to speech quality that perceptively changes [3]. H. Sanneck investigate the impact of frame loss at different positions within a speech signal on the quality and gained the knowledge that the loss of voiced frames at the beginning of a voiced signal segment leads to a significant degradation in speech quality while the loss of other frames are concealed rather well by the codec error concealment

algorithm [4]. There are also many research focused on the other factors such as jitter, delay, loss recovery techniques and codec type [5].

Scalability and performance of the VoIP traffic in the network based on some importance traffic parameters like end-to-end delay, end-to-end jitter, and maximum packet loss in the presence of QoS mechanisms. In table I shows the VoIP service requirements.

Advanced method should be developed to be used to assess the quality with consideration of quality features. The objective of measuring is to find problem and correcting them by collecting information to reach the communication QoS goal. The adequate measurement of QoS is important to the VoIP industry for various reasons, mainly for the cost reduction and high user satisfaction. All the items for the measuring purpose should be thoroughly reviewed and decided. Types of assessment method and quality assessment items are described in table II.

This paper is organized as follows. Section II briefly describes extend E-Model which add new parameter jitter buffer to standard E-Model. In section III describes the measurement and the experiment scenario, and analyses the comparison result among VoIP systems (Skype, Windows Live Messenger, Google Talk). In Section IV present our conclusions and future work.

II. THE EXTEND E-MODEL

Since the most popular objective measurement method is E-Model. E-Model, that is the abbreviation of “European Telecommunications Standards Institute (ETSI) Computation Model [6, 7], was developed by ETSI work group chosen by ITU [8]. It is different from other methods because it represents also a network simulation tool.

The E-Model’s R-factor is translated to a Mean Opinion Score (MOS), to suggest the network’s performance. On a scale from 0 to 100, the R-factor in the E-Model’s is defined as follows:

$$MOS = \begin{cases} 1 & R < 0 \\ 1 + 0.035 R + 7.10^{-6} R(R-60)(100-R) & 0 \leq R \leq 100 \\ 4.5 & R > 100 \end{cases} \quad (1)$$

The R factor can be obtained through the following expression:

$$R = R_o - I_s - I_d - I_e + A \quad (2)$$

where R_o represents the basic signal-to-noise ratio (SNR);

Is represents the combination of all impairments which occur more or less simultaneously with the voice signal; Id represents the impairments caused by delay; Ie represents impairments caused by low bit rate codec; and A is the advantage factor, that corresponds to the user allowance due to the convenience in using a given technology. Once the used codec is well-known (through the Ie factor), we only need to capture network (delay and loss) and application (jitter buffer delay, used codec) statistics for estimating the speech quality by means of the R factor expression (2).

TABLE 1. REQUIREMENTS OF VOIP TRAFFIC

	Performance Parameters	VoIP Service Requirements
1	delay	Less than 50ms in the network
2	loss	Less than 0.1% of packets
3	restoration	Within 50 msec
4	MOS	Greater than 3

TABLE 2. MAJOR MATRIX OF STANDARD QUALITY MANAGEMENT

Measuring item	MOS	R	Delay(ms)	Jitter(ms)	Loss(%)
Very satisfied	4.3-5	90-100	<130	<40	>0.4
satisfied	4. 0-4.3	80-90			
Some users dissatisfied	3. 6-4.0	70-80	130~400	40~60	0.4~1.0
many users dissatisfied	3.1-3.6	60-70	>400	>60	>1.0
Nearly all users dissatisfied	2. 6-3.1	50-60			
Not recommended	1-2.6	0-50			

This is the main reason in adopting the E-model as a measurement tool. The E-Model not only takes in account the transmission statistics (transport delay and network packet loss), but it also considers the voice application characteristics, like the codec quality, codec robustness against packet loss and the late packets discard.

According to [9], equation (2) can be reduced to the following expression:

$$R = 93.4 - Id(Ta) - Ie(codec, loss) \quad (3)$$

Where Id is a function of the absolute one-way delay and Ie is, in short, a function of the used codec type and the packet loss rate.

Since in the same network condition and codec condition, jitter buffer will lead to different MOS value, we add a new parameter, Ij, to standard E-Model.

$$R = Ro - Is - Id - Ie - Ij + A \quad (4)$$

If we ignore other impairments such as echo, factor can be simplified as:

$$R = Ro - Id - Ie - Ij + A \quad (5)$$

The default Ro is 93.2. Due to the affection of jitter buffer, Ij should be divide into two distinct parts. In the first part, the perceptual value slightly degrades when jitter increases because of jitter buffer. When jitter exceed a threshold, the jitter buffer is not helpful, so in the second part, the perceptual value degrade rapidly when jitter increase .so it has a threshold, when the stand deviation smaller than this value , we use liner function estimate the coefficient Ij while use Quadratic function if the stand deviation is bigger. The modified formula is

$$Ij = (\gamma_1 + \gamma_2 j)H(\gamma_0 - j) + (\gamma_3 + \gamma_4 j + \gamma_5 j^2)H(j - \gamma_0) \quad (6)$$

Where $\gamma_0, \gamma_1, \gamma_2, \gamma_3, \gamma_4, \gamma_5$ are coefficients, and γ_0 is the threshold .and H(x) is the function:

$$H(x) = \begin{cases} 0 & x < 0 \\ 1 & x \geq 0 \end{cases} \quad (7)$$

III. SIMULATION AND RESULT

A. Simulation Scenarios

We have developed different scenarios for the assessment of impairment parameters which affect the end to end VoIP call quality. The factors that we considered included: packet loss, packet delay and jitter buffer. It uses OPNET as simulation tools which is very powerful in network analysis and provides network designers useful results.

B. Experiment Result

Figure1 shows comparison between standard E-Model and extend E-Model in condition of packet loss among VoIP application. It is clear that extend E-Model MOS value is higher than standard E-Model. Extend E-Model is accuracy to evaluate QoS for VoIP applications such as Skype, Windows Live Message and Google Talker.

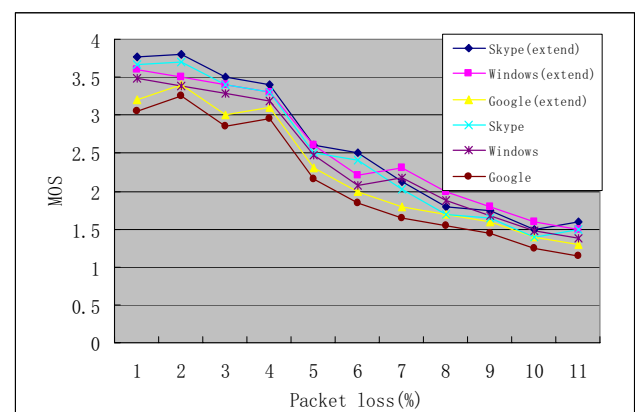


Figure 1. MOS Comparison among VoIP application in condition of packet loss using standard E-Model and extend E-Model.

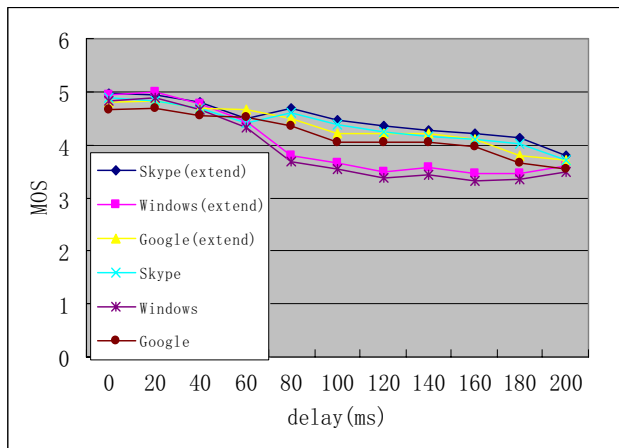


Figure 2. MOS Comparison among VoIP application in condition of packet delay using standard E-Model and extend E-Model.

Figure2 shows that comparison between standard E-Model and extend E-Model in condition of delay among VoIP application. It is clear that extend E-Model MOS value is higher than standard E-Model. Extend E-Model is accuracy to evaluate QoS for VoIP applications such as Skype, Windows Live Message and Google Talker.

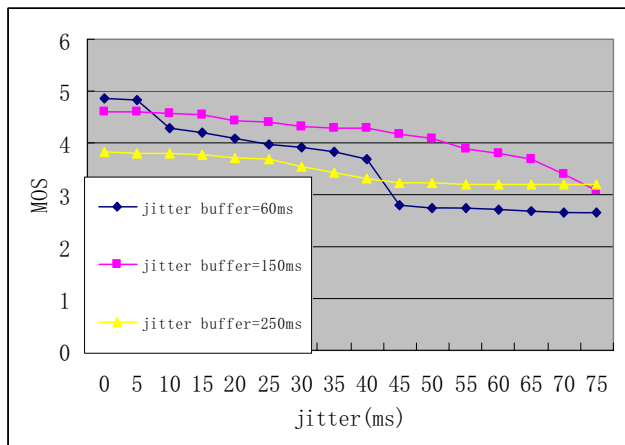


Figure 3. MOS Comparison in condition of jitter using various jitter buffer sizes.

Figure3 shows that comparison among various jitter buffer sizes in condition of jitter.

In this paper we have presented problems and proposed methods to solve the quality control of VoIP using extended objective quality of voice. Through this approach concluded in practice that Windows live messenger is superior in terms of listening, and Skype has consistently larger MOS, and Google

Talk generally has shorter MOS. Three soft of VoIP have different trend curves in condition of packet loss, delay and jitter.

IV. CONCLUSIONS

The adequate measurement of QoS is important to the VoIP industry for various reasons, mainly for the cost reduction and high user satisfaction. In this paper we have presented problems and proposed methods to solve the quality control of VoIP using extended objective quality of voice. Through this approach concluded in practice that Windows live messenger is superior in terms of listening, and Skype has consistently larger MOS, and Google Talk generally has shorter MOS. Three soft of VoIP have different trend curves in condition of packet loss, delay and jitter.

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