Optimizing Voice over Internet Protocol (VoIP) Networks Based-on Extended E-model

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Abstract—**In this paper, we optimize low speed networks access for designing an efficient Voice over Internet Protocol network, especially in 64 kbps, 128 kbps, and 256 kbps bandwidth capacities. We draw some analytical approach based on Extended E-model to quantify some levels of quality of service. Using the model, those levels are determined by network parameters such as delay, jitter, packet loss level, network utilization, and bandwidth capacity and by implementation configuration such as voice coder, packetization scheme, and size of jitter buffer. Our objective is to find maximum number of calls in some given bandwidth capacities while maintaining a certain level of the quality of service. Using numeric estimation, we found that the optimum solution for some given bandwidth capacities can be achieved by applying G.723.1 5.3 kbps voice coder, packet loss level less than 1%, jitter less than 80 ms, and network utilization less than 80%.**

Keywords—**quality of service, VoIP optimization, Extended E-model**

I. INTRODUCTION

Growth of Internet Protocol (IP) technology has a significant impact on global telecommunication today. A major thing of this is the use of Internet Telephony or Voice over IP (VoIP). Users demand this application as an alternative of expensive traditional telephone usage. Such in Public Switch Telephone Network (PSTN), users can make calls anywhere and anytime as long as they connect to IP connection, but with low cost charge and sometimes the usage of this telephony call is free.

In the same time, IP network architecture still lag in providing quality of service (QoS) for real-time applications. VoIP requires stricter quality of service than data applications such as email, World Wide Web, file transfer and chat. Like traditional telephone, VoIP requires high availability of resources, good quality of voices, low delay and low level of packet loss. If VoIP will be used as alternative of telephone applications or will be integrated to traditional telephone network, then it should provide the same or better quality of service [10][12][16].

Beside the requirement described above, implementation of VoIP networks faces the limitation of bandwidth capacity. Work on [18] propose a Softswitch-based Telephony System for Rural Next Generation Network (R-NGN) by applying Ethernet Radio who has range 128 or 256 kbps bandwidth between areas or villages. These connections are low speed IP networks if compared to the demand of users for multi-applications including email, World Wide Web, file transfer, chat and VoIP applications.

As long as VoIP features will be the basic services in R-NGN, we may have a question related to VoIP usage: how to maximize VoIP calls in low speed networks (64 kbps, 128 kbps, and 256 kbps) while maintaining some levels of their quality of service?

According to work on [11], there are three approaches in designing IP network to maintain quality of service and to achieve an optimal network. First, the use of trial and error method is very expensive and time consumed. Even this method can't guarantee to achieve an optimal network. Second, rule of thumb method sets certain values for quality of service requirements such as delay, jitter and packet loss. This method may cause over provisioning and inefficient network. Third, analytical method based on E-model (ITU-T Recommendation G.107) that used to predict voice quality from calculation of quality of service parameters. This method won't be valid when network characteristic change.

The next question is what method provides most efficient quality of service? To answer this question, it is needed to understand VoIP system organization and choose VoIP network parameters. Beside those, it is needed to understand about VoIP characteristics that are affected by voice coder, packet loss level, size of jitter buffer, and network utility. Understanding those characteristics of VoIP, optimization method applied to achieve maximum VoIP calls with minimum level of quality of service.

The rest of this paper is described as follows. Section 2 describes some research work related to this research. Here some characteristics of VoIP application that is comprised of delay, jitter and packet loss are described. In Section 3, we review E-model and Extended E-model. In this section, we describe analytical approach to find transmission rating factor R based on those models. In Section 4, we define our work on the optimization of low speed network and describe result and analysis of numeric estimation. Section 5 describes some conclusions.

II. RELATED WORK

Work on [18] identifies two important characteristic of data transmission in IP network. The first characteristic is delay required to transmit data from source to destination. The delay consists of two components, i) fixed delay which includes transmission and propagation delay, ii) variable delay which includes processing and queueing delay. The second characteristic is level of packet loss. Understanding those characteristic is important to design network properly.

There are other research projects focus on measurement of quality of service based on delay and packet loss parameters [12][17][18]. ITU-T Recommendation G.114 (05/2003) recommends delay (one way delay or latency) less than 150 ms for interactivity communications, and delay more than 400 ms will be unaccepted. Work on [13] describe that delay variation (jitter) less than 20 ms will be highly accepted, but it is will be unaccepted is delay variation more than 50 ms. Level of packet loss less than 10% is still tolerable [20].

Research work in [11] optimize VoIP network by selecting some parameters including voice coder, packet loss level and network utilization. This work show three scenarios for verifying and validating E-model optimization, i) finding optimal coder given link bandwidth, packet loss level and link utilization, ii) finding optimal voice coder and packet loss level given link bandwidth and link utilization, and iii) finding optimal voice coder and link utilization given link bandwidth and packet loss level.

Our research is the extension of work in [11], but we focus on low speed networks (64 kbps, 128 kbps, and 256 kbps). In this research, the optimization is based on Extended E-model approach where delay, jitter, packet loss level and voice coder are included in the analytical approach to predict the level of quality of service.

III. EXTENDED E-MODEL REVIEW

ITU-T G.107 [5] defines E-model, a computational model combining all the impairment parameters into a total value. The E-model is not a measurement tool, but an end-to-end transmission planning tool; the output can be transformed into a Mean Opinion Score (MOS) scale to indicate the subjectivity of voice quality.

As cited in [9], the E-model assesses the combined effects of varying transmission parameters that affect the conversation quality of narrow band telephony. The E-model is based on the assumptions that transmission impairments can be transformed into psychological factors and psychological factors on the psychological scale are additive. The output of the E-model is a transmission rating factor R that is defined in [5] as:

$$R = Ro - Is - Id - Ie + A \tag{1}$$

where Ro represents the basic signal-to-noise ratio, Is represents the impairments occurring simultaneously with the voice signal, Id represents the impairments caused by delay, and Ie represents the impairments caused by low bit rate voice coders. It also includes impairment due to packet losses of random distribution. A is an advantage factor than can be used for some compensation.

Work on [9] proposed Extended E-model, an extension of the E-model by including Ij that represents the impairments caused by jitter. The output of the Extended E-model is a transmission rating factor R that is defined as:

$$R = Ro - Is - Id - Ie - Ij + A \tag{2}$$

Next, we will draw analytical approach to find the transmission rating factor R..

A. Delay Impairment Id

Total delay in VoIP applications is comprised from packetization delay, queueing delay, serialization delay, propagation delay, encoding/decoding delay and jitter buffer delay. ITU-T G.107 [5] defines delay impairment Id as:

$$Id = Idte + Idle + Idd \tag{3}$$

In this research, we assume that echo cancellation work fine in our system. In this echo free environment, delay impairment of talker echo $Idte$ and that of listener echo $Idle$ is 0. Then, we will have $Id = Id$. Impairments cause by absolute delay Idd is defined as:

$$Idd = 25\left\{\left(1+X^6\right)^{1/6} - 3\left(1+\left[\frac{X}{3}\right]^6\right)^{1/6} + 2\right\} \tag{4}$$

where $X = log(Ta/100)/log2$. Ta is one way delay in echo free environment. If Ta is less than *100 ms,* then Idd is assumed 0.

Next, we will calculate queueing delay. Queueing process in packet data transmission contributes to queueing delay occurance [15]. Using queueing model M/M/1, queueing delay can be derived from distribution of sojourn time [3]. Sojourn time S is time spent in system (queueing + service) by customer k.

The probability of $S > Td$ is defined in [19] as:

$$Id = Idte + Idle + Idd \tag{5}$$

Packet loss *(PL%)* is assumed when the sojourn time S is greater then T_d. In other words, packet loss will be the point on the tail of queueing delay distribution where packets will be simply drop [11]. Then we will have $\%PL = P(S > Td)$. Substitute this to (5), queueing delay Td is determined by level of packet loss and affected by service rate μ and network utilization ρ as follows:

$$T_d = \frac{\ln(PL\%)}{-\mu(1-\rho)} \tag{6}$$

Applying N voice frame per packet data, the compression delay can be calculated using equation defined in [7] as:

$$comp\ delay = (N+1)*length\ of\ frame + look\text{-}ahead \quad (7)$$

In this research, we consider to use 3 hop systems for calculating total queueing delay. We assume the propagation delay is 25 ms and miscellaneous delays caused by switching and echo cancellation are 6 ms. The total delay Ta can be calculated using [11] as follows:

$$Ta = hop\ count * Td + compression\ delay + propagation\ delay + misc\ delay \quad (8)$$

B. Equipment Impairment Ie

ITU-T G.113 [6] provides guidance related to transmission impairments caused by voice compression and level of packet loss. *Ie* values can be determined trough subjective tests according to ITU-T P.833 (2001). Reference [1] describe that network congestion will be a major problem that affect the level of packet loss.

Work on [9] applying one frame per packet and using random packet loss distribution to find *Ie*. This impairment increased logarithmically with packet loss rate as follows:

$$Ie = Ie\_opt + C1 \cdot \ln(1 + C2 \cdot loss\_rate) \quad (9)$$

where *Ie_opt* is the optimum (without packet loss) *Ie* as defined in [6], *loss_rate* is the amount of packet loss in percent, and factors *C1, C2* are constant used to adjut the shape of the curve.

C. Jitter Impairment Ij

Work on [9] uses Pareto distribution to model network delay that is caused by jitter. The jitter impairment *Ij* is defined as:

$$Ij = C1 \cdot H^2 + C2 \cdot H + C3 + C4 \cdot e^{-Tj/K} \quad (10)$$

where *C1, C2,* C3, *C4* are coefficients, *Tj* is a fixed buffer and *K* is a time constant

IV. Optimization and Numeric Estimation

Our research is aimed to maximize calls in limited speed networks while maintaining a minimum level of quality of service. The constraint of the optimization is transmission rating factor R≥60. Table 1 in [5], R = 60 represents many users dissatisfied.

TABLE I. Relation Between R and User Satisfaction

R-value	MOS	User Satisfaction
90	4.34	Very satisfied
80	4.03	Satisfied
70	3.60	Some users dissatisfied
60	3.10	Many users dissatisfied
50	2.58	Nearly all users dissatisfied

As we stated above, our optimization model is based on selection of some parameters such as voice coder (G.723.1 5.3 kbps and 6.3 kbps, and G.729), packet loss level, the size of jitter buffer, and network utilization. The optimization algorithm is defined as follows:

Maximize: Number of calls in the IP conection;
Subject to: R (voice coder, packet loss level, jitter buffer, network utilization) >= 60;

A. Optimization of Voice Coder and Packet Loss Level

As shown in Fig.1, when the bandwidth was 64 kbps, optimum solution can be achieved by applying G.723.1 5.3 kbps voice coder, packet loss level less than 1%. The optimum solution was 2.46 calls.

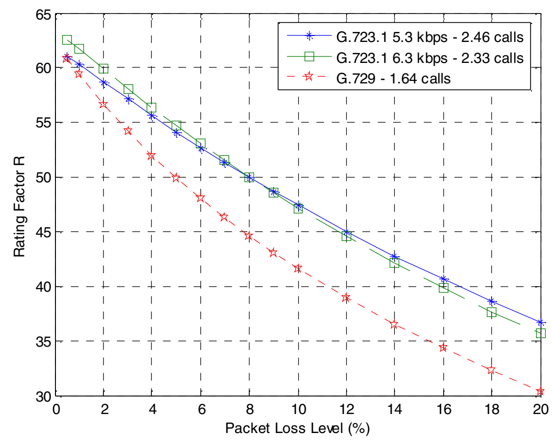

Figure 1. Factor R vs. packet loss level at 64 kbps

The result is similar when the bandwidth was 128 kbps and 256 kbps, but the optimum solution was 4.92 calls for 128 kbps and 9.85 calls for 256 kbps.

B. Optimization of Voice Coder and Jitter Buffer

As shown in the Fig. 2, when the bandwidth was 64 kbps, optimum solution can be achieved by applying G.723.1 5.3 kbps voice coder and jitter buffer was set between 30-80 ms. The optimum solution was 2.46 calls.

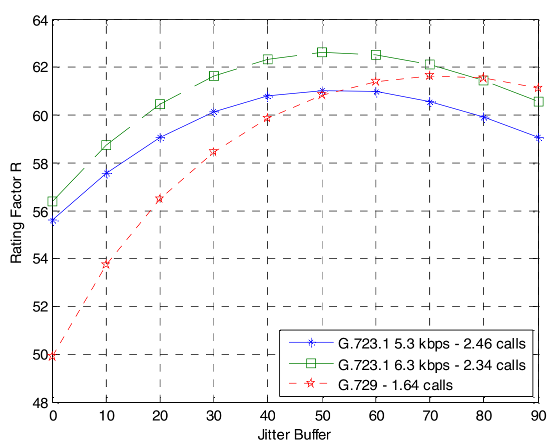

Figure 2. Factor R vs. jitter buffer at 64 kbps

When bandwidth was 128 kbps and 256 kbps, optimum solution can be achieved by the same voice coder but jitter buffer was set between 30-90 ms. The optimum solution was 4.92 calls and 9.85 calls.

C. *Optimization of Voice Coder and Network Utilization*

As shown in the Fig. 3, when the bandwidth was 64 kbps, optimum solution can be achieved by applying G.723.1 6.3 kbps voice coder, network utilization less than 80%. The optimum solution was 2.34 calls as shown in Fig.4.

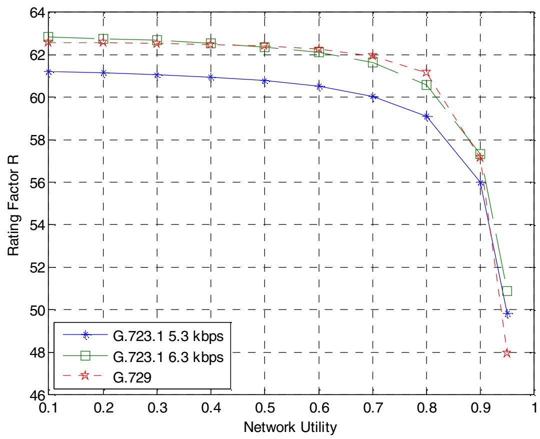

Figure 3. Factor R vs. network utilization at 64 kbps

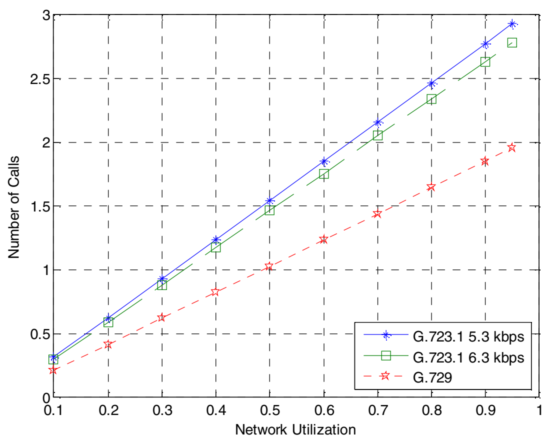

Figure 4. Number of calls vs. network utilization at 64 kbps

When bandwidth was 128 kbps, optimum solution can be achieved by applying G.723.1 6.3 kbps voice coder, network utilization less than 90%. The optimum solution was 5.26 calls. When bandwidth was 256 kbps, optimum solution can be achieved by applying G.723.1 5.3 kbps voice coder, network utilization less than 90%. The optimum solution was 11.08 calls.

V. CONCLUSION

This research is aimed to maximize number of calls while maintaining a minimum level of quality of service in low speed IP networks (64 kbps, 128 kbps, and 256 kbps). We utilize Extended E-model [9] that is based-on ITU-T G.107 E-model [5] as a tool to predict quality of voice based on various parameters in VoIP network including delay, packet loss and jitter.

Using analytical approach based on Extended E-model, we optimize the low speed networks based on combination of parameters voice coder, packet loss level, jitter buffer, and network utilization. The constraint of our optimization is transmission rating factor R ≥60 or similar to MOS≥3.1 where “many users dissatisfied”.

The result of applying numeric method are as follows. When the bandwidth was 64 kbps, optimum solution can be achieved by applying G.723.1 5.3 kbps voice coder, packet loss level less than 1%, jitter buffer 30-80 ms and network utilization less than 80%. When the bandwidth was 128 kbps, optimum solution can be achieved by applying G.723.1 5.3 kbps voice coder, packet loss level less than 1%, jitter buffer 30-90 ms and network utilization less than 80%. When the bandwidth was 256 kbps, optimum solution can be achieved by applying G.723.1 5.3 kbps voice coder, packet loss level less than 1%, jitter buffer 30-90 ms and network utilization less than 90%.

In general, optimum network performance in low speed networks can be achieved by applying voice coder G.723.1 5.3 kbps, packet loss level less than 1%, delay jitter less than 80 ms, and network utilization less than 80%

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