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VoIP Quality Optimization in IP–Multimedia Subsystem (IMS)

Wagdy A. Aziz*, Salwa H. Elramly † and Magdy M. Ibrahim †
* The Egyptian Company for Mobile Services-Mobinil/Technology Department, Cairo, Egypt, wanis@mobinil.com
† Faculty of Engineering/Electronics and Communication Engineering Department, Cairo, Egypt
sramlye@netscape.net and Magdy_Ibrahim@yahoo.com

***Abstract*—IP Multimedia Subsystem (IMS) is very important due to the critical role it plays in the Next Generation Network (NGN) of the Fixed and Mobile Networks. Voice traffic in IMS will be served using Internet Protocol (IP) which is called Voice over IP (VoIP). This paper uses the "E-Model", (ITU-T G.107), as an optimization tool to select network and voice parameters like coding scheme, packet loss limitations, and link utilization level in IMS Network. The goal is to deliver guaranteed Quality of Service for voice while maximizing the number of users served.This optimization can be used to determine the optimal configuration for a Voice over IP in IMS network OPNET is the optimization tools that is used in this paper. The paper also provides new equations can be added to enhance E-Model to relate packet loss to the level of Equipment Impairment (Ie) with different codecs.**

I. Introduction

The IP Multimedia subsystem (IMS) is an overlay system that is serving the convergence of mobile, wireless and fixed broadband data networks into a common network architecture where all types of data communications are hosted in all IP environments using the session initiation protocol (SIP) protocols infrastructure [1].

IMS is logically divided into two main communication domains, one for data traffic, i.e., real time protocol packets consisting of audio, video and data and the second one is for SIP signaling traffic.

This paper focuses on the VoIP Quality of Service over IMS using SIP as a signaling protocol.

Quality is a subjective factor, which makes it difficult to measure. Taking an end to end perspective of the network further complicates the QoS measurements. The reasons for low quality voice transmission are due to degrading parameters like delay, packet delay variation, codec related impairments like speech compression, echo and most importantly packet loss. Large research efforts have been made to solve the vital quality of service issues. In the VoIP end-to end QoS measuring and monitoring area this has resulted in various objective QoS measuring models. The output is generally a single quality rating correlated to the subjective MOS score.

Many of the developed models for measuring VoIP quality of service are inappropriate for smaller, private networks. They may take too much process resource, are intrusive on the regular traffic or contain very complicated test algorithms.

One of the best models used for measuring VoIP quality of service is the E-model, which is a parameter-based model.

The E-Model, (ITU-T G.107) [2], is a model that allows users to relate Network impairments to voice quality. This model allows impairments to be introduced and voice quality to be assessed. Three cases are considered to demonstrate the effectiveness of optimizing the VoIP over IMS network using E-Model.

New equations were also provided to enhance E-Model that can be used to relate packet loss to the level of Equipment Impairment (Ie) with different codecs.

The objective function for all cases is to maximize the number of calls that can be active on a link while maintaining a minimum level of voice quality (R> 70).

The cases considered are:

1. Find voice coder given link bandwidth, packet loss level, and link utilization level.
2. Find voice coder and packet loss level given link bandwidth and background link utilization.
3. Find voice coder and background link utilization level given link bandwidth and packet loss level

OPNET is the optimization tool that is used in this paper.

A. *IMS Architecture*

IP Multimedia Subsystem is defined by 3rd-Generation Partnership Project (3GPP) which defines IMS standards as a network domain dedicated to the control and integration of multimedia services [1].

IMS builds on Internet Engineering Task Force (IETF) protocols like Session Initiation Protocol (SIP), Session Description Protocol (SDP) [3] and Diameter protocol. Figure 1 shows the common nodes included in the IMS. These nodes are:

1) P-CSCF (Proxy-CSCF): The P-CSCF is the first point of contact between the IMS terminal and the IMS network. All the requests initiated by the IMS terminal or destined to the IMS terminal traverse the P-CSCF.

2) I-CSCF (Interrogating-CSCF): It has an interface to the SLF (Subscriber Location Function) and HSS (Home Subscriber Server). This interface is based on the Diameter protocol [4].The I-CSCF retrieves user location information and routes the SIP request to the appropriate destination, typically an S-CSCF.

3) S-CSCF (Serving-CSCF): It maintains a binding between the user location and the user's SIP address of record (also known as Public User Identity). Like the I-CSCF, the S-CSCF also implements a Diameter interface to the HSS.

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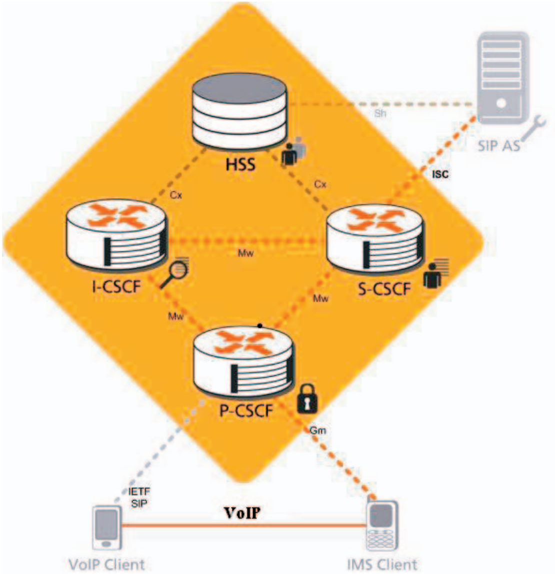

Fig. 1. IMS Functional Elements.

4) SIP AS (Application Server): The AS is a SIP entity that hosts and executes IP Multimedia Services based on SIP.

5) The Home Subscriber Server (HSS): contains all the user related subscription data required to handle multimedia sessions.

B. SIP in IMS

Session Initiation Protocol (SIP) is a prominent protocol (RFC 3261 [5]). SIP has been chosen in IMS to play the key role for setting up the session while inter-working with other protocol.

All IP voice and multimedia call signaling in IMS will be performed by SIP, end to end, providing a basis for rapid new service introductions and integration with fixed network IP services.

II. RELATED WORK

Quality is a subjective factor, which makes it difficult to measure. Taking an end to end perspective of the network further complicates the QoS measurements. The reasons for low quality voice transmission are due to degrading parameters like delay, packet delay variation, codec related impairments like speech compression, echo and most importantly packet loss. Large research efforts have been made to solve the vital quality of service issues. In the VoIP end to end QoS measuring and monitoring area this has resulted in various objective QoS measuring models. The output is generally a single quality rating correlated to the subjective MOS score.

These methods are subdivided in three categories: comparison or intrusive methods, absolute estimation models also called non-intrusive models and transmission rating models that derive quality estimations from knowledge about the network.

One of the best models used for measuring VoIP quality of service is the E-model, which is a parameter-based model.

III. DESCRIPTION OF E-MODEL OPTIMIZATION

The E-Model, (ITU-T G.107) [2] is extremely complex with 18 inputs that feed interrelated components. These components feed each other and recombine to form an output (R).

The recommendation ITU-T G.108 [6 gives a thorough description on how to carry out an E-model QoS calculation within VoIP networks

Due to the complexity of the E-Model, the approach used here is to try to identify which E-Model parameters are fixed and which parameters are not. In the context of this research the only parameters of the E-Model that are not fixed are:

- T and Ta – Delay variables
- Ie – Equipment Impairment

Where (T) is the mean one way delay of the echo path, (Ta) is the absolute delay in echo free conditions [4]. In addition, parameters that affect delay Id and Ie are introduced:

- PL - Packet Loss %
- ρ- Link Utilization
- Coder Type

Next, the relationship between these parameters is identified. Since, we are making the assumption that the echo cancellers are on the end and are very good, we can say that T = Ta and Ie is directly related to a particular coding scheme and the packet loss ratio.

According to the above assumption, R-Factor equation can be reduced to the following expression:

$$R = 93.2 - Id\ (Ta) - Ie\ (codec,\ packet\ loss) \qquad (1)$$

A. To Calculate The Delay Impairment Id [4]:

For $Ta < 100$ ms:

$$Id = Idd = 0 \qquad (2)$$

For $Ta > 100$ ms:

Id=

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

With:

$$X = \frac{\log\left(\frac{Ta}{100}\right)}{\log 2}$$

B. To Calculate Equipment Impairment Ie

The loss impairment **Ie** captures the distortion of the original voice signal due to low-rate codec, and packet losses in both the network and the play out buffer. Currently, the E-Model can only cope with speech distortion introduced by several codecs i.e. G.729 [7] or G.723 [8].

Specific impairment factor values for codec operation under random packet-loss have formerly been treated using tabulated, packet-loss dependent Ie-values. Now, the Packet-loss Robustness Factor Bpl is defined as codec specific value.

The packet-loss dependent Effective Equipment Impairment Factor Ie-eff is derived using the codec specific value for the Equipment Impairment Factor at zero packet-loss. Ie and the Packet-loss Robustness Factor Bpl, both listed in Table I for several codecs. With the Packet-loss Probability Ppl, $I_{e\text{-}eff}$ is calculated using formula (4)

$$Ie - eff = Ie + (95 - Ie)\frac{Ppl}{Ppl + Bpl} \qquad (4)$$

As can be seen from this formula (4), the Effective Equipment Impairment Factor in case of Ppl = 0 (no packet-loss) is equal to the Ie value defined in Table I.

Ie represents the effect of degradation introduced by codecs, Packet Loss. G.113 provides parameters for use in calculating Ie from codec type and Packet Loss rate [9].

TABLE I Provisional planning values for the equipment impairment factor Ie and for packet-loss robustness factor Bpl [9]

Codec	Rate (Kbps)	Packet Size (msec)	Ie (no packet loss)	Bpl
G.711	64	10	0	25.1
G.729A + VAD	8	20	11	19.0
G.723.1+VAD	6.3	30	15	16.1

Coming to the VoIP traffic Characterization, Human speech is traditionally modeled as sequence of alternate talk and silence periods whose durations are exponentially distributed and referred as to ON-OFF model.

On the other hand all of the presently available codecs with VAD (Voice Activity Detection) have the ability to improve the speech quality by reproducing Speakers back ground by generating special frame type called SID (Silence Insert Descriptor). SID frames are generated during Voice Inactivity Period.

C. Mapping R factor Into MOS Scale

We can map ***R*** into MOS scale by the following equations [2]

For R < 0:

MOS = 1 (5)

For 0 <R <100:

MOS = 1 + 0.035 R + 7.10-6 R(R-60)(100-R) (6)

For R > 100:

MOS = 4.5

(7)

IV. Project Setup

The simulation is done using OPNET simulation tool IT Guru Academic Edition 9.1 for VoIP in IMS network using SIP Protocol.

The network consists of IP-Telephones (VoIP or IMS Clients) connected to the Internet by routers which act as IP gateway, the network is managed by the SIP proxy server (act as P-CSCF) which uses the SIP protocol to establish the voice calls (VoIP) on the IMS network.

The links between the routers and the Internet are T1 with link speed 1.544 Mbps and the links between the dialer, dialed, Proxy Server and the routers are 1000 Base-x. The idea is to configure the network with a certain parameters and run the simulation then getting from the tool the result values which used in E-Model equations to measure the Quality of service Factor R.

The objective function for all cases is to maximize the number of calls that can be active on a link while maintaining a minimum level of voice quality (R). The cases considered are:

1. Find the optimal voice coder given link bandwidth, packet loss level, and background link utilization level.
2. Find the optimal voice coder and the optimal packet loss level given link bandwidth and background link utilization.
3. Find the optimal voice coder and the optimal background link utilization level given link bandwidth and packet loss level.

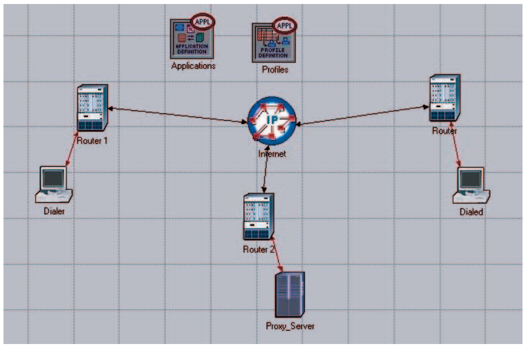

Fig .2. Network Topology

Table II shows standard parameters for each codec used in the analysis

TABLE II Codec Parameters

Codec Infor-mation	Bandwidth Calculations					
Codec & Bit Rate (Kbps)	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Number of Voice Frames per Packet	Packets Per Second (PPS)	Bandwidth with RTP/UDP/IP/ PPP Header	MTU Size (Bytes)
G.711 (64 Kbps)	160 Bytes	20 ms	1	50	82.8 Kbps	207 Bytes
G.729 (8 Kbps)	20 Bytes	20 ms	1	50	26.8 Kbps	67 Bytes
G.723.1 (6.3 Kbps)	24 Bytes	30 ms	1	34	18.9 Kbps	71 Bytes

V. RESULTS

The results are divided into three general cases. For all cases, the aim is to maximize the number of calls that can be carried on a link while maintaining a minimum voice quality level (R > 70).If two combinations produce the same number of calls, the highest R value will be considered the best selection.

A. Case 1 - Optimizing for Coder Selection

The goal of this case is to find the optimal voice coder given link bandwidth, packet loss level, and background link utilization level.

Table III and Table IV are containing the coding parameters used in Case1 1 of the simulation. OPNET is configured by these parameters which are according to the ITU-T G.107

TABLE III Codec Parameters for case1-1 [2] [9]

Standard	Type	Codec Bit Rate (kbps)	Voice Frame Length (ms)	Look ahead (ms)	Frame length (ms) Packet Length	Number of Voice Frames per Packet	I_E No PL
G.711	PCM	64	0.125	0	20	1	0
G.729	CS-ACELP	8	10	5	20	2	11
G.723.1	MP-MLQ	6.3	30	7	30	1	15

For Case 1 with a link speed of 1.544 Mbps, The simulation was run for 2 hours and 4 hours and in all cases G.723.1 gave the max. Number of calls with R value more than 70, so G.723.1 was selected as the optimum Coder.

For G.711 gave the max. Quality of service (Highest R value) but the lowest number of calls, G.729 gave middle number of calls between G.711 and G.723 and also middle R value. As shown in Table 7

The results of this case are shown in Figure3 not surprising, as G.723.1 is a more efficient but lower quality of voice.

TABLE IV Codec Parameters for case1-2

Codec Infor-mation	Bandwidth Calculations					
Codec & Bit Rate (Kbps)	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Number of Voice Frames per Packet	Packets Per Second (PPS)	Bandwidth with RTP/UDP/IP/ PPP Header	MTU Size (Bytes)
G.711 (64 Kbps)	160 Bytes	20 ms	1	50	82.8 Kbps	207 Bytes
G.729 (8 Kbps)	20 Bytes	20 ms	2	50	34.8 Kbps	87 Bytes
G.723.1 (6.3 Kbps)	24 Bytes	30 ms	1	34	18.9 Kbps	71 Bytes

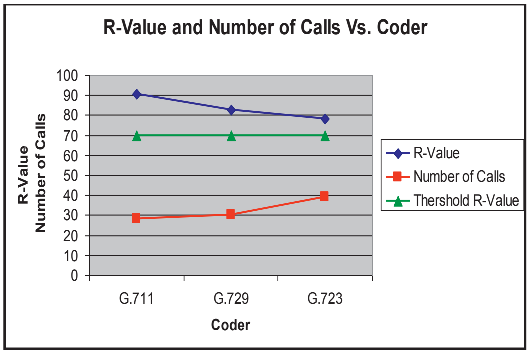

Fig. 3. R Value, Number of Calls vs. Coder – case (1)

B. Case 2 - Optimizing for Coder and Packet Loss Level Selection

The goal of this case is to find the optimal voice coder and the optimal packet loss level given link bandwidth and background link utilization.

Table V contains the parameters used for case 2 of the simulation which is according to the ITU-T recommendation G.113

Table V Codec Parameters for case 2 [9]

Codec	Rate (Kbps)	Packet Size (ms)	Ie With no PL	Bpl
G711	64	10	0	25.1
G.729A+VAD	8	20	11	19.0
G.723.1+VAD	6.3	30	15	16.1

The OPNET simulation is configured by the above parameters like the codec bit rate and the packet size and the number of voice frames per packet but other values like Ie and Bpl are coming from ITU-T G.107 and G.113 for the mentioned codecs. The simulation was run for 1 hour, 2 hours and 4 hours and for the 3 coders G.711, G.729 and G.723 with different values of packet loss ratio.

For the 3 coders G.711, G.729 and G.723 with different values of packet loss ratio (0.5 %, 1 %, 1.5 %, 2 % and 5 %) knowing that the maximum allowable ratio is 2% but the simulation was run for PL% equal 5% to observe the network behavior in case of big crisis as shown in Figure 6.The test was run with a link speed of 1.544 Mbps. The maximum number of calls was 29 calls. G.723.1 with packet loss of 0.5% was the combination chosen and the same combination was chosen till packet loss of 1.5 %.

When packet loss ratio reached 2 %, G.723.1 became not feasible as its R value is less than 70 and G.729 with packet loss 2% was the combination chosen.

For packet loss more than 2 % G.723.1 and G.729 became not feasible and the only feasible coder is G.711.G.711 with packet loss more than 2 % was the combination chosen.

It is noticed that the most affected parameter in this case is the Ie which is expected as the PL is affecting the Ie parameters as shown in Figure5.

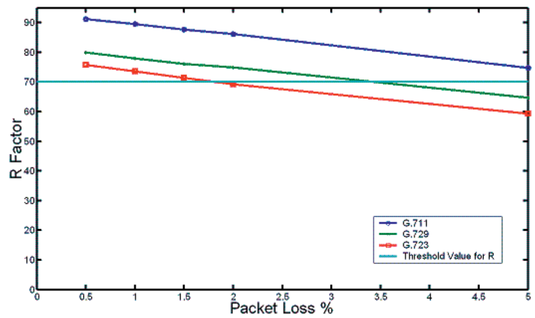

Fig. 4. R Value and PL % vs. Coder – case (2)

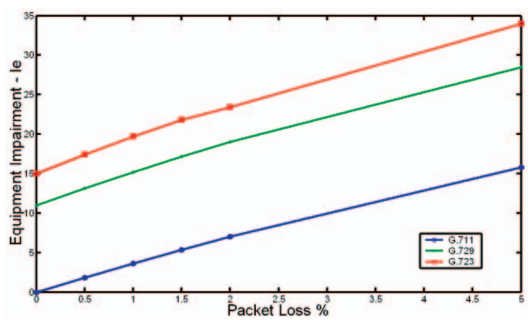

Fig.5. PL % and Ie vs. Coder – case (2)

The following equations were driven and could be added to enhance E-Model for some codecs.

For G.711, the following polynomial was generated, where x represents the level of packet loss and y represents the level of impairment (Ie).

$$y = 0.0046\, x^3 - 0.156x^2 + 3.8x - 0.00035 \quad (8)$$

For G.729A, the following polynomial was generated:

$$y = 0.0081x^3 - 0.22x^2 + 4.4x + 11 \quad (9)$$

For G.723.1, the following polynomial was generated:

$$y = 0.084x^3 - 0.74x^2 + 5.2348x + 15 \quad (10)$$

Figure 6, Figure 7 and Figure 8 show a graph of the observed versus the curve fit.

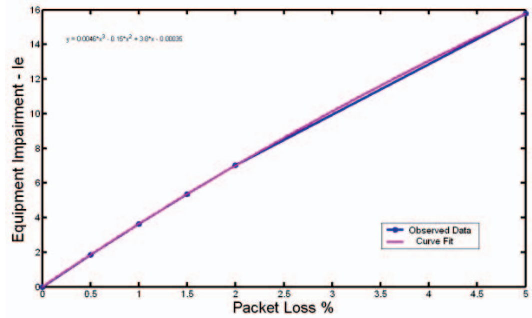

Fig. 6. G.711 Polynomial Fit

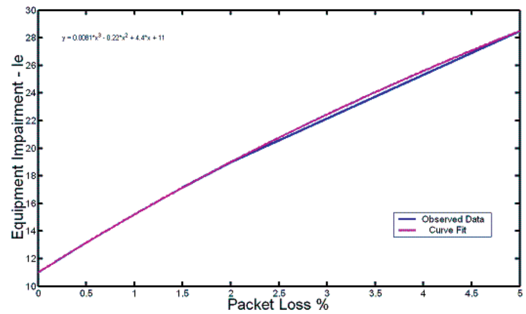

Fig .7. G.729A Polynomial Fit

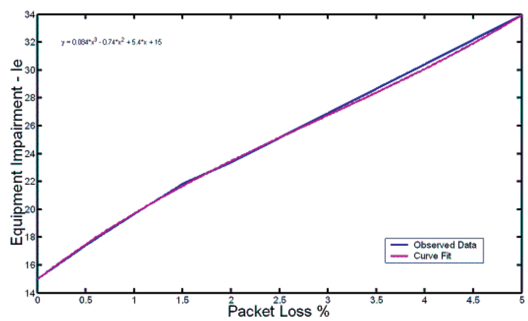

Fig.10. G.723.1 Polynomial Fit

The optimization Results for case (2) is listed in Table VI

TABLE VI Results of E-Model Optimization Case (2)

Case #	Packet loss%	Optimum Coder
1	0.5%	G.723.1
2	1%	G.723.1
3	1.5%	G.723.1
4	2%	G.729
5	> 3.5%	G.711

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