Impact of Network Jitter on Effective Equipment Impairment Factor

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Abstract: - The paper deals with a non-intrusive approach to the speech quality assessment and examines an impact of network jitter on effective equipment impairment factor in E-model. In order to improve currently used computational E-model (ITU-T G.107), we proposed an optimization based on numerous measurements and experiments with an objective intrusive method PESQ (Perceptual Evaluation of Speech Quality) under varying delay, packet loss, jitter and play-out buffer. Comparing achieved results in experiments and the computed values in E-model, we proposed modification which improves estimated MOS of E-model.

Key-words: - E-model, MOS, packet loss, jitter, jitter buffer, network traffic, call quality.

1 Introduction

The Internet, VoIP and in general IP traffic is known to possess the property of being self-similar, long-range dependent (LRD) or in other words “bursty”.

The behavior of a “bursty” traffic differs from ideal stochastic model of independent packets when trying to evaluate traffic interarrival times via well-known distributions. This property translates into the failure of general queuing models, such as M/M/1/k, which counts on Exponential and Poisson characteristics of input stream and service time, to describe the situation of incoming VoIP stream at buffer on receiver’s side.

In our article we analyze and improve original E-Model designed to give real-time estimate of VoIP call quality in MOS scale based solely on network performance parameters and codec type. We work with the 04/2009 version of E-model, which still after numerous updates, does not incorporate the effects of jitter. While the performance of the E-Model estimate is satisfactory under good network conditions, the E-Model MOS estimate becomes too optimistic under slightly and moderately impaired network conditions as shown in our previous work [1], [2] and [3].

2 E-model Description and Caveats

Mean opinion score (MOS) is a measure based on subjective user satisfaction with overall listening and conversational quality on five grade scale from 5 (best) to 1 (worst). MOS can be estimated by subjective methods based on physical listening tests or by objective methods relying on and working solely with real-time measured network performance parameters (delay, packet loss) which unfortunately does not include jitter and buffer size.

E-model defined by ITU-T G.107 [4] is widely accepted objective method used for estimation of VoIP call quality. E-model uses a set of selected input parameters to calculate intermediate variable – R factor, which is finally converted to MOS value. Input parameters contribute to final estimate of quality in additive manner as expressed in (1).

$$R = R_0 - I_s - I_d - I_{(e\text{-}eff)} + A. \qquad (1)$$

Where

- R_o is the basic SNR, circuit and room noise;
- I_s represents all recording impairments;

• I_d covers degradations caused by delay;
• $I_{e\text{-}eff}$ impairment factor presents all degradations caused by packet network transmission path, including end-to-end delay, packet loss and codec PLC masking capabilities;
• A is a technology advantage factor;

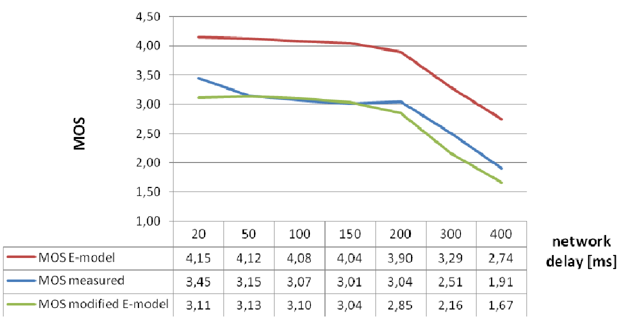
We focus on $I_{e\text{-}eff}$ parameter calculated as (2):

$$I_{(e\text{-}eff)} = I_e + (95 - I_e).P_{pl}/(P_{pl} + B_{pl}) \quad (2)$$

Where I_e represents impairment factor given by codec compression, B_{pl} is codec robustness characterizing codec's immunity to random losses.

The values are given for 8kHz sample rate codecs in ITU-T G.133 appendix [5]. P_{pl} parameter represents measured network packet loss in %. In this paper we propose a substitution of P_{pl} parameter for P_{plef} further described in section IV of the paper.

Our measurements and simulation show that the performance and estimate accuracy of E-Model deteriorates unacceptably beyond network jitter (calculated by RFC 1889) over 20 ms for codecs G.711 with and without PLC, G.723.1 ACELP and MP-MLQ, G.726 and G.729. Fig. 1 shows an example of measured E-Model MOS inaccuracy of VoIP network connection in following manner:
• "MOS E-Model" – represents MOS as estimated via software on receiving side by reading network performance from RTCP protocol not accounting for the effects of local jitter buffer.
• "MOS measured" – represents MOS estimated by measuring software – IX-Chariot – based of the net voice input packet stream entering the decoder behind buffer;
• "MOS modified E-Model" shows estimate performed via software using E-Model [4] incorporating the effects of jitter and buffer size based on actual codec configuration and data about network performance without physically observing or interfering with packet stream behind jitter buffer.



	20	50	100	150	200	300	400
MOS E-model	4,15	4,12	4,08	4,04	3,90	3,29	2,74
MOS measured	3,45	3,15	3,07	3,01	3,04	2,51	1,91
MOS modified E-model	3,11	3,13	3,10	3,04	2,85	2,16	1,67

Fig. 1. Comparison of MOS estimates for G.729 codec at 40 ms RFC jitter and 40 ms buffer size, ideal 0 % packet loss under varying network delay

2 Jitter Buffer Effects on MOS - Call Quality Estimate

Timescale of our interest is in order of seconds under practical real-time conditions what is supported by the following facts: Jitter J is calculated from 16 consequent interarrival times. Jitter buffer size is in order of tens to hundreds of milliseconds for practical VoIP call purposes. E.g., with standard packetization of 20ms we get 320 ms buffer size when considering buffering 16 packets.

Regarding the traffic, following holds true: the interarrival time is "exactly second-order self-similar" with Hurst parameter $H = 1-\beta/2$ and eq. (3) holds true.

$$r(k) = \frac{1}{2}\delta^2\left(k^{2-\beta}\right). \quad (3)$$

The variance of input packet stream can be considered constant for the short time-scale we operate on as induced from [7 and 12]. The Hurst parameter from short-term point of view in order of seconds is constant and can be put equal to H=1.

Voice packets are generated at sending device – IP phone – as a homogenous flow with constant transmit intervals depending mostly on packetization interval set in the codec. VoIP packets that traversed transport network have their regular spacing disrupted irregularly. Internet traffic arrival times and delay can be successfully statistically modelled by long-tailed Generalized Pareto distribution (GPD) [6, 7, 8, 9, 11]. We use GPD to further describe VoIP input packet stream. Delay distribution of received packets is in Fig. 2.

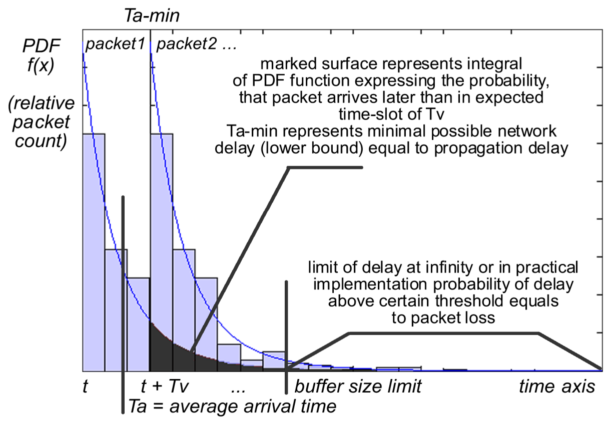


Fig. 2. Distribution of Pareto-related packet arrival times

Real-time change of network parameters causes variations in network delay. Differences between packet arrivals are not constant and arrival times oscillate between minimal delay *Ta-min* and infinite delay, which is effectively a lost packet. Mean value of the process exists and is interpreted as an End-to-End delay *Ta* (one of the input parameters for E-model). Real packet path usually consists of a mixture of different networks with different devices and technologies. Each device adds a degree of uncertainty in packet delivery time. Overall delay statistics is a sum of all partial statistics at each device. Pareto distribution is well suited to describe delay, which has lower bound, no upper bound and finite mean value. Probability density function of Pareto (PDF) is given by eq. (4) and cumulative distribution function (CDF) by eq. (5).

$$f_{(\xi,\mu,\sigma)}(x) = \frac{1}{\sigma}\left(1+\frac{\xi(x-\mu)}{\sigma}\right)^{\left(-\frac{1}{\xi}-1\right)} \quad (4)$$

$$F_{(\xi,\mu,\sigma)}(x) = 1-\left(1+\frac{\xi(x-\mu)}{\sigma}\right)^{-\frac{1}{\xi}} \quad (5)$$

Where σ = std. deviation, ξ = shape parameter, μ = location parameter (minimal value of random variable with Pareto distribution). μ is an offset of Pareto distribution from zero on time axis and represents minimal delay $T_{a\ min}$ (Fig. 2). The shape parameter must meet condition $\xi < 0$ and to get valid results from eq. (4) and (5) $\mu \le x \le \mu - \sigma/\xi$.

3 Proposed E-Model Modification to Impairment Factor

Based on simulation results and measurements, the optimal shape parameter ξ giving the smallest overall MSE error of differences between measured and estimated P_{loss} by equations (6) and (7), is published in our previous work [3].

P_{loss} denotes the probability of a packet arriving with greater delay than is the jitter buffer size. The delayed arrival does not immediately mean that the packet is lost. The buffer can start re-buffering and start a playback with a delay correction during the silent period of conversation, when the sequence of delayed packet is longer. The final effect is then just a short-term increased average two-way network delay.

$$P_{loss}(x,\xi,\mu,\sigma) = \int_{Tbuff}^{\infty}\frac{1}{\sigma}\left(1+\frac{\xi(x-\mu)}{\sigma}\right)^{\left(-\frac{1}{\xi}-1\right)}dx = 1-\int_{0}^{Tbuff} PDFdx \quad (6)$$

$$P_{loss} = \left(1+\frac{\xi(x-\mu)}{\sigma}\right)^{-\frac{1}{\xi}} \quad (7)$$

Where σ = scale, ξ = shape and μ = location parameter (min. value of random variable with Pareto distribution). μ is an offset of Pareto curve from zero on time axis and represents minimal network delay $T_{a\text{-}min}$ (Fig. 1.) and $x = T_{buff}$ is an actual size of jitter buffer in milliseconds.

Actual buffer loss of a packet occurs, when the two consequent packets are delayed and only a single such delay occurs in a short-term period. Then the probability of a packet lost on a buffer, P_{loss_buffer} is in relation of correlation of delays of the consecutive packets as shown in Fig. 3.

Optimal value of sought shape parameter ξ was proved to be between values – 0.1 and – 0.2 depending on actual network traffic characteristics giving good results across wide range of LAN IP networks. Our experiments and consequent analysis show, that the value of -0.1 is acceptable.

Experimentally, we have verified, that there is a possibility to find and describe actual packet loss on jitter buffer, regardless on the burstiness (could be measured by Hurst parameter) of the input packet stream, by upper and lower bound for loss P_{loss_buffer}.

Equation (8) represents lower bound of packet loss P_{LOWER_BOUND} when the autocorrelation of subsequently delivered packets' delay is highest (thus the function squared). This function after substitution, ξ = – 0.1 and μ = 0 according to our previous work [1], [2] and [3], where x = buffer size in [ms], becomes a compound function. To obtain correct results, a following condition must be obeyed: If $x \le 10\sigma$, then eq. (8) is valid; else $P_{UPPER\_BOUND} = 0$;

Equation (9) represents upper bound of packet loss P_{LOWER_BOUND} when the autocorrelation of subsequently delivered packets' delay is lowest (thus the function is not squared). This function after substitution, ξ = – 0.1 and μ = 0 according to our previous work [1], [2] and [3], where x = buffer size in [ms], becomes a compound function. To obtain correct results, a following condition must be obeyed: If $x \le 10\sigma$, then eq. (9) is valid; else $P_{LOWER\_BOUND} = 0$;

$$P_{LOWER\_BOUND}(x,\xi,\mu,\sigma) = \left[\left(1+\frac{\xi(x-\mu)}{\sigma}\right)^{-\frac{1}{\xi}}\right]^2 \cdot \frac{1}{2} \quad (8)$$

$$P_{UPPER\_BOUND}(x,\xi,\mu,\sigma) = \left[\left(1+\frac{\xi(x-\mu)}{\sigma}\right)^{-\frac{1}{\xi}}\right] \cdot \frac{1}{2} \quad (9)$$

Data from measurements of real packet loss on jitter

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buffer and respective lower and upper bounds are present in tabular form in the figure 3 for one selected data row of 21,121 ms jitter.

Jitter buffer size Tbuff [ms] = x	F(x)	1-F(x)	F(x)/2	Lower bound '(1-F(x))^2 / 2	Upper bound '(1-F(x)) / 2	HW measured loss
0	0,000000	1,000000	0,000000	0,500000	0,500000	N/A
10	0,386087	0,613913	0,193043	0,188445	0,306957	N/A
20	0,632427	0,367573	0,316214	0,067555	0,183786	0,140333
30	0,785942	0,214058	0,392971	0,022910	0,107029	N/A
40	0,879136	0,120864	0,439568	0,007304	0,060432	0,022667
50	0,934082	0,065918	0,467041	0,002173	0,032959	N/A
60	0,965428	0,034572	0,482714	0,000598	0,017286	0,003000
70	0,982658	0,017342	0,491329	0,000150	0,008671	N/A
80	0,991735	0,008265	0,495868	0,000034	0,004132	0,000333
90	0,996288	0,003712	0,498144	0,000007	0,001856	N/A
100	0,998445	0,001555	0,499222	0,000001	0,000778	0,000000
110	0,999400	0,000600	0,499700	0,000000	0,000300	N/A
120	0,999791	0,000209	0,499895	0,000000	0,000105	0,000000
130	0,999936	0,000064	0,499968	0,000000	0,000032	N/A
140	0,999983	0,000017	0,499992	0,000000	0,000008	0,000000
150	0,999996	0,000004	0,499998	0,000000	0,000002	N/A
160	0,999999	0,000001	0,500000	0,000000	0,000000	0,000000

Fig. 3. Measured packet loss vs. calculated upper and lower bounds for 21,121 ms jitter and varying buffer size

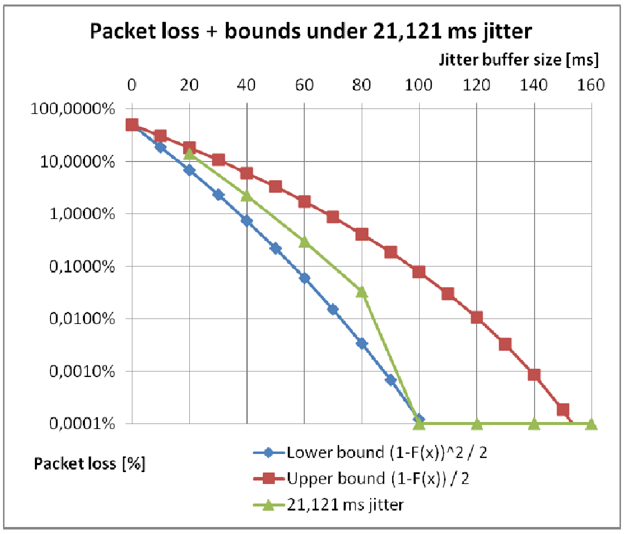

Fig. 4. Measured packet loss vs. calculated upper and lower bounds for 21,121 ms HW jitter and varying buffer size in a lin-log graph showing waterfall-like loss curves up to the expected measurement accuracy

4 Results

As has already been shown in our previous work [1, 2, 3] and several studies in the field of Internet and IP traffic [6, 7, 8, 9, 11] the distribution of packet arrival and interarrival times is long-tailed with long-range dependency (LRD). When considering suitable function for E-model improvement to simulate PESQ results of MOS, it is proficient to simplify the function (10) and find a descriptive function with parameters between upper and lower bounds as stated previously.

Based on local time invariance and presumptions in section A, supported by the results in [2, 3], we consider distribution functions of interarrival times of two consecutive packets to be in the ratio of 1:1 hence eq. (10) can be rewritten to (11).

$$P_{loss\_wo}(x,\xi,\mu,\sigma) = \int_{Tpacket}^{\infty} \frac{1}{\sigma}\left(1+\frac{\xi(x-\mu)}{\sigma}\right)^{\left(-\frac{1}{\xi}-1\right)} dx \qquad (10)$$

$$= 1 - \int_{0}^{Tbuff} PDF dx$$

$$P_{loss\_wo}(x,\xi,\mu,\sigma) = \left[\left(1+\frac{\xi(x-\mu)}{\sigma}\right)^{-\frac{1}{\xi}}\right] \cdot \frac{1}{2} \qquad (11)$$

Figure 5 depicts preliminary measurements performed on IxiaXM2 hardware voice quality testbed. We can see the comparison of MOS estimates as given by an E-model relying on RTP data packets, PESQ intrusive model and E-model calculated by independent IxChariot software probe knowing only the network characteristics.

The aim of proposed method is to improve E-model estimate via inclusion of jitter effects without the need of time- and resource- consuming PESQ model under real network conditions with non-zero jitter.

Pareto Sigma Parameter (Traffic)	RFC 3550 jitter (calculation)	RFC 3550 jitter (HW)	One-Way Delay Average (ms) (software)	RFC 1889 Jitter Average (ms)	MOS - PESQ mix (HW)	MOS - RTP E-model (HW)	MOS - E-model (Software)
1	1,186	0,746	2	0,367	4,459	4,410	4,37
5	5,93	4,530	5	4,433	4,300	4,400	4,37
10	11,86	8,572	9	6,571	3,614	3,940	4,33
15	17,79	11,204	17	8,269	3,065	3,052	4,17
20	23,72	14,065	28	9,917	2,558	2,997	3,63
25	29,65	17,087	28	11,696	2,236	2,921	3,44
30	35,58	20,109	30	12,136	1,843	2,872	2,83
35	41,51	21,121	28	12,714	1,779	2,754	2,67
40	47,44	23,570	28	13,5	1,554	2,306	2,2
45	53,37	25,378	37	15,632	1,311	1,983	2,19
50	59,3	28,153	41	14,684	1,300	1,361	2,08
55	65,23	29,311	45	16,667	1,266	1,276	1,98
60	71,16	30,542	51	16,882	1,169	1,237	1,31
65	77,09	32,374	50	18,059	1,169	1,221	1,41
70	83,02	34,749	60	19,188	n/a	1,200	1,26
75	88,95	36,495	69	18,875	n/a	1,180	1,05
80	94,88	37,739	64	21,2	n/a	1,160	1,07

Fig. 5. Preliminary results of MOS given by E-model compared to PESQ estimates

5 Conclusion

Proposed change in equipment impairment factor calculation leads to improved MOS estimate of E-model when network jitter is present. Proposed method is useful for MOS prediction under real network conditions with jitter. Discovered dependence of buffer packet loss at different jitter strengths for different buffer sizes is results in different MOS estimates for E-model and PESQ methods. Proposed equations and modifications to

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E-model add simply an improved estimate of MOS based on real PESQ results for different jitter, jitter-buffer size and codecs.

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References:
[1] A. KOVAC, M. HALAS, M., Analysis of Influence of Network Performance Parameters on VoIP Call Quality. In Proc. *Knowledge in Telecommunication Technologies and Optics 2010*, Ostrava, pp 26-30.
[2] A. KOVAC, M. HALAS, M., ORGON, M., VOZNAK, E-model MOS Estimate Improvement through Jitter Buffer Packet Loss Modelling, In *Journal Advances in Electrical and Electronic Engineering*, Volume 9, Number 5, December 2011, pp. 233-242.
[3] A.KOVAC, M. HALAS, M. ORGON, Determining Buffer Behaviour under Different Traffic Conditions as MMPP/D/1/K System. In Proc. *Research in Telecommunications Technology 2011*, Techov, September 2011, pp. 143-147.
[4] ITU-T G.107, The E-model, a computational model for use in transmission planning, ITU-T Recommendation G.107, ITU-T. Geneva, Switzerland: April 2009.
[5] ITU-T G.133, Appendix I: Provisional planning values for the equipment impairment factor Ie and packet-loss robustness factor Bpl, ITU-T Recommendation G.133, ITU-T. Geneva, Switzerland: September 1999.
[6] Y. KOH, K. KISEON, Loss probability behavior of Pareto/M/1/K queue. In: *Commun Letters*, *IEEE*, vol.7, no.1, p. 39-41, January 2003.
[7] S. KASAHARA, Internet traffic modeling: a Markovian approach to self-similar traffic and prediction of loss probability for finite queues. *IEICE Transactions on Communications: Special Issue on Internet Technology*. vE84-B i8. 2134-2141.
[8] S. MIRTCHEV, R. GOLEVA, *Evaluation of Pareto/D/1/K Queue by Simulation.* International Book Series "Information Science and Computing". Technical University of Sofia. Sofia, Bulgaria: June 2008.
[9] M. CROVELLA, A. BESTAVROS, Self-Similarity in World Wide Web Traffic: Evidence and Possible Causes. IEEE/ACM Transactions on Networking, vol. 5, No. 6, December 1997
[10] S. KLUCIK, A.TISOVSKY, Queuing Systems in Multimedia Networks. In Journal *Elektrorevue*. Vol 15, November 2010 (2010), art. no 99.
[11] W. ZHANG, J. HE, Statistical Modeling and Correlation Analysis of End-to-End Delay in Wide Area Networks. In Proc. *Eighth ACIS International Conference on Software Engineering, Artificial Intelligence, Networking, and Parallel/Distributed Computing*: IEEE, 2007.
[12] H. TORAL, D, TORRES, C. HERNANDEZ, L. ESTRADA. Self-Similarity, Packet Loss, Jitter, and Packet Size: Empirical Relationships for VoIP. In Proc. *Electronics, Communications and Computers,* 2008. CONIELECOMP 2008, 18th Int. Conference, pp.11-16.
[13] W. HUERLIMANN, From the General Affine Transform Family to a Pareto Type IV Model. In *Journal of Probability and Statistics,* Volume 2009, Article ID 364901, Hindawi Publishing Corporation 2009.