E-MODEL MOS ESTIMATE PRECISION IMPROVEMENT AND MODELLING OF JITTER EFFECTS

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Abstract. *This paper deals with the ITU-T E-model, which is used for non-intrusive MOS VoIP call quality estimation on IP networks. The pros of E-model are computational simplicity and usability on real-time traffic. The cons, as shown in our previous work, are the inability of E-model to reflect effects of network jitter present on real traffic flows and jitter-buffer behavior on end user devices. These effects are visible mostly on traffic over WAN, internet and radio networks and cause the E-model MOS call quality estimate to be noticeably too optimistic. In this paper, we propose a modification to E-model using previously proposed Pplef (effective packet loss) using jitter and jitter-buffer model based on Pareto/D/1/K system. We subsequently perform optimization of newly added parameters reflecting jitter effects into E-model by using PESQ intrusive measurement method as a reference for selected audio codecs. Function fitting and parameter optimization is performed under varying delay, packet loss, jitter and different jitter-buffer sizes for both, correlated and uncorrelated long-tailed network traffic.*

Keywords

Call quality, E-model, jitter, jitter buffer, MOS, network traffic, packet loss.

1. Introduction

The Internet, VoIP and in general IP traffic are 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 the 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. Our work is applicable to the E-model of version 04/2009 and newer, 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].

Our measurements and simulation showed that the performance and estimate accuracy of E-Model deteriorates unacceptably beyond network jitter (calculated by RFC 1889) over 20 ms for all tested codecs including G.711 with and without PLC, G.723.1 ACELP and MP-MLQ, G.726 and G.729. Figure 1 shows an example of E-Model MOS inaccuracy of VoIP network connection in the 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 from RTCP without physically observing or interfering with packet stream behind jitter buffer.

As we can observe, the actual discrepancy of E-Model estimate, being around 1.00 MOS scale under 40 ms jitter is unacceptable for all purposes. These network conditions are not unreal and are common on WiFi and mobile connections.

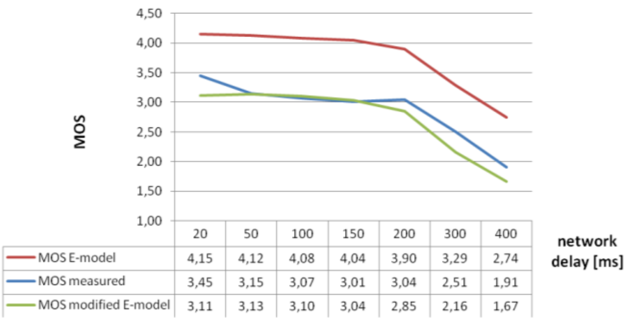

Fig. 1: Comparison of MOS estimates for G.729 codec at 40 ms RFC jitter and 40 ms buffer size, 0 % packet loss under varying network delay.

2. Brief E-model Description

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 jitter 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 the final estimate of quality in an additive manner as expressed in (1):

$$R = R_0 - I_s - I_d - I_{e-eff} + A, \tag{1}$$

where R_0 represents the basic SNR, circuit and room noise, I_s represents all impairments related to voice recording, I_d covers degradations caused by delay of audio signal, $I_{e\text{-}eff}$ 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 an advantage factor of particular technology. We focus at $I_{e\text{-}eff}$ parameter, which is calculated as (2):

$$I_{e-eff} = I_e + (95 - I_e)\frac{P_{pl}}{P_{pl} + B_{pl}}, \tag{2}$$

where I_e represents impairment factor given by codec compression and voice reproduction capabilities, B_{pl} is codec robustness characterizing codec's immunity to random losses.

The values are given for 8 kHz sample rate codecs in ITU-T G.133 appendix [6]. 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.

3. Jitter Buffer Effects on MOS

3.1. Model Implementation Presumptions

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 20 ms 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:

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

The variance of input packet stream can be considered constant for the short time-scale we operate on as induced from the results from [9] and [15]. The Hurst parameter value from short-term point of view in order of seconds is constant and can be put equal to $H = 1$.

3.2. Network Delay Description and Statistics

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 modeled by long-tailed Generalized Pareto distribution (GPD), [8], [9], [10], [12] ,[14]. We use GPD to further describe VoIP input packet stream. Delay distribution of received packets is in Fig. 2.

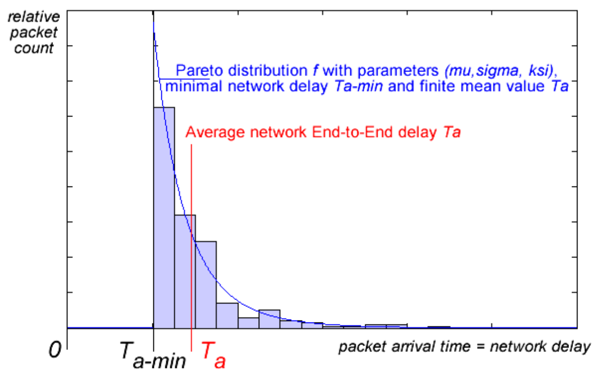

Fig. 2: Distribution of Pareto-related packet arrival times.

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 $T_{a\text{-}min}$ 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):

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

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

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 the time axis and represents minimal delay $T_{a\ min}$ (Fig. 2). The shape parameter must meet the condition $\xi < 0$ and to get valid results from Eq. (4) and (5) $\mu \le x \le \mu - \sigma/\xi$.

4. 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 being 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.

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

$$=1-\int_{0}^{Tbuff}PDFdx\,, \tag{6}$$

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

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 the 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 Tab. 1. Optimal value of sought shape parameter ξ was pro to be between values -0,1 and -0,2 depending on actual network traffic characteristics giving good results across a 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}. These bound can be described by equations (8) and (9) as follows.

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 $\mu = 0$ 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 $\mu = 0$ 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 $P_{LOWER\_BOUND} = 0$.

After substitution of actual values of parameters into Eq. (2), and with reordering capability where x = packet size in [ms]:

$$P_{LOWER\_BOUND}(x,\xi,\mu,\sigma)=$$

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

$$P_{UPPER\_BOUND}(x,\xi,\mu,\sigma)=$$

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

$$P_{loss\_buffer\_LOWER}=\left(1+\frac{-0{,}1\cdot x}{\sigma}\right)^{20}\cdot\frac{1}{2}, \tag{10}$$

$$P_{loss\_buffer\_UPPER}=\left(1+\frac{-0{,}1\cdot x}{\sigma}\right)^{10}\cdot\frac{1}{2}. \tag{11}$$

Data from measurements of real packet loss on jitter buffer and respective lower and upper bounds are present in tabular form in Tab. 1 for one selected data row of 21,121 ms jitter.

Tab.1: Measured packet loss vs. calculated upper and lower bounds for 21,121 ms HW jitter and varying buffer size.

Jitter buffer size [ms]	F(x)	1-F(x)	F(x)/2	Lower bound (1-F(x))^2 / 2	Upper bound (1-F(x)) / 2	Hardware 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

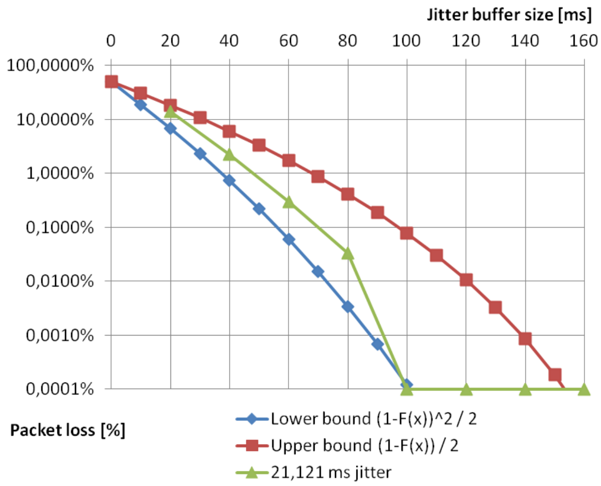

Fig. 3: Measured packet loss vs. calculated upper and lower bounds for 21,121 ms HW jitter and varying buffer size in a lin-log graph showing waterfall-like loss curves up to the expected measurement accuracy.

As has previously been shown in our previous work [1], [2], [3] and several studies in the field of Internet and IP traffic [8], [9], [10], [12], [14] 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 (12) and find a descriptive function with parameters between upper and lower bounds as stated previously.

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

$$=1-\int_{0}^{Tbuff} PDFdx\,. \qquad (12)$$

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. (12) can be rewritten to (13),

$$P_{loss\_wo}(x,\xi,\mu,\sigma)=\left[\left(1+\frac{\xi(x-\mu)}{\sigma}\right)^{-\frac{1}{\xi}}\right]\cdot\frac{1}{2}\,. \qquad (13)$$

Tab.2: Preliminary results of MOS given by E-model compared to PESQ estimates for G.711 codec and 20 ms packetization.

Pareto Sigma (Traffic)	RFC 3550 jitter (calcula-tion) [ms]	RFC 3550 jitter (HW) [ms]	One-Way Delay Average [ms] (software)	RFC 1889 Jitter Average [ms]	MOS - PESQ mix (HW)	MOS - RTP E-model (HW)	MOS - E-model (SW)
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

5. Test of Results

Iterative distribution fitting was performed using various distributions to find the best fit parameters. These parameters and distributions were put under Kolmogorov-Smirnov and Chi-Squared tests to find best descriptive statistics of Pareto-distributed stream time differences with applied jitter. Results of finding best descriptive statistics with optimal iteratively found parameter set with error of 10e-5 are sorted in Tab. 3.

Tab.3: Best fit parameters of tested distributions.

Distribution	Best fit distribution parameters
Generalized Pareto (GPD)	k = 0,19328 σ = 0,0224 μ = -0,00306
Generalized Extreme	k = 0,36239 σ = 0,01384 μ = 0,00909
Weibull	α = 0,39981 β = 0,01718
Gen. Gamma	k = 0,98444 α = 0,40502 β = 0,05293
Log-Pearson 3	α = 6,081 β = -1,175 γ = 1,641
Laplace	λ = 39,109 μ = 0,0247
Weibull (3P)	α = 0,4745 β = 0,01408 μ = 1,3003e-5
Gamma	α = 0,46676 β = 0,05293
Logistic	σ = 0,01994 μ = 0,0247
Lognormal	σ = 2,8971 μ = -5,504

Statistical tests showed as a proof of concept, that GPD Pareto distribution is also the most suitable one for describing interarrival times of general long-tailed LAN/WAN packet streams impaired by random jitter with equal distribution. This shows also Pareto distribution to be the best compromise between calculation complexity (compared to fractal modelling methods) and statistical significance for modelling also jitter buffer loss behaviour under variable jitter.

To explain best fit parameters of GPD from Tab. 3:

- σ in all equations corresponds to optimised σ in Tab. 3. Proposed relation between σ and actual jitter J substituted can be expressed in the ratio $J/\sigma \in$ <1;2>. For actual imposed 40 ms network jitter the optimized parameter was σ = 0,0224 s = 22,4 ms what would field J/σ ratio = 22/14 $\in$ <1;2>. Actual parameter substitution ratio needs further testing.

- ξ = shape parameter corresponds to optimised k. Actual shape parameter for our model was chosen to be $\xi = [-k]$ rounded to one tenth in order to maintain exponent in all equations of integer value for computational effectiveness.

- μ = location parameter corresponds in Tab. 3 to μ = -0,00306. It was chosen as $\mu = 0$ with negligible effect.

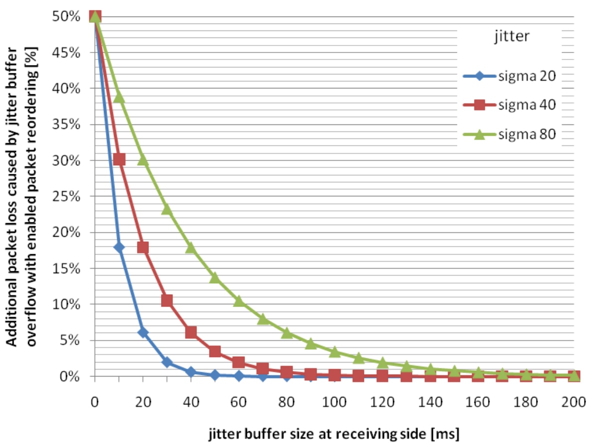

Fig. 4: Jitter buffer packet loss P_{jitter} graph for different jitter.

6. 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 E-model improve voice quality MOS estimate accuracy when network jitter is present. We use a simplified estimate to calculate expected packet loss on jitter buffer of the receiving device, which is superimposed in a multiplicative way to network packet loss. Resulting packet loss is commonly greater when the jitter buffer is not large enough. The model is able to give results, which are more in concordance with expected results of PESQ intrusive method of quality testing.

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