Jitter Buffer Loss Estimate for Effective Equipment Impairment Factor

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***Abstract*—**The paper deals with a proposal of original E-model modification, the new model includes improvements giving more precise results when a jitter is present in transmission channel. Our idea is based on the fact that the network jitter can affect overall delay in delivery or a packet loss due to a limitation of play-out buffer in IP phones or VoIP gateways. Delay is incorporated in an impairment factor I_d of E-model whereas losses are issue of $I_{e\text{-}eff}$ parameter. We examined how the estimated speech quality is affected by jitter and compared computed results with MOS values gained by PESQ objective intrusive method (Perceptual Evaluation of Speech Quality). Experiments were performed in circumstances of varying delay, packet loss, jitter and play-out buffer. We have proved that the proposed modification improves estimated MOS computed by E-model.

***Keywords*—** E-model, MOS, packet loss, jitter, jitter buffer, network traffic, call quality, QoS, function, fitting.

I. 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 [1]. 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].

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II. E-model Description and Its Prediction 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 [3].

Objective methods can be sub-divided into two groups, Intrusive and Non-intrusive. The core of intrusive (also referred to as input-to-output) measurements is the comparison of the original sample and the degraded sample affected by a transmission chain. Contrary to intrusive methods which require both the output (degraded) sample and the original sample, non-intrusive methods do not require the original sample. This is why they are more suitable to be applied in real time. Yet, since the original sample is not included, these methods frequently contain far more complex computation models. Intrusive methods are very precise but their application in real-time measurement is unsuitable because they require sending a calibrated sample and both endpoints of the examined communication.

Nevertheless, we usually need to assess the speech quality in real traffic and be able to record its changes, especially degradation. The important non-intrusive method was standardized in recommendation ITU-T P.562 (INMD) and in ITU-T G.107, so-called E-model. INMD measurement (In-service Non-intrusive Measurement Devices) is applied primarily to measure voice-grade parameters such as speech, noise and echo. The E-model is a computation model which takes into account all the links between transmission parameters and is 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 [1] 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):

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

Where I_e represents impairment factor given by codec compression, B_{pl} is codec robustness characterizing codec's immunity to random losses. Factor B_{pl} is defined as a codec-specific value. The packet-loss dependent effective equipment impairment factor $I_{e\text{-}eff}$ is derived using the codec-specific value for the equipment impairment factor at zero packet-loss I_e and the packet-loss robustness factor B_{pl}, both listed in Appendix I of ITU-T G.113 for several codecs.

The denominator $(P_{pl}+B_{pl})$ in equation (2) can be expressed as $(B_{pl}+B_{pl}/BurstR)$ where *BurstR* is the so-called burst ratio, defined as ratio between "Average length of observed bursts in an arrival sequence" and "Average length of bursts expected for the network under random loss".

Where the packet loss is random, *BurstR*=1 and in case of packet loss burstiness, *BurstR*>1. As packet loss distributions correspond to the Gilbert-Elliot model with transition probabilities p between the found and a loss state, and q between the loss and the found state, the burst ratio can be calculated as $p/(p+q)$. The packet loss P_{pl} is expressed as p/(p+q) and *BurstR* can be calculated as $(1- P_{pl})/q$. As can be seen from equation (5), the effective equipment impairment factor in case of P_{pl} = 0 (no packet-loss) equals *Ie* value defined in ITU-T G.113.

In this paper we propose a substitution of P_{pl} parameter for P_{plef} described further in the paper in more detail.

III. Previous Results and Indications

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.

Figure 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 [9].
- "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 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 [2], [3].

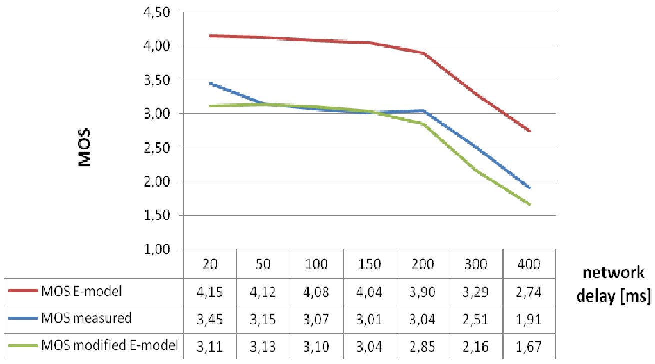

Fig. 1 Comparison of MOS estimates for G.729 codec at 40 ms RFC jitter and 40 ms buffer size, ideal 0 % packet loss under varying network delay

IV. Jitter Buffer Effects on MOS - Call Quality Estimate

A. A Timescale of Interest

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 formula (3) holds true.

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

The variance of input packet stream can be considered constant for the short time-scale we operate on as induced from [4], [7] and [11]. 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 [10]. Internet traffic arrival times and delay can be successfully statistically modeled by long-tailed Generalized Pareto distribution (GPD) [7], [11]. We use GPD to describe VoIP packet stream. Delay distribution of received packets is in Fig. 2.

Real-time change of network parameters causes variations in network delay [5], [6], [8]. 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 T_a , one of the input parameters for E-model, so-called I_{dd} impairment factor. Factor *Idd* represents the impairment caused by too-long absolute delay *Ta* which occurs even with perfect echo cancelling. For Ta ≤ 100 ms we can assume *Idd = 0* because a negligible influence appears in the R-factor but with the delay increasing the overall R-factor is affected.

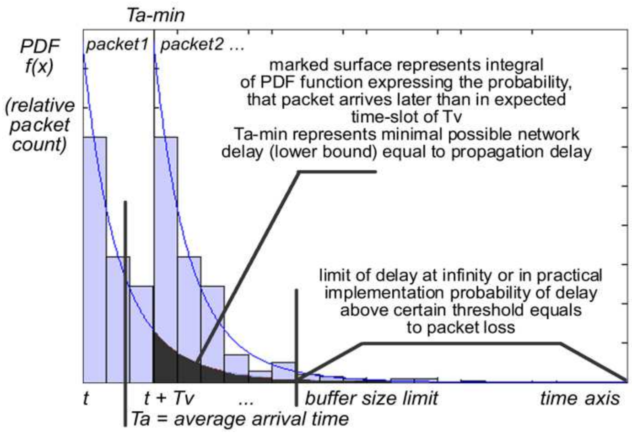

Fig. 2 Distribution of Pareto-related packet arrival times

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 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) $\mu \le x \le \mu - \sigma/\xi$.

V. 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 [2].

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.

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

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

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 time axis and represents minimal network delay T_{a-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 experimentally determined to lie in the interval <-1;1> depending on actual network traffic characteristics giving good results across a wide range of LAN IP networks. We have discovered, 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 and establish an empirical lower and upper bound for loss P_{loss}.

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, $\xi = -0.1$ 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). In our previous work [2], [12] and [14] we have successfully used following set of parameter values for substitution, $\xi = -0.1$ and $\mu = 0$, where x = buffer size in [ms]. To obtain correct results, a following condition had to be obeyed: If $x \le 10\sigma$, then eq. (9) is valid; else $P_{LOWER\_BOUND} = 0$;

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

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

Data from measurements of real packet loss on jitter buffer and respective lower and upper bounds are present in tabular form in Fig. 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

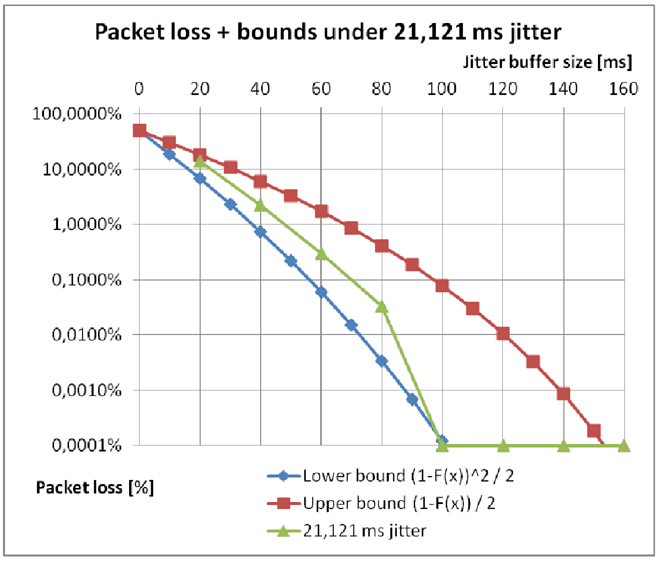

Fig. 4 Measured packet loss vs. calculated upper and lower bounds for 21.121 ms HW jitter and varying buffer size in a lin-log graph showing waterfall-like loss curves up to the expected measurement accuracy

VI. Function Fitting and Goodness of Fit Statistical Testing

In our previous work [12], [14], we have proposed an estimate for jitter buffer losses. We will further elaborate over the possible fitting function estimating the jitter buffer packet loss P_{LOSS_EST} (10):

$$P_{LOSS\_EST}(x,\xi,\mu,\sigma)=\left[\left(1+\frac{\xi(x-\mu)}{\sigma}\right)^{-\frac{1}{\xi}}\right]\cdot\frac{1}{2} \qquad (10)$$

Despite the good fit results over the tested data range, the parameter values (namely the shape parameter ξ = – 0.1) imposes restrictions to the range of values for x and σ, where the function is defined. We have conducted further optimization of eq. (10) followed by statistical tests of goodness of fit using F-statistics, both in 3D curve fitting software by Systat. We sought for optimal values of parameters $\xi = A$ and exponent = B of eq. 10 generalized into eq. 11.

$$P_{LOSS\_EST}(x,A,\mu,\sigma,B)=\frac{1-\left[1-\left(1+\frac{A(x-\mu)}{\sigma}\right)^{-\frac{1}{A}}\right]^{B}}{2} \qquad (11)$$

We imposed following restrictions on parameters and conditions on the sought-for function:

- $\xi = A < 0$; $\mu = 0$;
- $1 \le B \le 5$ (initial value in our previous work was B = 2);
- Non-linear robust fitting method – Pearson VII Limit with minimization function (12);

$$f_{min}(P_{LOSS\_EST})=\sum \ln\sqrt{1+residual^{2}} \qquad (12)$$

- The function must tend towards P_{LOSS_EST} = 0.5 when jitter is significantly greater than buffer, i.e. $x << \sigma$;
- The function must be defined for all buffer sizes $\Rightarrow x > 0$ and jitter values $\Rightarrow \sigma > 0$;

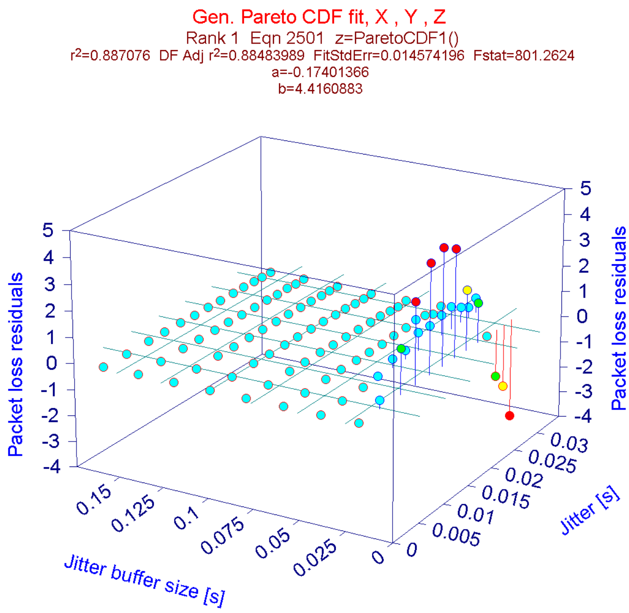

Fig. 5 Residuals of P_{LOSS_EST} function with $\xi < 0$

Generalized Pareto Distribution (4), (5), as a function for peaks and threshold analysis, takes one of the three forms for certain ξ shape parameter values [13]:

- If $\xi < 0$, GPD transforms to Pareto Type II distribution and its domain is defined for $\mu \leq x \leq \mu - \sigma / \xi$;
- If $\xi = 0$, GPD degenerates to Exponential distribution;
- If $\xi > 0$, GPD transforms to Beta distribution of the first kind and its domain is defined for all $x > 0$ and $\sigma > 0$;

For regression testing fitness we assume following hypotheses using F-statistics:

- 95% confidence interval;
- H_0 is a hypothesis stating that there is no relationship between variables of measurements (dependent) and a model function being fitted (independent – predictor). A small F, with a big p-value indicates that there is no relationship;
- H_A is an alternative hypothesis stating that there is a relationship between variables of measurements and a model.

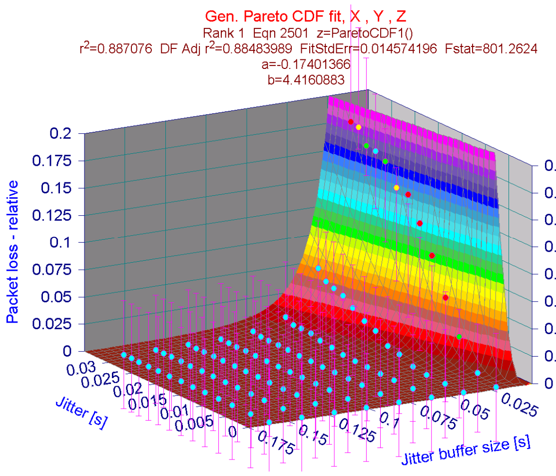

Fig. 6 Optimized P_{LOSS_EST} function with $\xi < 0$

A large value of F, with a small p-value, means that H_0 is discredited, and we would assert that there is a general relationship between the measurements and model function.

Our first optimization run yielded a function in Fig. 5. with parameter values $A = -0.174$, $B = 4.416$ and a coefficient of determination $R^2 = 0.887076$.

As we can see in Fig. 6., the proposed function lacks universality over broader range of input parameters (observe the undefined region of the function for large jitter values against small buffer sizes in Fig. 6.).

Also large errors of jitter buffer loss prediction (too optimistic) are present for small buffers, see Fig. 7 for error function of residuals. F-statistics showed F=801.2624 and $P>F$ of 0.000, what discredits H_0 and asserts H_A hypothesis.

We were however not satisfied with the general parameters of the function and we wanted to obtain better fit with $R^2 > 0.90$. We have therefore investigated the possible range of $\xi > 0$, where the GPD P_{LOSS_EST} function would take on the shape of Beta function and would be defined continuously for all $x > 0$ and $\sigma > 0$.

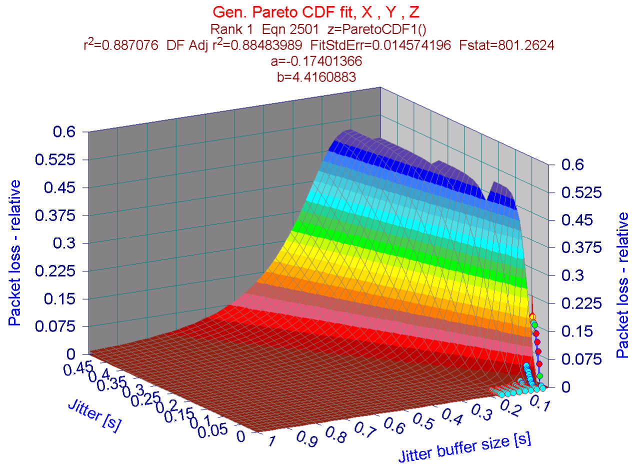

Fig. 7 Larger view on P_{LOSS_EST} function with $\xi < 0$

We imposed following restrictions on parameters and conditions on the sought-for function:

- $5 > \xi = A > 0$; $\mu = 0$;
- $1 \leq B \leq 5$ (initial value in our previous work was $B = 2$);
- Non-linear robust fitting method – Pearson VII Limit with minimization function (12);
- The function must tend towards $P_{LOSS\_EST} = 0.5$ when jitter is significantly greater than buffer, i.e. $x << \sigma$ and must be defined for all $x > 0$ and values $\Rightarrow \sigma > 0$;
- The 0.5 upper limit for loss is the most extreme case, where the jitter buffer reinitializes when after each lost packet it receives one into an empty buffer. The limit case is that this could theoretically happen after each second packet.

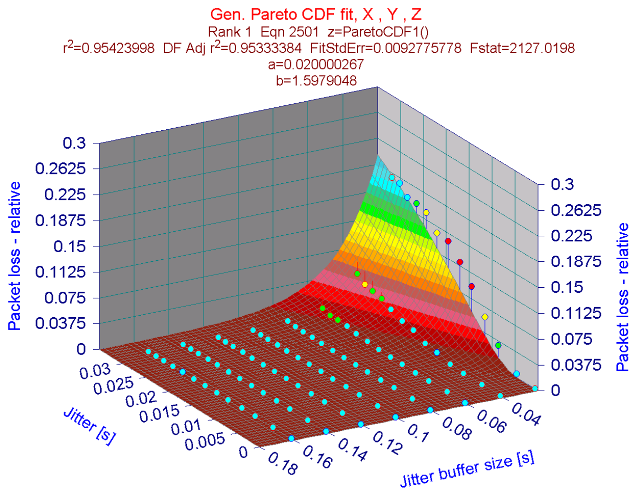

Fig. 8 Optimized P_{LOSS_EST} function with $\xi > 0$

Our next optimization run yielded a function in Fig. 8 with parameter values $A = 0.020$, $B = 1.598$ (to three decimal places of accuracy) and a significantly improved coefficient of determination $R^2 = 0.95423998$. For computational simplicity we decided to round the A and B parameter values to two decimal places, which can be written as a rational numbers as follows A = 0.02 = 1/50, B = 1.60 = 8/5. We will be using these rounded values onwards.

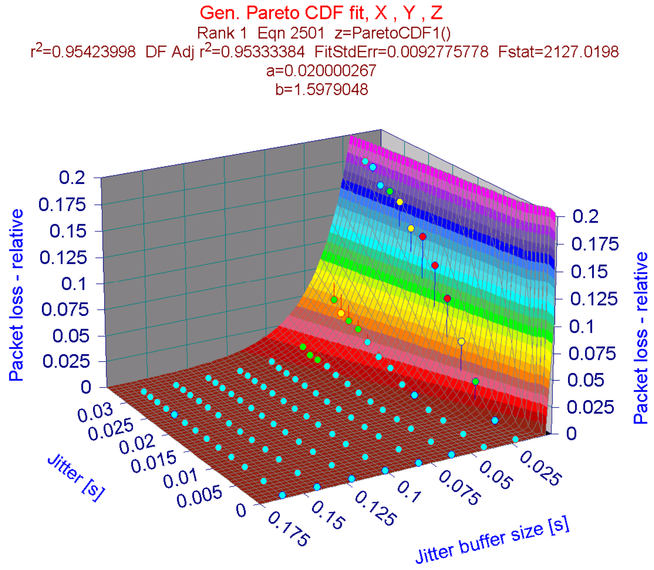

Fig. 9 Larger view on P_{LOSS_EST} function with $\xi > 0$

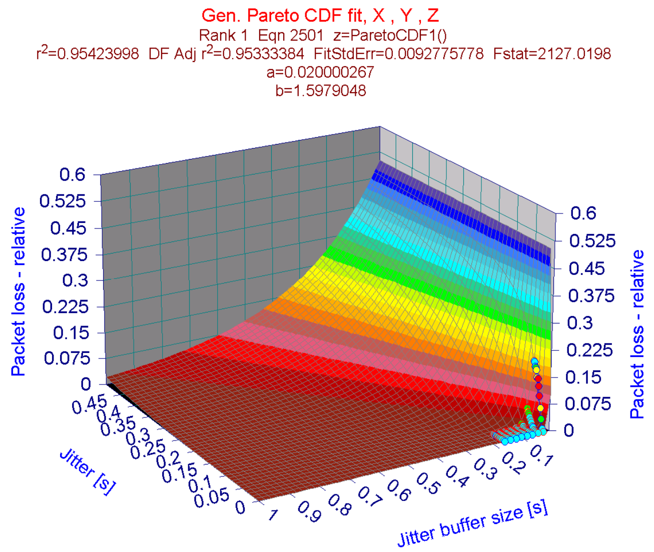

Fig. 10 Larger view on P_{LOSS_EST} function with $\xi > 0$

As we can see in the Fig. 9 and Fig. 10, the proposed function is smooth, continuous and offers the universality of estimate over broader range of input parameters as opposed to function P_{LOSS_EST} function with $\xi < 0$. The absolute errors of jitter buffer loss prediction were lowered and are now a little too pessimistic, overestimating the packet loss in the most extreme conditions about around 3% (only in the region of jitter buffer smaller than 40 ms, which is practically useless). The error function of residuals is depicted in Fig. 10.

F-statistics showed even improved F=2127.0198 and $P>F$ of 0.000, what discredits H_0 and asserts H_A hypothesis. Therefore, we prefer the new set of parameters for further modeling of jitter buffer losses under network jitter conditions over broader range of parameters.

Based on our new experiments and fitting of P_{LOSS_EST} function with $\xi > 0$, we discovered several caveats with upper (8) and lower (9) bounds empirically set for P_{LOSS} (7) in our previous work [12], [14]. Upper bound does not fit all cases of measured loss on jitter buffer, mainly in extreme cases, where the buffer is less than 40 ms. Lower bound still loses accuracy when the loss approaches zero (P_{LOSS} below 0.01%) due to measurement precision and error (0.01% loss means one lost packet from 10000, or in other words one lost packet during 200 seconds of phone call assuming codec packet generation rate of 50 pps under standard 20 ms packetization.

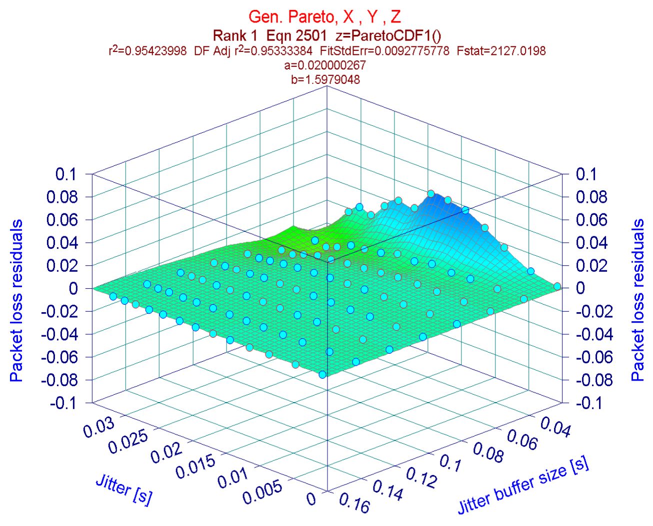

Fig. 11 Absolute residuals of P_{LOSS_EST} function with $\xi > 0$

Therefore we propose a new set of lower $P_{LOWBOUND}$ eq. (14) and upper $P_{LOWBOUND}$ eq. (15) bounds, which satisfy the experiments over a wider range of values up-to the precision limit of 0.01%.

$$P_{LOWBOUND}(x,A,\sigma)=\frac{1-\left[1-\left(1+\frac{A(x)}{\sigma}\right)^{-\frac{1}{A}}\right]^{3}}{2} \tag{14}$$

$$P_{UPBOUND}(x,A,\sigma)=\frac{1-\left[1-\left(1+\frac{A(x)}{\sigma}\right)^{-\frac{1}{A}}\right]^{1}}{2} \tag{15}$$

Upper bound then simplifies to eq. (16):

$$P_{UPBOUND}(x,A,\sigma)=\frac{\left(1+\frac{A(x)}{\sigma}\right)^{-\frac{1}{A}}}{2} \tag{16}$$

VII. Results

As has already been shown in our previous work [2], [12], [14] and several studies in the field of Internet and IP traffic [4], [7] and [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) with the found optimal and computationally simple integer values of parameters (which were our preference over real values). The descriptive function (17) will after substitution of found optimal parameters substituted into equation (11), a following form:

$$P_{LOSS\_EST}(x,\sigma) = \frac{\left[\left(1+\frac{x}{50\sigma}\right)^{-50}\right]^{8/5}}{2} \tag{17}$$

Where σ (scale) = measured network jitter in seconds and $x = T_{buff}$ = size of a buffer in seconds.

Proposed change to the E-model focuses at $I_{e,eff}$ parameter, which is calculated as in eq. (18):

$$I_{e,eff} = I_e + (95 - I_e).\frac{P_{pl}}{P_{pl} + B_{pl}} \tag{18}$$

I_e represents impairment factor given by codec compression and voice reproduction capabilities, B_{pl} is codec robustness describing how immune is particular codec against random losses and what are its PLC masking qualities. In order to take jitter buffer losses on receiver's side into account, there is a need to multiply packet transmittances of network and jitter buffer into P_{plef}. Packet transmittance in this sense is a complement to packet loss either on network or jitter buffer to 1 (or 100%). Eq. (19) expresses how to calculate effective packet loss:

$$P_{plef} = P_{pl} + P_{LOSS\_EST} - P_{pl}.P_{LOSS\_EST} \tag{19}$$

Where P_{pl} is a standard measured network packet loss and P_{LOSS_EST} is an estimated loss on receiver's buffer under non-zero network jitter.

$$\begin{aligned} I_{e,eff} &= I_e + (95 - I_e).\frac{P_{plef}}{P_{plef} + B_{pl}} = \\ &= I_e + (95 - I_e).\frac{P_{pl} + P_{LOSS\_EST} - P_{pl}.P_{LOSS\_EST}}{P_{pl} + P_{LOSS\_EST} - P_{pl}.P_{LOSS\_EST} + B_{pl}} = \\ &= I_e + (95 - I_e). \\ &.\frac{P_{pl} + \frac{\left[\left(1+\frac{x}{50\sigma}\right)^{-50}\right]^{8/5}}{2} - P_{pl}.\frac{\left[\left(1+\frac{x}{50\sigma}\right)^{-50}\right]^{8/5}}{2}}{P_{pl} + \frac{\left[\left(1+\frac{x}{50\sigma}\right)^{-50}\right]^{8/5}}{2} - P_{pl}.\frac{\left[\left(1+\frac{x}{50\sigma}\right)^{-50}\right]^{8/5}}{2} + B_{pl}} \end{aligned} \tag{20}$$

The relation (20) is the final proposed equation for modified effective equipment impairment factor $I_{e\text{-}eff}$ calculation in E-model and incorporating jitter buffer loss and jitter buffer size through effective packet loss P_{plef}.

VIII. Conclusion

Figure 12 depicts preliminary measurements performed on voice quality test-bed. The experiment was based on IxiaXM2 tester with IxLoad SW and VQM (Voice Quality Module). The IxLoad is a tool for performance testing of VoIP devices emulating various types of traffic in IP network including simultaneously established VoIP calls. Ixia's VQM module provides real-time evaluation of end-user quality of experience (QoE) and we evaluated the quality by an objective PESQ method in accordance to ITU-T P.862. The PESQ algorithm requires high processing power to access audio signals and perform in depth signal analyzes. The VQM analyzes RTP audio streams and communicates with the application load module through the chassis backplane of IxiaXM2 tester, the module is able to measure PESQ up to 300 simultaneous RTP streams.

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

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

Proposed changes of estimated parameters of P_{LOSS_EST} function provide better data fitting over observations and offer more accurate prediction of jitter buffer loss over a broad range of input values (buffer sizes and jitter).

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