E-MODEL MOS ESTIMATE IMPROVEMENT THROUGH JITTER BUFFER PACKET LOSS MODELLING

Adrian KOVAC[1], *Michal HALAS*[1], *Milos ORGON*[1], *Miroslav VOZNAK*[2]

[1]Department of Telecommunications, Faculty of Electrical Engineering and Information Technology, Slovak University of Technology, Ilkovicova 3, 812 19 Bratislava, Slovakia
[2]Department of Telecommunications, Faculty of Electrical Engineering and Computer Science, VSB-Technical University of Ostrava, 17.listopadu 15, 803 00 Ostrava, Czech Republic

fotoadrian@gmail.com, halas@ut.fei.stuba.sk, orgon@ut.fei.stuba.sk, miroslav.voznak@vsb.cz

Abstract. *Proposed article analyses dependence of MOS as a voice call quality (QoS) measure estimated through ITU-T E-model under real network conditions with jitter. In this paper, a method of jitter effect is proposed. Jitter as voice packet time uncertainty appears as increased packet loss caused by jitter memory buffer under- or overflow. Jitter buffer behaviour at receiver's side is modelled as Pareto/D/1/K system with Pareto-distributed packet interarrival times and its performance is experimentally evaluated by using statistic tools. Jitter buffer stochastic model is then incorporated into E-model in an additive manner accounting for network jitter effects via excess packet loss complementing measured network packet loss. Proposed modification of E-model input parameter adds two degrees of freedom in modelling: network jitter and jitter buffer size.*

Keywords

VoIP, E-model, MOS, QoS, packet loss, jitter buffer, loss estimate, Generalized Pareto, packet interarrival, Pareto/D/1/K.

1. Introduction

Mean opinion score - MOS - represents user satisfaction with overall listening and conversational quality on five grade scale from 5 (best) to 1 (worst). MOS can be estimated by either subjective methods based on user listening tests or by objective methods based on measured network parameters such as delay, packet loss and jitter. E-model defined by ITU-T G.107 [1] is widely accepted objective method used for estimation of VoIP call quality. E-model uses set of selected input parameters to calculate intermediate variable – R factor, which is consecutively converted to MOS value using (3). Input parameters contribute to final estimate of quality in additive manner as expressed in Eq. (1).

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

where R_o represents the basic SNR, circuit and room noise; I_s represents all impairments related to voice recording such as quantization distortion, low voice volume; I_d covers degradations caused by delay of audio signal including side-tone echo; $I_{e,eff}$ 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.

Tab.1: Impairment factors for selected codecs [5].

Audio Codec	Codec bitrate	Impairment Factor	
		I_e	B_{pl}
G.711 w/o PLC	64 kb/s	0	10
G.711 with PLC	64 kb/s	0	34
G.723.1	5,3 kb/s	19	24
G.723.1	6,3 kb/s	15	20
G.726	16 kb/s	40	69
G.726	24 kb/s	25	38
G.726	32 kb/s	12	24
G.726	40 kb/s	7	24
G.728	16 kb/s	16	27
G.729	8 kb/s	10	18
G.729 A	8 kb/s	11	17
GSM FR 6.10	13,2 kb/s	26	43

We focus at $I_{e,eff}$ parameter, which is calculated as in Eq. (2):

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

where I_e represents impairment factor given by codec compression and voice reproduction capabilities, B_{pl} is codec robustness, which describes how immune is particular codec against random losses and what are its PLC masking qualities. These values are given for narrowband codecs in ITU-T G.133 appendix [5]. Table 1 gives a summary of I_e and B_{pl} parameters for selected codecs. P_{pl} parameter represents measured network packet loss in %.

2. Relations between Network Delay, Jitter and Packet Loss

An easiest way to comply with the conference paper formatting requirements is to use this document as a template and simply type your text into it.

2.1. Properties of Network Delay

Voice packets being sent from VoIP device – IP phone – can be considered as a regular flow with constant transmit intervals and transmit duration. VoIP packets after transport network traversal have their regular time spacing disrupted in irregular way. Internet traffic arrival times and delay can be successfully statistically modelled by long-tailed Generalised Pareto distribution (GPD) [7], [9], [10] which we use to describe incoming VoIP packet stream from transport network to IP phone and refer to it like "Pareto distribution" further in this paper. Network delay distribution of received packets is depicted in Fig. 1.

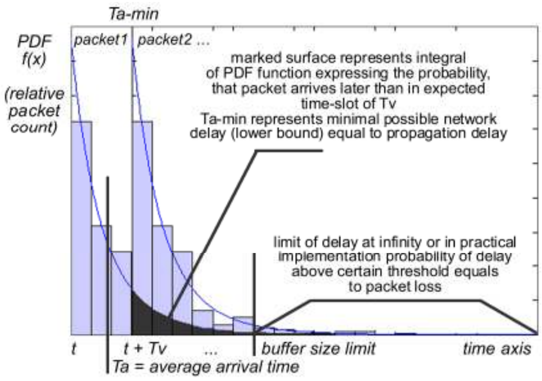

Fig. 1: Packet transmission delay distribution.

Real-time change of network operational and performance parameters cause variations in delay. Differences between consecutive packet arrivals are not constant and arrival times oscillate between minimal possible network delay *Ta-min* and infinite delivery time (lost packet). Mean value of the described process exists and is called and measured as an End-to-End network delay *Ta* (one of the input parameters for E-model).

2.2. Network Delay Statistics

Real packet path usually consists of a mixture of different networks with different devices and technologies. Each device adds certain degree of uncertainty in packet processing and transmitting time. Overall network delay statistics is a result of all partial statistics at each device. Aforementioned Pareto distribution is well suited to describe time as a variable, which has lower bound, no upper bound and finite mean value. It is better suited to describe internet packet flow than Weibull distribution as it yields lower MSE in practical measurements of real traffic [7], [9], and [10]. Pareto has different shape than exponential distribution with steeper slope and less occurrences in higher values. Probability density function of Pareto (PDF) is given by Eq. (3) and cumulative distribution function (CDF) by Eq. (4).

$$f_{(\xi,\mu,\sigma)}(x) = \frac{1}{\sigma}\left(1+\frac{\xi(x-\mu)}{\sigma}\right)^{\left(-\frac{1}{\xi}-1\right)}, \tag{3}$$

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

where σ = std. deviation, ξ = shape parameter, μ = location parameter (minimal value of random variable with Pareto distribution). μ represents offset of Pareto distribution from zero on time axis and represents minimal network delay *Ta_min* also illustrated at Fig. 1. Shape of Pareto PDF for varying σ is depicted at Fig. 2. For Pareto PDF to converge faster than exponential, shape parameter must meet condition $\xi < 0$ and to get valid results from Eq. (3) and (4) $\mu \leq x \leq \mu - \sigma/\xi$.

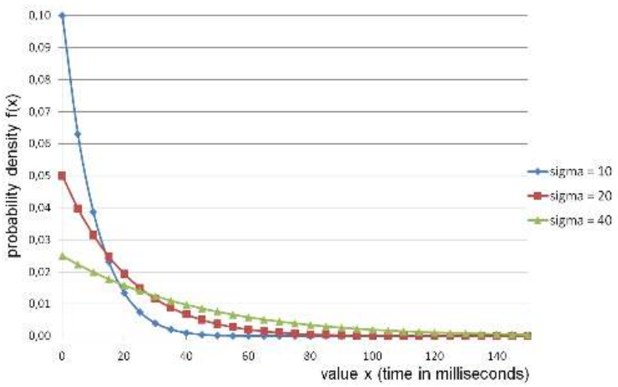

Fig. 2: Packet transmission delay distribution.

2.3. Network Jitter

Jitter J [ms] is calculated in real-time as floating average of differences between interarrival times called "Timestamps" of consecutively received packets contained in RTP protocol header. Calculation of J is given by Eq. (5). Each particular difference is given by

Eq. (6), both according to RFC 1889, where R are timestamps when packet was received, S when packet was sent and indices i,j are consecutive packet numbers. Jitter value is transferred in RTCP protocol header as one of the QoS parameters. For correct measures of jitter correct synchronization of clocks in network is needed.

$$J = J + \left(\left|D_{(i-1,i)}\right| - J\right)/16[ms], \quad (5)$$

$$D_{(i,j)} = \left(R_j - R_i\right) - \left(S_j - S_i\right)[ms], \quad (6)$$

During VoIP communication, voice packets are generated in regular intervals with fixed period Tv assuming that no VAD is active and packetization interval is constant. Deviations of placing packets in correct timeslots on transport medium can occur when other traffic is present, e.g. when sending files through FTP and simultaneously making VoIP calls, traffic has to be interleaved. These issues are further amplified at active layer 2 and 3 network elements when routing, switching and traffic shaping occurs. Receiver expects packets at regular timeslots but to compensate disturbances in delivery time receiver has to use buffer memory called "jitter buffer" at the expense of increasing overall end-to-end delay at least by length of buffer in milliseconds / 2. Result of the process at receiver's side is an irregular packet stream which can be described through Markov process as Pareto/D/1/K queue [7], [9] depicted in Fig. 3.

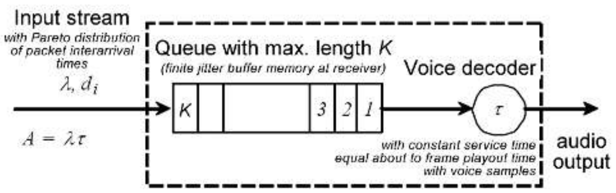

Fig. 3: Markov chain model of VoIP packet receiver with buffer.

3. Analysis of Jitter Effects

3.1. Effective Packet Loss

Packet Loss in packet data network can occur either in transport network or at user device. Transport network can suffer from congestion, load imbalance and also mixture of several transport technologies, which behave differently.

Jitter itself is not one of the input parameters for E-model calculation, but its influence is not negligible under real network conditions. Effects of jitter on packet loss appear as increased net total packet loss which is equal or greater than measured network packet loss expressed as P_{pl}. In this document we refer to "Modified E-model" as to model based on original ITU-T G107 [2] recommendation with P_{pl} packet loss parameter in Eq. (2) substituted by proposed P_{plef}. Codec robustness factor B_{pl} in Eq. (2) expresses resilience of certain codec to random and burst packet loss and its masking capabilities through packet loss concealment (PLC) algorithm. Overall effective packet loss P_{plef} can be expressed as a complement to product of packet transmittance in network and in receiver jitter buffer as given by Eq. (7).

$$P_{plef} = 1 - \left(1 - P_{pl}\right) \cdot \left(1 - P_{jitter}\right) \quad \in \langle 0;1 \rangle, \quad (7)$$

that can be rewritten to Eq. (8) as a jitter buffer loss:

$$P_{jitter} = \left(P_{plef} - P_{pl}\right) / \left(1 - P_{pl}\right) \quad \in \langle 0;1 \rangle, \quad (8)$$

where P_{pl} is network packet loss as measured by receiving device and P_{jitter} is an additional packet loss occurring at finite jitter buffer memory when compensating time fluctuations of arriving voice packets. Packet transmittance is a complement to packet loss and is expressed by terms in brackets in Eq. (7).

3.2. Jitter Buffer Behaviour

We have used Pareto/D/1/K queue [7], [9] depicted in Fig. 3 to model jitter buffer behaviour. Markov chain Pareto/D/1/K behaves like M/D/1/K because memory-lessness of previous states of input stream is ensured. Zero correlation of Poisson input stream is maintained with Pareto distribution.

Figure 4 shows sample timelines of transmitter and receiver of VoIP packets, expected arrival timeslots as well as deviations from expected delivery time, for which there is a jitter buffer. Jitter buffer has threshold levels of how much ms of audio samples it has to receive before starting playback of stream, which is usually 50 % of its length and it equals to additional end-to-end delay added by buffer. If receiving buffer behaves like ordinary queue, packets that are severely influenced by jitter and delayed so much that they arrive later than packets sent after them, such exchange in packet order cannot be treated and late packet is treated as lost even when its timeslot has not passed (Fig. 4). Real jitter buffer can repair or reorder swapped packets in input stream by their RTP sequence number when their timeslot or time to play has not passed. This process significantly lowers additional packet loss in networks with high jitter comparable with packet length (jitter usually 30 ms and higher).

For calculation of P_{jitter} at jitter buffer with reordering capability in Eq. (8) we have to recalculate P_{jitter} in raw Markov chain-type buffer without reordering capability represented by PDF Pareto function $F_{wo}(x, \mu, \xi, \sigma)$ (transmittance of buffer without reordering capability) to F_{wr} (transmittance of buffer with packet reordering capability) according to Eq. (9). Formula (9) is based on autocorrelation function of packet loss P_{jitter} without reordering which equals to probability, that packet will not be in jitter buffer present at the moment of processing in its expected timeslot. Real measured values of F_{wo}, F_{wr} as well as their complements – jitter buffer packet losses P_{wo} and P_{wr} are present in Tab. 2 and 3. These values were used to find the best fitting shape parameter with minimal MSE error of packet losses on buffer with reordering ability.

$$P_{jitter} = F_{wr} = \frac{\left(1 - F_{wo}\left(x, \mu, \xi, \sigma\right)\right)^2}{2} \in \left\langle 0;0{,}5 \right\rangle . \quad (9)$$

In Eq. (9) P_{jitter} reaches the maximum of 0,5 which shows an extreme, that every packet is mixed. From every pair one packet would be discarded and additional packet loss on jitter buffer reaches theoretical maximum of 50 % (transmittance factor 50 % equal to behavior of binary symmetric channel).

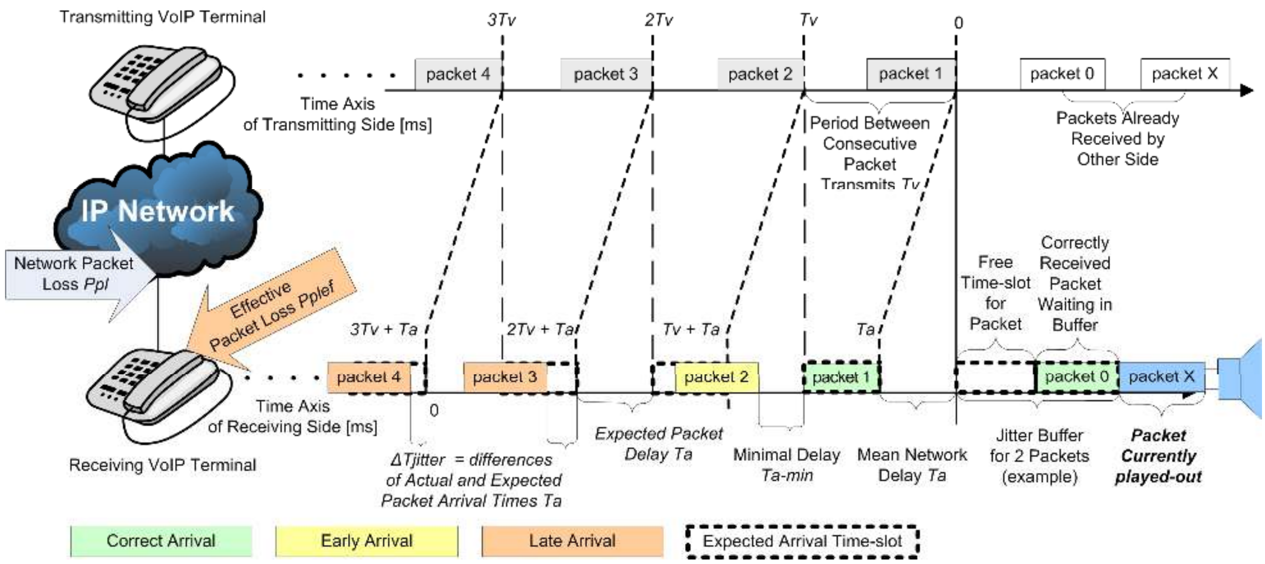

Fig. 4: Packet transmission delay, arrival time and jitter buffer function.

4. Simulation and Measurement of Jitter Effects

Measurements were carried out to model and prove jitter buffer behaviour and to find correct shape parameter ξ for Eq. (3) and (4) to find best fitting function modelling jitter buffer packet loss *Pjitter*. We have simulated MOS dependence of following codecs on QoS parameters:

- G.711 with PLC and without PLC, 64 kb/s, 40 ms jitter buffer, 20 ms/packet,
- G.723.1 ACELP, 5,3 kb/s, 60 ms jitter buffer, 30 ms/packet,
- G.723.1 MPMLQ, 6,3 kb/s, 60 ms jitter buffer, 30 ms/packet,
- G.726, 32 kb/s, 40 ms jitter buffer, 20 ms/packet,
- G.729, 8 kb/s, 40 ms jitter buffer, 20 ms/packet.

Key QoS network parameters were chosen as follows:

- One-way end-to-end delay $Ta \in \{0, 20, 50, 100, 150, 200, 300, 400\}$ [ms],
- Network packet loss $Ppl \in \{0, 1, 2, 3, 5, 7, 10, 15, 20\}$ [%],
- Network jitter of 20, 40 and 80 ms with Pareto distribution.

Special set of measurement and simulation was dedicated to statistics testing and was performed with WANem, EasyFit and StatAssist software suite as follows:

- Generic codec without PLC (valid for PCM or ADPCM codecs family G.711 and G.726) with jitter buffer from 2 to 6 packets (when packetization = 20 ms, then the buffer would be from 40 to 120 ms) and RFC 1889 jitter from 0 to 100 ms in steps of 10 ms. Delay was fixed to 100 ms and network packet loss P_{pl} set to ‘0’ to show pure influence of buffer on resultant real loss.

VoIP traffic was simulated using IxChariot endpoint on dedicated computer with 100 BASE-TX Ethernet network card, switched through computer with two network cards emulating transport network by imposing QoS parameters on relayed packets by WANem software. VoIP stream was received with third computer with IxChariot endpoint.

For each codec and combination of QoS parameters values was simulated a packet flow of 10000 packets corresponding to 200 seconds of continuous VoIP call with 20 ms packetization. Altogether more than 189 combinations of parameters were simulated. Test runs, where *Ta.2 < jitter* were omitted as e.g. mean end-to-end of 20 ms with 80 ms jitter cannot exist when using aforementioned statistics and jitter calculation methods.

Measured dependencies of MOS on QoS parameters were processed as 3D graphs and in tabular form. Estimate of MOS for G.711 codec is through E-model, proposed modified E-model and measurement is enclosed in section 6. Data and net packet losses at jitter buffer are analyzed for particular jitter strength and buffer sizes. Resultant additional losses are in Tab. 2 and 3. Pure jitter buffer loss measurements and packet flow statistic tests are present in section 7.

Tab.2: Measured jitter buffer loss without reordering capability.

Jitter [ms]	Jitter buffer size [ms]	Average loss on jitter buffer P_{wo}*(jitter,buffer-size)*	Complement to loss – transmittance of buffer F_{wo}*(jitter,buffer-size)*
20	40	0,116114	0,883886
20	60	0,029393	0,970607
40	40	0,307519	0,692482
40	60	0,180276	0,819725
80	40	0,509000	0,491000
80	60	0,399790	0,600206

Tab.3: Measured jitter buffer loss with RTP reordering capability.

Jitter [ms]	Jitter buffer size [ms]	Average loss on jitter buffer P_{wr}*(jitter,buffer-size)*	Complement to loss – transmittance of buffer F_{wr}*(jitter,buffer-size)*
20	40	0,001049	0,998951
20	60	0,000316	0,999684
40	40	0,059720	0,940280
40	60	0,014509	0,985491
80	40	0,147420	0,852580
80	60	0,078991	0,921009

5. Equipment Impairment Factor Calculation Modification Using Effective Packet Loss

Based on simulation results and measurements we have determined optimal shape parameter ξ giving the smallest overall MSE error of *Pwr* for all jitter and jitter buffer size combinations from Tab. 2 and 3. Optimal value for our simulation is ξ = - 0,1 with relative MSE of MOS of only 12,0 %. We add two parameters to E-model through Eq. (10) which incorporates jitter buffer size x [ms] and network jitter σ [ms].

$$P_{jitter} = \frac{\left(1-\left(1-\left(1+\frac{\xi(x-\mu)}{\sigma}\right)^{-\frac{1}{\xi}}\right)\right)^{2}}{2}. \qquad (10)$$

After substitution of parameters ξ = -0,1and μ = 0 we get equation for jitter buffer packet loss as Eq. (11):

$$P_{jitter} = \frac{\left(1+\frac{-0{,}1x}{\sigma}\right)^{20}}{2}. \qquad (11)$$

Further we rewrite Eq. (7) as Eq. (12) and substitute P_{jitter} into Eq. (12). We get Eq. (13) which expresses effective packet loss P_{plef} incorporating network and jitter buffer packet loss. P_{plef} from Eq. (13) substitutes P_{pl} in Eq. (2) which leads to Eq. (14):

$$P_{plef} = P_{pl} + P_{dejitter} + P_{pl}\cdot P_{dejitter}. \qquad (12)$$

$$P_{plef} = P_{pl} + \frac{\left(1+\frac{-0{,}1x}{\sigma}\right)^{20}}{2} - P_{pl}\cdot\frac{\left(1+\frac{-0{,}1x}{\sigma}\right)^{20}}{2}. \qquad (13)$$

$$I_{e,eff} = I_e + (95 - I_e)\cdot\frac{P_{plef}}{P_{plef} - B_{pl}}. \qquad (14)$$

Equation (14) is the final proposed equation for equipment impairment factor calculation in E-model including jitter buffer loss and jitter buffer size through effective packet loss P_{plef}.

6. Results

Proposed change in equipment impairment factor calculation leads to improved MOS estimate of E-model when network jitter is present. Network jitter effects as increased packet loss were mostly negligible when network jitter was smaller than approx. ½ of jitter buffer length. Real jitter buffer have always reordering capability through RTP packet numbering and queue without reordering serves as a basic concept for formulating appropriate equations. In our application marginal value for jitter was 20 ms, thus our simulations and measurements were conducted with 20, 40 and 80 ms jitter, see Tab. 3 for results. Discovered dependence of jitter buffer packet loss at different network jitter strengths for different buffer sizes is illustrated at Fig. 5.

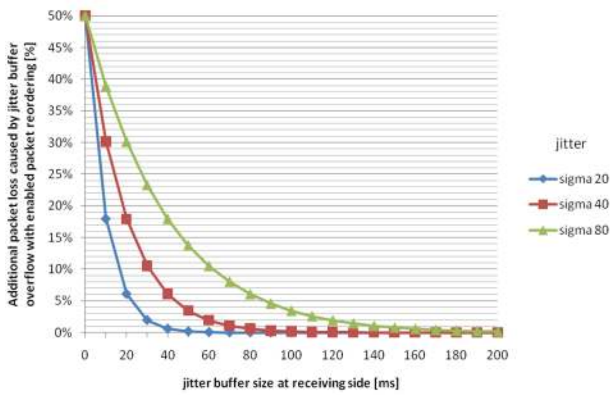

Fig. 5: Jitter buffer packet loss P_{jitter} graph for different jitter.

Table 4 summarizes MOS estimate accuracy by comparison of original and modified E-model with measurements from simulation through average differences in MOS category values. Figure 6 shows comparison of MOS estimates of original E-model, modified E-model and measurements. Offset of MOS estimate caused by presence of jitter is successfully compensated with slight local deviations caused by smaller size of statistical set of measurements. Effective packet loss and resulting end-to-end delay at receiver were greater than objectively measured network QoS parameters due to buffer behaviour and buffering delay.

Tab.4: MOS estimate improvement and comparison of estimation errors of original and modified E-model.

	Original E-model estimate		Proposed Modified E-model estimate		MOS estimate
Jitter [ms]	**MOS diff. (absolute)**	**MOS diff. (%)**	**MOS diff. (absolute)**	**MOS diff. (%)**	**Modified model has**
G.711 codec without PLC, 64 kb/s					
20	-0,17	-8,8	-0,01	-2,5	**improved**
40	-1,03	-37,6	-0,14	-9,9	**improved**
80	-1,12	-39,8	0,01	-2,9	**improved**
G.711 codec with PLC, 64 kb/s					
20	0,00	0,5	0,06	2,2	**worsened**
40	-0,64	-19,4	-0,16	-6,1	**improved**
80	-1,43	-45,3	-0,32	-17,2	**improved**
G.723.1 ACELP, 5,3 kb/s					
20	-0,37	-14,6	-0,35	-13,8	**improved**
40	-0,52	-20,3	-0,33	-13,9	**improved**
80	-1,06	-41,7	-0,33	-20,1	**improved**
G.723.1 MPMLQ, 6,3 kb/s					
20	-0,25	-9,3	-0,23	-8,5	**improved**
40	-0,41	-15,6	-0,19	-8,0	**improved**
80	-1,00	-38,8	-0,16	-12,2	**improved**
G.726 codec, 32 kb/s					
20	-0,89	-32,4	-0,83	-30,7	**improved**
40	-1,43	-49,2	-0,88	-37,4	**improved**
80	-1,75	-58,5	-0,93	-43,7	**improved**
G.729 codec, 8 kb/s					
20	0,04	3,1	0,12	6,2	**worsened**
40	-0,63	-21,2	0,02	2,1	**improved**
80	-1,45	-53,0	-0,52	-32,5	**improved**

Proposed ITU-T E-model calculation modification improved MOS estimate significantly in most cases. Further simulations and improvements to emulation setup would obtain statistically more reliable base data for function fitting with more data points.

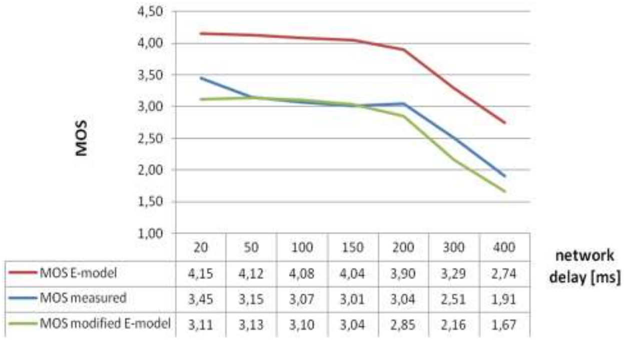

Fig. 6: Comparison of MOS estimate for G.729 codec at 40 ms jitter, 40 ms jitter buffer, 0 % packet loss at various network delay.

Potential improvements in MOS estimation by adding extra parameter describing burstiness of packet stream are further tested in next section by imposing 0 % delay correlation (absolutely random Pareto-distributed packet stream) and 50 % delay correlation to obtain maximum burstiness across larger time-scale still maintaining Pareto-distributed delay.

7. Experiment, Statistic Tests

In this experimental part of the paper we present measurements and statistical tests of VoIP input streams generated according to the most generally accepted knowledge of general IP packet streams [8], [9], [10], [12] to support our model proposed in previous sections.

7.1. Relation between Jitter, Buffer Size and Additional Packet Loss on Jitter Buffer

Figure 7 shows dependency of packet loss occurring on jitter buffer under jitter with varying intensity. Buffer size is normalised relative to jitter amount. It means that if jitter buffer is 2 packets and jitter intensity is 1,5 packets, for 20 ms packetization it would refer to 40 ms jitter buffer and 30 ms RFC 1889 network jitter and vice versa.

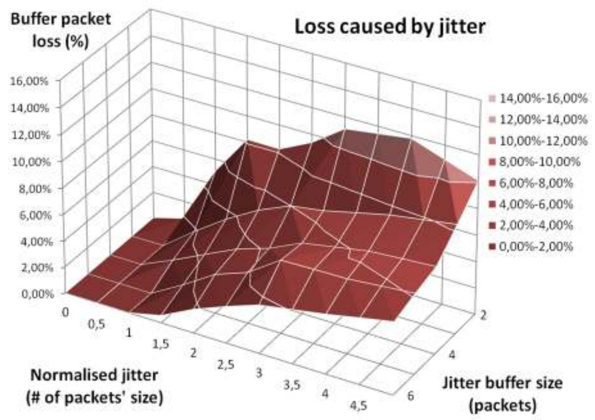

Fig. 7: Packet loss on buffer on relative buffer size to jitter.

7.2. MOS, Jitter and Buffer Size Relations

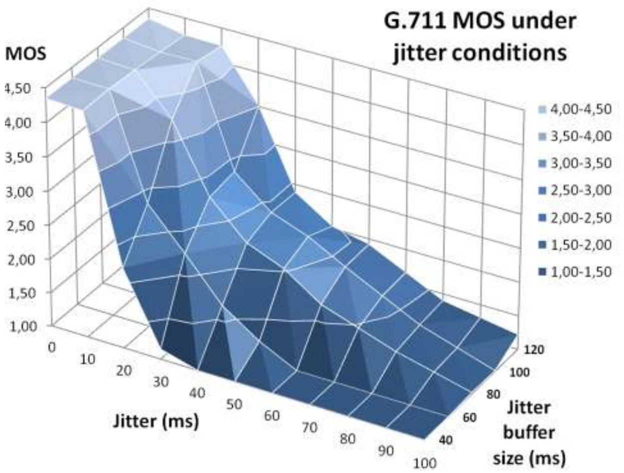

Fig. 8: Actual MOS performance of G.711 codec without PLC under varying jitter and buffer size as measured on Pareto stream.

7.3. Test of Packet Interarrival Distribution

We have chosen Pareto distribution for its long-tailed property for modelling packet input streams from intranet / internet network based on several rigorous studies of IP traffic [8], [9], [10], [12]. Packet interarrival times are defined as timestamp differences. To keep statistical moments as median and variance finite [12], we abstract from network by defining two losses – on the network and on the buffer - as described in Eq. (7) and (8). The input packet stream in size of 10000 packets with Pareto-distributed delay was generated by IxChariot. Stream was sent through WANem, which applied Pareto delay distribution and various jitter to incoming packet stream. The stream was captured by Wireshark packet sniffer at receiver side along with IxChariot performing MOS estimate. RTP timestamp data were loaded into spreadsheet. Then the interarrival times, the first differences of packet RTP timestamps, were calculated according to RFC 1889 recommendation (Eq. 6). Iterative distribution fitting was performed using various distributions to find best fit parameters. These parameters and distributions were put under Kolmogorov-Smirnov, Anderson Darling 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. 5.

Table 6 summarizes best fitting distributions based on aforementioned three statistical tests on dataset. The quest was to test, whether Pareto distribution was suitable not only to describe packet delay distribution of each individual VoIP stream extracted from larger traffic, but also its differences – interarrival time distribution affected by equally distributed jitter and various delay correlation (i.e. burstiness) averaged over timescale of thousands of packets – the magnitude corresponding to common call duration times.

Tab.5: 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
Gamma (3P)	α=0,3377; β=0,08127; γ=1,3000e-5
Log-Pearson 3	α=6,081; β=-1,175; γ=1,641
Generalized Gamma (4P)	k=1,6025; α=0,20097; β=0,09338; γ=1,3000e-5
Laplace	λ=39,109; μ=0,0247
Weibull (3P)	α=0,4745; β=0,01408; γ=1,3003e-5
Gamma	α=0,46676; β=0,05293
Logistic	σ=0,01994; μ=0,0247
Lognormal	σ=2,8971; μ=-5,504

Tab.6: Best fit distribution ranking.

Distribution	Kolmogorov-Smirnov Test		Anderson Darling Test		Chi-Squared Test	
	Stat.	**Best Fit Rank**	**Stat.**	**Best Fit Rank**	**Stat.**	**Best Fit Rank**
Gen. Pareto	*0,12738*	*1*	*71,243*	*1*	*574,28*	*2*
Gen. Extreme	0,13416	2	86,466	2	377,48	1
Weibull	0,15924	3	188,58	4	2293,7	8
Gen. Gamma	0,16413	4	217,81	6	N/A	N/A
Gamma (3P)	0,16819	5	237,21	8	N/A	N/A
Log-Pearson 3	0,17897	6	220,33	7	2531,8	12
Gen. Gamma (4P)	0,1814	7	186,34	3	1798,2	7

Laplace	0,19224	8	274,75	9	574,51	3
Weibull (3P)	0,19363	9	345,11	13	N/A	N/A
Gamma	0,20779	10	207,83	5	1675,6	6
Logistic	0,22469	11	280,74	10	668,76	4
Lognormal	0,22694	12	343,23	12	4324,8	13

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 conditions. Dataset CDF is illustrated in Fig. 9 and represents normalised jitter buffer packet transmittance under chosen jitter (in our case 40 ms) dependent on jitter buffer size on x-axis (from 0 to 1 second = 0 to 1000 ms).

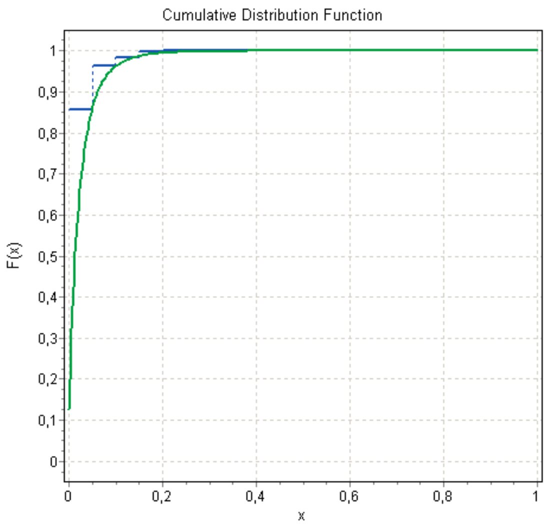

Fig. 9: GPD distribution fitting to real network measured interarrival times on CDF plot: green line – GPD CDF (best fit, parameters from Tab. 5); blue histogram – real interarrival times.

To explain best fit parameters of GPD from Tab. 5, the relation between these parameters to parameters in Eq. (9), (10), (11) and (13) is following:

- σ in Eq. (9) = std. deviation of bounded GPD Pareto realization corresponds to optimised σ in Tab. 5. Proposed relation between σ and actual jitter J substituted can be expressed in 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 yield J/σ ratio = 22/14 $\in$ <1;2>. Actual parameter substitution ratio needs further testing and finding acceptable compromise for various jitter sizes,
- ξ in Eq. (9) = shape parameter and corresponds to optimised $-k$ in Tab. 5. Actual shape parameter for our preliminary model in Eq. (9), (10), (11) and (13) was chosen to be $\xi = \lfloor -k \rfloor$ rounded to one tenth in order to maintain exponent in Eq. (9), (10), (11) and (13) an integer. The resulting error in jitter buffer loss estimate does not exceed 1 % in our case, but further optimisation and testing could lead to better approximation while maintaining simplicity of calculation,
- μ in Eq. (9) = location parameter corresponds to μ= – 0,00306 in Tab. 5. Actual shape parameter for our preliminary model in Eq. (9), (10), (11) and (13) was chosen to be $\mu = 0$ with negligible effect.

Selected common distributions with their best fit parameters were chosen to show main reason of their unsuitability in packet interarrival time and jitter buffer behaviour estimates as visualised in Fig. 10 below.

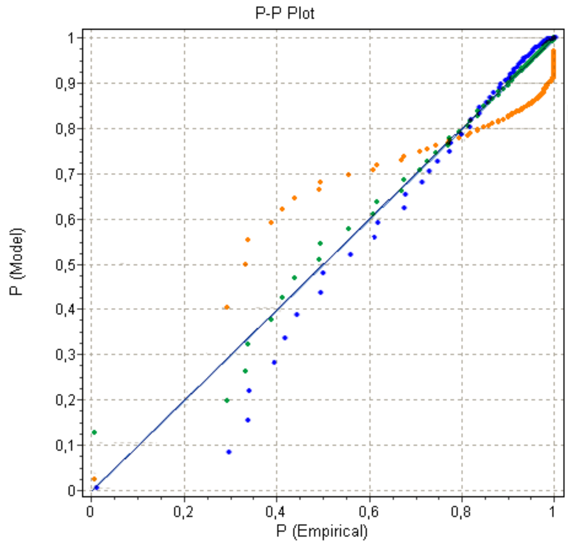

Fig. 10: Comparison of fitting on interarrival time on P-P plot for several distributions: orange – Log-Normal; blue – Exponential (2P); green – Generalized Pareto (best fit).

7.4. Burstiness Effect on Jitter Buffer Loss

Analysed set of data showed no significant effect of interarrival time correlation, i.e. burstiness on proposed modified E-model estimate. The most significant property is the statistical distribution of input stream that if maintained from long-term view (order of hundreds of packets) leads to stable per call MOS estimate. Measured difference with and without correlation was less than 0,5 % average for all combinations of jitter and buffer sizes tested. Difference can be seen in Fig. 11 as a crossection of red (buffer loss on RTP stream with delay correlation of 50 %) and blue plane (correlation = 0 %).

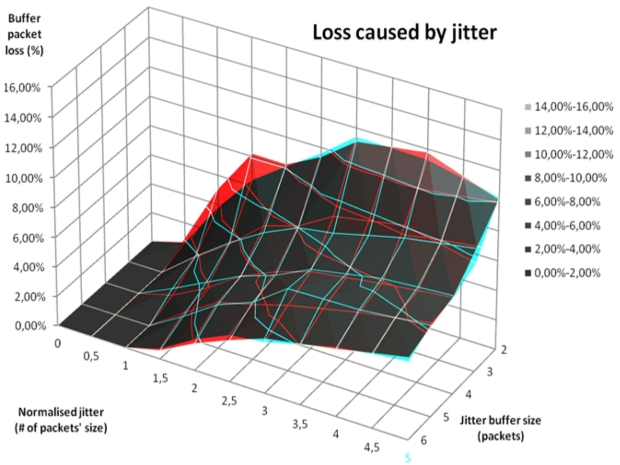

Fig. 11: Difference of jitter buffer packet loss on 50 % bursty (red) and random uncorrelated (blue) Pareto packet stream.

8. Conclusion

Proposed E-model parameter modification by substitution of actual network packet loss yield statistically improved and more coherent MOS VoIP call quality estimates extending to larger network packet losses and greater overall packet losses caused by real network with jitter. Jitter induced packet loss cannot be supported by RTP and RTCP protocols as one of the QoS parameters, because it is purely a matter of user end device, not the network part. Our testing showed great significance and importance of accounting for receiving jitter buffer behavior, for it affect perceived VoIP call quality more than network itself. Proposed modification can be applied generally on WAN or LAN networks and can describe RTP VoIP input stream from various networks with high enough statistical significance to model buffer behavior under Pareto/D/1/K input stream and improve E-model estimate over wide range of jitter and buffer sizes.

Acknowledgements

This paper and whole underlying research has been sponsored by national research projects VEGA 1/0565/09 and SMART II: ITMS 26240120029.

References

[1] ITU-T G.107. The E-model, a computational model for use in transmission planning, *ITU-T Recommendation G.107, ITU-T.* Geneva, Switzerland: May 2000.

[2] ITU-T G.107. The E-model, a computational model for use in transmission planning, *ITU-T Recommendation G.107, ITU-T.* Geneva, Switzerland: 2002.

[3] ITU-T G.107 – Amendment 1, Provisional Impairment Factor Framework for Wideband Speech Transmission, *ITU-T Recommendation G.107, ITU-T.* Geneva, Switzerland: June 2006.

[4] ITU-T G.109. Definition of categories of speech transmission quality, *ITU-T Recommendation G.109, ITU-T.* Geneva, Switzerland: September 1999.

[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] ITU-T STUDY GROUP 12 – DELAYED CONTRIBUTION 106: *Estimates of Ie and Bpl parameters for a range of CODEC types.* Geneva, Switzerland: 27-31 January 2003.

[7] KOH, Y.; KISEON, K. Loss probability behavior of Pareto/M/1/K queue. *Communications Letters, IEEE*, January 2003. Vol.7, No.1, p. 39-41. ISSN 1089-7798.

[8] KASAHARA, S. 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*. 2001. Vol. E84-B, No. 8. pp. 2134-2141. ISSN 1745-1345.

[9] MIRTCHEV, S.; GOLEVA, R. Evaluation of Pareto/D/1/K Queue by Simulation. In *International Book Series "Information Science and Computing"*. Technical University of Sofia. Sofia, Bulgaria: June 2008. pp. 45-52. ISSN 1313-0455.

[10] DANG, T., D.; SONKOLY, B.; MOLNÁR, S. Fractal analysis and modeling of VoIP traffic. In *Telecommunications Network Strategy and Planning Symposium. NETWORKS 2004, 11th International conference*, 13-16 June 2004. pp. 123- 130. ISBN 3-8007-2840-0.

[11] KLÚČIK, S.; TISOVSKÝ, A. Queuing Systems in Multimedia Networks. *Elektrorevue*. Vol. 15, 16.11.2010 (2010), No 99. ISSN 1213-1539.

[12] CROVELLA, M. E.; BESTAVROS, A. Self-Similarity in World Wide Web Traffic: Evidence and Possible Causes. *IEEE/ACM Transactions on Networking*, vol. 5, No. 6, December 1997. ISSN 1063-6692.

[13] TIJMS, H., C. Approximations for the Overflow Probability in Finite Buffer Queues, *Serie Research Memoranda*. Amsterdam: Faculteit der Ekonomische Wetenschappen en Econometrie, August 1992.

[14] ZHANG, W.; HE, J. Statistical Modeling and Correlation Analysis of End-to-End Delay in Wide Area Networks. In *Eighth ACIS International Conference on Software Engineering, Artificial Intelligence, Networking, and Parallel/Distributed Computing*. IEEE, 2007. Vol. 3, pp. 968 – 973. ISBN 978-0-7695-2909-7.

[15] JIANG, W.; SCHULZRINNE, H. Comparison and Optimization of Packet Loss Repair Methods on VoIP Perceived Quality under Bursty Loss. In *NOSSDAV '02 Proceedings of the 12th international workshop on Network and operating systems support for digital audio and video*, May 12-14, 2002, Miami Beach, Florida, USA. pp. 73-81. ISBN 1-58113-512-2.

About Authors

Adrian KOVAC, born in 1986, has received master's degree in Telecommunications in 2010 at the Slovak University of Technology, Faculty of Electrical

Engineering and Information Technology in Bratislava, where he currently pursues his Ph.D. studies and provides lectures about communication network security and services. In 2008 he received Bachelor's degree in Business Informatics at the University of Economy in Bratislava. Adrian specialises in network security, performance testing and modelling of communication systems behaviour. His current research topic is modelling and measurement of VoIP call quality and secure communication.

Michal HALAS was born in 1978 in Slovakia. He graduated from Slovak University of Technology and received the electronic engineering degree in 2003. Since this year he has started postgraduate study at Department of Telecommunications STU Bratislava. In 2006 he received his Ph.D. from Slovak University of Technology Nowadays he works as a lecturer in Department of Telecommunications of FEI STU in Bratislava and topics of his research interests are Speech quality, Next Generation Networks and IP telephony.

Milos ORGON was born in 1956. He received his Master degree and Ph.D. degree from the Faculty of Electrical Engineering and Information Technology, Slovak University of Technology in Bratislava in 1980 and 1988, respectively. Nowadays he works as an associate professor at the Department of Telecommunications of FEI STU Bratislava. He has been engaged in the research and development of telecommunication networks and services in liberalized environment for the area of convergent technologies.

Miroslav VOZNAK was born in 1971. He has been studying Telecommunications engineering at VSB-TU Ostrava and he holds a doctorate in Telecommunications. He received Ph.D. degree in 2002 at the Faculty of Electrical Engineering and Computer Science where he works as an associate professor nowadays. He gives the lectures at the Department of Telecommunications of VSB – TU in Ostrava.