Effective Packet Loss Estimation on VoIP Jitter Buffer

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Abstract. The paper deals with an influence of network jitter on effective packet loss in dejitter buffer. We analyze behavior of jitter buffers with and without packet reordering capability and quantify the additional packet loss caused by packets dropped in buffer on top of the measured network packet loss. We propose substitution of packet loss parameter P_{pl} in ITU-T E-Model by effective packet loss P_{plef} incorporating network jitter, a jitter buffer size and a packet size as additional input parameters for E-Model.

Keywords: Jitter buffer; Dejitter buffer; MOS; E-Model; Packet loss; Packet drop; Pareto distribution; Pareto/D/1/K; VoIP.

1 Introduction

Voice over IP communication gains still greater importance in telecommunications industry. Lack of synchronization in comparison to TDM (Time-division Multiplexing) brings concerns about variable conditions on network which cause packet loss and fluctuations in delay (jitter). Jitter causes excess packet loss on receiving buffers depending on the buffer size and delay variance. When using E-Model as an objective method based purely on network parameters for speech quality assessment, effects of jitter are not incorporated in original E-Model. Estimated MOS (Mean Opinion Score) can be positively biased or be too optimistic about call quality under real IP network conditions. In our paper we propose methods of numerical approximation of general jitter buffer behavior. These approximations are used to quantify effective network packet loss P_{plef} taking three additional network parameters on the input: jitter buffer size [ms]; voice packet size [ms] and network delay variance (jitter) [ms].

2 Selected Time-Related IP Network Parameters

Under time-related network parameters we understand packet transmission delay, mean interarrival time difference – jitter and secondarily packet loss, which can be understood as infinite delay of packet delivery. Based on practical experience of voice perception and human conversational model, ITU-T G.114 [1] defines the recommended value of Mouth-to-Ear delay which consists of partial delays occurring at different stages of communication path. The delay distribution on IP networks can be successfully modeled and described by Pareto distribution which give better results compared to Weibull, Poisson or Log-Normal distribution [2]-[4], [11]. Pareto distribution belongs to family of geometrical and long-tailed distributions that characterize statistical set where most values are clumped around the beginning of the interval. Equation (1) shows PDF and equation (2) CDF functions of generalized Pareto distribution (GPD).

$$F_{(\xi,\mu,\sigma)}(x) = 1 - \left(1 + \frac{\xi(x-\mu)}{\sigma}\right)^{\left(-\frac{1}{\xi}\right)} \qquad f_{(\xi,\mu,\sigma)}(x) = \frac{1}{\sigma}\left(1 + \frac{\xi(x-\mu)}{\sigma}\right)^{\left(-\frac{1}{\xi}-1\right)} \qquad (1), (2)$$

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 and represents minimal network delay $T_{a\text{-}min}$. Jitter J [ms] is calculated in real-time as floating average of 16 samples of differences between interarrival times (timestamps) of consecutively received packets contained in RTP (Real Time Protocol) header, the calculation is defined in RFC 1889 and given by equation (3) where each difference is calculated according to equation (4). Jitter value is transferred in RTCP (Real Time Control Protocol) protocol header as one of the QoS parameters.

$$J = J + (|D_{(i-1,i)}| - J) / 16 \ \ [ms] \quad \text{and} \quad D_{(i,j)} = (R_j - R_i) - (S_j - S_i) \ \ [ms] \qquad (3), (4)$$

Where R are timestamps of packet reception time, S when packet was sent and indices i,j are consecutive packet numbers.According to G.1020 [5] an alternative approach of jitter based on determination of the Mean Absolute Packet Delay Variation with regard to a short term average or minimum value (adjusted absolute packet delay variation) offers more accurate calculation short-term jitter better describing relationship to jitter buffer behavior. In [5] the short term jitter is computed for current packet *(i)* whose delay is designated t_i. Packet *(i)* is compared to a running average estimate of the mean delay using 16 previous packet delays and assigned either a positive or negative deviation value. These characteristics are calculated according to equations (5), (6) and (7) [5], where a Mean Delay is expressed in relation (5), a Positive dealy deviation in (6) and Negative deviation in (7).

$$D_i = (15 \times D_{i-1} + t_i - 1) / 16 \qquad (5)$$

$$P_i = t_i - D_i \quad if\ t_i > Di \quad \text{and} \quad N_i = D_i - t_i\ if\ t_i < Di \qquad (6), (7)$$

If $t_i = D_i$, then both P_i and N_i are zero. Mean Absolute Packet Delay Variation 2 (*MAPDV2*) for packet i is computed as (8) [5]:

$$MAPDV2 = mean(P_i) + mean(N_i) \qquad (8)$$

Variance of input stream packet delivery causes problems with synchronous playback. Receiver can wait finite amount of time for data to be delivered [12], [13]. All data that are not lost in the network but arrive later than expected are considered lost, hence we use the term *Effective packet loss* to describe all losses in data path. Effective packet loss is encountered at the input of audio decoder and is equal to or greater than measured network packet loss advertised through RTCP packets. We analyze additional loss on jitter buffer and its inclusion into E-Model further.

3 E-Model as Objective Measurement Method

E-Model defined by ITU-T G.107 is widely accepted objective method used for estimation of VoIP call quality [6], [7]. E-Model uses set of selected input parameters to calculate intermediate variable, *R-factor*, which is mostly converted to MOS-CQ value. Input parameters contribute to the final estimation of quality in additive manner as expressed in (9).

$$R = R_o - I_s - I_d - I_{e\text{-}eff} + A \,. \tag{9}$$

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, compression artifacts; I_d covers degradations caused by audio signal delay including side tone echo; $I_{e\text{-}eff}$ impairment factor represents all degradations caused by network transmission path, including end-to-end delay, packet loss, codec compression artifacts and PLC (Packet Loss Concealment) masking capabilities and A is an advantage factor of particular communication technology. We focus at $I_{e\text{-}eff}$ parameter, which is calculated as in (10):

$$I_{e\text{-}eff} = I_e + (95 - I_e) \times P_{pl} / (P_{pl} + B_{pl}) \,. \tag{10}$$

Where I_e represents impairment factor given by codec compression and voice reproduction capabilities, B_{pl} is codec robustness describing immunity of particular codec against random losses and quantifying its PLC masking qualities. These values are listed for narrowband codecs in ITU-T G.113 appendix [8]. We propose to substitute P_{pl} parameter for an overall effective packet loss P_{plef} according to equations (11) and (12). In this document we refer to "Modified E-Model" as to model based on original ITU-T G.107 with P_{pl} packet loss substituted by proposed P_{plef}. Both P_{pl} and P_{plef} values lies in interval <0,1>.

$$P_{plef} = 1 - (1 - P_{pl}).(1 - P_{jitter}) \quad \text{and} \quad P_{jitter} = (P_{plef} - P_{pl})/(1 - P_{pl}) \qquad (11), (12)$$

Where P_{jitter} is net packet loss on jitter buffer. According to the type of buffer used, P_{plef} should be substituted by either P_{loss_wo} or P_{loss_wr} respectively.

3.1 Jitter Effects Simulation and Measurements of Effective Packet Loss

Measurements, which were carried out, proved the jitter buffer behavior described in chapter 2 and helped to find a best fitting function modeling the jitter buffer packet loss P_{jitter}. We have simulated MOS dependence of several codecs such as G.711 both μ-law and A-law with PLC, G723.1 (ACELP and MP-MLQ), G.726, G.729 on QoS parameters which were presented by One-way delay $T_a \in \{0, 20, 50, 100, 150, 200, 300, 400\}$ [ms]; Network packet loss $P_{pl} \in \{0, 1, 2, 3, 5, 7, 10, 15, 20\}$ [%] and Network jitter of 20, 40 and 80 ms with Pareto distribution.VoIP traffic was emulated using IxChariot software and the transport network presented a computer with two network cards with WANem software implementing Pareto distribution on an output packet stream. The stream was received with third computer with IxChariot endpoint, the situation is depicted in Fig. 1. For each codec and combination of QoS parameters values were simulated one-minute VoIP calls and repeated three times, from which data for evaluation were collected, overall 189 combinations of parameters were simulated [9], [10].

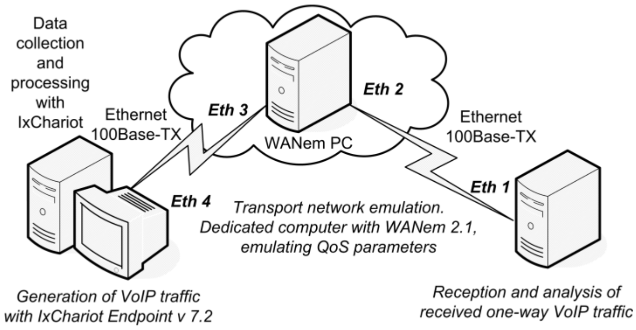


Fig. 1. Test bench for VoIP MOS evaluation with E-Model.

Based on simulation results and measurements we have determined optimal shape parameter ξ giving the smallest overall MSE error of differences between measured and estimated P_{loss_wo} and P_{loss_wr} by equations (13) and (14). Optimal value of sought shape parameter ξ is around values – 0.1 to – 0.2 depending on actual network traffic characteristics, but in general value of – 0.1 proved itself to give good results across wide range of LAN IP networks.

$$P_{loss\_wo} = \left(1 + \frac{\xi(x-\mu)}{\sigma}\right)^{-\frac{1}{\xi}} \qquad P_{loss\_wr} = \left(1 + \frac{\xi(x-\mu)}{\sigma}\right)^{-\frac{1}{\xi}} \cdot \frac{1}{2} \qquad (13), (14)$$

After substitution of equations (13) and (14) into (12), parameters $\xi = -0.1$ and $\mu = 0$ we get equation for jitter buffer packet loss without reordering (equation (15), x = buffer size in [ms]) and with reordering capability (16) where x = packet size in [ms]:

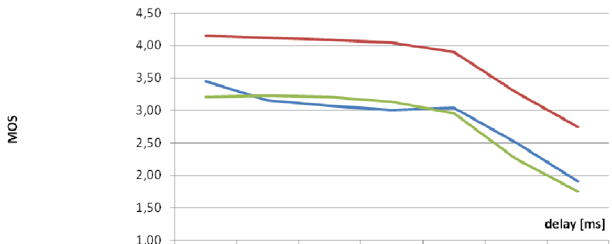
$$P_{loss\_wo} = \left(1 + \frac{-0.1x}{\sigma}\right)^{10} \qquad P_{loss\_wr} = \left(1 + \frac{-0.1x}{\sigma}\right)^{10} \cdot \frac{1}{2} \qquad (15), (16)$$

To use these calculated losses in E-Model we propose to use equation (17), which should be used in place of parameter P_{pl} (network packet loss). Equation (18) is final proposed equation for equipment impairment factor calculation in E-Model $I_{e\text{-}eff}$.

$$P_{plef} = 1 - (1 - P_{pl}) \times (1 - P_{dejitter}) = P_{pl} + P_{dejitter} - P_{pl} \times P_{dejitter} \quad (17)$$

$$I_{e,eff} = I_e + (95 - I_e) \times P_{plef} / (P_{plef} - B_{pl}) \quad (18)$$

Graph in Fig. 2 shows comparison of: measured MOS, MOS estimated by original implementation of ITU-T E-Model and MOS estimate using E-Model with proposed modifications in case of G.729 codec. Modified E-Model takes into account three additional parameters: network jitter, buffer size [ms] and audio packet length [ms]. Proposed modifications give more accurate MOS showing good correlation with real network conditions with jitter as opposed to too optimistic original E-Model, we received similar results for other measured codecs.



	20	50	100	150	200	300	400
MOS original 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	3,21	3,23	3,20	3,14	2,95	2,26	1,75

Fig. 2. MOS, G.729 codec, network jitter 20 ms, buffer 40 ms

4 Conclusions

We propose the E-Model improvement based on the substitution of packet loss parameter P_{pl} by the effective packet loss P_{plef} and we perceive it as the main contribution of this paper. Since the jitter J is calculated as floating average of 16 packets, we consider the characteristics constant if they are similar over recent 16 samples. When considering codec with 20 ms audio per packet, so all values are balanced after 320 ms. The transient response of real-time E-Model shows sufficiently fast recovery rate for practical purposes below 1 second.

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