INFLUENCE OF REDUNDANT PACKET SENDING ON THE QUALITY OF VOIP CONNECTION

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Abstract

In this paper we analyze an influence of redundant packet sending on the quality of packetized voice connections. The analysis starts from a real trace record of error distribution on a link and this record is used as an input set of data for simulation process realization. It is proved that, for two-fold content repeating, voice quality is improved, but, if voice content is repeated more than two times, effect of additional delay in packet transmission on the voice quality decrease overcomes positive effect of redundant packet sending to its loss probability and burstiness. The analysis includes subjective variation of connection rating factor: it does not change immediately to its calculated (or estimated) value, but it exponentially tends to this value.

Keywords: VoIP, voice connection quality, redundant packet sending, subjective rating factor.

INTRODUCTION

Today Internet is more and more present in our lives. New services with always increasing throughput demand are continuously introduced. But, voice communication, as the fundamental service, remains very important and each analysis, dealing with its better quality, deserves to devote it our attention.

In this paper we analyze possibility to improve voice connection quality by redundant packet sending. Redundancy means that voice packets, containing coded signal samples from the same voice segments, are transmitted in more consecutive frames. In this way packet loss probability is decreased, thus increasing voice connection quality. But, in the same time, signal delay is increased, leading, as a consequence, to the decrease of voice connection quality.

STARTING ELEMENTS OF THE ANALYSIS

VoIP connection quality may be expressed in several ways. One possibility is the use of E-model, which is one of starting elements in our analysis.

E-model is calculation model defined in [1]. According to its definition, the starting element in an analysis is the quality of ideal connection ($R0$=94). Starting from this value, connection rating factor (R), which is the measure of connection quality, may be expressed as:

$$R=R0-I_S-I_D-I_e+A \tag{1}$$

where it is:

I_S – simultaneous impairment factor;

I_D – factors expressing influence of transmission delay and echo;

I_e – equipment impairment factor: it collects influence of mutually independent elements: codec (compressor) type, packet loss and packet loss concealment (PLC);

A – advantage factor, representing connection quality improvement, if connection quality is better than it is expected.

Influence of I_e (i.e. its corrected value effective equipment impairment factor - $I_{e\text{-}eff}$ in (1)) can be modelled by the equation, [1]:

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

where it is:

P_{pl} – packet loss probability (in percents);

$BurstR$ – burst ratio: relation of average burst loss length in an arrival sequence to the average length of random (single) packet loss;

B_{pl} – packet-loss robustness factor: it expresses coder (compressor) resistance to packet loss.

The value of $BurstR$ is approximately equal to the mean number of successive lost packets.

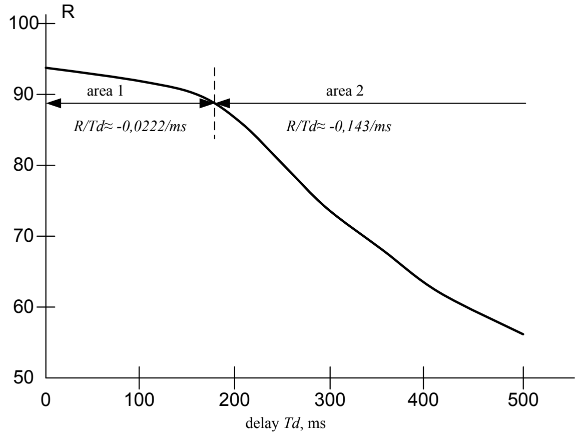

Fig. 1. *The variation of connection rating factor as a function of packet delay and the slope of this factor decrease*

The second starting element of our analysis is the variation of connection rating factor (R) as a consequence of packet transmission delay (T_d). This variation is presented by graph in Fig. 1, sketched on the base of Fig. 2 from [2], i.e. Fig. 1 from [3]. Connection rating factor is decreased when packet delay is increased. The slope of this decrease is smaller for smaller values of delay, but for greater values of delay it becomes greater. The approximate slope value is presented in Fig. 1.

The third starting element of our analysis is modelling subjective connection quality estimation as a consequence of impairment variation, [4]. When, for example, data transmission conditions on a line changes from a gap (errorless or small error value period) to a burst (packet loss period), this variation is not perceived immediately. Subjective voice quality estimation changes exponentially from instantaneous rate level towards the new rate level. This estimation of impairment variation is presented in Fig. 2. As a consequence, subjective rating factor variation may be also estimated on the basis of (1).

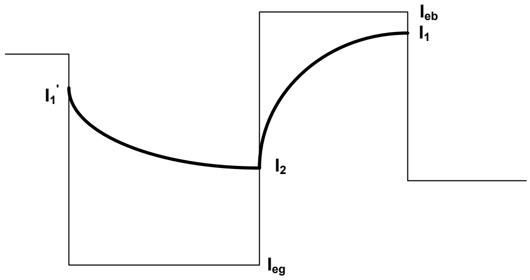

Fig. 2. *Voice connection impairment variation when transmission conditions on a line are changed*

According to designations from Fig. 2, estimated impairment values in the moment of transition from burst to gap (I_1) and vice versa (I_2) can be expressed by equations, [4]:

$$I_1 = I_{eb} - (I_{eb} - I_2) \cdot e^{-\frac{b}{t_1}} \tag{3}$$

$$I_2 = I_{eg} + (I_1 - I_{eg}) \cdot e^{-\frac{g}{t_2}} \tag{4}$$

where I_{eb} is impairment value in a burst period and I_{eg} is impairment in a gap period. Typical time constants of impairment decay are t_1=5s in a burst period and t_2=15s in a gap period.

REDUNDANT PACKET STRUCTURE

Fig. 3 presents packet structure for redundant sending of voice signal over Internet. More detailed structure of transmitted packets in this case is presented in [5]. Each Internet packet consists of standard protocol headers and ending content. Packet payload (in this case coded voice) is situated between headers and ending. In the example from Fig. 3 it is presented that content of each signal segment is transmitted three times. In the first packet we have signal from the last segment in that moment (*n*), but also from previous two segments (*n-2* and *n-1*). If we consider three consecutive packets, it is obvious that voice signal from segment *n* is transmitted three times, but the first time as the signal from the last segment, the second time as the signal from the previous segment, and the third time as the signal from the segment two packetization intervals ago. In this way, necessary channel bandwidth is increased. The voice signal from segment *n* is transmitted correctly if at least one of three presented packets is transmitted correctly. In other words, if the burst loss length is less than three, voice signal is transmitted successfully. At the receiving side of a connection packet is considered lost if content with error is received or if a packet arrives with successive delay, thus causing too great rating factor decrease, according to Fig. 1.

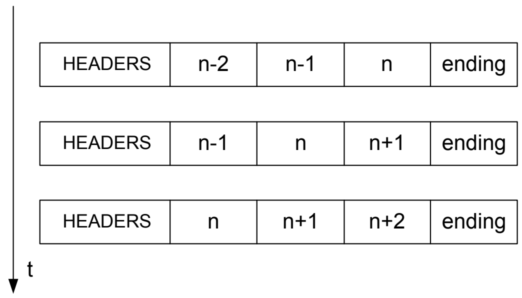

Fig. 3. *Redundant sending of voice signal packets over Internet*

SIMULATION PROCESS

Influence of redundant packet sending is analyzed starting from the distribution of packet loss, presented in figures 7-10 from [6]. For our purposes we used Fig. 7 to determine the probability to lose smaller number of consecutive packets (till 9 of them) and Fig. 10 to lose 10 or more than 10 consecutive packets. This distribution has been already used to determine distribution of voice connection quality during some time period, [7].

The flow-chart of simulation process is presented in Fig. 4. It starts by the simulation of the link event: whether the next packet (or packets) is (are) lost. It is two-step random number generation process. Implemented random numbers are uniformly distributed in the range (0,1). If the first generated random number is

$$RN1 < \frac{P_{plm}}{B_m \cdot 100} \quad (5)$$

where P_{plm} is simulated mean packet loss and B_{mean} is the mean number of consecutively lost packets, the next packet (or packets) will be lost (otherwise, the following packet is not lost). In a case of packet loss, the second generated random number is used to determine the burst length, in such a way that this random number is compared to the cumulative distribution function (CDF) of burst loss length. The complete range of random numbers is separated to *K* segments, where *K* corresponds to total number of possible burst lengths. If the generated random number is in the range $0<RN2<CDF_1$, the following loss corresponds to random loss (of only one packet). If the generated number is in the range $CDF_1<RN2<CDF_2$, the following loss corresponds to burst loss of two consecutive packets, and so on. If $CDF_{i-1}<RN2<CDF_i$, *i* consecutive packets are lost.

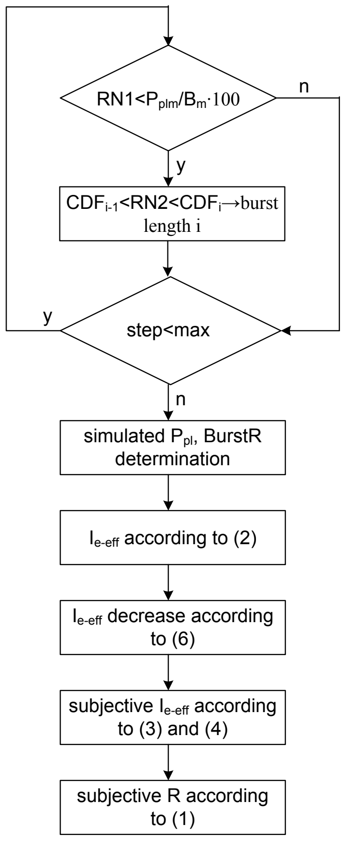

Fig. 4. *Flow-chart of the simulation process*

This simulation process continues until the number of generated events on a link approaches the packet segment length (designation max in Fig. 4). After this part of simulation is finished, the detailed record of link loss is used to determine simulated instantaneous P_{pl} and *BurstR*. Simulation is finished by applying (2) to determine $I_{e\text{-}eff}$ value in the considered connection segment and, after that, applying (3) and (4) to calculate subjective $I_{e\text{-}eff}$. The value of subjective rating factor R may be obtained from (1) directly when $I_{e\text{-}eff}$ is determined and if influence of I_s, I_d and A is neglected.

In our analysis, presented simulation process is applied three times: first at the originally generated loss sequence, than at the sequence with two times sent redundant content, and at the end for the sequence with three times sent redundant content. In the case of two times sent content all single lost packets are improved and when two consecutive packets are lost, they are converted to random packet loss. When the same content is sent three times, error may be improved in the case of random packet loss and two consecutive packet loss. In the same time, an error in three consecutive packets is converted to random loss.

When redundant packet sending is considered, connection quality decrease as the consequence of additional packet delay must be also included in the analysis. According to Fig.1, greater connection quality decrease is in the region where packet delay is greater than 180ms (slope -0.143/ms), comparing to the packet delay region smaller than 180ms (slope -0.0222/ms), [8]. A connection quality decrease as a consequence of redundant sending, i.e. $I_{e\text{-}eff}$ increase, may be expressed as:

$$\Delta I_{e-eff} = n_{red} \cdot T_{pack} \cdot sl \qquad (6)$$

where n_{red} is number of consecutive sending of the same content, T_{pack} is packetization interval (period of one packet duration), and sl expresses slope of the characteristic. In our analysis we considered T_{pack}=20ms and the worse case when we are considering the slope (-0.143/ms).

RESULTS

The results of subjective voice quality estimation (when redundant packet sending is implemented) are presented in figures 5-8 for packet loss probability P_{pl}=2%. The results are obtained after simulation process implementation, according to method from Fig. 4, for packet loss distribution from [6].

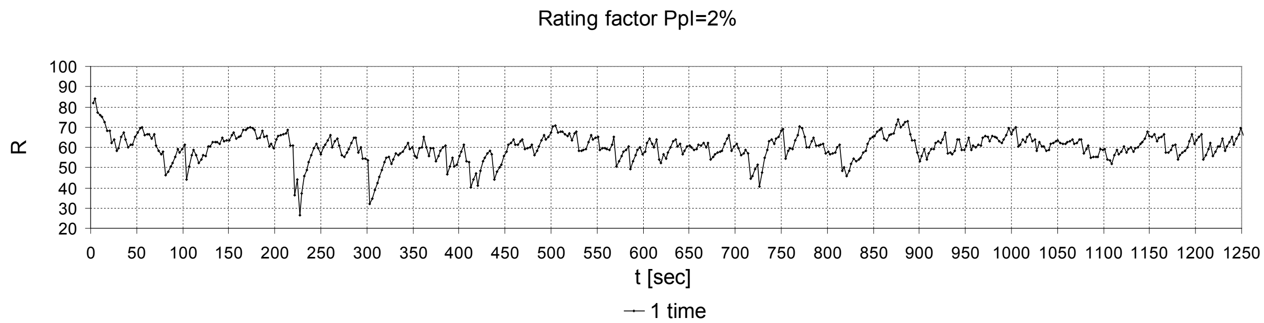

Fig. 5. *Variation of subjective rating factor in standard packet sending*

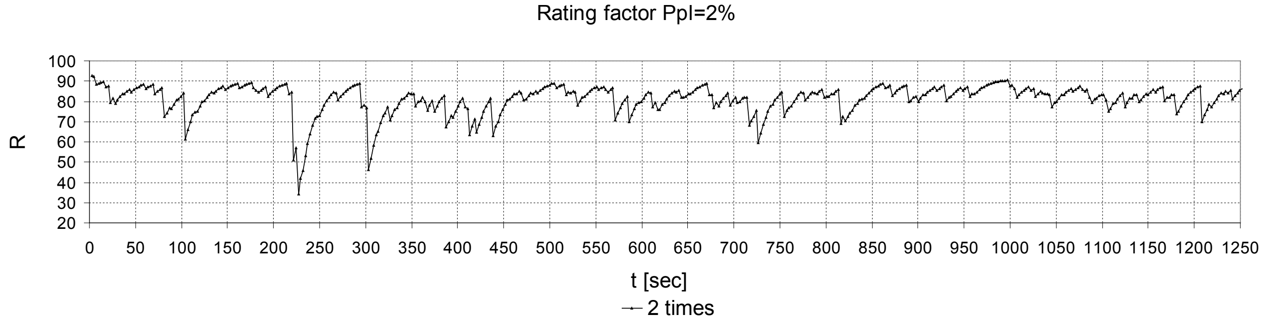

Fig. 6. *Variation of subjective rating factor for redundant packet sending (two times repeated content)*

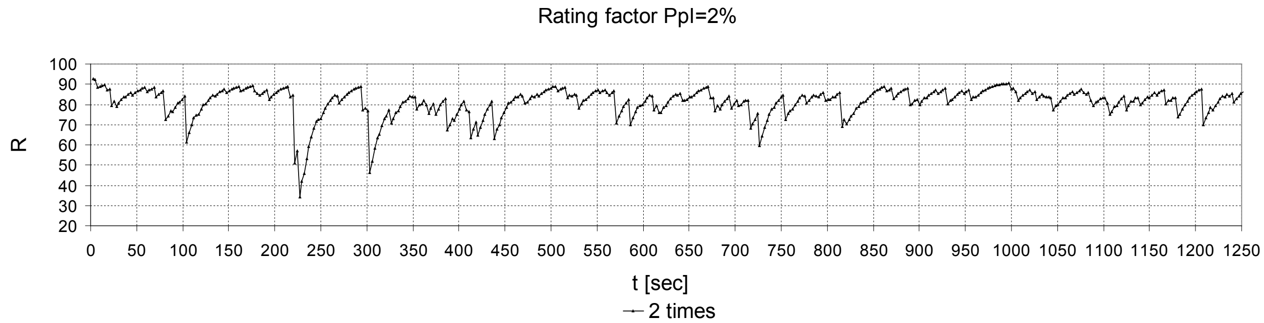

Fig. 7. *Variation of subjective rating factor for redundant packet sending (three times repeated content)*

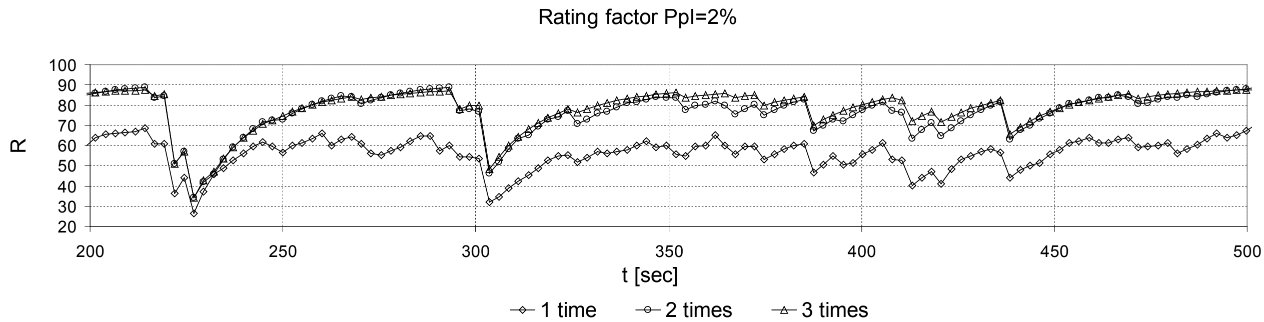

Fig. 8. *Comparison of subjective rating factor variation when the content is sent once and repeated two and three times*

Fig. 5 corresponds to standard packet sending (with no redundancy implementation). The variation of connection rating factor when redundancy is introduced is presented in Fig. 6 (each voice packet content repeated twice) and Fig. 7 (each voice packet content repeated three times). After that, in Fig. 8 one part of simulated rating factor trace from figures 5-7 is collected on one graph to facilitate comparison. An improvement, which is achieved by two-fold redundant content sending, is 15-20 rating units in this case. This is significant contribution to the value of subjective rating factor. Speaking in the other way, time periods when rating factor is lower than some threshold (for example, below 50 units, when quality becomes unsatisfactory) are shorter when redundant packet sending is implemented.

Transition to triple repetition does not significantly improve connection quality, comparing to two-fold transmission. In some periods of time connection quality is even worse, because its decrease as a consequence of additional packet delay overcomes the contribution of the decrease of packet loss probability and burstiness as a consequence of greater number of repetitions when packets are transmitted.

CONCLUSION

In this paper we estimate voice connection quality (rating factor *R*) improvement as a consequence of redundant packet sending. This analysis is performed by simulation, based on a trace from practice. Besides determination of $I_{e\text{-}eff}$, we included effect of additional packet delay and modelled *R* variation to present *R* subjective flow.

Implementation of redundant packet sending is justified, when each voice packet is sent two-times. In a case when voice content is repeated three times, improvement is very small, or, sometimes, it does not exist. This small improvement does not justify repeating voice content more than two times, because redundant packet sending increases necessary bandwidth for voice signal transmission.

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