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A SIMULATION METHOD FOR DETERMINATION OF LOSS IMPAIRMENT FACTOR IN INTERNET SPEECH CONNECTION

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Professional paper
This paper presents a research on the influence of packet loss on quality of Internet speech connection. A novel simulation method is proposed to determine time distribution of the loss impairment factor, which is one of the parameters necessary to determine connection quality according to E-model. Several examples illustrate the use of method for the analysis of links with random and burst packet loss and with the loss as one recorded real trace. Use of the presented method can provide better insight and more data about the quality of Internet telephony.

Keywords: *cumulative distribution function of loss impairment factor, E-model, Internet, quality of speech connection, simulation program*

Simulacijska metoda za određivanje čimbenika umanjenja koji uključuje gubitke u Internetskoj govornoj vezi

Stručni članak
Ovaj rad predstavlja istraživanje o utjecaju gubitka paketa na kvalitetu Internetske govorne veze. Nova metoda simulacije je predložena da se utvrdi vremenska raspodjela čimbenika umanjenja kvalitete, koji uključuje gubitke (i utjecaj kompresora), što je jedan od parametara potrebnih za određivanje kvalitete veze prema E-modelu. Nekoliko primjera ilustriraju korištenje metode za analizu linkova sa slučajnim i grupnim gubitkom paketa i s jednim realno zabilježenim gubitkom. Uporabom predstavljene metode može se pružiti bolji uvid i više podataka o kvaliteti Internetske telefonije.

Ključne riječi: *kumulativna funkcija raspodjele čimbenika umanjenja koji uključuje gubitke, E-model, Internet, kvaliteta govorne veze, program za simulaciju*

1 Introduction

Modern telecommunications are related more and more to the communication over Internet. The method for determination of communication quality over Internet is related to the quality of speech connection and to the quality of other services, which are used in the communication over Internet. The literature, related to this theme, is very numerous, and some of the great numbers of references on the speech communication are given in [1, 2, 3] deals with the other services.

A number of factors have influence on the quality of internet speech connection. Some of them were unknown in classic telecommunications, or their influence was significantly lower, than in the communication over Internet. Factors, as delay, delay variation, packet loss, noise, echo, quantization distortion and so on, have influence on speech signal quality in Internet telephony. One of the most important factors is late packets, which come at the receiving side of the connection with excessive delay. That is why the conditions of speech real time transmission cannot be fulfilled, and these packets must be missed. Besides these, there are factors connected with the voice signal processing, which have influence on speech signal quality. These factors are applied type of compression, type of implemented coding, multiple speech signal coding/transcoding, noise and echo suppression, methods of packet loss concealment and delay equalization, and so on.

In this paper we investigate the influence of packet loss on the quality of Internet speech connection. We analyze the influence of random packet loss and burst packet loss. The analysis of packet loss when realizing the speech connection is especially important, because retransmission is not used when packet is transmitted incorrectly. As it is well known, retransmission of incorrectly transmitted packets disables the satisfaction of the conditions for the real time transmission. The methods, used for the analysis of loss impairment factor (LIF), mainly consider its mean value and represent the way how to determine this mean value [4-8]. But, the mean value is not always enough to estimate LIF. This situation especially happens when parameters, which have influence on LIF, often change during the connection. Then it is difficult to measure or estimate its value. The purpose of this paper was to find easier method for LIF estimation. This method is based on simulation program and it determines cumulative distribution function (CDF) as the better measure of LIF value.

2 Method for the determination of voice signal quality

Few methods for the determination of voice signal quality are developed. One of these methods is based on the use of the E-model [9, 10]. E-model is used in this paper, because it allows the estimation of speech connection quality considering users' perception, which is, in fact, real measure of subjective impression. All the factors, which have influence on speech connection quality, and which are mentioned in the introductory section, are unified in one value, which simply (as one number) presents voice signal quality. If some important factor for speech quality impairment has greater influence (as, for example, delay), the other important factor (as, for example, packet loss) must have smaller influence. The formulas for calculation are especially simple when considering packet loss, which is the main subject of analysis in this paper.

The main equation of the E-model is:

$$R = R0 - I_{\mathrm{s}} - I_{\mathrm{d}} - I_{\mathrm{e-eff}} + A, \tag{1}$$

where is:
R – transmission rating factor of one connection;

$R0$ – basic signal-to-noise ratio: the quality of an ideal connection, reduced by the basic room noise and circuit noise. This value is assumed to be about 94 and it is considered that it represents the quality of the local ISDN connection, where all other factors are negligible;
I_s – simultaneous impairment factor which integrates simultaneous impairments, such as the deviation of user sides from standard values, too great influence of speaker's own voice, and the quantizing distortion;
I_d – delay impairment factor which represents the influence of the voice signal delay and the influence of echo (on both talker's and listener's side) on the decrease of connection quality;
I_{e-eff} – effective loss impairment factor (ELIF) which represents the influence of the compressor, packet loss and packet loss concealment on the decrease of voice signal quality;
A – advantage factor which is the psychological factor representing subjective feeling of connection improvement if bad connection quality is expected, or if its setup is not expected.

The factor I_{e-eff} can be expressed by equation:

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

where is:
I_e – equipment impairment factor which represents the decrease of voice signal quality, caused by the use of voice signal compression;
P_{pl} – packet-loss probability, expressed in percents;
$BurstR$ – burst ratio which represents the influence of packet-loss burstiness;
B_{pl} – packet-loss robustness factor which represents the resistivity of the used coding and decoding technique (i.e. compression and decompression) of a voice signal on packet loss.

In the case when voice signal compression is not used, it will be I_e=0, i.e. Eq. (2) becomes:

$$I_{e-eff} = \frac{95\cdot P_{pl}}{\dfrac{P_{pl}}{BurstR} + B_{pl}}. \tag{3}$$

In this paper we analyze the situation when speech signal compression is not used.

The number of factors, which have influence on speech signal quality, is great, as presented in previous section. Among them delay and delay variation are the most important. In this paper analysis is limited to packet loss, which is also important and analyzed according to Eq. (3).

3 Flow-chart of the simulation program

Two important elements for the determination of the speech signal quality and, therefore, for the simulation process, are packetization interval and considered speech signal segment. Packetization interval represents collection of all consecutive speech signal samples, which are sent by one message packet over Internet. Speech signal segment represents the collection of a number of packetization intervals (packets), for which we determine the quality of speech signal by simulation.

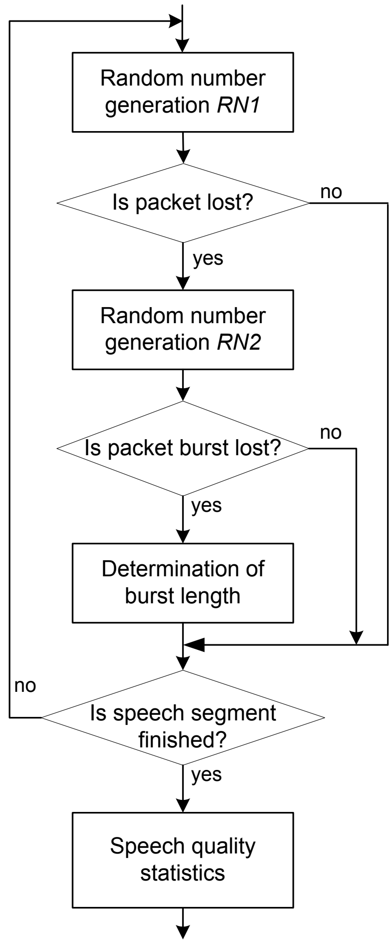

Figure 1 Flow chart of the simulation program

The simulation program is realized in Excel. Its flow-chart is presented in Fig. 1. This flow-chart illustrates the simulation, which is performed for each speech signal segment included in the determination of speech signal quality by simulation.

Before starting the simulation, the initial parameters of simulation are specified: the probability of message packet loss, the probability of random packet loss (i.e. loss of isolated packet), the distribution of burst packet loss lengthwise (i.e. according to the number of consecutively lost packets), the duration of packetization interval and the duration of speech signal segment for which the speech signal quality is determined. The typical durations of packetization interval are 10 ms or 20 ms, and for speech signal segment 10 s.

After that the process of simulation starts, according to Fig. 1. The first step in simulation process is random number ($RN1$) generation. This random number has uniform distribution in the range between 0 and 1. Depending on the value of generated $RN1$ it is decided whether the packet will be lost, or it will be correctly transmitted. If the packet is lost, the second random number ($RN2$) with the uniform distribution is generated with the aim to determine the kind of loss, i.e. whether it is the random loss or it is the loss of few consecutive packets.

According to the generated $RN2$ the length of burst loss is also determined.

The presented simulation process is repeated for each packet in the considered speech signal segment. For example, it is repeated 1000 times if the packetization time is 10 ms. When the simulation for the whole segment is finished, the impairment of speech signal quality in the considered segment is determined by Eq. (3), and after that the simulation for the next segment starts.

Determination of the event in the simulation process is illustrated by Fig. 2.

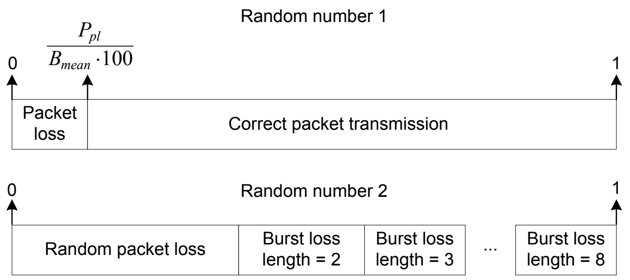

Figure 2 Determination of the event type in the simulation: whether the packet is lost or not and what is the number of consecutively lost packets

It is already pointed out that the first generated $RN1$ determines whether the packet is lost or it is correctly transmitted. If the value of generated random number $RN1$ is

$$RN1 < \frac{P_{\mathrm{pl}}}{B_{\mathrm{mean}} \cdot 100},$$

packet is lost. If $RN1$ satisfies inequality

$$\frac{P_{\mathrm{pl}}}{B_{\mathrm{mean}} \cdot 100} < RN1 < 1,$$

packet will be correctly transmitted. In these equations B_{mean} represents the mean number of consecutively lost packets, calculated according to the desired distribution of packet loss length, i.e.:

$$B_{\mathrm{mean}} = \sum_{i=1}^{n} P_i \cdot i \,. \tag{4}$$

In Eq. (4) i is the number of consecutively lost packets (the length of burst loss), P_i is the probability that series of i consecutive packets is lost among all series of lost packets, and n is the greatest number of consecutively lost packets.

If the packet is lost, the second generated $RN2$ is compared with the probability distribution function of the number of consecutively lost packets. It is allowed in the simulation that packets are separately lost (random packet loss) or that they are lost in bursts of length 2, 3, ..., 8, and more consecutive packets.

4 Simulation results

Simulation results are presented in Figs. 3 ÷ 6. In these figures we present the CDF of ELIF, i.e. probability that impairment factor is smaller or equal to the value on x axis. In other words, it is the percent of time during one long connection when the impairment factor is smaller than the value on x axis.

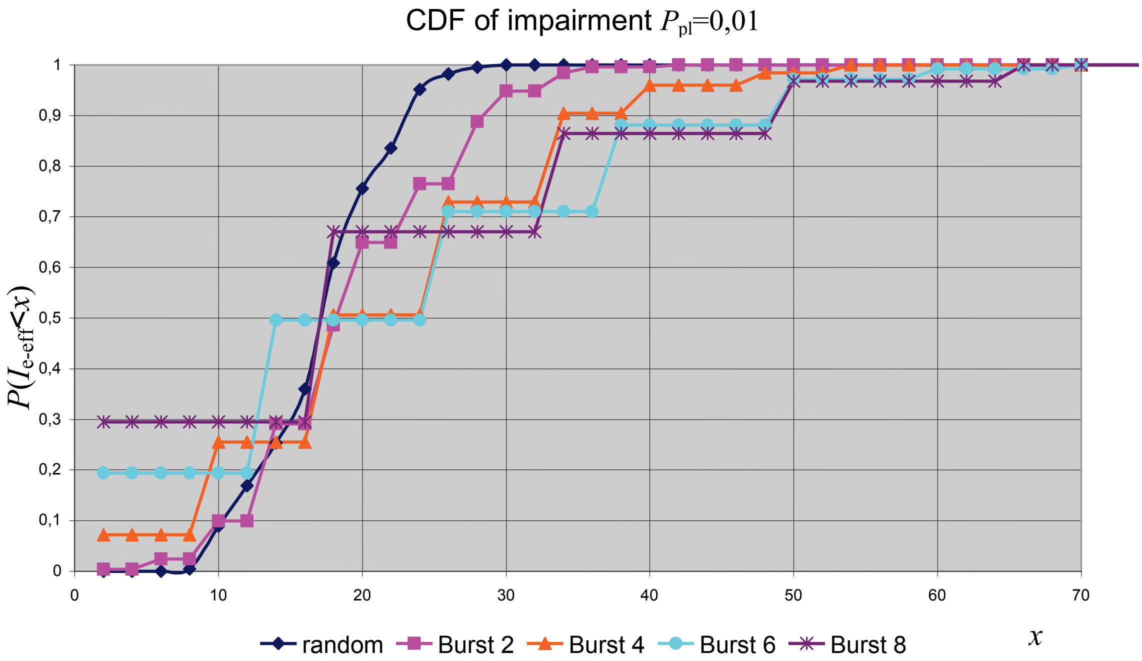

Figure 3 Cumulative distribution function of the voice quality impairment for P_{pl}=1 % and burst length as the parameter

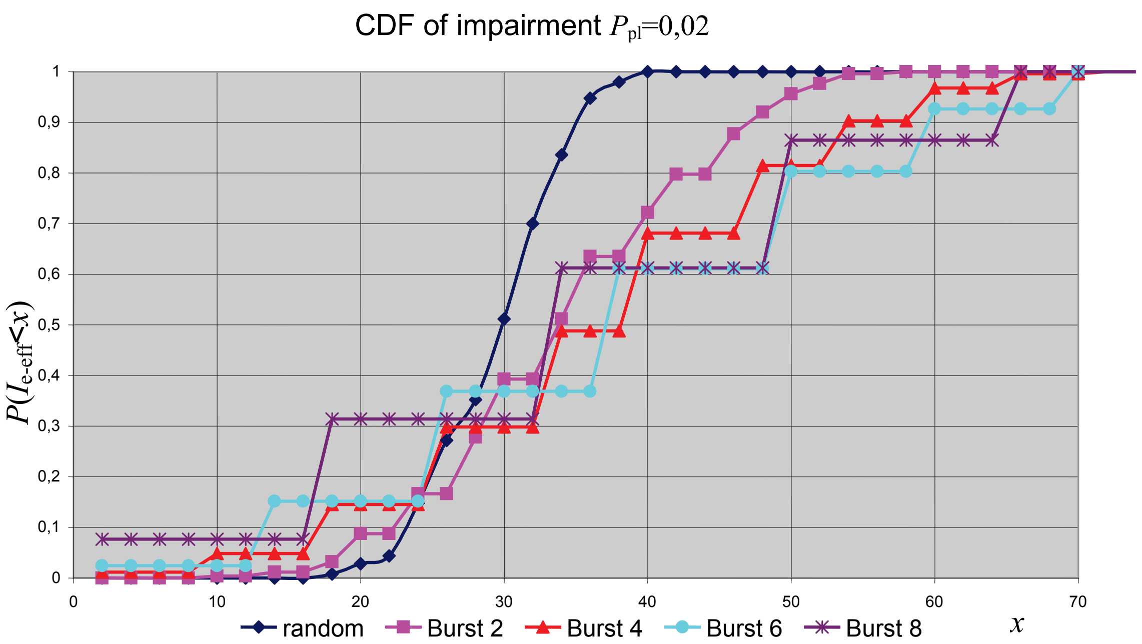

Figure 4 Cumulative distribution function of the voice quality impairment for P_{pl}=2 % and burst length as the parameter

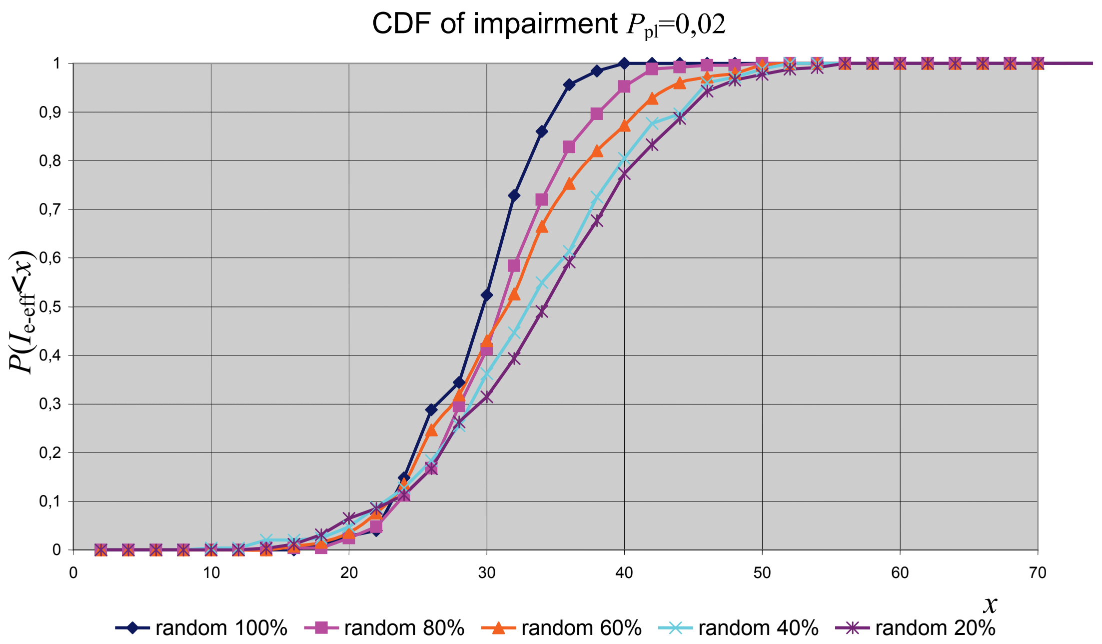

Figure 5 Cumulative distribution function of the voice quality impairment for P_{pl}=2 % and part of random packet loss in total packet loss as the parameter

Figs. 3 and 4 present CDF of ELIF when packet loss probability is 1 % (Fig. 3) and 2 % (Fig. 4). The parameter on graphs is the number of consecutively lost packets in each group of lost packets (one separately lost packet for random, two consecutively lost packets for Burst 2, etc.). This, practically, means that all lost packets are lost separately when simulating random packet loss (before and after these lost packets are correctly transmitted packets), when simulating Burst 2 packet loss, all lost packets are lost as two consecutive packets (before and after these two consecutively lost packets are correctly transmitted packets), etc.

Fig. 5 presents the influence of the change in the relationship between random packet loss and burst packet loss in total packet loss. This is the most realistic case. The parameter in this figure is part of random packet loss in total packet loss.

From the graphs in Figs. 3, 4 and 5 it can be concluded that the distribution of ELIF is such that their values are nearer to the mean value in the case of random packet loss than when burst packet loss is considered. When increasing the length of burst packet loss (Figs. 3 and 4), i.e. when increasing the part of burst packet loss in total packet loss (Fig. 5), the range of expected values of connection quality spreads on both sides around its mean value.

The simulation program is verified on one example, which uses data from reference [11], Figs. 7 and 10.

In Fig. 6 we presented the CDF of the ELIF, obtained by simulation for the link with packet loss distribution as in [11], Figs. 7 and 10. These figures refer to the same trace with packet loss distribution for the series of N_p=443 000 packets. Total approximate number of lost packets (N_{pl}) is determined according to the distribution, presented in Fig. 7 in [11]. Packet loss probability, calculated on the basis of these considerations, is approximately:

$$P_{pl} \approx \frac{N_{pl}}{N_l} = 0{,}035 = 3{,}5\ \%.$$

Fig. 7 [11] is used to determine the distribution of lost packets number when smaller number of consecutive packets is lost (till 9 of them). Beside this, from Fig. 10 [11] can be concluded that there are some isolated cases of few tens of consecutively lost packets. That is why the simulation is made to allow the loss of up to 50 packets.

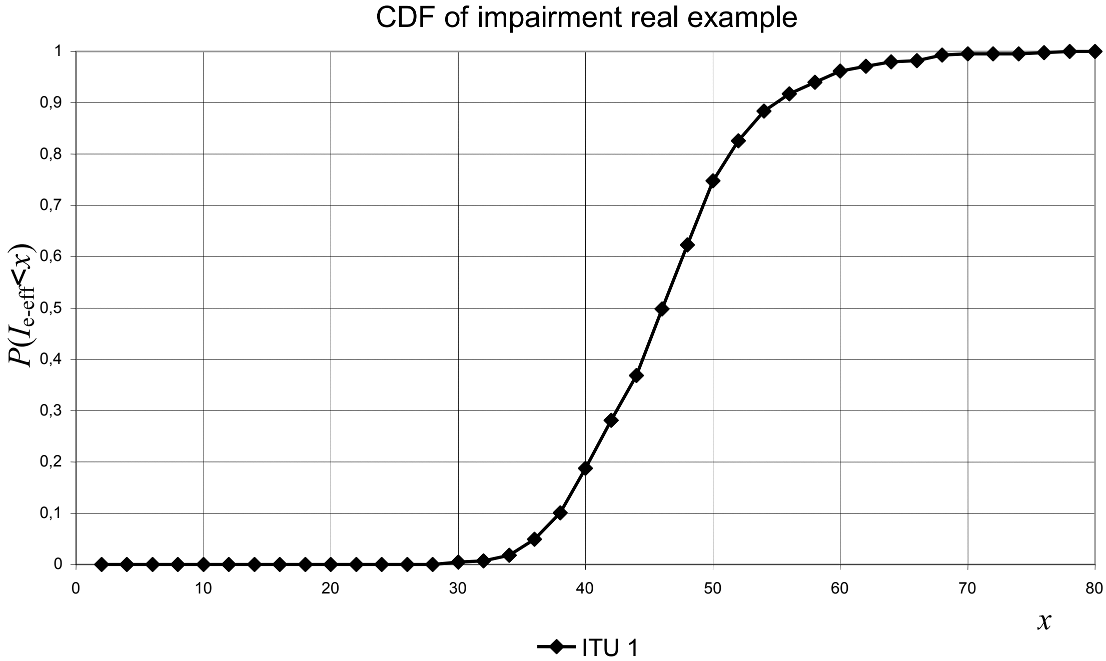

Figure 6 Cumulative distribution function of the voice quality impairment for one real example of packet loss distribution

Example: We wish to determine what is the probability that impairment factor for speech connection is better than 45 for the link with packet loss characteristics, presented in [11]. The value of 45 is often considered as the limit for acceptability of speech connection quality. From the graphic in Fig. 6 it can be determined that this probability is about 0,44. Or, in other words, the impairment factor on this link will be better than 45 during 44 % of connection duration, i.e. the speech connection quality will be poor, but acceptable during 44 % of time.

5 Conclusion

The method, presented in this paper, allows detailed insight in the Internet speech LIF. Using this method, we can determine percent of time when the impairment factor of speech connection is better or equal to some threshold. The method also allows us to compare LIF in the function of the number of consecutively lost packets, i.e. in the function of the relationship between random loss and burst packet loss in total packet loss. The analysis can be done easily without using some special equipment for testing distribution of packet loss, and without considering determination of packet loss by implementing possibilities of Internet protocols. It is based on simulation program, which is realized in simple program surrounding (Excel), that is available on each computer.

The paper introduces CDF as the better measure of ELIF estimation than it is its mean value, which is usually used. CDF gives more data for analysis of smaller segments of voice connection. The implementation of the method is illustrated on several examples. Among these examples, the most important is the analysis on the data collected by recorded trace from practice [11]. This example shows that mean value of impairment factor may be poor, although great percent of time impairment factor may be acceptable (in the considered example 44 % of time). Beside these, examples show that values of ELIF spread more and more around its mean value as the part of burst packet loss increases in total packet loss and as the length of burst packet loss increases.

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