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Method for the Determination of Effective Loss Impairment Factor when Sending Short Messages over the Internet

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In this paper we present the method for the determination of Effective Loss Impairment Factor in the case of sending short voice messages over the Internet. The method can be used for random packet loss and for burst packet loss. In the paper, the method is illustrated for the group packet loss. The method provides a possibility to determine the distribution of quality of different short-term voice connections.

Key words: **Internet, speech signal, voice transmission, message transmission, voice connection quality, speech analysis, signal quality**

Introduction

MODERN telecommunication networks are increasingly based on the communication over the Internet. Migration from the Public Switched Telephone Network (PSTN) to the Internet protocol network (IP network) is successive, and during this transition period, it is necessary to provide the appropriate functionality over both networks. The consequence of using the IP network for providing telephone services is the existence of several factors which decrease the quality of voice signals. These factors were unknown or of less importance in the PSTN. The unknown factors in the PSTN were: speech signal compression, multiple speech signal coding/transcoding, and speech packet loss. Delay and echo, very important factors in the IP telephone technology, are only important in satellite connections when considering the classic telephone technology. The influence of speech signal delay and the influence of echo (listener echo and talker echo) are well known and studied [1-6]. The influence of a coder (compressor) is also known from the recommendation G.113, version from 1996 [7]. The influence of the packet loss on the quality of transmitted signals is still studied, and new recommendations and standards are edited. Recommendation G.113, Appendix I, 2001 [8], defines the influence of the packet loss on speech signal quality, but still does not define the packet loss robustness factor of different compressor types. Recommendation G.113, Appendix I, 2002 [9], introduces the packet loss robustness factor in the calculation. The version of recommendation G.113, Appendix I, 2007 [10], presents a detailed calculation of the influence of packet loss on the quality of packetized speech signals. The data about the influence of packet loss on speech signal quality in the telephone network of Electric Power Utility (where the level of packet loss is increased) can be found in [11-13].

In this short paper, we will consider sending short telephone messages in public networks [14]. In the packet part of the public telephone network of the Telekom Serbia the negligible influence of delay and echo is achieved, and a compressor does not exist. The speech signal flow rate is 64kbit/s.

Sending a priori defined voice messages increases in modern networks. These are short voice messages. The number of different possible messages is high, and one voice message is obtained as a combination of a number of different shorter messages. Each voice message can be thus treated as a combination of messages. As messages are short, the quality of voice signals can differ greatly from one connection to another and it can deviate significantly (it can be much better, or much worse) in these connections, from the average voice quality.

This paper deals with the method for calculating the voice connection quality in the equipment for short voice messages sent over the packet part of the IP network.

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 [1]. The main equation of the E-model is:

$$R = R0 - I_s - I_d - I_{e-eff} + A \qquad (1)$$

where is [15]:

- 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;

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- 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;
- $I_{e\text{-}eff}$ - effective loss impairment factor 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 $I_{e\text{-}eff}$ can be expressed by equation [1]:

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

where is [12]:

- 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. equation (2) becomes:

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

As P_{pl} expresses packet-loss probability in percents, its value in (3) can be changed with $100\cdot p$, where p is the mean probability of packet loss.

Method for voice quality determination when sending short messages

The factor $I_{e\text{-}eff}$, calculated using equations (2) and (3), represents the mean value during one connection. The packet-loss probability can be changeable over time, as in, for example, [16]. Short time intervals of 5s or 10s, corresponding to one voice message, can be with great error probability, or they can elapse without errors in packet transmission. The aim of this paper is to determine error probability distribution for voice messages and to determine the corresponding distribution of voice connection quality.

Many different models are used in modelling packet loss (Bernoulli, Gilbert, Gilbert-Elliot, Markov). The survey of possible models can be found, for example, in [17].

The literature dealing with the voice signal quality in VoIP connections considers the connections of longer duration. That is why this quality can be presented in a satisfactory way by its mean value. The presentation of time varying of speech connection quality and its value reported at the end of connection (effect of recency) can be found in [18-19]. However, for short connections the mean value and the effect of recency are not good enough presentations. It is necessary to take into account differences in voice signal quality between various connections. Therefore, we developed a new method for voice signal estimation. Our goals were to take into account differences between different connections and to achieve the desired voice signal quality for a certain percent of connections (usually 95%).

In this paper we suppose that the packet loss is bursty. In the calculations, the following parameters are used:

- n - number of packetization intervals in one message (for example, when the message duration is 5s, and the packetization interval is 10ms, n=500);
- l - length of the lost group of packets; for the calculation we suppose that all lost groups are of the same length, l=1 would mean that packet loss is random;
- k - number of lost groups of packets (each of the duration l) within the scope of n intervals of packetization;
- p - mean probability of packet loss; mean probability of burst packet loss would be p/l.

If the lost packet bursts can appear independently at each place in the series of n packetization intervals, then we can use binomial distribution to express the packet loss. Considering one time interval consisted of these n packetization intervals, we can calculate the probability that k packet groups are lost as:

$$P(n,k,l) = \left(\frac{p}{l}\right)^k \cdot \left(1-\frac{p}{l}\right)^{n-k\cdot l} \cdot \frac{(n-k\cdot(l-1))!}{(n-k\cdot l)!\cdot k!} \tag{4}$$

In this equation the first element $\left(\frac{p}{l}\right)^k$ is the probability that one specific combination of k packet groups of duration l is lost. The remaining $n-k\cdot l$ packets are received and this probability is expressed by the second element $\left(1-\frac{p}{l}\right)^{n-k\cdot l}$, where $1-\frac{p}{l}$ is the probability that one considered packet is received. The total number of combinations of lost and received packets in n packetization intervals is expressed by the third term in the equation.

The cumulative probability that in the sequence of n packetization intervals appear $k \le m$ groups of lost packets, each group of duration l, can be calculated as:

$$P(m) = \sum_{k=0}^{m} P(n,k,l) = \sum_{k=0}^{m} \left(\frac{p}{l}\right)^k \cdot \left(1-\frac{p}{l}\right)^{n-k\cdot l} \cdot \frac{(n-k\cdot(l-1))!}{(n-k\cdot l)!\cdot k!} \tag{5}$$

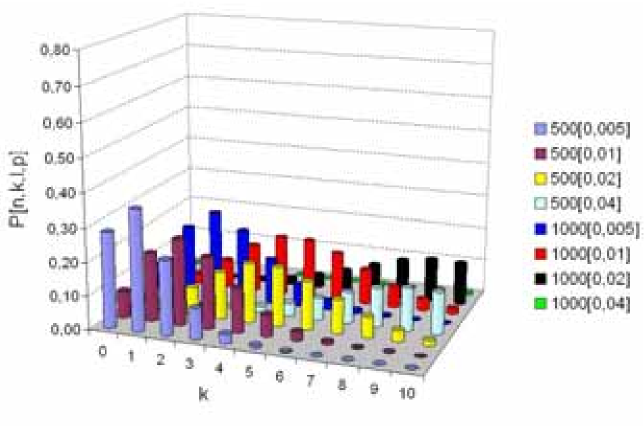
The decrease of voice signal quality in one connection, considering equation (3), will be:

$$I_{e-eff}(n,k,l) = \frac{95\cdot\dfrac{k\cdot l}{n}\cdot 100}{\dfrac{k\cdot l}{n\cdot BurstR}\cdot 100 + B_{pl}} \tag{6}$$

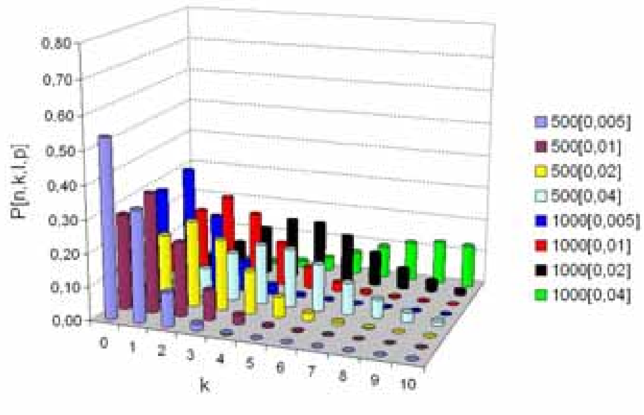
where the packet-loss probability (p) is the ratio of the total number of lost packets ($k\cdot l$) and the number of packetization intervals (n). According to the state mentioned earlier, in (6) we used the fact that P_{pl}=$100\cdot p$.

Discussion of the obtained results

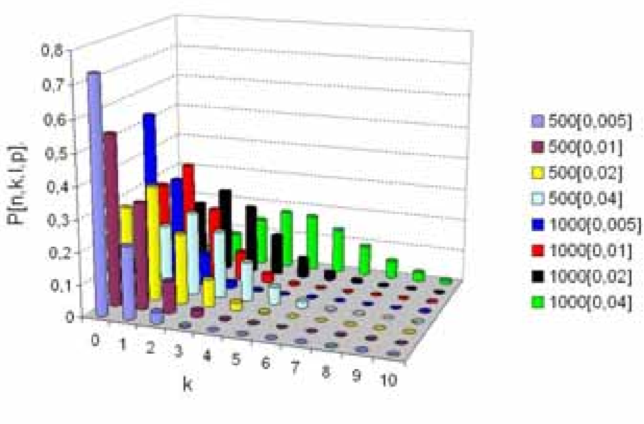
Figs. 1a), 1b) and 1c) present the distribution of the packet loss probability in the function of the number of packet loss bursts (k), if the packets are lost in the groups of l=2 (Fig.1a)), l=4 (Fig.1b)) and l=8 (Fig.1c)). These probabilities are calculated using equation (4). We use the



a) l = 2

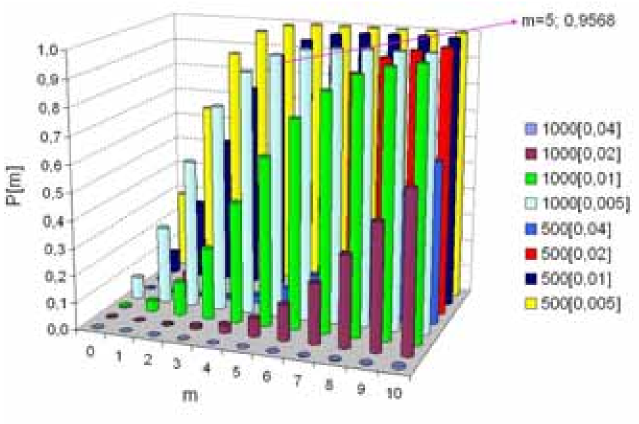


b) l = 4

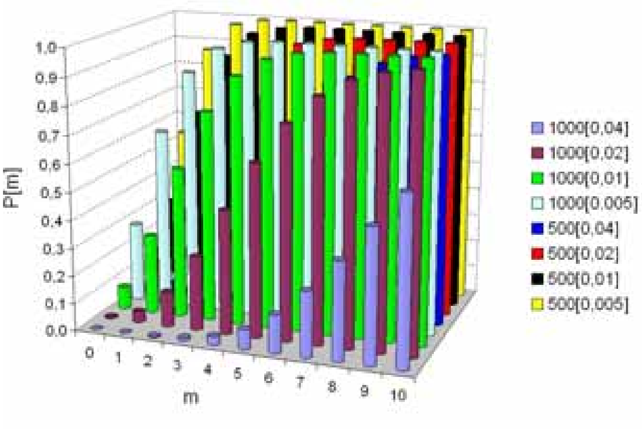


c) l = 8

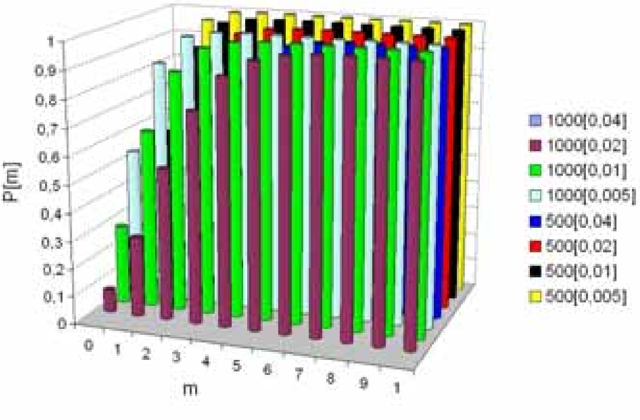
Figure 1. Distribution of the packet-loss probability when packets are lost in groups of 2 (a), 4 (b) and 8 (c)



a) l = 2



b) l = 4



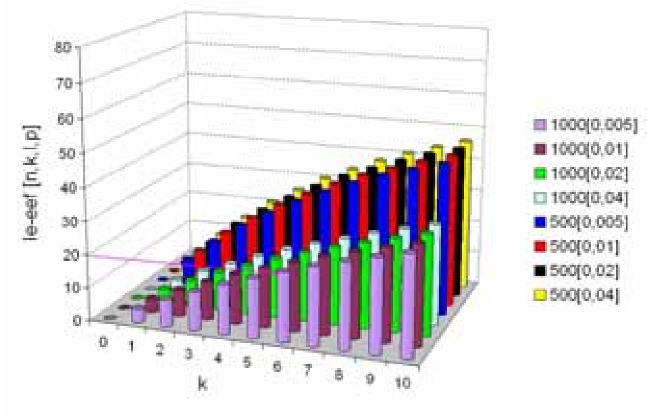
c) l = 8

Figure 2. Cumulative distribution of the packet-loss probability when packets are lost in groups of 2 (a), 4 (b) and 8 (c)

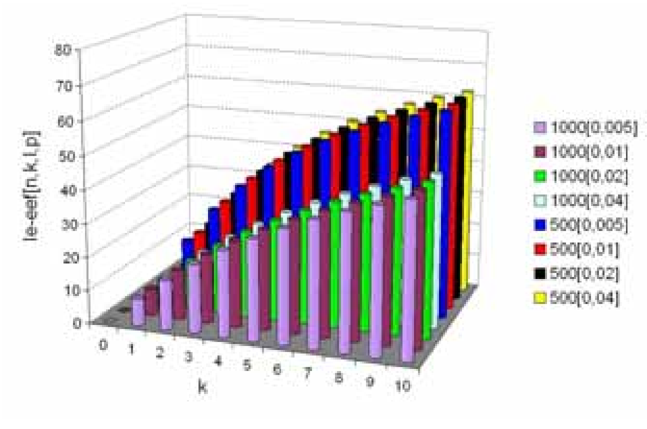
number of packetization intervals (n=500 and n=1000) and the packet-loss probability (from p=0.005 to p=0.04) as the parameters in these figures.

Figs.2a), 2b) and 2c) present cumulative probability, calculated using equation (5), that not more than m packet groups are lost in the function of the maximum number of lost packet groups (m), if the packets are lost in the groups of l=2 (Fig.2a)), l=4 (Fig.2b)) and l=8 (Fig.2c)).

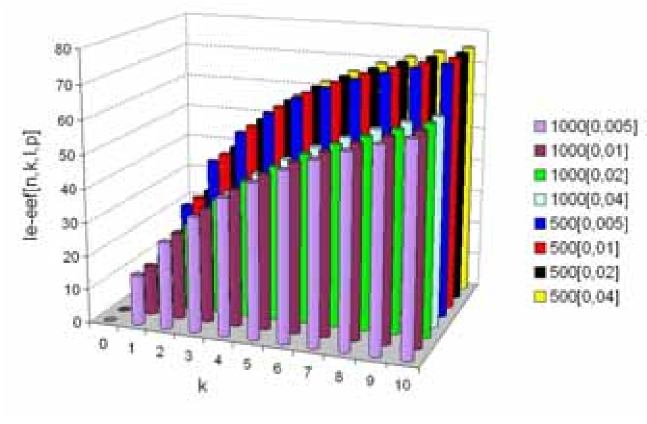
Figs.3a), 3b) and 3c) represent the decrease of voice connection quality in the function of the number of lost packet groups (k), if they are lost in the groups of l=2 (Fig.3a)), l=4 (Fig.3b)) and l=8 (Fig.3c)). This decrease is calculated according to equation (6). As mentioned earlier, the number of packetization intervals and the probability of packet loss are used as the parameters in these figures.



a) l = 2



b) l = 4



c) l = 8

Figure 3. Effective loss impairment factor when packets are lost in groups of 2 (a), 4 (b) and 8 (c)

We can estimate the distribution of $I_{e\text{-}eff}$ in the equipment for sending short voice messages, by using the presented graphics. First, we choose the desired value of $I_{e\text{-}eff}$, which still provides the satisfactory voice connection quality. Then, according to one of the graphics from Fig.3a) or Figs.3b) or Fig.3c), depending on the length of the group of lost packets, we determine the limiting value of the number of lost packet groups, which still provides the desired voice connection quality. And, finally, using one of the graphics in Fig. 2(a), b) and c)), we determine the percent of realized connections, which have the connection quality better than required.

Considering Fig.1 to Fig.3, the following facts are obvious:

- in the case of small values of error probabilities in packet sending, the most probable occurrence is that there are no errors at all, i.e. that there is no decrease of voice signal quality ($I_{e\text{-}eff}=0$);
- distributions of probability of packet loss agree for $n{\cdot}p=const$;
- when increasing the packet loss, $I_{e\text{-}eff}$ also increases; when k increases, $I_{e\text{-}eff}$ tends to the value of 95;
- quality of voice signal from one short-term connection to the other can differ greatly, depending on the number of lost packets.

Numerical example

Let us suppose that in short voice messages, where voice signal is packetizatized, the interval of packetization is n=1000, and that it is necessary to achieve the quality of voice signal, such that effective equipment impairment factor, $I_{e\text{-}eff}$, is lower than 20. The lengths of lost bursts of message packets are l=2, and the packet-loss probability, p, is 0.5%. We want to determine the percent of voice connections which will have satisfactory voice connection quality, i.e. $I_{e\text{-}eff} \leq 20$.

For l=2 in Fig.3 a) the line is drawn through $I_{e\text{-}eff}$=20 until its intersection with the vertical lines which correspond to the number of n=1000 packetization intervals. The corresponding number of lost packet bursts, with the length l=2, is k>5. As the value k must be an integer, we adopt k=5.

The next step is that in Fig.2a), which presents the cumulative probability of packet loss for l=2, the vertical line corresponding to the parameter values m=5, n=1000 and p=0.005 is chosen. On the vertical axis it can be read that the cumulative distribution of the probability of packet loss, when the groups of 2 packets are lost, is approximately 95%, which means that the voice connection quality for 95% connections is desired or better than desired. In other words, connection quality impairment, when groups of 2 packets are lost, is less or equal to 20, $I_{e\text{-}eff} \leq 20$ for 95% connections (Fig.3a)).

Conclusion

In this paper, we presented a new method for voice connection quality estimation. The method is usable, above all, for short voice connections, because the voice quality in different connections can differ greatly from the mean value, obtained for all connections. The method considers burst packet loss and independent appearance of burst packet losses. The possibility to present the variations of voice connection quality in different connections is an advantage of this method.

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Postupak određivanja činioca umanjenja kvaliteta govorne veze pri slanju kratkih poruka preko Interneta

U ovom radu prikazuje se postupak određivanja činioca umanjenja kvaliteta govorne veze u slučaju slanja kratkih govornih poruka preko Interneta. Postupak je primenljiv kad su gubici paketa poruka slučajni i grupni. U radu je postupak ilustrovan za grupne gubitke paketa. Postupak omogućava određivanje raspodele kvaliteta različitih kratkih govornih veza.

Ključne reči: **internet, govorni signal, prenos govora, prenos poruka, razumljivost govora, analiza govora, kvalitet signala.**

Порядок – метод определения фактора нарушения качества голосовой связи при отправке коротких текстовых сообщений голосом через Интернет

Эта статья представляет собой способ – порядок определения факторов понижающих качество голосовой связи при отправке коротких текстовых сообщений голосом через Интернет. Эта процедура применяется, когда потери пакетов сообщений голосом являются случайными и групповыми. В статье показана процедура для групповых потер пакетов. Процедура позволяет определять распределение качества различных коротких голосовых соединений.

Ключевые слова: **интернет, речевой сигнал, передача человеческой речи, передача данных, понятность речи, анализ речи, качество сигнала.**

La méthode pour la détermination du facteur diminuant la qualité de la connexion d'appels pendant l'envoi des messages vocaux par Internet

Dans ce papier on a présenté la méthode pour la détermination des facteurs qui font diminuer la qualité de la connexion d'appels lors de l'envoi de courts messages vocaux par Internet. On peur utiliser cette méthode lorsque les pertes des paquets de messages sont accidentelles ou en groupes. Dans ce travail la méthode est illustrée pour les pertes des paquets en groupes. Cette méthode permet la détermination de la distribution de la qualité chez les différentes brèves connexions d'appels.

Mots clés: **Internet, signal de parole, transmission de parole, transmission des messages, compréhension de parole, analyse de parole, qualité du signal.**