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About Delay Loss Equiquality Characteristics in Packet Telephony

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Original scientific paper

In this paper we present the combined influence of packet delay and packet loss on speech quality in packet telephony. We calculate equiquality lines, which demonstrate the joint influence of these factors on the packetized signal quality. The use of equiquality lines to determine the optimal size of de-jitter buffer is illustrated by couple of numerical examples.

Key words: Packet telephony, E-model, Packet loss, Packet delay

O ekvikvalitetnim značajkama kašnjenja i gubitaka paketa u paketskoj telefoniji. U radu se analizira združeno djelovanje kašnjenja i gubitaka paketa. Izračunate su ekvikvalitetne linije koje pokazuju utjecaj ovih čimbenika na kvalitetu paketiziranog govora. Korištenje ekvikvalitetnih linija za određivanje optimalne veličine *de-jitter* međuspremnika prikazano je na nekolicini primjera.

Ključne riječi: paketska telefonija, E-model, gubici paketa, kašnjenje paketa

1 INTRODUCTION

Transmission of human voice in telecommunication network suffers different impacts. The estimation of these impacts was always subject of international recommendations. The estimation of total impact of all factors in classic telephone technique was very difficult. Permissible limits of different impairment factors were expressed using different units. The limits can be expressed, for example, in: decibels (attenuation) or in (mili)volts (noise), and it was difficult to compare these factors. The only way to determine the common impact of some factors was to experimentally examine their mutual dependence, as it is done in the case of delay and echo, [1-3]. The calculation model for planning the transmission, the so-called E-model [4,5], introduces estimation of all impacts as dimensionless numbers. The parameter R in E-model integrates influence of all factors (delay, packet loss, echo, compression, etc.) by subtracting (or adding) each influence. These influences are expressed by dimensionless numbers. The measures expressed in the calculation of R are: I_d - influence of delay and echo; I_e - influence of compression; I_s - influence of impairments which occur simultaneously with the voice transmission, A - advantage factor. They can be calculated for each specific condition. In this way we can compare some mutually different impacts on the quality of speech signal transmission. Very useful feature of E-model that the estimation of the impact of different components can be added and subtracted is designated additivity. This feature allows not to strictly determine recommended limits of the impact of different factors, but to define the limits for their sum. In this paper we pay attention on the possibility to create equiquality (or isoquality) characteristics, i.e. contour lines, for some pairs of factors, which have common impact on the speech signal transmission. We present this possibility on the example of delay and packet loss in the packetized speech signal transmission.

In the Section 2. we present the common influence of delay and echo. In the Section 3. we consider common influence of delay and packet loss in packet telephone technics. Equiquality (Delay-Loss) lines (D-L lines) are defined in Section 4, and their usage presented in Section 5.

2 DELAY AND ECHO

One of rare cases of considering common impact of two factors on the speech signal in classic telephone techniques is a long time ago published recommendation G.131, [1]. In this recommendation on Fig. 2 appear the curves presenting the same speech signal quality for different pairs of values of delay and echo signal. The measure of echo signal in this recommendation is designated Overall Loudness Rating (ORL) of the echo path, and the measure of the speech signal quality is probability of encountering objectionable echo (1% and 10%). Equiquality lines differ also according to the number of 4-wire circuits. The same equiquality lines are presented in the versions of recommendation G.131 [2] and [3], with following changes in relation to [1]:

- the measure of echo signal is designated Talker Echo Loudness Rating (TELR), and
- the quality measure (1%) is designated "acceptable", and the measure (10%) "limiting" quality.

After the introduction of E-model in the last version of recommendation G.131, [3], Fig 1., the curve "acceptable" is estimated by R factor 74, and the curve "limiting" is estimated by R factor 60. Figures 2a and 2b in [3] present equiquality lines for the values $R = 50, 60, 70, 74, 80$ and 90. Equiquality lines present what limits must satisfy the common impact of echo signal and speech signal delay. These curves provide possibility that we, for example, in the networks with lower delay, allow greater echo signal, i.e. lower TELR value.

3 DELAY AND PACKET LOSS

In packet telephone techniques (VoIP) packet delay, packet delay jitter and packet loss for the packets carrying speech signal present the factors of speech signal quality decrease at the receive side. The problem of packet delay jitter is solved using de-jitter buffer, which changes the impact of the delay variation to the increased total delay and the increased part of lost calls. That is to say, in order to send speech packets on the reproduction (playout) in regular time intervals, fast packets must be delayed and their delay must be brought to the level of slow packets. On the other hand, packets with too great delay, i.e. delay, which can't be covered by de-jitter buffer, are lost (late and lost packets). It is clear that dependencies of delay and packet loss on the de-jitter buffer size are opposite: greater buffer causes increased delay, but decreases packet loss, and vice versa. As it is well known, the speech signal delay causes speech signal quality decrease at the receive side, [4-6].

The packet loss also decreases the speech signal quality, according to the equation (3-29) from [5]:

$$I_{e-eff} = I_e + \frac{(95 - I_e) \cdot P_{pl}}{\frac{P_{pl}}{BurstR} + B_{pl}}, \qquad (1)$$

where the meaning of the variables is:

- I_e - Equipment Impairment Factor; it expresses impairment caused by the use of low bit-rate codec; some values of I_e are given in Table I.1., [8];

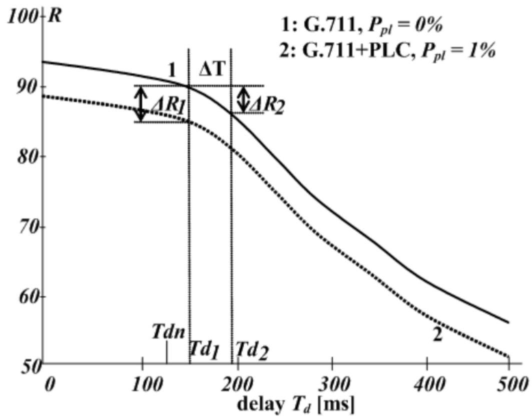

Fig. 1: Influence of packet delay and packet loss on the value of R factor

- I_{e-eff} - Effective Equipment Impairment Factor. It is the value of I_e corrected by the influence of packet loss.
- P_{pl} - Packet-loss Probability, expressed in percents;
- $BurstR$ - Burst Ratio, the factor which expresses the burstiness of packet loss;
- B_{pl}- Packet-loss Robustness Factor; it is stated, [1], Subsection 3.5, that B_{pl} has codec specific value; some values of B_{pl} are suggested in [7].

Calculation of value $BurstR$ is explained in [5].

It is clear that the demands for delay decrease and packet loss decrease are opposite when considering de-jitter buffer, Fig. 1.

In Fig. 1 we present one example of the dependence of speech signal quality, coded using G.711 coder, without network packet loss, according to E-model. This is the dependence of R factor, from the total delay between talker and listener, $R(Td)$, curve 1 (full line). In order to remove the impact of variable delay, we can use one of two de-jitter buffers on the receive side: BF1 with small and BF2 with great delay. The sum of network delay (Tdn) and the delay of buffer BF1 is Td_1, and the sum of network delay and the delay of buffer BF2 is Td_2. If we use buffer BF1, the amount of late and lost packets is 1% ($P_{pl} = 1\%$).

If we use buffer BF2, practically there is no loss of packets, ($P_{pl} = 0\%$). Sensitivity of G.711 coder on packet loss is the reason why Packet Loss Concealment (PLC) is used. The dependence $R(Td)$ in the case of speech signal coded using G.711 coder, if packet loss is 1% and PLC exists, is presented by dotty line, curve 2, in Fig. 1. The speech signal quality decrease on the receive side ΔR_1 is caused by packet loss. In the case of buffer BF2, quality

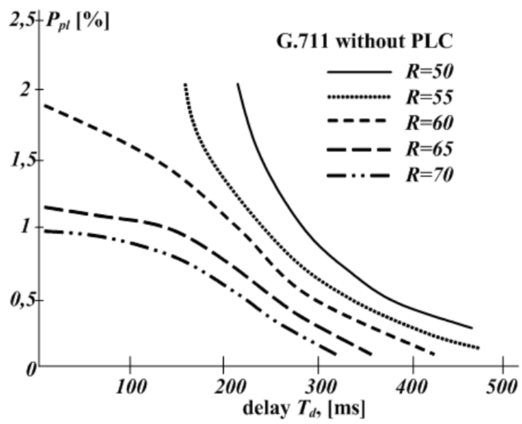

Fig. 2: Equiquality lines for G.711 coder without PLC

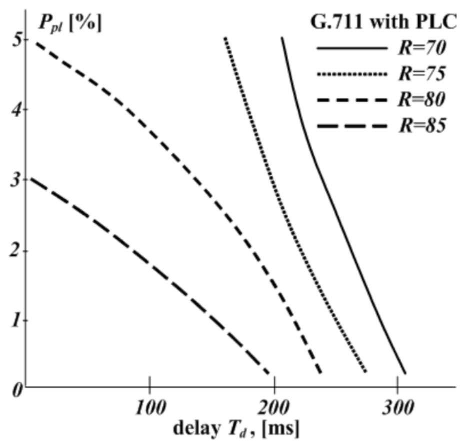

Fig. 3: Equiquality lines for G.711 coder with PLC

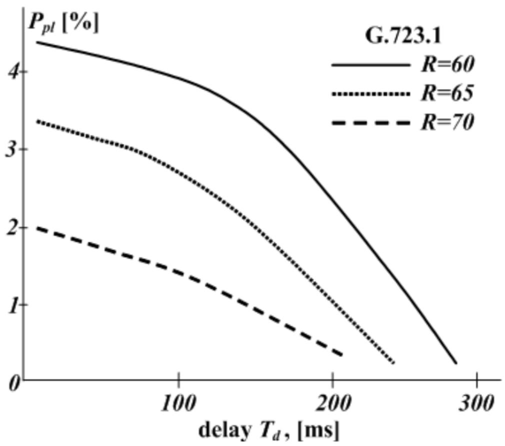

Fig. 4: Equiquality lines for G.723.1 coder

decrease ΔR_2 is caused by the increased delay. The decision whether to use buffer with smaller or greater delay is solved taking into account the relation between the two values of speech signal quality decrease (ΔR_1 and ΔR_2).

The relation between ΔR_1 and ΔR_2 depends on the slope of the curves and it is different for the different delays of the network. In order to determine the optimal delay in de-jitter buffer, we suggest creating of graphs with equiquality lines, which present pairs of delay and packet loss values with the same speech signal quality on the receive side. In the following part of the paper these lines are designated equiquality D-L lines.

4 EQUIQUALITY D-L LINES

Equiquality D-L lines can be easily constructed from lines $R = f(Td)$, which are presented for the constant values of P_{pl}. The sources for determination of D-L lines are: equation (1), tables from [8] and diagrams from [9]. The smallest quantity of data exists for the impact of packet loss on speech signal quality in the case of G.711 coder without PLC, Fig. 2. Coder G.711 is intended for digital networks with channel switching and it has not included the mechanism for the protection from the loss of speech signal part (packets).

The great impact of packet loss on the speech quality in this case is obvious from Fig. 2. For example, if packet loss is 1%, and the expected quality is $R \geq 65$, the total delay can't be greater than 150 ms. If we define again $R = 60$ as the "limiting value", as in [2, 3], it can be seen that this value can not be achieved for packet loss $P_{pl} > 2\%$. "Acceptable value" $R = 74$ can not be achieved in this case for $P_{pl} > 1\%$, despite the value of delay.

Equiquality lines for coder G.711 with PLC are presented in Fig. 3. It is obvious from Fig. 3. that "acceptable value", i.e. $R = 74$ can be achieved for very high percent of packet loss, even if total delay is 200ms. It can be, also, seen that satisfactory quality ($R \geq 80$) can be achieved when $P_{pl} \leq 3\%$ and $Td \leq 150$ ms.

Equiquality lines for coder G.723.1 are presented in Fig. 4. Equiquality lines for coder G.729+VAD (Voice Activity Detector) are presented in Fig. 5.

5 USE OF EQUIQUALITY LINES

Equiquality D-L lines can be used for the determination of de-jitter buffer size when sending packetized speech signal in the following way: we consider the distribution of the probabilities of packet delay, T, on the receive side. The delay time consists of constant and variable part. The variable delay part can be compensated using de-jitter buffer. The important question is: what delay has to be compensated by this buffer? If the buffer is smaller, the impact of packet loss will be greater, and if the buffer is greater, the impact of increased delay will be dominant.

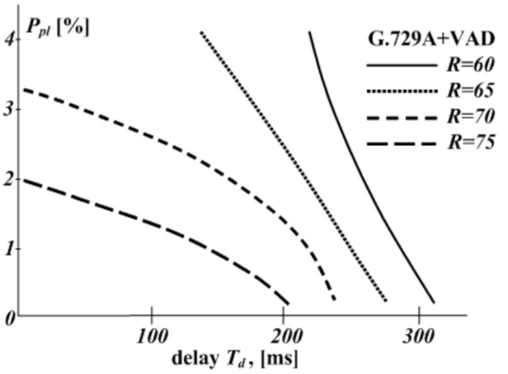

Fig. 5: Equiquality lines for G.729+VAD coder

Let us suppose that cumulative probability distribution of total packet delay, $F(Td)$, is known, such that:

$$F(Td) = P\{T \leq Td\}, \tag{2}$$

where $P\{T \leq Td\}$ means probability that random packet delay, T, is less than Td. Let us, also, suppose that:

$$Q(Td) = 1 - F(Td) = P\{T > Td\}. \tag{3}$$

The sketch of the functions $F(Td)$ and $Q(Td)$ is presented in Fig. 6. The distribution $Q(Td)$ presents which part of packets will be lost if the buffer is limited, i.e. we allow total delay to be at most Td.

Figure 7 presents more precisely the situation in the region of small values of $Q(Td)$ in the case of using G.711+PLC. It is obvious that we get the best speech signal quality for the values P_{pl} and Td where $Q(Td)$ enters the region of the greatest quality.

Let us consider two examples, where the numerical values are selected to illustrate very clearly determination of packet delay (i.e. buffer delay) to obtain greatest speech signal delay. In these examples we use the equiquality lines and the dependence $Q(Td)$ to determine the optimal buffer size, i.e. to obtain the greatest quality of signal on the receive side.

Example 1: Speech packets are sent without compression and with Packet Loss Concealment. The minimal delay through network is about 80ms, Fig. 7. This figure is obtained from Fig. 3. by adding the distribution $Q(Td)$.

It is obvious from Fig. 7. that small buffer (i.e. if total delay is $Td \approx 100$ ms) causes the great packet loss (about 4%), and the speech signal quality about $R = 80$. The great buffer, $Td \approx 250ms$, causes small packet loss (about 0.2%), but produces too great delay, and the quality is again about $R = 80$. It can be seen from the Fig. 7 that the greatest speech signal quality ($R \approx 84$) can be obtained if the buffer size limits total delay Td on the value 130 ms, i.e. if buffer delay is about 50 ms.

Example 2: For the same packet network as in example 1 and the same distribution Td, but if compressor G.729A+VAD is used, it can be seen that the greatest speech signal quality ($R \approx 74$) will be obtained when the total delay is about 170 ms, i.e. if the buffer delay is 90 ms, Fig. 8.

As can be concluded from these examples, equiquality D-L lines are very suitable mean for the determination of delay value in the receiving de-jitter buffer.

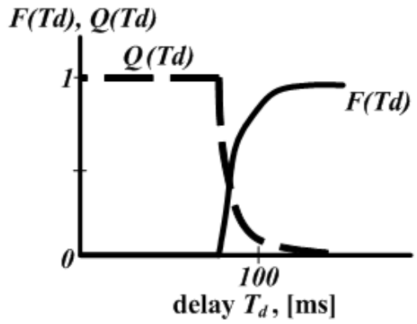

Fig. 6: Graphs of the functions $F(Td)$ and $Q(Td)$

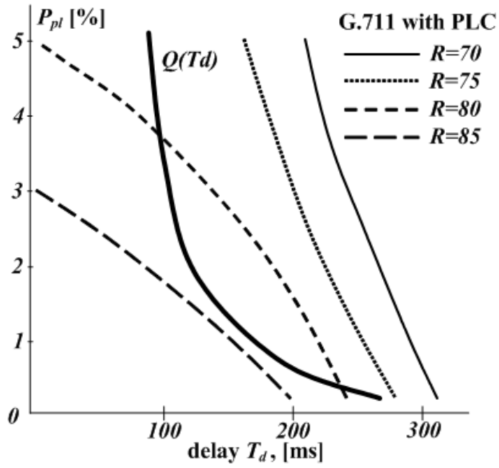

Fig. 7: Determination of optimal value of packet delay to obtain greatest speech signal quality in the case of G.711 coder with PLC

6 CONCLUSION

Many factors have the impact on the quality of transmitted packetized speech signal. Due to the additivity of factors which have the impact on the speech signal quality according to E-model, it is possible to calculate equiquality lines. These lines present common and, in the sum, same impact of several factors. This feature offers, also, the possibility to change some impacts at the expense of other impacts and to estimate how to improve in this way the total

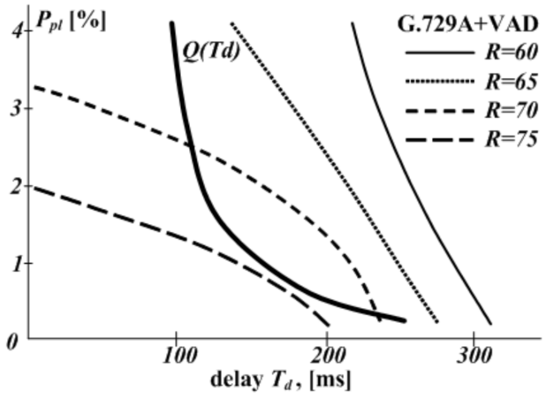

Fig. 8: Determination of optimal value of packet delay to obtain greatest speech signal quality in the case of G.729A+VAD coder

impact. In this paper we present the common impact of packet loss and packet delay on the speech signal quality at the receive side. We present how equiquality lines can be used to choose for one network the pair of values (packet delay and packet loss) to obtain the greatest signal quality on the receive side. Thus we determine the optimal size of de-jitter buffer in packet telephone network.

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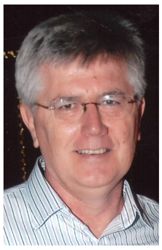

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