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How to estimate the impact of burst packet loss on speech quality in packet network of Electric Power Utility

***Abstract**. In this paper influence of burst packet loss on the quality of telephony speech signal in the packet network of Electric Power Utility (EPU) is calculated. Due to transients, the burst packet loss in EPU network is more probable than random packet loss. We can calculate the impairment factor in two ways: as the mean value of impairments in all nodes, or as the impairment for the mean value of burst duration in the network. Jensen's inequality is used to prove that the second method of calculation gives the pessimistic (conservative) estimate of the speech signal impairment.*

***Streszczenie.** W artykule analizuje się wpływ utraty pakietu na jakość telefonicznego sygnału w pakietowej sieci EPU. Rozerwanie pakietu jest bardziej prawdopodobne niż przypadkowa strata pakietu. Analizowano współczynnik zniekształcenia na dwa sposoby: jako średnia wartość zniekształceń we wszystkich węzłach lub średnia wartość trwania zniekształcenia. (**Jak określić wpływ utraty pakietu na jakość rozmowy w pakietowej sieci EPU**)*

Keywords: Packet Network, E-model, Electric Power Utility, Jensen's Inequality.
Słowa kluczowe: sieć pakietowa EPU, utrata pakietu..

Introduction

The modern public and private telecommunication networks are migrating to IP solutions. The network of Serbian Electric Power Utility (EPU) is also migrating to IP [1].

The method for the calculation of the influence of power system on the quality of IP telephone signal in telephone network of Serbian EPU is presented in this short paper. The calculation is based on the special characteristics of EPU and on the classic mathematical knowledge.

The packet network of EPU is similar to classic packet network. However, it has, at least, three features, which differentiate it from other packet networks:
- the equipment is located in the power system objects [2];
- the influence of power system on the telecommunication system is significant because power or overvoltage disturbances in the power system cause packet loss in the telecommunication system;
- the overvoltage disturbances are of long duration [3], so the packet loss is almost always bursty.

In one telephone connection, established by the network of EPU, each node can produce burst packet loss of different duration.

Different influences (speech signal delay, echo, speech signal compression, packet loss, transcoding) may reduce the quality of the packetized speech signal. This paper deals with the packet loss only. We explain the method how to estimate the speech signal quality degradation, caused by the burst packet loss, if the mean duration of packet loss varies from node to node.

Model, designations and assumptions

It is necessary to estimate the speech signal quality of the telephone connection in the packet network of EPU (E-model) [4]. The connection passes through k network nodes. The speech signal is not compressed (G.711), and speech signal is transmitted without significant delay and echo.

The power equipment and the telecommunication equipment are located in all network nodes. The power equipment in each network node causes disturbances in the packet network, i.e. burst packet loss. The occurrence of disturbances is random, i.e. disturbances generate Poisson process. The mean number of packets in one burst of impaired (lost) packets in the node i is n_i, $i = 1, 2,..., k$. The overall packet loss probability is expressed by P_{pl} [4].

The robustness of speech signal to packet loss is expressed by B_{pl} and it depends only on the type of coder and compressor. For the G.711 coder this value is 4.3 – 4.8 [5].

BurstR – (*Burst Ratio*) is the factor which expresses the influence of packet loss burstiness on the speech signal impairment. When packet loss is random, the value of *BurstR* is *1*, and when packet loss is bursty, the values of *BurstR* are greater than *1*. The value of *BurstR* can be calculated using the equation (3-30) from [4]:

$$BurstR = \frac{1}{p+q} \tag{1}$$

where p and q are the transition probability between a "found" state (correct packets received) and a "loss" state (incorrect packets received, i.e. packets loss) in the Gilbert model with two states. It can be shown easily that the value of *BurstR* is proportional to the number of lost packets in one burst.

Calculation of speech signal impairment

The speech signal impairment, caused by the compressor and burst packet loss, can be calculated using the equation (3-29) from [4]:

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

where I_e is the equipment impairment factor due to compressor and $I_{e\text{-}eff}$ (*Effective Equipment Impairment Factor*) is the value of I_e corrected by the influence of (burst) packet loss.

In the case described in this paper the equation (2) is simplified, because the compression doesn't exist. That's why the degradation of speech signal quality caused by the compression doesn't exist, $I_e = 0$, and the equation (2) becomes:

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

In the packet network of EPU, the connection passes through several (k) nodes where burst packet loss with different mean number of lost packets in one burst is generated. We suppose that each network node of one connection has different mean number of lost packets. Due to this fact, in each network node the different value of *BurstR* will exist. If $BurstR_i$ is the value of *BurstR* in network node i, then the mean value of *BurstR* for the entire connection is:

$$BurstR_m = \sum_{i=1}^{k} v_i \cdot BurstR_i \tag{4}$$

where v_i (i=1,2,...,k) is the part of lost packets in the node i. This variable presents also the probability, and, so, the sum of all v_i values is equal to 1.

The speech signal impairment of the first kind, $I_{e\text{-}effm1}$, can be calculated using this mean value $BurstR_m$:

$$I_{e\text{-}effm1} = \frac{95 \cdot P_{pl}}{\dfrac{P_{pl}}{BurstR_m} + B_{pl}} \tag{5}$$

On the other hand, the mean value of speech signal impairment of the second kind, $I_{e\text{-}effm2}$, can be calculated as the mean value of all impairments using equation:

$$I_{e\text{-}effm2} = \sum_{i=1}^{k} v_i \cdot I_{e\text{-}effi} \tag{6}$$

where $I_{e\text{-}effi}$ is the impairment in the network node i with $BurstR = BurstR_i$:

$$I_{e\text{-}effi} = \frac{95 \cdot P_{pl}}{\dfrac{P_{pl}}{BurstR_i} + B_{pl}} \tag{7}$$

The main question of this paper is: which estimate of the speech signal impairment, caused by burst packet loss, is better: $I_{e\text{-}effm1}$ or $I_{e\text{-}effm2}$. This question can be answered bearing in mind two facts:

- The first one is the shape of the function expressed by the equation (3). This function represents the speech signal impairment depending on the packet loss burstiness. The equation (3) can be written in the simpler form:

$$I_{e\text{-}eff}(BurstR) = f(x) = \frac{A}{\dfrac{B}{x} + C} \tag{8}$$

The function $f(x)$ has the following characteristics:
1. The function is, obviously, monotonically increasing.
2. The function is continuous and two times differentiable.
3. The second derivation of the function is negative:

$$\frac{d^2 f(x)}{dx^2} = -\frac{2 \cdot A \cdot B \cdot C}{(B + C \cdot x)^3} < 0 \tag{9}$$

because all the values (A, B, C, x) in (8) are positive. According to the characteristics of function $f(x)$ and equation (9), it is clear that the function (3), i.e. (8) is concave relative to x axis.

- The second important fact is explained by Jensen's inequality [6]. Jensen's inequality determines the relation between the mean value of real function, $f(x)$, and the mean value of independent variable, x.

Let us observe the function $f(x)$ on the interval $a \le x \le b$. If the function is concave between a and b, i.e. if:

$$f\left(\frac{x_1 + x_2}{2}\right) \ge \frac{f(x_1) + f(x_2)}{2}, \quad a \le x_1 \le b, a \le x_2 \le b \tag{10}$$

then:

$$f\left(\frac{x_1 + x_2 + ... + x_n}{n}\right) \ge \frac{f(x_1) + f(x_2) + ... + f(x_n)}{n}, \quad a \le x_1 \le b, a \le x_2 \le b, ..., a \le x_n \le b \tag{11}$$

i.e. the value of the function for the mean value of independent variable is not less than the mean value of function for the observed values of independent variable. (The reverse inequality stands for convex function.)

According to these:

$$\frac{95 \cdot P_{pl}}{\dfrac{P_{pl}}{\sum_{i=1}^{k} v_i \cdot BurstR_i} + B_{pl}} = \frac{95 \cdot P_{pl}}{\dfrac{P_{pl}}{BurstR_m} + B_{pl}} = I_{e\text{-}effm1} \ge \sum_{i=1}^{k} v_i \cdot I_{e\text{-}effi} = I_{e\text{-}effm2} \tag{12}$$

We can conclude that the impairment of the speech signal quality, calculated for the mean number of lost packets in the bursts, is not less than the mean value of impairments calculated over all nodes. It means that the estimate $I_{e\text{-}effm1}$, i.e. equality (5), gives the conservative estimate of speech signal impairment, i.e. the estimate which is on the safe side.

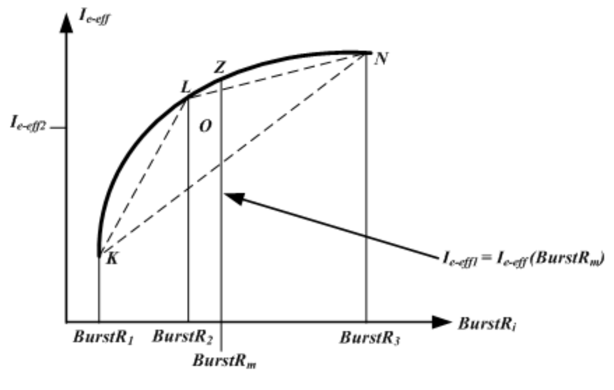

Fig.1. Illustration of the method for *BurstR* calculation in the network of three nodes figure inserted into the text

Figure 1 presents graphically the equation (12). According to the measurements, the lost packets appear in one of three different groups.

The probability that n_i packets are lost in the burst is v_i ($i = 1, 2, 3, v_1 + v_2 + v_3 = 1$). If n_i packets are lost in the burst, the corresponding value of *BurstR* is $BurstR_i$. The mean value of *BurstR* ($BurstR_m$) can be calculated using equation (4).

The value of the function $I_{e\text{-}eff}(BurstR_m) = I_{e\text{-}effm1}$ is presented by the point *Z*.
The mean value $I_{e\text{-}effm2}$, expressed by the equation (6), is represented by one point from the area *O*, surrounded by the lines *KL*, *LN* and *NK*. This value is always under the graph of the function $I_{e\text{-}eff}(BurstR_i)$, except in the trivial case when 2 of 3 values v_i ($i = 1, 2, 3$) are equal zero.

Example: Let us consider the packet network of EPU where the uncompressed speech signal (G.711) is transferred without packet loss concealment ($B_{pl} = 4.8$). Because of the different influence of power system, there are $k=3$ kinds of burst packet loss: with $n_1 = 2$, $n_2 = 6$ and $n_3 = 8$ lost packets in burst. The overall packet loss is $P_{pl} = 1\%$ and all three groups equally participate in packet loss. As the packet loss rate is small, it can be shown that $n_i \approx BurstR_i$, $i = 1, 2, 3$.

According to equation (7):
In the case of bursts of 2 packets, the value $I_{e\text{-}eff1} \approx 17.92$.
In the case of bursts of 6 packets, the value $I_{e\text{-}eff2} \approx 19.13$.
In the case of bursts of 8 packets, the value $I_{e\text{-}eff3} \approx 19.29$.

The mean value of *BurstR* is:

$$BurstR_m = \frac{1}{3} \cdot \sum_{i=1}^{3} BurstR_i = 5.33$$

From the equation (12) follows:

$$I_{e\text{-}eff}(BurstR_m) = 19.05 > 18.78 = \frac{1}{3} \cdot (I_{e\text{-}eff1} + I_{e\text{-}eff2} + I_{e\text{-}eff3})$$

We can conclude that the first estimate of the packetized speech signal impairment, which uses the mean value of *BurstR* ($BurstR_m$), is conservative. It is also very approximate to the real value because of the light curvature of function $I_{e\text{-}eff}(BurstR)$.

Conclusion

Packet telephone network of EPU is under the influence of power system and the packet loss is bursty. The speech signal impairment can be calculated using two methods: using the mean value of the impairment and using the mean values of the number of lost packets. The main contribution of this paper is the proof that the second method gives the results, which are on the safe side, i.e. they are conservative. Jensen's inequality is used to prove that this calculation gives the worst results in the calculation of speech signal quality.

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