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Peculiarities of Testing the Impact of Packet Loss on Voice Quality

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Introduction

In daily life speech is one of the major communication applications. In telecommunications systems the significant growth in the volume of multimedia and data transmission do not reduce the importance of voice and speech. The phone conversation is the process where information is transmitted. Voice carries not only semantic information. Individual voice properties are of no less importance. Any part of communication may identify the other using redundant information in her voice. Voice also transmits emotional state of a person, his reactions to details in the conversation.

The human speech is an analogue signal $v(t)$ that varies slowly in time. Current communication means digitize voice signal, divide it into segments of length T, and encode these segments (Fig.1). The i-th segment of voice signal $v(t)$ is depicted as code V_i and transmitted as i-th data packet V_i. In other words data packet V_i convey i-th frame of voice signal $v(t)$

$$V_i \Rightarrow v(t) \text{, where } t_0 + (i-1)T \le t \le t_0 + iT . \qquad (1)$$

Sophisticated speech coding techniques created for phone communication purposes are now viable: G729, G723.1, GSM 06.10, AMR-*n*. It is obvious that different codes correspond to differen codecs. Every speech coder in general is characterized by two main attributes: bit rate and measure of degradation of the coded speech signal named as quality.

The quality of transmitted speech is dependent on many factors including bandwidth, signal level, echo, delay, signal-to-noise ratio, codec type. In wireless networks such as GSM, WCDMA, and voice-over-IP, there are also additional factors affecting speech quality such as packet loss. The packet loss is particular distortion, where loss of i-th data packet V_i, is lost i-th frame of voice signal $v(t)$.

Concepts of speech quality evaluation have changed with the development of new phone systems. Review of papers concerning speech quality is given in [1]. Relations between spectral bandwidth, signal and noise levels and speech quality were established [1].

Signal waveform generated by modern coders, may differ significantly from original signal. Nowadays quality of the speech in the phone system is understood as an interaction between an acoustic signal and a listener. The most reliable way to measure the voice quality is through use of subjective testing, i.e. a group of qualified listeners are asked to score the speech they just heard on to a scale from 1 to 5. The average of these scores is the subjective Mean Opinion Score, MOS [2]. This method of speech quality assessment is very expensive and time consuming. The problem with MOS is that this data is hard to verify and most of the results are not repeatable. Due to these reasons models have been developed to identify audible distortions through an objective process based on human perception. Objective methods are implemented in computer programs to automate speech quality measurement in real time. The survey of published works on speech quality assessment has been presented in the paper [1].

PESQ [3], the ITU-T's Perceptual Evaluation of Speech Quality is among the most widely used objective assessment tools in telecommunications and IP networks. PESQ was developed by combining the two advanced speech quality measures PSQM+ and PAMS.

The purpose of this paper is twofold: to investigate the impact of lost packets on distortions of voice signal and to analyze the peculiarities of PESQ by estimating the impact of lost packets on voice quality.

Concealment of Packet Loss

Packet is set to be a lost packet, if it has failed cyclic redundancy code (CRC) check. When speech frame is lost, some voice information in that signal part is lost, too. If the decoder simply mutes the output for this time interval (zero stuffing) the resulting distortion can be very disturbing for the listener. This type of muting creates impulse disturbances and high degradation of quality.

Modern decoders are supplemented with a packet loss concealment algorithm to fill the gaps in the output speech signal. Various methods have been proposed for generation of an estimate of the lost speech segments included in the lost packets. The simplest method assumes that the difference between two consecutive speech frames is small. Hence, the lost packet is replaced by repeating the previous packet. This method degrades quality more compared to the more advanced methods.

For investigation, we use AMR codec that was developed for GSM cellular systems and has been chosen as the mandatory codec for 3G cellular systems. AMR packet loss concealment includes active and passive forms: parameter repetition/substitution, and attenuation/muting [4]. The AMR packet loss concealment function changes not only signal in the lost frame, but also signal in some subsequent correctly transmitted frames. Fig.1 shows original voice signal $v(t)$ and by AMR decoder from time point $(i\text{-}1)T$ substituted signal $v^*(t)$, because i-th frame was lost.

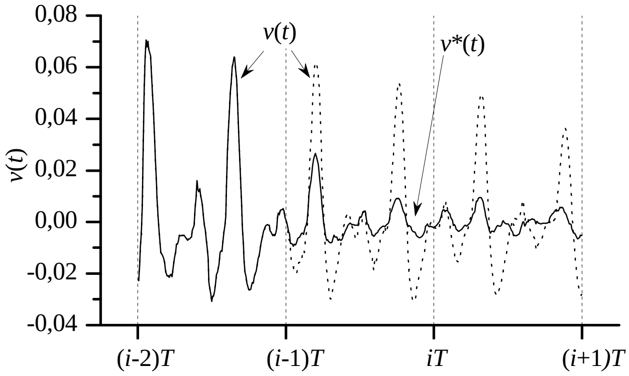

Fig. 1. Original and substituted voice signals

Generally speaking, the packet loss concealment technique transforms signal in lost frames into artificial signal, which is similar to the last frame signal. This way clipping is avoided. Such waveform becomes similar to natural voice signal and the same quality evaluation means may be applied as used for other distortion types [4 - 6].

Influence of lost frame on subsequent signal

Investigation of the influence of lost frame on waveform of speech signal was performed using AMR coder. This way we obtained original $v(t)$ and degraded signals $v^*(t)$. For testing the similarity of waveform $v^*(t)$ to waveform $v(t)$ the short term correlation coefficients $\rho_{i,j}$ were calculated

$$\rho_{i,j} = \frac{\int_{(i-1+j)T}^{(i+j)T} v(t)\cdot v^*(t)\cdot dt}{\sqrt{\int_{(i-1+j)T}^{(i+j)T} v^2(t)dt \cdot \int_{(i-1+j)T}^{(i+j)T} v^{*2}(t)dt}}, \qquad (2)$$

where index i labels a number of lost packet and index j indicates number of substituted packets. Fig. 2 shows some examples of variations of $\rho_{i,j}$ as function of index j calculated for several sentences where one frame is lost.

As we see from the Fig. 2 the influence of lost frame is distinct for various sentences. The durations of this influence also can be quite different.

The nominal continuance of influence on waveform of lost frame was estimated as time JT where J is a maximum number j of substituted frames when

$$\rho_{i,j} \le 0.95 \ \text{for} \ \ j \le J\,. \qquad (3)$$

Executing such calculations for one thousand randomly selected segments of voice we obtain the mean rate of continuance JT of influence of lost frame. Fig. 3 shows this histogram. As we see the continuance of influence of lost frame mostly is 1 or 2 frames but sometimes may be 10 and more frames.

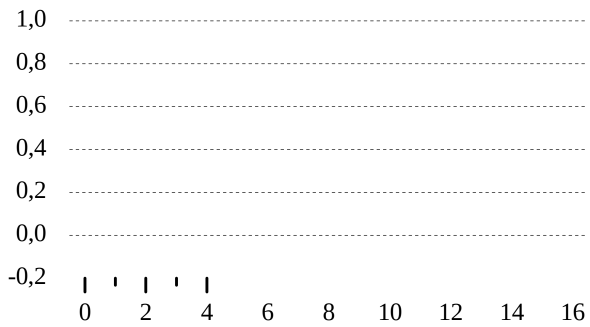

Fig. 2. Examples of variations $\rho_{i,j}$ as function of index j

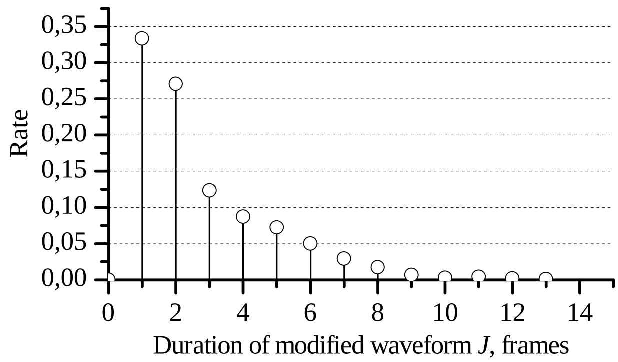

Fig. 3. Influence of lost frame

On peculiarity of PESQ algorithm

Subjective listening tests according to MOS must be performed using meaningful, short (2 – 3 s) sentences [2]. For example, 2 seconds sentence is divided into 100 frames by GSM coder. When in test sentence was lost only one frame there are 100 different positions. Because every lost frame at different position may have different impact on voice quality in this case 100 listening tests shall be performed.

When in 2s test time was lost 2 frames there are near 5000 different positions. Obviously, when growing the number of lost frames the required number of listening experiments increases rapidly. It is evident that listening tests of impact of lost frames on voice quality are extremely time-consuming and expensive to use. Therefore some published works presented different MOS scores. For example, 1% of lost frames degrades mean MOS score by 0.07 [5], and in [6] the same 1% of lost frames gives 0.177 degradation.

Currently the most reliable objective voice quality measurement method is considered PESQ [3, 7]. The PESQ scores are calibrated using a large database of subjective tests. In terms of accuracy, i.e. correlation with subjective assessments, PESQ has an advantage over all the other purely objective quality metrics. Benchmark tests in [3] of PESQ have yielded an average correlation of 0.935 with the corresponding MOS values.

The PESQ is accepted as an objective measurement tool that predicts the results of subjective listening tests on telephony systems. This method is implemented in many commercially available testing devices and monitoring systems. The general speech quality evaluation scheme using PESQ is shown in Fig. 4. PESQ uses a sensory model to compare the original, unprocessed signal with the degraded signal from the network or network element.

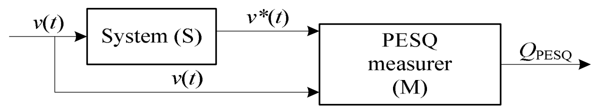

Fig. 4. Voice quality measurement main scheme

The voice signal $v(t)$ is sent to measurer by two ways. To the first input of PESQ measurer is sent signal degraded by system under investigation S. Another way voice signal goes directly to second input of PESQ measurer. The typical system S under investigation consists of encoder, channel model and decoder.

PESQ measurer works using several stages: time alignment, level alignment to a calibrated listening level, time frequency mapping, frequency warping, and compressive loudness scaling. Thus minor, steady state differences between original and degraded are compensated. Later on differences between signals $v(t)$ and $v^*(t)$ are calculated using algorithm which imitates human listening properties.

It is known [8] that the human ear performs a time-frequency transformation. In PESQ this is implemented by a short-term FFT with a window (also this window is called phoneme) size of $T_{\mathrm{ph}} = 32$ ms. Overlap between successive time windows (phonemes) is 50 percent. 20 overlapped phonemes amount one $T_{\mathrm{sl}} = 320$ ms long syllable (Fig. 5).

At the first step by integration over frequency the measure of the perceived disturbance D_{n} and the measure of the asymmetric disturbance DA_{n} for every n-th phoneme are estimated.

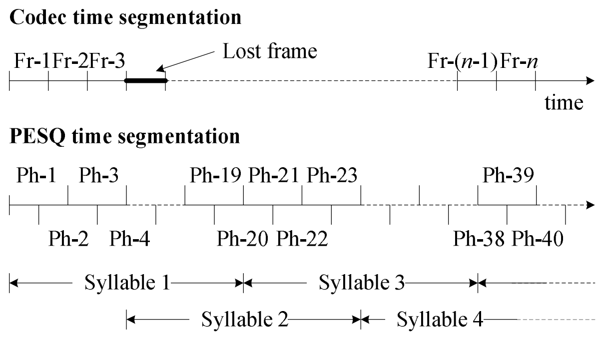

Fig. 5. Time segmentations in coder and PESQ calculator

At the next step phoneme disturbance values D_n and DA_n are aggregated for every syllable using the following formula:

$$L_D = \left(\frac{1}{20} \sum_{n=1}^{20} D_n^6 \right)^{1/6}, \; L_{\mathrm{DA}} = \left(\frac{1}{20} \sum_{n=1}^{20} DA_n^6 \right)^{1/6}. \qquad (4)$$

Finally prosecuted integration over the whole speech signal – the aggregation of syllable disturbances using typical mean square algorithm:

$$d_{\mathrm{sym}} = \left(\frac{1}{N} \sum_{i=1}^{N} L_{\mathrm{D}}^2(i) \right)^{1/2}, \; d_{\mathrm{asym}} = \left(\frac{1}{N} \sum_{i=1}^{N} L_{\mathrm{DA}}^2(i) \right)^{1/2}, \qquad (5)$$

where N – number of syllables in PESQ measurement window. At the last PESQ value is calculated using formula:

$$Q_{\mathrm{PESQ}} = 4.549 - 0.1 \cdot d_{\mathrm{sym}} - 0.0309 \cdot d_{\mathrm{asym}}. \qquad (6)$$

When using quality degradation rating dQ_{PESQ} value is calculated according to formula

$$dQ_{\mathrm{PESQ}} = 4.549 - Q_{\mathrm{PESQ}} = 0.1 \cdot d_{\mathrm{sym}} - 0.0309 d_{\mathrm{asym}}. \qquad (7)$$

PESQ demonstrates good accuracy when main factors affecting speech quality are background noise, coding distortions [3]. Many mentioned factors, if presented, affects voice signal constantly all measurement time and are stationary. Disturbances D_n and DA_n arising from these factors have almost equal value over all syllable duration. Therefore the mean square formula (5) for aggregation of such syllable disturbances is logical.

Distortions of voice signal when packet or frame is lost are particular. These particularities can be structured as follows:

1. Let assume that lost frame is Fr-i (Fig. 5). By substitution of lost frame, decoder may modify the voice transmitted on next j frames;
2. The influence of i-th lost frame may be perceptible at particular time interval only from 0 to jT_{Fr};
3. Under such circumstances, disturbance values D and DA can be definitely not zeros, when thus are calculated for phonemes which are calculated at above-mentioned time interval from 0 to jT_{Fr};
4. Disturbance values D_n and DA_n vary by moving the PESQ measurement start time or moving start time of phonemes;
5. The aggregated disturbance values L_{D} and L_{DA} vary by moving start time of syllables;
6. As consequence the strings of disturbances L_{D} or L_{DA} created by single lost frame or small group of lost frames can not be considered as stationary;
7. Thus application of mean square algorithm for aggregation of syllable disturbances is dubious.

The mentioned particularities of frame loss may impact some uncertainty results concerning measurements of voice quality by using PESQ measure.

Laboratory testbed

For examination of influence of PESQ measurement specifications such as measurement window duration and window start position on estimated dQ_{PESQ} value was developed laboratory testbed. This testbed used in our experiments is shown in Fig. 6.

Voice signal is coded by the encoder E1. Encoded frames are sent to decoder D1 and to channel model. The last one simulates only one frame loss, always at the same given n-th position. The purpose of decoder D2 is decoding sequence of frames (with one loss).

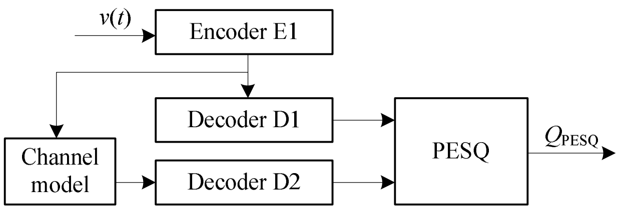

Fig. 6. PESQ examination testbed

PESQ measurer receive reference signal from decoder D1 and degraded signal from decoder D2. This measure scheme eliminates a codec influence on voice quality and leaves the channel influence only.

Influence of measurement window size

In calculation of syllable disturbances d_{sym} and d_{asym} using formula (5) parameter N depends on the length of PESQ measurement window T,

$$N \approx 2 \cdot \frac{T}{T_{\mathrm{sl}}}\ . \tag{8}$$

ITU recommendation P.862 [3] does not recommend strict value of T. Because of this, when lost packets are considered, length of measurement window has a big influence on PESQ measurement result.

Let us say that only one factor affecting voice quality is the packet loss. The one randomly lost frame subject to position of loss may impact from one to three syllables. Other components of (4) i.e. LD(i) and LDA(i) are equal to zero. In this case (if tree syllables are not zero) formulas (5) can be transformed into

$$d_{\mathrm{sym}} = \frac{L_{\mathrm{sym}}}{\sqrt{N}}, d_{\mathrm{asym}} = \frac{L_{\mathrm{asym}}}{\sqrt{N}} \tag{9}$$

where N is a number of syllables and

$$L_{\mathrm{sym}} \cong \left(L_{\mathrm{D}}^2(j) + L_{\mathrm{D}}^2(j+1) + L_{\mathrm{D}}^2(j+2)\right)^{1/2}, \tag{10}$$

$$L_{\mathrm{asym}} \cong \left(L_{\mathrm{DA}}^2(j) + L_{\mathrm{DA}}^2(j+1) + L_{\mathrm{DA}}^2(j+2)\right)^{1/2}. \tag{11}$$

The formula (9) clearly shows that impact of single lost frame on PESQ score depends on measurement window length N. The longer the measurement time T, the smaller quality degradation is noticeable.

For verification of influence of PESQ measurement time experimental simulation using scheme Fig. 6 was performed. The channel model simulates only one lost frame at the same given n-th position. The measurement of dQ_{PESQ} value was performed using different length of PESQ measurement window T. Experiments was performed for two different encoder and decoder systems – AMR-12.2 and G.711.

Voice quality degradation dQ_{PESQ} results tested by PESQ measurer are presented in Fig. 7. dQ_{PESQ} value is calculated according to formula (7). Every dot on the Fig. 7 diagram was calculated using the same voice signal in witch one frame at the same position was lost. The quality degradation of G.711 coded voice is higher than AMR because in G.711 decoder zero stuffing mechanism for lost frame was used.

The results of voice quality degradation presented in Fig. 7 confirm theoretical conclusion followed on formula (9). The impact of lost frame on dQ_{PESQ} score measured by PESQ measurer depends on measurement window length T. Increasing measurement time T decreases value of measured degradation level.

This implication seems not very faithful and trustworthy. When voice frame is lost, some information is lost to. The following frames can't compensate this loss.

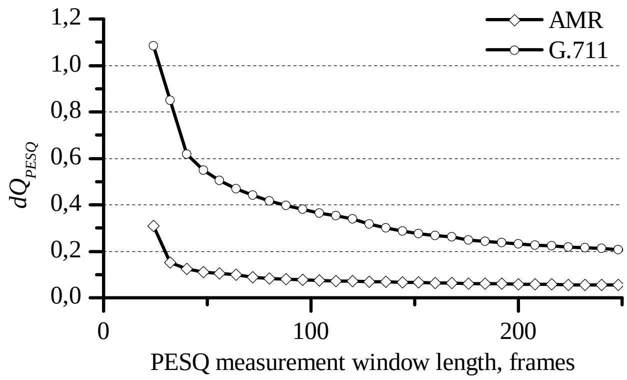

Fig. 7. dQ_{PESQ} score dependence on measurement window length

Therefore we consider that PESQ measurer as presented in ITU recommendation P.862 is not well-chosen for testing the impact of lost frame on voice quality. We propose some modification of this measurer by choice of measurement window size with cover one, to or three PESQ syllables. Such measurement window sizes may be 480, 620 or 960 ms.

Influence of measurement window position

By pursuing experiments with PESQ measurer we have perceived that the impact of lost frame depends not only on duration of measurement window but depends on packet loss position in measurement window. For confirmation or disconfirmation of this hypothesis we performed the special experiment and for this reason special test signal was constructed. The explanation of the test signal we perform using time stamps as presented at Fig. 8.

For test signal three spaces are significant. The first one is silence (t_0-t_1). At the second space, from t_1 to t_3, active voice signal is generated. The third space beyond t_3 is silence too. The duration of active voice space was 480 ms. Packet loss is simulated at time point t_2. In experiments AMR coder was used.

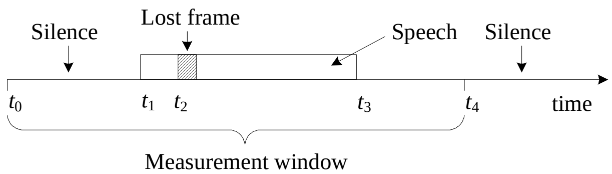

Fig. 8. Scheme for evaluation of influence of window position

The measurement experiment of dQ_{PESQ} value was performed many times. By repeating measurement, the measurement window was moved from start position t_0 to stop position t_1 by steps of 1 ms. The curve A in Fig. 9 shows example of variations of quality degradation dQ_{PESQ}

values. The cause of these fluctuations lies in movement of voice signal disturbance position in respect of overlapping phoneme windows in PESQ algorithm. Obtained dQ_{PESQ} values are pseudo-periodic with period equal to 16 ms or time of overlapping of neighboring phonemes.

It is important to point out that voice quality based on listening tests as obvious are not depending on position of active voice signal space in listening window. The variations of dQ_{PESQ} values as function of position of measurement window in reality causes uncertainty of PESQ measurements. The absolute uncertainty of quality degradation dQ_{PESQ} is quite high. For example presented in Fig. 9 the absolute uncertainty is ± 0.17 .

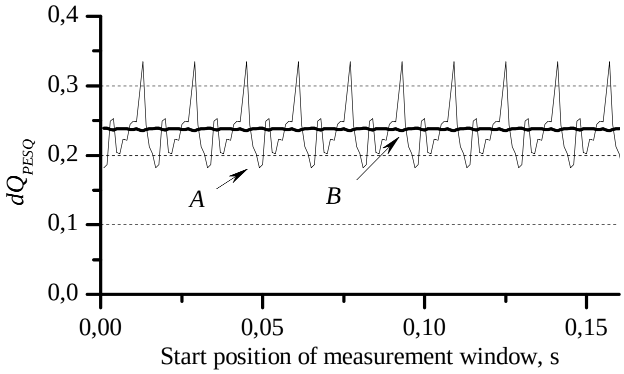

Fig. 9. Variations of quality degradation values

For comparative studies well fitted relative uncertainty with expresses the relative size of the uncertainty of a measurement. The relative uncertainty of dQ_{PESQ} measurements caused by changes position of measurement window and evaluated by several experiments was estimated in the rage $\pm$ (0.2 … 0.55).

Another illustration of influence of measurement window position on dependence of dQ_{PESQ} on location of lost frame is presented at Fig. 10.

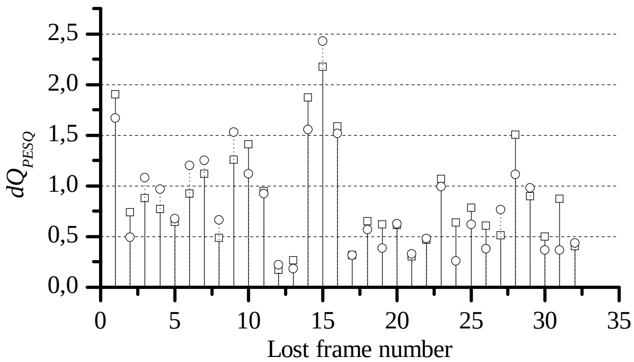

Fig. 10. Dependence of dQ_{PESQ} on location of lost frame.

By simulating two positions of measurement window was used. At Fig. 10 estimated minimum and maximum of dQ_{PESQ} are shown. The averaged uncertainty of this measurement expressed as standard deviation of this experiment is 0.21.

Means for reduction of uncertainty

Striving to reduce the uncertainty of measurements of voice quality degradation caused by start position of measurement window we design special test signal. This design is based on ITU recommendation P.862 proposition "real speech test signal may be constructed by concatenating short fragments of real speech while retaining a representative structure of speech and silence".

Using this we construct prolonged signals which consist of replicated original voice segments (Fig. 11). The purpose of signal prolongation was the attempt for randomization of positions on lost frames in the measurement window.

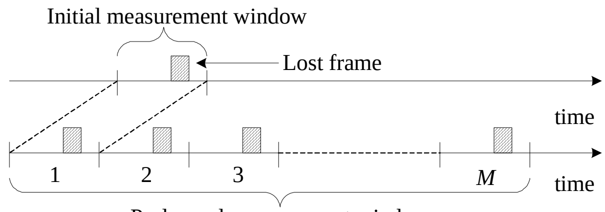

Fig. 11. The structure of prolonged test signal

By composing of prolonged test signal we start from initial measurement window, with sizes as mentioned above can be 480, 620 or 960 ms. For initial window voice signal $v_{\mathrm{w}}(t, T_{\mathrm{w}})$ shall be prepared and the lost frame shall be simulated:

$$v_{\mathrm{w}}(t, T_{\mathrm{w}}) = \begin{cases} 0, & t < 0, \\ v(t), & 0 \le t \le T_{\mathrm{w}} \ , \\ 0, & t > T_{\mathrm{w}} \ , \end{cases} \tag{12}$$

where T_{w} – initial measurement window. The randomization of positions on lost frames in the measurement window will occur when repetition period is not iterative to 16 ms or namely, not iterative to period of variations as shown in Fig. 9. The prolonged composite signal we can write as

$$v_{\Sigma}(t) = \sum_{j=0}^{M} v_{\mathrm{w}}(t - jT_{\mathrm{r}}, T_{\mathrm{w}}) \ , \tag{13}$$

where $T_{\mathrm{r}} = T_{\mathrm{w}} + \Delta T$, and ΔT is not iterative to 16 ms. In our simulations we used $\Delta T = 1$ ms.

For the evaluation of influence of packet loss position in measurement window on quality degradation values dQ_{PESQ} by using prolonged test signal repeated experiment according scheme shown in Fig. 8. The result of this experiment represented in Fig. 9 (line *B*). The uncertainty of this measurement is ten times lower than represented by curve A.

By using prolonged test signal for performing a similar experiment as shown in Fig.10, the averaged uncertainty of measurement was reduced significantly, the standard deviation of this experiment is 0.01.

Conclusions

The packet loss is particular distortion. Some segment of voice signal is distorted under the influence of packet loss. By performing investigations of the impact of packet losses on degradation of voice quality by using PESQ algorithm, the uncertainty of measurement results was pointed out. The results of measurements depend on location of lost packet and measurement window. Some modifications of PESQ algorithm for the reduction of measurement uncertainty using prolonged test signal were proposed in this paper.

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Some packets may be lost or corrupted when voice is transmitted using radio channel. Current voice coders use complex techniques for masking of lost segments. Thus overall speech quality is improved. This work shows experimentally that masking technique changes not only signal corresponding to lost frame, but also subsequent signal. Work investigates properties of PESQ algorithm when quality degradation is due to localized frame errors. The impact of measurement window length and position is investigated using theoretical and experimental methods. It is shown that longer measurement window artificially decreases the impact of localized frame error. Also it is shown that position of measurement window influences measured result. These experiments show high uncertainty may result from using standardized PESQ algorithm for the signals with localized errors. We propose a method for reduction of measurement uncertainty witch is based on artificially constructed voice signal and standard PESQ algorithm. Ill. 11, bibl. 10 (in English; summaries in English, Russian and Lithuanian).

А. Каяцкас, А. Анскайтис, Д. Гуршнис. Особенности тестирования влияния стертых пакетов на качество голоса // Электроника и электротехника. – Каунас: Технология, 2008. – № 2(82). – С. 35–40.

При передаче сигналов голоса в виде пакетов по радиолиниям из-за искажений некоторые пакеты стираются. Современные декодеры способны маскировать стертые сегменты голоса и таким образом улучшить качество голоса. В данной работе показано, что при маскировании стертых пакетов модифицируется достаточно продолжительный отрезок сигнала. В работе также исследуются особенности алгоритма PESQ в случае его применения для оценки качества звука, когда причиной деградации являются стертые пакеты. Теоретически и путем моделирования оценивается влияние ширины и положения окна измерения на результат измерения. Показано, что при увеличении ширины окна измерения искусственно уменьшается влияние стертых пакетов на качество звука. Также показано, что положение стертого пакета в окне измерения влияет на результат измерения. Из-за этого неопределенность измерения качества голоса стандартным способом PESQ становится достаточно большой. Для уменьшения неопределенности измерения предлагается модификация PESQ алгоритма путем синтеза сигнала, составленного из стандартной длины повторяющихся отрезков. Ил 11, библ. 10 (на английском языке; рефераты на английском, русском и литовском яз.).

A. Kajackas, A. Anskaitis, D. Guršnys. Prarastų paketų įtakos balso kokybei vertinimo ypatumai // Elektronika ir elektrotechnika. – Kaunas: Technologija, 2008. – Nr. 2(82). P. 35–40.

Perduodant balsą paketais radijo ryšio linijomis, kai kurie duomenų paketai iškraipomi ir ištrinami. Šiuolaikiniai balso koderiai naudoja sudėtingas ištrintų segmentų maskavimo technikas ir šitaip pagerina bendrą dekoduoto balso kokybę. Šiame darbe realiais bandymais parodoma, kad prarastų paketų maskavimas pakeičia signalą ne tik ištrinto segmento laikotarpiu, bet ir modifikuoja vėlesnių segmentų signalo formą. Darbe tiriamos PESQ algoritmo savybės vertinant balso kokybės degradavimą, atsiradusį dėl lokalizuotų paketų praradimų. Teoriškai ir modeliavimo būdu nagrinėjama matavimo lango pločio ir lango pozicijos įtaka balso kokybės vertinimo rezultatams. Parodoma, kad, ilginant PESQ matavimo langą, pavienių ištrintų paketų įtaka yra dirbtinai sumažinama. Taip pat parodoma, kad prarasto paketo padėtis matavimo lange turi įtakos matavimo rezultatui. Tai reiškia, kad standartizuotu PESQ būdu vertinant ištrintų paketų įtaką atlikto matavimo rezultatų neapibrėžtis yra gana didelė. Balso kokybės matavimų neapibrėžčiai mažinti siūloma modifikuoti PESQ algoritmą, taikant sintetinius balsą imituojančius signalus, sudarytus iš apibrėžtos trukmės kartotinių atkarpų. Il. 11, bibl. 10 (anglų kalba; santraukos anglų, rusų ir lietuvių k.).

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