Darius GURŠNYS

VOICE QUALITY EVALUATION METHODS AND MEANS FOR MOBILE COMMUNICATIONS

Summary of Doctoral Dissertation
Technological Sciences,
Electrical and Electronic Engineering (01T)

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Vilnius LEIDYKLA TECHNIKA **2008**

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VILNIAUS GEDIMINO TECHNIKOS UNIVERSITETAS

Darius GURŠNYS

BALSO KOKYBĖS VERTINIMO METODAI IR PRIEMONĖS MOBILIOJO RYŠIO SISTEMOMS

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1. Introduction

Topicality of the problem. Currently it is a widespread practice when the quality of mobile communication services, similarly to the wired connection, is assessed statistically, by applying the same parameters: service accessibility, assessed as ratio of between the number of successful connections and the whole number of trials, the percentage of dropped calls and others. In the networks of wired communication, the conditions of all users are more or less the same, and such a method of evaluation by establishing the common quality indicators for the entire network can be justifiable. Whereas in case of mobile communication where the communication conditions of every user are different, indicators estimated by general statistical methods actually describe the level of development and quality of the entire network, but have nothing in common with the quality of service received by a specific user.

In the new generation of mobile communication systems the quality of services actually received by the user should actually be measured and analysed. All quality deviations from the established standard quality indicators should be registered and accounted. The service charging should be formed on the basis of service quality reports.

The evaluation using one quality score for user‘s conversation quality is not informative enough. Thus, the method using several quality scores for user’s conversation quality is analysed in this work.

Unfortunately, the quality of services used in modern telecommunication systems is neither measured nor assessed. One of the reasons is purely technical: there are no technical tools for the measurements of quality of service received by the user.

Analysing short voice signals (duration of several seconds) with local defects in the telecommunication science centre of Vilnius Gediminas Technical university (VGTU) it has been determined that the quality scores of perceptual evaluation of speech quality (PESQ) depend on the measuring conditions. To identify PESQ score uncertainty is one of the main tasks of this work.

The patterns of lost frames and models in the circuit switched data (CSD) channel of GSM communication system and in others wireless networks have been much studied. Unfortunately, flows of lost voice frames are not well described in references. The purpose of this work was to create implements for voice frames flows picking, to explore picked voice frames and to make mathematical models of them.

There is no possibility to determine the influence of particular lost voice frame on the voice quality. The research in order to establish the significance distributions of voice frames was implemented in this work.

Aim and tasks of the work. The aim of this work is to investigate voice frame losses in GSM and to propose the method for voice quality evaluation in mobile communication.

The tasks of the work:

1. To create the method for lost frame detection.
2. To add channel parameters to existing frame trace mathematical models.
3. To investigate possibility to use PESQ algorithm for quality evaluation of short duration voice signals.
4. To investigate the effectiveness of prolonged duration signals for reducing PESQ segmentation influence on quality score.
5. To propose the method for voice quality evaluation in mobile communications.

Scientific novelty

1. Proposed the method for the detection of lost voice frames in the GSM mobile communication system and calculation of their positions in the time axis, by applying the end user equipment.
2. The prepared mathematical models of GSM lost voice frame traces are related with the parameters of physical channel.
3. Proposed the method for quality measurements of short duration voice signals by PESQ algorithm.
4. Proposed the method for voice quality evaluation in mobile communication, based on voice quality categorisation.

Methodology of research

While developing the measurement tool for the voice frame traces and preparing trace models, the theoretical models, correlation analysis and mathematical statistics were applied. The statements of theoretical analysis were verified by experiments. All initial information important for the problem was obtained experimentally.

The work conclusions are based on the real measurements and the results of modelling.

Practical value

The method proposed for quality assessment of short voice signal fragments by PESQ algorithm has a significant practical value. The method allows significant reduction of uncertainty in the calculated voice quality scores.

The proposed method for the assessment of voice quality experience by the user in mobile communication system can be implemented in the current end user equipment. The quality calculated in the initial stage might have an informational character. The quality scores calculated in subsequent applications might be related with the payment for services.

Defended propositions

1. The method for the detection of lost voice frame positions based on the application of special test signals with correlational lost frame identification.
2. The models of lost voice frame traces in the mobile communication system based on experimental measurements.
3. New signal constructing method based on new experiments and algorithm analysis, minimising the score uncertainties created by the PESQ voice quality assessment algorithm.
4. The method for the quality assessment of transmitted voice by mobile communication systems, based on voice quality categorisation.

The scope of the scientific work. The scientific work consists of the introduction, 4 chapters, conclusions, list of literature, list of publications and addendum. The total scope of the dissertation – 78 pages, 39 pictures, 6 tables and 1 addendum.

2. Voice quality and its evaluation methods

In addition to the technical issues of signal transmission, the problems of quality of the provided services were solved in all generations of telecommunication systems.

The established practice of the development of telecommunication systems and quality assessment of services provided by them is targeted to the statistical user. The general system parameters and quality indicators are established as average values. They are calculated by processing the quality indicators of services rendered to a big number of users. Such practice was justified in case of fixed communication conditions when such conditions were similar for all users. But the conditions of mobile communication for every user are different, and the actually experienced quality of services is found to be very different in individual approach.

When the digital mobile communications systems were developed, the quality score became dependent on the whole number of new factors. The main of them include the frame loss in the channel, and the applied signal compression method – codec.

Unsatisfactory quality or radio channel determines the data frame losses. Research indicates that in case of low level of loss (up to 2 %), individual lost frames and short groups of lost frames (several lost frames in success) similarly affect the voice quality. With the increasing frequency of losses, the grouped losses affect the voice quality far more significantly. In case of frame group losses, not only parts of words but the entire words can be deleted from the transmitted speech.

Voice codec is a second factor by importance, affecting the voice quality. The adaptive multi-rate (AMR) codec is applied in the third generation cellular systems. With worsening communication conditions, a codec with lower rate of generated data is connected, and the remaining trace is used for additional channel encoding. This allows the increase in resistance for interferences, and reduction of the frame loss possibility. Unfortunately, when the flow of useful data reduces, the quality of transmitted voice deteriorates, as well.

According to the general ITU provisions, the quality of service (QoS) is a conditional subject. The concept of service quality is subjective, and depends on many factors, but the main of them is the degree of user’s satisfaction. The reduced quality of transmitted speech determines the user’s dissatisfaction with the service.

Voice quality assessment is a complex task. Human speech consists of active speech and silence intervals. Frames can be lost both in the intervals of speech and silence. Most often, the loss of silence intervals does not affect the voice quality. But the quality of transmitted voice depends on the location of lost frame in the active part of the speech.

Algorithms analysing the voice signal quality can be divided into intrusive and non-intrusive ones. The non-intrusive algorithms analyse only the parameters of communication channel, or the signal transmitted and distorted by this channel. In real communication systems, these algorithms are easier to adapt, but they are not susceptible to calculations, and not always feature the sufficient correlation of results with mean opinion score (MOS) assessments.

Intrusive type of algorithms (PAMS, PSQM+, PESQ) assess the voice quality by comparing the original signal with the signal degraded in the system.

Research indicates that the intrusive algorithms feature a strong correlation with expert MOS assessments.

PESQ algorithm becomes increasingly common as the main method of objective voice quality assessment. But the analysis of literature indicates that the quality score calculated under the same conditions in different works is obtained as rather different. It presupposes scrutinised interpretation of the obtained results, and more detailed analysis of the circumstances determining the uncertainties of the results.

3. Analysis of voice frame losses in GSM system

The standard, engineering measurement equipment of GSM network parameters (for instance: Ericsson TEMS Investigation 4.1, Motorola Cell Optimization Platform) provides the whole number of physical channel parameters. But they do not include the voice frame trace parameters, necessary for the analysis of frame loss consistencies, and the evaluation of the transmitted voice quality.

The method and tools for the collection of information about voice frame traces were developed. Fig 1 demonstrates a scheme for voice frame traces collection.

Fig 1. Data collection scheme of voice frame traces

Computer AK1 forms measurement signal $S(t)$, which is transmitted to the cell phone MS1 via the microphone circuit of hands-free equipment. A good communication channel with small number of errors is selected between the MS1 and BS1. In this point, the connection line should be an alternative, guaranteeing the reliable channel. Voice frame traces are measured in the downlink channel between the BS2 and MS2. Voice signal $S^*(t)$ in the computer AK2 is recorded to the file. MS2 can be an ordinary cell phone in such a case the information only about the frame traces will be accumulated. There is also a possibility to replace the MS2 with engineering telephone (for instance, TEMS Investigation 4.1). This would enable some additional measurements of other channel parameters, and their relation with the voice frame traces.

The analysis of accepted signal $S^*(t)$ indicates whether a specific voice frame was accepted or lost. GSM 06.10 decoder applies a Lost Frame Substitution function, which repeats the last correctly accepted voice frame instead of the lost one. For this reason, strongly correlated fragments develop in the signal. In case of several lost frames, the last correctly accepted frame is repeated instead of them, and their energy is additionally reduced. Increased correlation and reduced signal energy are the two criteria which make it possible to conclude that the frame was lost.

Speech signal features the varying energy, and the adjacent signal fragments are correlated, therefore the speech is not suitable as a measurement signal $S(t)$.

For the separation of every speech frame it is necessary that the measurement signal would change its characteristics every 20 ms (the duration of one GSM voice frame). The simplest solution is to apply the sinusoid signal of several frequencies and 20 ms duration. Good results were achieved by formation of signal $S(t)$ according to formula (1). It consists of 11 segments of sinusoid of various frequencies.

$$S(t) = s_1(t) + s_7(t-T) + s_2(t-2T) + \ldots + s_{11}(t-10T) + s_6(t-11T)\,, \qquad (1)$$

where $T = 20$ ms – duration of one sinusoid and

$$s_i(t) = \begin{cases} \sin(\Omega_i \cdot t), & \text{when } t \in [0,T] \\ 0, & \text{when } t \notin [0,T] \end{cases}. \qquad (2)$$

Sinusoid frequencies of the measurement signal should be multiple:

$$\left|\Omega_i - \Omega_{i-1}\right| = 2 \cdot \pi \cdot f_0 \cdot k\,, \qquad (3)$$

where $f_0 = 1/T = 50$ Hz, k is an integer number. This will ensure the full number of the sinusoid periods in the transmitted voice frame and minimal correlation between the adjacent frames. Taking into consideration these requirements, Ω_i is calculated according to

$$\Omega_i = 2 \cdot \pi \cdot (300 + 100 \cdot (i-1)), \text{ when } i = 1, 2, \ldots 11. \qquad (4)$$

Fig 2 shows a fragment of measurement signal. The signal period consists of 11 fragments of the 20 ms duration sinusoids. Frequencies of each of them are positioned in such a way that the adjacent segments differ by 500 or 600 Hz.

In real experiments, the measurement signal $S(t)$ is not synchronised with the formation of voice frames in the GSM network. The signal and the network frame are shifted in respect to each other by the time interval λ. In such a case the fragments of two sinusoids fall into one network frame. We will obtain copies of these fragments in the accepted signal $S^*(t)$, in place of the lost frame. For this reason it becomes rather complicated to apply the spectral method of analysis of the accepted signal.

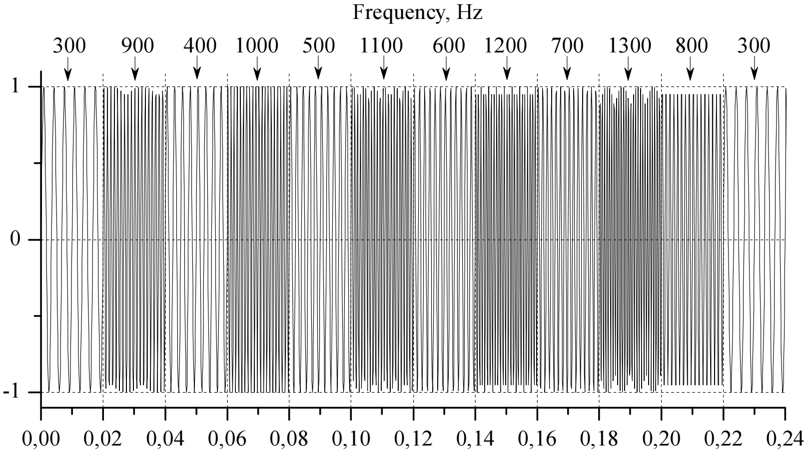

Fig 2. Fragment of measurement signal. Period duration – 220 ms

The 2 state model, N state, Markov chains, Hidden Markov models and others are applied in wireless data transfer networks for the models of lost frames. Mobile communication conditions are continuously changing, therefore the frame loss is non-stationary process, whereas, the analysed models generate a stationary process. The trace generated by these models represents only the average communication conditions that have occurred during the measurements of frame traces. The sequences of lost frames generated in this way are distributed evenly across the entire flow, and do not represent changes in communication conditions.

By classifying the communication conditions, the transfer probabilities of models should be calculated for each class. Then, by specifying the communication conditions that occurred for a specific user, more precise generation of frame traces and the forecast of expected quality of transmitted voice are possible.

GSM power control is carried out every 480 ms, which corresponds to 24 frames. During this period the power of the mobile and base station does not change, therefore it is possible to presume that the channel parameters vary insignificantly. On the basis of this assumption, we divide the frame traces into 24 frame fragments. Table 1 shows the communication quality areas that were formed considering the measurement results of average distribution of the lost frame numbers at different values of C/I.

Table 1. Parameters of communication areas

Communication conditions	Number of lost frame in the zone	FER, %	C/I, dB
“Good“	1	≤ 4	> 15
“Average“	2 ... 4	4 < FER ≤ 17	11 ... 15
“Bad“	5 ... 13	17 < FER ≤ 58	5 ... 10
“Unacceptable “	> 13	> 58	< 5

Fig 3 shows the measured and generated voice frame traces. One may note that the generated sequences of lost frames are unevenly distributed across the entire flow and accumulated only in the same areas as those in the measured flow.

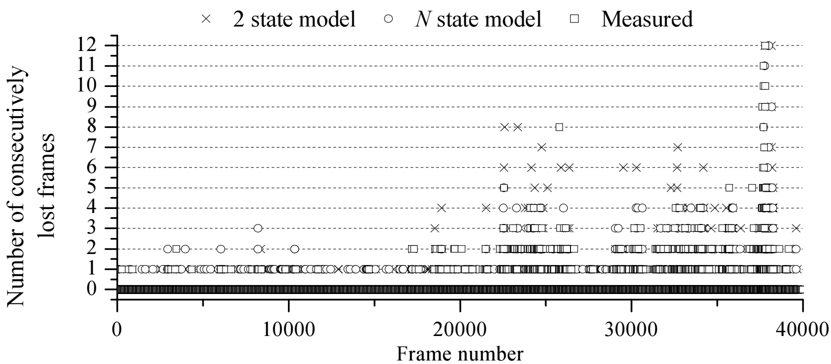

Fig 3. Comparison of the measured and model-generated traces

When evaluating the voice quality, it is important to have information about the occurring sequences of lost frames. Therefore, the distribution of the sequences of lost frames could be a criterion for the comparison of model efficiency. Fig 4 demonstrates the distribution of sequences of lost frames for measured and modelled traces. One may note that even the simplest 2 state

Markov chain generates a similar flow, in case of assessment of communication quality areas.

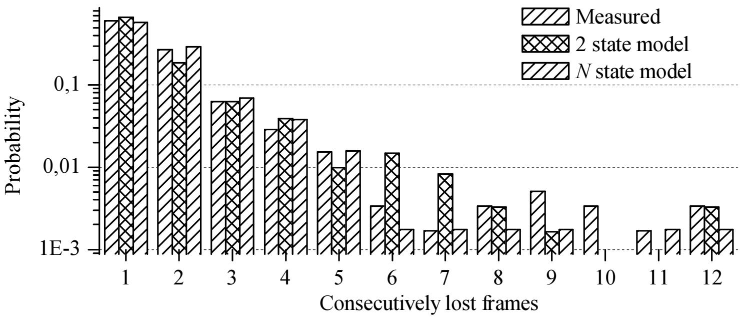

Fig 4. Comparison of the measured flow and the flows generated with 2 state and *N* state models with communication conditions assessment

4. Analysis for the evaluation of the effect of localised defects by PESQ algorithm

It was noted that high distribution of PESQ result values is obtained in the evaluation of the localised defects to the voice quality in short signals (2 – 3 s). One of the reasons is signal division into "frame" windows (T_f = 32 ms) where the signal quality degradations are calculated, used in the algorithm.

The local defect – 20 ms lost voice segment – may affect the signal quality in two or three frame windows. It depends on the position of the lost segment in the PESQ measurement window. The voice quality score calculated by PESQ algorithm becomes dependent on the position of defect in the measurement window. This determines the uncertainty of results.

Fig 5 contains the scheme of experiment enabling to show the effect of PESQ score uncertainty, when the defect changes its position in the frame windows.

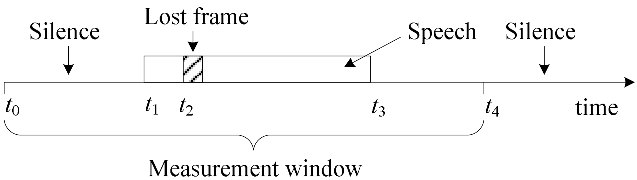

Fig 5. Composition of measurement signal and positioning of measurement window

For this experiment, the measurement signal is specially constructed and consists of 3 parts. From t_0 to t_1 the signal consists of silence. From t_1 to t_3 there is a voice signal and from t_3 again the silence starts.

The packet loss is modelled on the time moment t_2. The beginning of the measurement window is moved in 1 ms steps from t_0 to t_1 and the Q_{PESQ} scores are measured. Fig 6 shows the Q_{PESQ} for three voice signals, when the duration of the speech signal is $t_3 - t_1 = 1700$ ms, the position of lost packet in the speech signal is $t_2 - t_1 = 1000$ ms and the duration of measurement window is $T_{\mathrm{PESQ}} = 2000$ ms. Changes of Q_{PESQ} score are obtained as periodical, with the period of 16 ms and coincide with the overlapping time of frame windows.

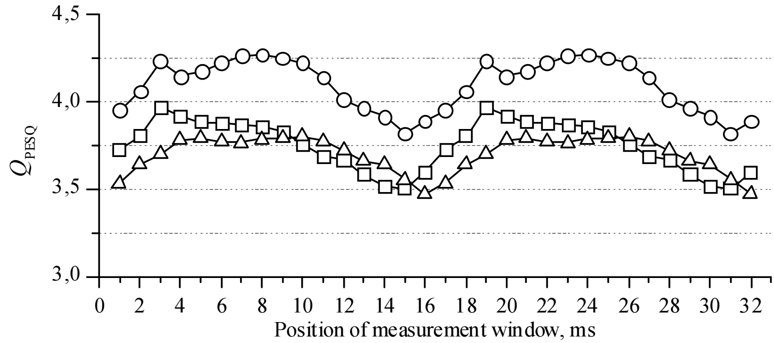

Fig 6. Dependency of Q_{PESQ} score on the position of measurement window for three speech fragments

It is important to note that the position of measurement window changes, the ratio between the signal silence in the beginning and end of durations redistributes, but the voice segment with the lost packet remains unchanged. While assessing this experiment from the MOS positions (when the quality assessment is carried out by an experts) it is obvious that the change of silence in the duration of several dozen of milliseconds does not affect, or affects the result very insignificantly.

P.862 recommendation indicates that the signal for voice quality evaluation can consist of several fragments of signals, by maintaining the natural structure of speech. After the evaluation of reasons for the uncertainties of measurements, the special scheme for preparing the signal for measurements was proposed.

The initial voice signal is replicated several times (Fig 7). The procedure of signal duration "extension" should be carried out both for the original (signal without distortions) and distorted signals. When selecting T_{prad} – the duration of the initial voice signal, it is intended that by repeating the signal, the lost packets would fall into different frame window positions every time.

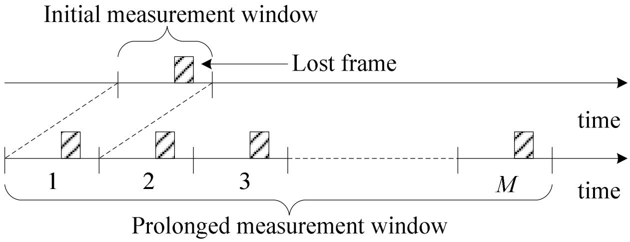

Fig 7. Preparation of the signal with extended duration

This condition is met, when the duration of the initial signal is selected according to

$$T_{\text{prad}} = \left(k + \frac{1}{M}\right) \cdot 0{,}016\,, \tag{9}$$

where M – the number of repetitions, k is an integer number. The necessary duration of the initial signal is selected by changing k. Time 0,016 s is the duration of overlap of the adjacent “frame” windows.

The number of repetitions M is also a selected value. It is possible to expect that a higher value of M shall guarantee the lower dependency of results on the measurement window position. The effect of M on the uncertainty of results is demonstrated in Table 2.

Table 2. The largest uncertainty (E) that occurred during the assessment of 100 voice fragments

	M							
	1	2	3	4	5	6	7	8
E	0,697	0,264	0,182	0,134	0,106	0,054	0,039	0,033

These data clearly testify the method efficiency and the fact that the uncertainty of measurements reduces, by increasing the number of repetitions M. For better demonstration of the efficiency of the signal extension method, Fig 8 shows the dependencies of Q_{PESQ} score on the position of measurement window, when applying the method of signal duration extension (M = 8) and without this method.

As we see, when the measurement window is moved, the dependency of quality of the extended duration signal on the measurement window position is virtually unnoticeable.

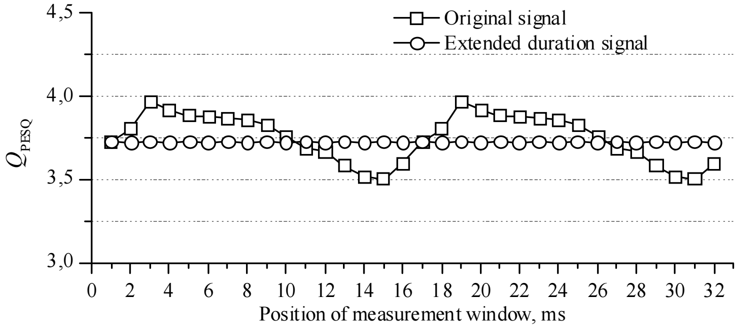

Fig 8. Dependencies of Q_{PESQ} score on the position of measurement window, with/without the method of signal duration extension

For the objective evaluation of the effect of single lost packet for the voice quality, the measurement experiments with the voice codec G.711 were carried out by applying the method of extended measurement signal (duration of the initial measurement window – 2 s, $M = 8$). Packets one by one were deleted in sequence and every time the quality measurements were performed. The result histogram is given in Fig 9. The range of quality degradations affected by single packets varies from 0 to 1,6. Unfortunately, it is not possible to identify any other regularities and significances of packets.

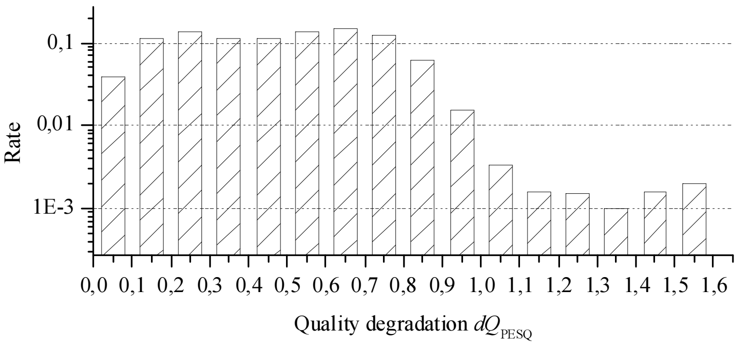

Fig 9. Histogram of voice quality degradation by G.711 codec, when one packet is lost

When evaluating the voice quality actually experienced by the user, it was identified that some packets were lost during the transmission. But such information does not enable unambiguous conclusion about what is the value of real voice quality degradation. As we can see from the provided examples, only the degradation range and the frequency of occurrence in the selected degradation value can be identified.

5. Voice quality evaluation in the mobile communication systems

It is proposed to express the experienced voice quality by the distribution of standard quality categories and not by one indicator. The path of voice quality categorisation was formulated in the ITU recommendation G.107, where E – model and quality rating R, related to the level of user's satisfaction, were introduced. The categories of the transmitted speech quality were named in the ITU recommendation G.109 according to the degree of user's satisfaction.

Voice quality result is expressed in 5 digits, indicating a relative part of each quality category. It allows saving information about possible short, low quality intervals during the conversation, which might not be noticed by applying the elementary calculated averaging of quality scores. The longer the conversation, the lower the effect of short low quality intervals on average will be. When the quality score is described by the distribution of quality categories, the duration of conversation does not affect the results and the obtained distribution enables a more detailed assessment of the quality experienced by the user.

Table 3. Voice quality categories

Voice quality category, q	Q_{PESQ} upper limit	Q_{PESQ} lower limit	Width
Best	4,50	4,34	0,16
High	4,34	4,03	0,31
Medium	4,03	3,6	0,43
Low	3,6	3,1	0,5
Poor	3,1	1	2,1

Lost frames affect the voice quality in a high variety of ways. Direct calculations of lost frames during a certain period do not allow us to judge about the voice quality reduction. But it is possible to calculate the probability of each quality category in case of specific number of lost frames.

A method of statistical score identification of the voice quality is implemented in several stages:

- According to the recommendation P.800, the duration of conversation is divided into intervals of 2 s. The lost frames numbers $i(j)$ are identified in the end user's equipment, in every 2 s (the j – th) interval.

- The array $\{i(j)\}$ of numbers collected throughout the entire conversation (divided into N intervals) is sorted and the number of intervals containing 0, 1, 2, 3, 4, ... lost frames is calculated.
- After marking the numbers of such intervals with the symbol n_i, the discreet constituents of the experienced voice quality distribution are calculated according to the formula

$$P(q) = \frac{1}{N}\sum_{i=1} n_i \cdot p(q \mid i), \quad (10)$$

where N – the number of 2 s intervals, n_i – the number of intervals n, where i frames were lost, $p(q|i)$ – probability for the quality category q, when i frames are lost.

For every voice codec, only the set of probabilities $p(q|i)$ calculated for it is necessary. The sets of probability distribution for AMR codec are calculated in the work.

6. General conclusions

1. Frame losses in the mobile communication systems are the main factors reducing the quality of transmitted voice. The voice frame traces have not been analysed in scientific literature widely. The method for collecting the voice frame traces in GSM was developed. Mathematical models were adapted for the voice frame traces, by relating their parameters with the parameters of physical channel condition.
2. The PESQ algorithm recommended in ITU P.862 is the most widely applied for the evaluation of voice quality, but it was proven that the uncertainty of short voice fragment measurement results is very high (up to 0,6 MOS). It was proposed to solve this problem by forming signal from fragments of the primary signal repeated M times. This way the uncertainty of measurement results can be reduced up to 20 times.
3. Multiple researches have shown that there is no unambiguous relation between the number of lost voice frames and the value of voice quality reduction. Only the relations between the lost frames and the probability that voice quality will be reduced by the value dQ, can be identified.
4. When evaluating the voice quality in mobile communication by one parameter, the real information about short quality reductions is hidden. For this reason it was offered to measure the quality actually experienced by the user via the distributions of standardised quality categories.

List of published works on the topic of the dissertation in the reviewed scientific periodical publications

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BALSO KOKYBĖS VERTINIMO METODAI IR PRIEMONĖS MOBILIOJO RYŠIO SISTEMOMS

Mokslo problemos aktualumas

Šiuo metu mobiliojo ryšio paslaugų kokybė kaip ir fiksuotojo (laidinio) ryšio vertinama statistiškai, taikant tuos pačius parametrus: paslaugos pasiekiamumas, išreiškiamas sėkmingų sujungimų santykiu su visų bandymų skaičiumi, nutrauktų skambučių skaičius procentais ir kiti. Fiksuotojo laidinio ryšio tinkluose, kai visų vartotojų ryšio sąlygos yra maždaug vienodos, toks vertinimo būdas, nustatant visam tinklui bendrus kokybės rodiklius gali būti pateisinamas. Mobiliojo ryšio sąlygose, kai kiekvieno vartotojo ryšio sąlygos yra skirtingos, bendri statistiniais metodais nustatyti rodikliai apibūdina viso tinklo išvystymo bei tinklo kokybės lygį, bet nieko nesako apie konkretaus vartotojo gautos paslaugos kokybę.

Naujos kartos mobiliojo ryšio sistemose turėtų būti matuojama ir analizuojama vartotojo individualiai gautų paslaugų kokybė. Visi kokybės nukrypimai nuo numatytų normatyvinių kokybės rodiklių turėtų būti registruojami ir apskaitomi. Pagal paslaugų kokybės ataskaitas formuojamas paslaugos apmokestinimas. Vartotojo pokalbio kokybės vertinimas vienu kokybės įverčiu yra nepakankamai informatyvus. Todėl šiame darbe nagrinėjama balso kokybės vertinimo keliais parametrais metodika.

Deja, dabartinėse telekomunikacijų sistemose individualiai patirta paslaugų kokybė nėra nei matuojama, nei vertinama. Viena iš priežasčių yra techninė – nėra vartotojo gautos paslaugos kokybei vertinti būtinų techninių instrumentų.

Vilniaus Gedimino technikos universiteto (VGTU) telekomunikacijų mokslo centre tiriant trumpas (kelių sekundžių trukmės) balso signalo atkarpas su lokaliais defektais buvo pastebėta kad, PESQ (*angl.* Perceptual Evaluation of Speech Quality) algoritmu nustatyti kokybės įverčiai priklauso nuo matavimo sąlygų. Vienu pirmųjų šiame darbe sprendžiamų uždavinių yra nustatyti, kas lemia PESQ įverčio neapibrėžtumus.

Paketų praradimų dėsningumai ir modeliai GSM mobiliojo ryšio sistemos duomenų perdavimo CSD (*angl.* Circuit Switched Data) kanale ir kitose bevielėse duomenų perdavimo sistemose yra neblogai ištirti. Deja, balso kanale prarastų paketų srautai mokslinėje literatūroje nėra išsamiai aprašyti. Darbe siekiama sukurti priemones balso paketų srautams rinkti. Ištirti sukauptų balso paketų srautus ir sudaryti jų matematinius modelius.

Nėra galimybės realiu laiku nustatyti konkretaus prarasto paketo įtakos kalbos balso kokybei. Darbe vykdomi tyrimai, kuriais siekiama nustatyti balso paketų reikšmingumo skirstiniai.

Darbo tikslas ir uždaviniai

Darbo tikslas – ištirti GSM mobiliojo ryšio sistemos balso paketų srautus ir pasiūlyti balso kokybės vertinimo būdą ir priemones mobiliojo ryšio sistemoms.

Darbo tikslui pasiekti sprendžiami šie uždaviniai:

1. Sukurti prarastų balso paketų pozicijų aptikimo būdą ir priemones.
2. Pritaikyti žinomus matematinius modelius prarastų balso paketų srautams, įvertinant ryšio sąlygas.
3. Ištirti galimybę vertinti mažos trukmės balso signalų atkarpų, su pavieniais prarastais paketais, kokybę PESQ algoritmu.
4. Sukurti atstojamojo signalo sudarymo iš trumpų balso signalų atkarpų būdą ir ištirti jo efektyvumą vykdant kokybės vertinimą PESQ algoritmu.
5. Įvertinus gautus balso paketų srautų modelius ir prarastų paketų įtaką, sukurti mobiliojo ryšio sąlygoms tinkamą balso kokybės vertinimo metodą.

Mokslinis naujumas

1. Pasiūlytas GSM mobiliojo ryšio sistemoje prarastų balso paketų aptikimo ir jų pozicijų laiko ašyje apskaičiavimo metodas, panaudojant galinę vartotojų įrangą.
2. Sudaryti GSM prarastų balso paketų srautų matematiniai modeliai susieti su fizinio kanalo parametrais.
3. Pasiūlytas metodas leidžiantis efektyviai vertinti mažos trukmės balso signalų kokybę PESQ algoritmu.
4. Sukurtas mobiliojo ryšio sąlygoms pritaikytas balso kokybės vertinimo būdas, pagrįstas balso kokybės skirstymu į kokybės kategorijas.

Tyrimų metodika

Darbe taikoma kalbos signalų apdorojimo metodai, koreliacinė analizė, matematinė statistika. Eksperimentais tikrinami teorinio tyrimo teiginiai. Darbo išvados grindžiamos matavimais ir modeliavimo rezultatais.

Praktinė vertė

Pasiūlyti su ryšio kanalo parametrais susieti GSM balso paketų srautų matematiniai modeliai leidžia modeliuoti individualaus vartotojo ryšio sąlygas.

Praktinę vertę turi pasiūlytas metodas trumpų balso signalo atkarpų kokybės vertinimui PESQ algoritmu. Metodas leidžia daugiau kaip 20 kartų sumažinti apskaičiuotų balso kokybės įverčių neapibrėžtį.

Pasiūlytas mobiliojo ryšio sistemose vartotojo patirtos balso kokybės vertinimo metodas gali būti diegiamas dabartinėje vartotojo galinėje įrangoje. Pradinėje fazėje apskaičiuota kokybė galėtų būti informacinio pobūdžio. Vėlyvesniuose taikymuose apskaičiuoti kokybės įverčiai galėtų būti siejami su paslaugos apmokestinimu.

Ginamieji teiginiai

1. Prarastų balso paketų padėties aptikimo būdas, pagrįstas specialių testinių signalų panaudojimu ir koreliaciniu prarastų paketų identifikavimu.
2. Eksperimentiniais matavimais paremti prarandamų balso paketų srautų modeliai mobiliojo ryšio sistemai.
3. Naujas eksperimentais ir algoritmo analize paremtas signalo konstravimo metodas, minimizuojantis PESQ balso kokybės vertinimo algoritmo sukuriamus įverčių neapibrėžtumus.
4. Mobiliojo ryšio sistemomis perduoto balso, kokybės vertinimo metodas, pagrįstas balso kokybės skirstymu į kokybės kategorijas.

Darbo apimtis

Darbą sudaro bendra darbo charakteristika, 5 skyriai, išvados, literatūros sąrašas, publikacijų sąrašas ir priedai. Bendra disertacijos apimtis – 78 puslapiai, 38 iliustracijos, 6 lentelių ir 1 priedas.

Pirmajame skyriuje nagrinėjamas problemos aktualumas, formuluojamas darbo tikslas ir uždaviniai. Aptariami tyrimų metodai ir mokslinis darbo naujumas. Pristatomas autoriaus indėlis.

Antrasis skyrius skirtas literatūros apžvalgai. Jame pateikiama paslaugų kokybės sampratos evoliucija. Apžvelgiama nuo ko priklauso perduodamo balso kokybė. Susisteminami kalbos kokybės vertinimo metodai. Analizuojami atlikti balso kokybės vertinimo tyrimai. Skyriaus pabaigoje formuluojamos išvados.

Trečias skyrius skirtas balso paketų GSM ryšio sistemoje tyrimams. Pateikiamas originalus prarastų balso paketų pozicijų laike nustatymo metodas. Balso paketų praradimus kanale aprašantys modeliai papildomi ryšio kanalo būvį įvertinančiais parametrais.

Ketvirtame skyriuje nagrinėjama galimybė PESQ algoritmu vertinti mažos trukmės balso signalų kokybę. Pateikiami segmentavimo įtakos kokybės įverčiui tyrimų rezultatai. Pasiūlomas metodas mažos trukmės signalų kokybei vertinti PESQ algoritmu, leidžiantis sumažinti segmentavimo įtaką kokybės įverčiui.

Penktame skyriuje pasiūlomas originalus vartotojo patirtos balso kokybės vertinimo metodas, pagrįstas kokybės priskyrimu kokybės kategorijoms. Pateikiami, metodui būtini, kokybės kategorijų tikimybių rinkiniai, skirti skirtingiems AMR kodeko spartos režimams.

Šeštame skyriuje pateikiamos galutinės išvados.

Bendrosios išvados

1. Mobiliojo ryšio sistemose prarasti paketai blogina perduoto balso kokybę, tačiau mokslinėje literatūroje balso paketų praradimai nebuvo išsamiai ištirti. Disertacijoje pristatomas balso paketų srautų rinkimo metodas ir aprašomos priemonės, skirtos GSM balso paketų srautams tirti. Markovo grandinių matematiniai modeliai pritaikyti prarastų balso paketų srautams imituoti. Šių modelių ypatumas yra tas, kad jų parametrai susieti su ryšio kanalo parametrais.

2. Darbe įrodyta, kad balso kokybei vertinti plačiai taikomas, ITU P.862 rekomendacijoje įteisintas PESQ algoritmas, nėra tinkamas matuoti trumpų balso atkarpų kokybę, nes gaunamų matavimo rezultatų neapibrėžtis gali siekti daugiau nei 0,6 MOS balo. Problemai spręsti pasiūlyta naudoti dirbtinai suformuotą tiriamąjį signalą, sudarytą iš M kartų pakartotų pirminio signalo atkarpų. Tai leidžia iki 20 kartų sumažinti matavimo rezultatų neapibrėžtį.

3. Daugkartiniais tyrimais įrodyta, kad nėra vienareikšmio ryšio tarp prarastų balso paketų skaičiaus ir balso kokybės pablogėjimo vertės. Galima nustatyti tik sąsajas tarp prarastų paketų ir tikimybės, kad balso kokybė bus sumažinta dydžiu dQ_{PESQ}.

4. Nustatyta, kad vertinant mobiliojo ryšio balso kokybę vienu parametru paslepiama informacija apie trumpalaikius kokybės pablogėjimus. Todėl pasiūlyta vartotojo patirtą balso kokybę išreikšti standartizuotų kokybės kategorijų skirstiniais.

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VOICE QUALITY EVALUATION METHODS AND MEANS FOR MOBILE COMMUNICATIONS

Summary of Doctoral Dissertation
Technological Sciences, Electrical and Electronic Engineering (01T)

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