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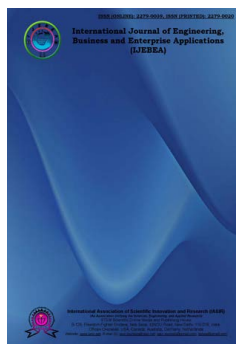
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Indexing





Simulation Analysis of Speech Quality Dependence from Communication Channel Type and Channel Coding Methods

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***Abstract:** Speech quality estimation methods are a part of general audio or sound quality estimation methods developed for establishment and guaranties the quality of the received speech signal transmitted over the voice or multimedia communication systems and networks. There are several methods and standards developed for speech quality estimation. A precise analysis of these methods leads to conclusions of existing wide spread subjective and objective methods and also a lot of corresponding standards based on these methods. These methods and standards are applied for testing new methods for speech coding or for improvements of the existing ones. On the whole, the speech quality depends not only from the speech coding methods, but also from a lot of various factors. Some of these factors are related to the characteristics of the communication channels using for transmission of speech in voice or multimedia systems and networks. The main reason of speech quality degradation in these real communication systems are noise level and typical distortions in each type of communication channels. Therefore, the goal of this article is focused to the precise analyze of the speech quality dependence from the above mentioned channel characteristics. Attention of speech quality analysis is directed also to the corresponding channel coding methods, which must be in accordance to the appropriate channel characteristics and to leads of decreasing the errors in received speech or voice signals and therefore to increase the speech quality. All of these analyses are based on the proposed in this article simulation model for voice transmission in multimedia systems. The structure of this model allow a number of configurations to investigate the numerous of variations and combinations between chosen communication channel and the corresponding type of channel coding method and then analyzed the speech quality of each of these variants.*

***Keywords:** speech quality, multimedia systems, voice communication system*

I. Introduction

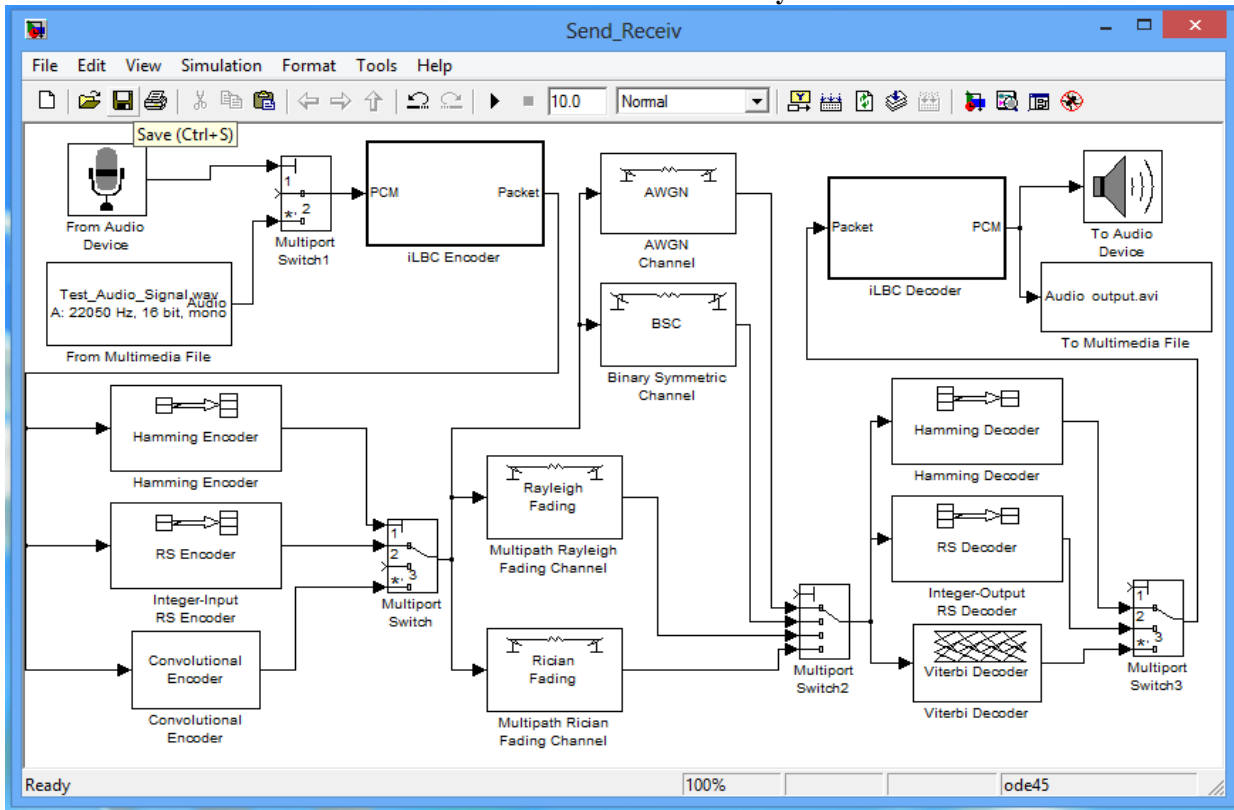
Sound quality can be considered as a psychoacoustic category [1], because of human ability to precise estimate the perceived original sounds and their reproductions after transmission in multimedia systems and networks. Therefore, there are subjective methods and standards [2] for speech quality estimation based on this above mentioned human ability to speech sounds estimation. These methods are very precise, but must be prepared with big number of people and spend long time i.e. they are expensive. There exist also a group of objective methods and standards [3], which are based on objective measurements and estimation of chosen speech signal parameters. These methods can be prepared easier than subjective methods, are cheaper, but with less accuracy than subjective methods. The function of the Introduction is to establish the context of the work being reported. This is accomplished by discussing the relevant primary research literature (with citations) and summarizing our current understanding of the problem you are investigating. State the purpose of the work in the form of the hypothesis, question, or problem you investigated; and, Briefly explain your rationale and approach and, whenever possible, the possible outcomes your study can reveal.

There are many aspects of speech quality estimation, for example testing new methods for speech coding or improvements of the existing ones, etc. One of these numerous applications of speech quality estimation is the investigation of voice or multimedia communication channels testing about speech quality of the received after speech signals transmission. This is chosen as a goal of this article: to develop an appropriate simulation model of different voice or multimedia communication channels and with the eligible specific channel characteristics and standards for speech information coding. The proposed simulation model can be applied for the purpose of comparative analyses of estimated speech quality for each of the chosen during the simulation voice or multimedia communication channels and their corresponding channel coding methods.

II. Development of the simulation models for speech quality estimation and analysis

In Fig. 1 is presented the proposed simulation model to prepare various scenarios of speech quality analysis and estimation in voice or multimedia communication systems. The main goal of this model is possibility to examine the dependence of speech quality from different types of multimedia communication channels and also from the wide spreads channel coding methods.

Figure 1 The proposed simulation model for various scenarios of speech quality analysis and estimation in voice or multimedia communication systems.



A brief description of the simulation model is the following:

- input test audio signals in real time from microphone (**From Audio Device** in Fig.1);
- input test audio signals preliminary saved as audio files (**From Multimedia File** in Fig.1);
- switch the input test signals (**Multiport Switch 1** in Fig.1);
- apply a chosen speech coder for the input test signals based on the existing standard speech coding methods for voice and multimedia communication systems (**iLBC Encoder** in Fig.1);
- switch (**Multiport Switch** in Fig.1) the different channel coders to analysis the dependence of speech quality from their ability of errors correction if there are the noise and distortions added in the communication channel (**Hamming Encoder, Integer-Input RS Encoder, Convolutional Encoder** in Fig.1);
- switch (**Multiport Switch 2** in Fig.1) the different channel types to analysis the dependence of speech quality from their noise and distortion characteristics (**AWGN Channel, Binary Symmetric Channel, Multipath Rayleigh Fading Channel, Multipath Rician Fading Channel** in Fig.1);
- in the receiving part to switch (**Multiport Switch 3** in Fig.1) the appropriate channel decoder (**Hamming Decoder, Integer-Output RS Decoder, Viterbi Decoder** in Fig.1) and speech decoder (**iLBC Decoder** in Fig.1) for correct restoration (**To Audio Device or To Multimedia File** in Fig.1).

It is proposed to use the above described simulation model to prepare various scenarios of speech quality analysis and estimation in voice or multimedia communication systems. The main goal of this model is possibility to examine the dependence of speech quality from different types of multimedia communication channels and also from the wide spreads channel coding methods.

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III. Estimation of the received after simulation speech signals to analyze their of speech quality dependence from channel type and channel coding methods

For estimation of the speech quality of the received after simulation (Fig. 1.) here it is assumed that the main part of distortions in the received speech signals is in form of noise added from the voice or multimedia communication channel. Therefore, the measurement and estimation of the noise in received speech signals can be carried out in two ways: subjective or objective estimation. There are several objective measures [4] used for estimation of the noise in speech signals. Here in this article the estimation of speech quality is focused around

some set of objective speech quality measures and especially around the following versions of SNR, i.e. speech energy to non-speech energy ratio: NIST STNR (National Institute of Standards and Technology Signal To Noise Ratio), WADA SNR (Waveform Amplitude Distribution Analysis Signal To Noise Ratio), BASS (Blind Audio Source Separation), PESQ (Perceptual Evaluation of Speech Quality) and SNR_VAD (Signal To Noise and Voice Activity Detection) and it is chosen to apply the Waveform Amplitude Distribution Analysis Signal To Noise Ratio WADA SNR as it is mentioned in [5] of their advantages: significantly less bias and less variability with respect to the type of noise compared to the standard NIST STNR algorithm and in addition quite computationally efficient algorithm.

The probability density function f_s of a discrete original (without noise) speech signal $osig(n)$ with amplitude x can be approximated [6] well as a symmetric gamma distribution Γ with the shaping α_x and the rate β_x parameters:

$$f_s(x|\beta_x) = \frac{\beta_x}{2\Gamma(\alpha_x)} (\beta_x|x|)^{\alpha_x-1} \exp(-\beta_x|x|) \quad (1)$$

The original (without noise) speech signal $osig(n)$ can be corrupted by discrete additive Gaussian noise signal $nsig(n)$ with amplitude n and with the probability density function f_n expressed as:

$$f_n(n) = \frac{1}{\sqrt{2\pi}\sigma_n} \exp\left(-\frac{n^2}{2\sigma_n^2}\right) \quad (2)$$

From the probability density function f_s of the original (without noise) speech signal $osig(n)$ (1), the probability density function f_n of the additive Gaussian noise $nsig(n)$ (2) and the probability density function of gamma distribution Γ can be obtain the following parameter G_{cr} :

$$G_{cr} = \ln(\alpha_{cr}) - \psi_0(\alpha_{cr}) = \ln\left(\frac{1}{N} \sum_{n=0}^{N-1} |crsig[n]|\right) - \frac{1}{N} \sum_{n=0}^{N-1} \ln(|crsig[n]|) \quad (3)$$

where $crsig(n)$ is the corrupted with additive Gaussian noise $nsig(n)$ original (without noise) speech signal $osig(n)$:

$$crsig(n) = osig(n) + nsig(n) \quad (4)$$

Equation (3) can be expressed using observations of original speech signal $\tilde{osig}(n)$, observation of noise signal $\tilde{nsig}(n)$, Gaussian distribution parameter σ_n of the noise signal and shaping parameter β_x :

$$G_{cr} = \ln\left(E\left[\tilde{osig}(n) + \beta_x \sigma_n \tilde{nsig}(n)\right]\right) - E\left[\ln\left[\tilde{osig}(n) + \beta_x \sigma_n \tilde{nsig}(n)\right]\right] \quad (5)$$

For a fixed shaping parameter α_x , which indirectly exist in equation (5), is possible to claim that the values of G_{cr} are uniquely related to the signal to noise ratio (SNR) and therefore, can be used as an objective evaluation of the received speech signal quality in voice or multimedia communication systems.

IV. Experimental results from simulation

With the described above simulation model of voice or multimedia communication system (Fig.1) and the appropriate equations (1 - 5) are prepared numerous experiments with various scenarios to examine the dependence of speech quality from different types of multimedia communication channels and also from the wide spreads channel coding methods. For comparative speech quality estimation between transmitted and received after simulation speech signals first are calculated their spectrums. In the Fig.2 are presented together the high frequency parts of the transmitted and the received speech signals for one of the various simulations carried out. The reason to present and use only these high frequency parts of the spectrums is that usually the noise presence is in the high frequency part of the noisy speech signals. From Fig. 2 is not seen clearly an essential difference between two spectrums: of the transmitted-original speech signal $tss(n) = osig(n)$ and the received speech signal $rss(t) = crsig(n)$. But if the difference between them is calculated:

$$diff(n) = tss(n) - rss(n) = osig(n) - crsig(n) \quad (6)$$

it is seen that there exist an essential difference $diff(t)$ presented in Fig. 3 for two cases of the defined signal to noise ratio (SNR=10 dB - above in Fig. 3 and SNR=40 dB - below in Fig. 3).

It is proposed here in this article to apply this difference $diff(t)$ for a preliminary coarse estimation of speech quality between transmitted $tss(t)$ or original speech signal $osig(n)$ and the received $rss(t)$ or corrupted with additive Gaussian noise $nsig(n)$ speech signals.

Figure 2 The high frequency parts of the spectrums of the transmitted and the received speech signals for one of the various simulations carried out.

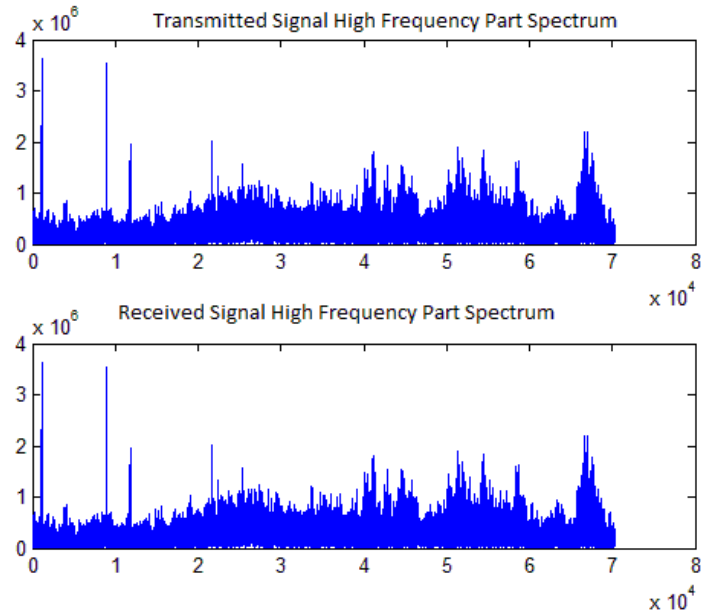
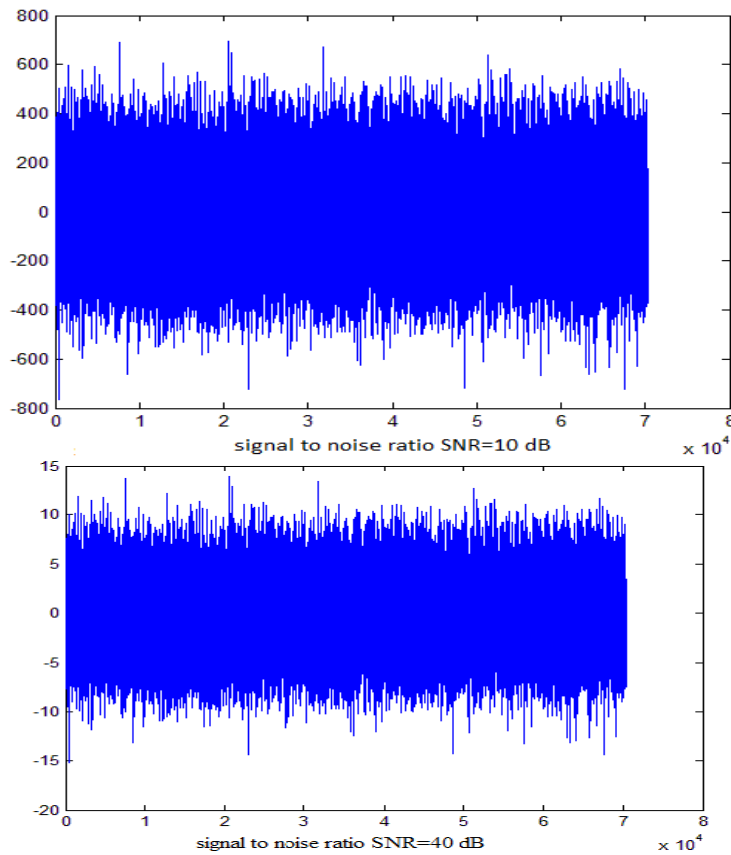


Figure 3 The difference calculated between high frequency parts of the spectrums $h_{sptss}(f)$ of the transmitted $t_{ss}(t)$ and $h_{sprss}(f)$ of the received $r_{ss}(t)$ speech signals for two cases of the defined signal to noise ratio: SNR=10 dB – above and SNR=40 dB – below.



The results as relative amplitudes of coarse estimation of speech quality from these experiments are summarized in the Table 1 for different communication channels and appropriate channel coding methods and for the defined in each case the signal to noise ratio between SNR=10 dB and SNR=40 dB.

Table 1

Relative amplitudes of coarse estimation of speech quality for SNR 10 to 40 dB	AWGN Channel	BSC Channel	Rayleigh Fading Channel	Rician Fading Channel
Hamming Encoder/Decoder	900 to 20	800 to 15	1100 to 26	1300 to 30
RS Encoder/Decoder	600 to 5	300 to 1	700 to 16	850 to 21
Convolutional Encoder/Decoder	760 to 8	420 to 3	810 to 18	930 to 27

The analysis of the results from Table 1 leads to conclusion of a real possibility to use the proposed coarse speech quality estimation and gives the clarity of the robustness and resistance to noise of the channel encoders/decoders (In Table 1 - minimal relative values 300 to 1 in the range of 10 to 40 dB for RS Encoder/Decoder and BSC Channel).

The verification of the above results and conclusion for the proposed coarse estimation of speech quality are prepared using also the described above simulation model of voice or multimedia communication system (Fig.1), but the speech quality estimation is based on the calculation of G_{cr} (equation 5) applying WADA SNR (Waveform Amplitude Distribution Analysis Signal To Noise Ratio). The results from these speech quality estimations (values for SNR in dB) are presented in Table 2.

Table 1

Relative amplitudes of coarse estimation of speech quality for SNR 10-40 dB	AWGN Channel	BSC Channel	Rayleigh Fading Channel	Rician Fading Channel
Hamming Encoder/Decoder	8 to 36	9 to 38	7 to 37	8 to 35
RS Encoder/Decoder	9 to 39	13 to 41	8 to 16	7 to 38
Convolution Encoder/Decoder	7 to 39	12 to 40	10 to 38	8 to 37

Again it is seen from Table 2 that the best result (SNR 13 to 41 dB) is for RS Encoder/Decoder and BSC Channel as it is obtained with the proposed coarse estimation of speech quality (Table 1).

V. Conclusion

The proposed a simulation model of voice or multimedia communication system and coarse estimation of speech quality, which is proven using comparative speech estimation of speech quality with the existing WADA SNR (Waveform Amplitude Distribution Analysis Signal To Noise Ratio). It is shown the ability of coarse speech quality estimation in the analyze of the speech quality dependence from the channel characteristics and channel coding methods and therefore to reduce the calculations, which is an important advantage of the proposed speech quality estimation method.

VI. References

- [1] H. Fastl, Psychoacoustics and sound quality metrics. In: Davies P, Ebbitt G (eds) Proc Sound-Quality Symp 3-10, Ypsilanti MI, 1998.
- [2] ITU-R Recommendation BS.1284: Methods for the subjective assessment of sound quality – general requirements ITU, Geneva, 1997.
- [3] Objective quality measurement of telephone-band (300-3400 Hz) speech codecs, ITU-T Rec. P.861 (1998).
- [4] C.Plapous and C. Marro, "Improved signal-to-noise ratio estimation for speech enhancement," IEEE Trans. Speech Audio Processing, vol. 14, no. 6, Nov. 2006, pp. 2098-2108C. Kim, RM Stern Robust Signal-to-Noise Ratio Estimation Based on Waveform Amplitude Distribution Analysis- INTERSPEECH, September 22-26, Brisbane Australia, pp.2580-2601.
- [5] R. V. Hogg, A. Craig, and J. W. McKean, Introduction to Mathematical Statistics, 6th edition, Upper Saddle River, NJ: Prentice-Hall, 2005.

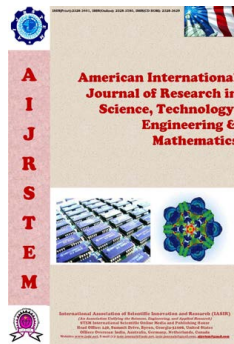
Acknowledgments

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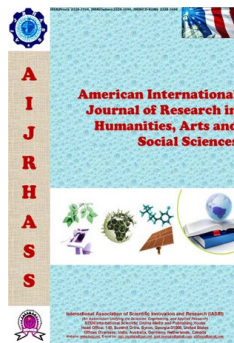
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