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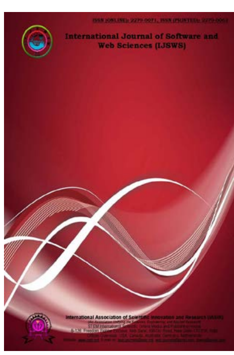

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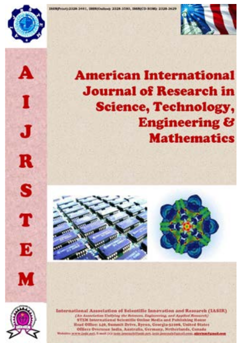

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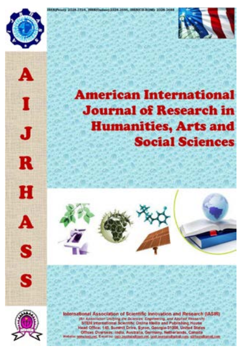

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Simulation of Different Types of Voice Communication Systems Used for Speech Quality Estimation with Applying Speech to Text as Objective Criterion of Audio Quality

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Abstract: *This article is about objective methods of speech quality estimation in voice communications systems. There are a lot of methods and algorithms for audio quality measurements and estimations. Most of them are based on subjective tests or on objective methods and algorithms trying to obtain the precision of subjective methods. In all cases of speech quality estimation it is necessary to create and use an appropriate simulation model of communication system suitable for transmission of the chosen test speech signals, which are object of quality estimation. Here in this article are prepared and used simulation models with different types of voice communication systems with applying different methods for source and channel encoding. This gives the possibility to test and estimate the quality of the decoded speech according to important voice communication systems characteristics – errors and noise. The main goal of this article is to apply the created voice communication systems simulation models in a proposed objective method for speech quality determination, combining the advantages of subjective methods to estimate the speech of receiving quality, but changing the estimator to be not a person, but a speech to text system.*

Keywords: *voice communication systems, speech and audio quality, speech to text*

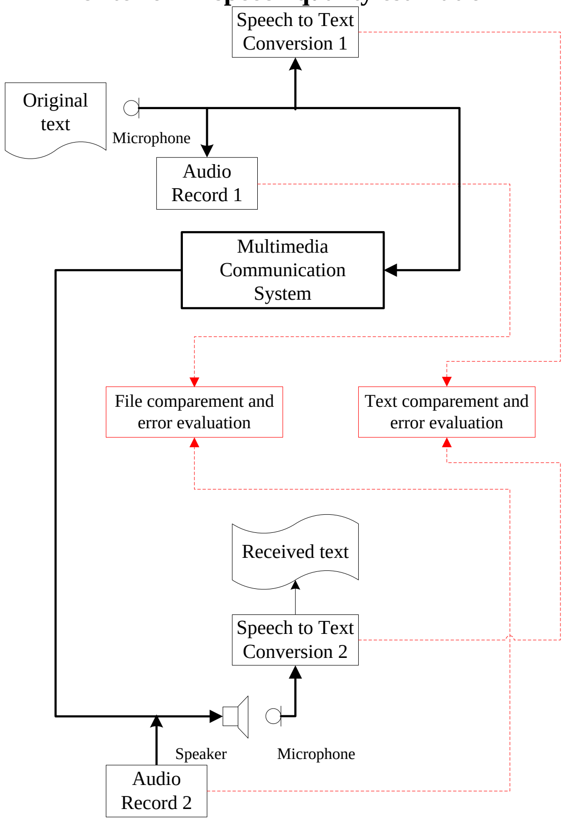
I. Introduction

The audio signal quality is important in voice communication systems, in which generally the information is carried out from speech signals. There are subjective and objective speech quality measurements and estimation method [1, 2, 3, 4] and standards [5, 6, 7]. Each of these methods or standards is prepared for specific cases of speech signals coding and concrete characteristics of communication channel in voice communication systems. The goal of this article is first to create and use the different simulation models of voice communication system and second to use these models in the proposed here method of objective speech quality estimation replacing person as estimator in subjective methods with text to speech and speech to text methods as criterion in speech quality estimation [8]. The main advantages from this proposition are to eliminate the human subjective factor in speech quality estimation process and to approach the precision of objective speech quality methods to the higher precision of subjective methods.

II. The proposed method of objective speech quality estimation using text to speech and speech to text methods as criterion to replace person as estimator in subjective methods of speech quality estimation

The proposed method developed of objective speech quality estimation with replacing person as estimator in subjective methods with text to speech and speech to text methods as criterion in speech quality estimations is presented in Fig.1 in form of an algorithm. The main difference of the proposed method from the existing methods consists in the proposition to convert speech signals into text file. In the beginning of the algorithm is used an original text (marked as block “Original text”) from printed document or computer file, which is read into a microphone device connected to the computer system and is converted into a speech signal. The input speech signal is recorded as audio file (marked as block “Audio Record 1”) in the computer system and simultaneously is converted into a digital text file (referred as block “Speech to Text Conversion 1”). The speech signal is transmitted via multimedia communication channel (block “Multimedia Communication System”) and is received from the receiver part of the multimedia system and is reproduced by loudspeaker device (presented as “Speaker” in Fig 1). At the same time the received speech signal is recorded on the computer as audio file (marked as block “Audio Record 2”). In front of and nearby the speaker device is placed another microphone, which transmits the speech signal for conversion into a new text file (referred as block “Speech to Text 2”).

Figure 1 Block algorithm of the proposed method developed of objective speech quality estimation with replacing person as estimator in subjective methods with text to speech and speech to text methods as criterion in speech quality estimation



After execution of the algorithm presented in Fig.1, the goal is to prepare the comparison (marked as block "Text compartment and error evaluation" in Fig.1.) and calculation of the number of incorrect received words between the two digital text files:

- one generated in the transmission part (text document created after speech to text transformation in transmission part and saved as text file **stt.txt**) ;
- second created in the receiver part (text document created after speech to text transformation in receiving part and saved as text file **rev_stt.txt**).

As a result of error evaluation is defined an exact objective quality assessment of the speech signal:

$$OSQE_D = DNErW = NErW_{re} - NErW_{tr} \tag{1}$$

or

$$OSQE_R = \frac{NErW_{tr}}{NErW_{re}}, \tag{2}$$

where

$OSQE_D$ and $OSQE_R$ are the objective speech quality estimations defined as difference (DNErW) or as ratio (RNErW) between the number of erroneous words in receiving ($NErW_{re}$) part and the number of erroneous words in transmission ($NErW_{tr}$) part of the voice or multimedia communication system.

In addition to the proposed above new objective speech quality estimations is possible to prepare an extra or additional estimation function for more precise objective speech quality assessment of the method and algorithm presented in Fig.1. This additional estimation is proposed to prepared as a possible comparison (marked as block "Audio comparison and error evaluation" in Fig.1.) between the two audio records "Record 1" and "Record 2".

- original speech signal saved as speech file **orig.wav**,
- received speech signal saved as speech file **rev.wav**.

The proposed above block schema of objective speech quality estimation with replacing person as estimator in subjective methods with text to speech and speech to text methods as criterion in speech quality estimations is presented in Fig.1 in form of an algorithm.

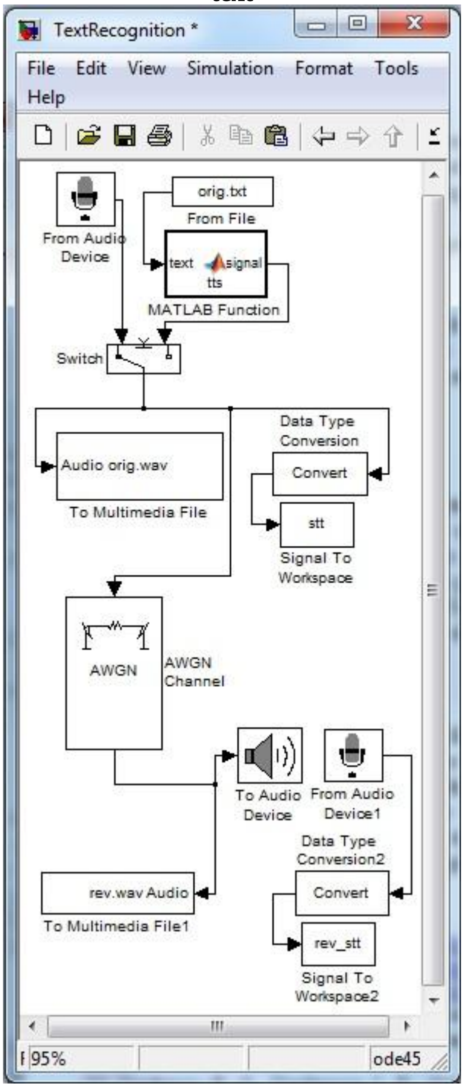
The block schema from Fig.1 is developed as simulation model for using in experimental modeling and analysis of the proposed in this article objective speech quality estimation with replacing person as estimator in subjective methods with text to speech and speech to text methods as criterion in speech quality estimations. The development of the simulation model is made in following two modifications:

- simulation model of audio input part with text to speech and audio output part with speech to text;
- simulation model of communication channel with the ability to switching and choosing the channel type and channel coding methods.

III. Development of the simulation model of audio input part with text to speech and audio output part with speech to text

The simulation model of audio input part with text to speech and audio output part with speech to text is presented in Fig. 2 and is developed using Simulink in Matlab.

Figure 2 Simulation model of audio input part with text to speech and audio output part with speech to text



There are presented on Fig. 2 two types of possibilities to choose the source of the speech signal:

- real speech signal direct from microphone (From Audio Device);
- speech signal converted from a speech to text system (Data Type Conversion)Estimation of the received after simulation speech signals to analyze their of speech quality dependence from channel type and channel coding methods.

In the transmission part on Fig. 2 the speech signal is saved as audio file (To Multimedia File) and in the same time is transmitted via communication channel of the Multimedia Communication System, in which is possible to define the level of noise and disturbances. In the receiving part are prepared similar operations like as in the transmission part. The received speech signal is saved back as audio file (To Multimedia File 1) and in the same time is reproduced with speaker (To audio device). With a microphone (From audio device 1), placed in front of the speaker is possible to made an inverse speech to text conversion (Data Type Conversion 2). This text is saved as a new received text document, which is used in the next step of the proposed method – the relative objective measures or estimations of speech quality in the multimedia system, described in next paragraph.

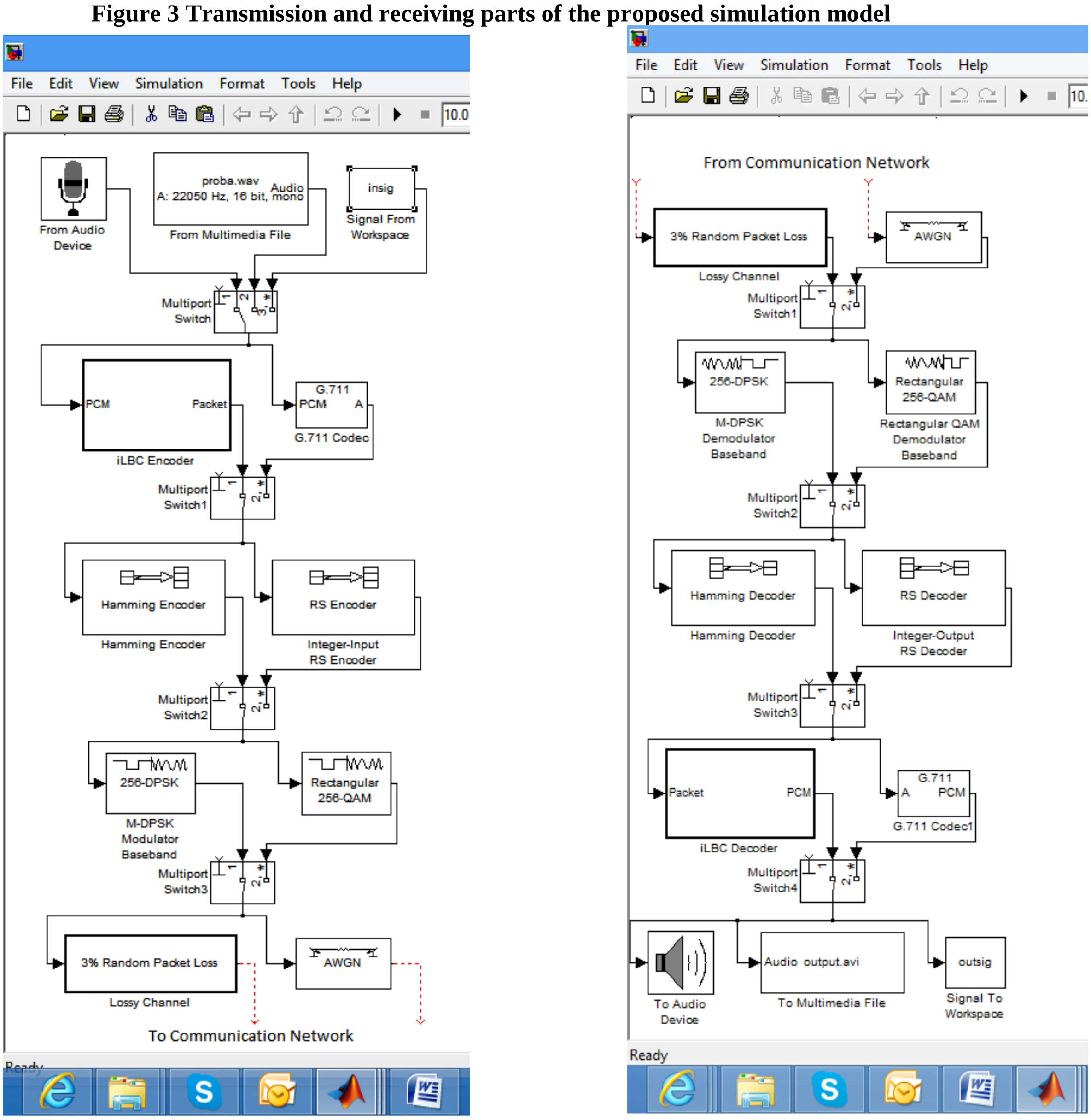
IV. Development of the simulation model of communication channel with the ability to switching and choosing the channel type and channel coding methods

In Fig.2 is shown only a simple case of a voice communication channel (an AWGN Communication Channel block from Matlab Simulink Library). To extend the application of the proposed in Fig. 2 simulation model with different types of multimedia communication channels and also with wide spreads channel coding methods it is developed the second modification of the proposed simulation model shown in Fig. 3. In this modification are used and can be switched blocks from Matlab Simulink Communications System Toolbox for different encoding, decoding blocks, blocks with different type of modulation, communication channels with different noise and error rate characteristics.

V. Simulations and experimental results of the proposed objective speech quality estimation based on original and received texts comparison

The proposed above equations (1 and 2) are two possible relative objective measures or estimations of speech quality in the multimedia or voice system. They are used in experiments with the developed in Fig. 2 and Fig. 3 two modifications of simulation models for objective measures or estimations of speech quality in the multimedia or voice communication systems.

Figure 3 Transmission and receiving parts of the proposed simulation model



These equations gives as results in experiments the values of relative objective speech quality estimations, which can be compared with some of existing subjective methods of speech quality estimation. Some of the important results from the simulations are shown in Fig.4. On this figure are shown the results from a simple example of one of the simulations:

- the original text after speech to text transformation **stt.txt**;
- text document created after speech to text transformation in receiving part **rev_stt.txt**.

It can be seen from Fig. 4, that there are differences of the number of erroneous words in the text documents **stt.txt** and **rev_stt.txt** after speech to text transformation in transmission and in receiving parts of the simulation model. This difference is used to calculate with the equations (1 and 2) the values the objective speech quality estimation (OSQE) as difference (DNErW) or as ratio (RNErW) between the number of erroneous words in receiving ($NErW_{re}$) part and the number of erroneous words in transmission ($NErW_{tr}$) part. For this example the concrete values of $NErW_{re}$ and $NErW_{tr}$ are: $NErW_{re} = 10$ and $NErW_{tr} = 5$.Then from the equations (1 and 2) are calculated the values:

$$OSQE_D = DNErW = NErW_{re} - NErW_{tr} = 10 - 5 = 5 \qquad OSQE_R = \frac{NErW_{re}}{NErW_{tr}} = \frac{10}{5} = 2 \quad (3)$$

Figure 4 Text documents stt.txt rev_stt.txt after speech to text in transformation and receiving parts

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1. add the text2speech folder to your Matlab path
2. Test your new function:
Get started, if you use SAPI (before .NET)...
1. Make sure SAPI is installed on your computer
a) get the Speech SDK 5.1 (86MB) for free from Microsoft:
b) test your default computer voice
2. add the text2speech folder to your Matlab path
3. Test your new function: ('This is a test.')
I would like to thank "Desmond Lang" for his Text-To-Speech tutorial
and my wife for letting me play with the computer ;).

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The values calculated in equations (3) are only a demonstration of the methodology necessary to apply the proposed method in simulations and real applications using the texts with larger number of words to achieve the more realistic and precise results. These results exist, but are not shown here a cause of limited size of this article.

VI. Conclusion

In this article is proposed the application of text to speech method in transmission part and speech to text method in receiving part of a multimedia system, as means to replace human as speaker in transmission part and human as listener in receiving part of the multimedia system. The proposed method is developed as simulation model and a lot of simulations are prepared from which it is seen that the proposition of using text to speech and speech to text methods gives good results for objective speech quality estimation in multimedia system with the advantage of elimination the human subjective factor in speech quality estimation and of achievement of a near to in objective speech quality methods near to the precision of subjective methods.

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