INTUITIONISTIC FUZZY ESTIMATIONS OF NOISE FILTERING IN SPEECH

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ABSTRACT

The use of applications for noise reduction often changes the characteristics of the input signal. The successful methods for eliminating noise unfortunately require high computational complexity, and thus the process of obtaining the final result is slow.

The present article studies a model demonstrating the cording signal of human speech, visualizing the noise in signal and its reduction through the Matlab application.

It is necessary to definite the level of detail for performance of each step of the algorithm, as well as the provisions for possible exits and rules for making decisions. An ongoing analysis of the processes occurring during the execution algorithm is carried out. Intuitionistic fuzzy sets are used to estimate noise reduction in speech.

Key words: *voice signal, filter, intuitionistic fuzzy estimations, signal discretization, noisy voice signals, Matlab*

INTRODUCTION

Speech is a basic means of communication between people. Human speech is a fundamental way for humans to convey information to one another. With a bandwidth of only 4 kHz; speech can convey information with the emotion of the human voice. Some properties of the speech signal are: a one-dimensional signal, time as its independent variable, random in nature, non-stationary and the frequency spectrum not constant in time. Although human beings have an audible frequency range of 20 Hz to 20 kHz, human speech has significant frequency components only up to 4 kHz. The effect of interference noise in speech signals is the most common problem in speech processing.

Speech is distributed in the form of a sound wave which spreads from the source in all directions and which is perceived by the auditory system. Speech wave is seen as an analog signal controlled by the photon apparatus of man. It is presented in the form of a signal with a certain amplitude and length (wavelength). Speech signals consist of analog speech models based on undergoing of the process of converting a continuous analog signal into a signal with discrete values called discretization, analysis and symbolic presentation of the speech (Figure 1) [3-4]. Each sound wave can be represented as a spectrum of the incoming into its composition frequencies.

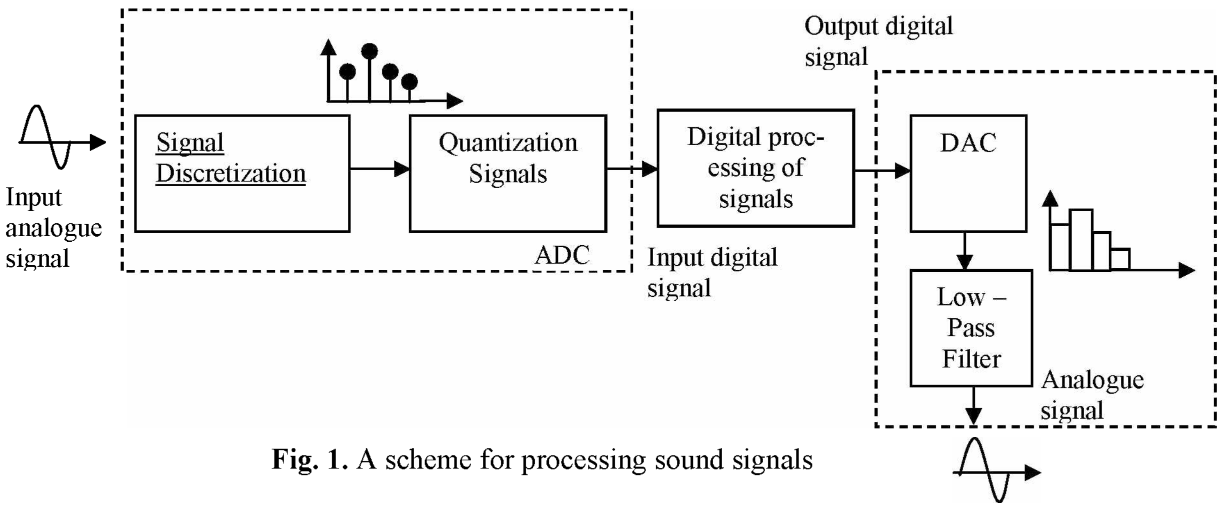

Fig. 1. A scheme for processing sound signals

When doing research on any spoken language, first of all the individual phones must be identified. Then it is determined in what position and what sound environment they meet each other and what their distribution is. The phoneme as the smallest segmental unit of speech serves as a differentiation of morphemes and word forms. The phoneme system of Bulgarian language consists of 45 phonemes: 6 vowels (vocals) and 39 consonants. In speech it is realized in the form of various options. The sensing characteristics of human speech have been estimated using the phonemes of the phoneme system of Bulgarian language.

This article is the first of a series of articles exploring many problems associated with noise filtering and recognition of Bulgarian speech. It examines how to apply filters to reduce noise and improve the original source voice recording. Then it will be used **OM** to describe the flow of the processes during training and at the end it will be used **IRO** to compile the assessment of the applied filters in the current examination. With the Matlab application a large number of experiments are conducted to maximize noise reduction. Figures 2 and 3 present a noisy speech signal and an estimated speech signal.

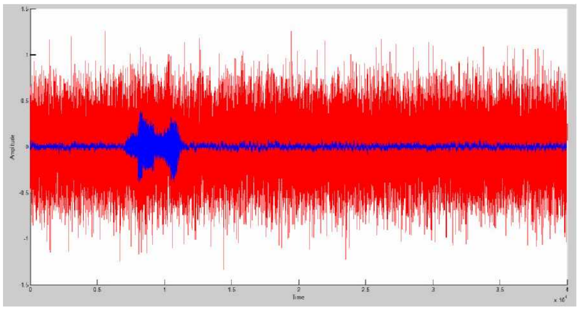

Fig. 2. Noisy speech signals

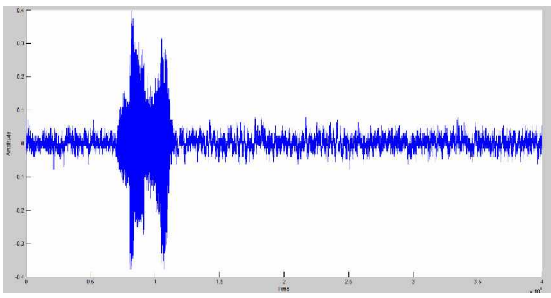

Fig. 3. Filtered speech signals

Interference noise masks the speech signal and reduces its comprehensibility. It can come from acoustical sources such as ventilation equipment, crowds, traffic, reverberation and echoes. It is possible to arise electronically from thermal noise, distortion products or tape hiss. If the sound system has unusually large peaks in its frequency response, the speech signal can even end up masking itself.

GENERALIZED NET MODEL

The generalized net model describes the process of applying algorithm and estimations in terms of IFS [1-2]. Initially, the following tokens enter the generalized net with the respective information characteristics:

In place l_{11}, a token enters with a characteristic "*speech signals*";

In place L_{2A} a token enters with a characteristic "*algorithms for noise filtering*";

In place L_{3A} a token enters with a characteristic "*compared algorithms*";

In place L_{4A} a token enter swith a characteristic "*calculating of intuitionistic fuzzy estimations*".

A generalized net model with an introduced set of transitions A: $A = \{Z_1, Z_2, Z_3, Z_4\}$ is developed, where the transitions describe the following processes, respectively:

Tasks performed by **speech recorder** – transition Z_1;

Tasks performed by **noise filtering algorithms** – transition Z_2;

Tasks performed by **compared input and filtered signals** – transition Z_3;

Calculation of intuitionistic fuzzy estimations – transition Z_4.

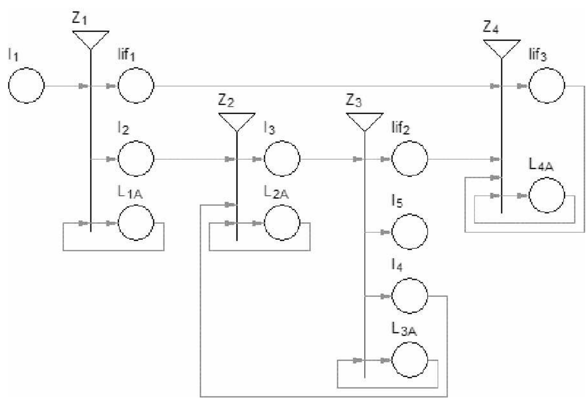

Fig. 4. GN-Model of noise reduction in speech

The Z_1 = $\{l_1, L_{1A}\}$, $\{l_{If1}, l_2, L_{1A}\}$, R_1, (l_1, L_{1A}) index matrix of the transition conditions is:

$$R_1 = \begin{array}{c|ccc} & l_{if1} & l_2 & L_{1A} \\ \hline l_1 & false & false & true \\ L_{1A} & W_{1A,if1} & W_{1A,2} & true \end{array} \qquad (1)$$

where:

$W_{1A,if1}$ = "is available signal";

$W_{1A,2}$ = "is available signal to noise filtering".

The token entering place l_2 obtains new characteristics "signal for noise filtering".

The Z_2= $\{l_2, l_4, L_{2A}\}$, $\{l_3, L_{2A}\}$, R_2, (l_2, l_4, L_{2A}) index matrix of the transition conditions is:

$$R_2 = \begin{array}{c|cc} & l_3 & L_{2A} \\ \hline l_2 & false & true \\ l_4 & false & true \\ L_{2A} & W_{2A,3} & true \end{array} \qquad (2)$$

where:

$W_{2A,3}$ = "noise filtering is complete".

The Z_3 = $\{l_3, L_{3A}\}$, $\{l_{if3}, l_4, l_5, L_{3A}\}$, R_3, (l_3, L_{3A}) index matrix of the transition conditions is:

$$R_3 = \begin{array}{c|cccc} & l_{if3} & l_4 & l_5 & L_{3A} \\ \hline l_3 & false & false & false & true \\ L_{3A} & W_{3A,if2} & W_{3A,4} & W_{3A,5} & true \end{array} \qquad (3)$$

$W_{3A,if2}$ = "good noise filtering";

$W_{3A,4}$ = "additional application of filters";

$W_{3A,5}$ = "noised or rejected signal".

The index matrix of the transition conditions is:

Z_4= $\{l_{If1}, l_{If2}, l_{If3}, L_{4A}\}$, $\{l_{If3}, L_{4A}\}$, $R3$, $(l_{If1}, l_{If2}, l_{If3}, L_{4A})$

$$R_4 = \begin{array}{c|cc} & l_{if3} & L_{4A} \\ \hline l_{if1} & false & true \\ l_{if2} & false & true \\ l_{if3} & false & true \\ L_{4A} & W_{4A,if3} & true \end{array} \qquad (4)$$

where:

$W_{4A,if3}$ = "results are estimated".

The token entering at place L_{4A} obtains characteristic "estimations μ_i, ν_i".

All calculated estimations take initial values of $\{0, 0\}$, when $i \geq 0$, the current $(i+1)^{-st}$ estimation is calculated on the basis of the previous estimations according to the iterative formula:

$$(\mu_{i+1}, \nu_{i+1}) = (\frac{i.\mu_i + \mu}{i=1}, \frac{i.\nu_i + \nu}{i=1}), \qquad (5)$$

where are the previous estimations and (μ, ν) is the latest estimation of the messages for $\mu, \nu \in [0, 1]$ and $\mu + \upsilon \leq 1$.

In this way the token at place l_{if3} forms the final estimation of noise reduction on the basis of the previous and the last calculated algorithm.

CONCLUSIONS

Efficient algorithms are applied for noise filtering through Matlab partial filtering of noise from speech signals. Tools for visualization of the signals in different stages of the obtained results are used.

GRATITUDES

The authors are thankful for the support provided by the Bulgarian National Science Fund under Grant Ref. No. DFNI-I-02-5 "InterCriteria Analysis: A New Approach to Decision Making".

REFERENCES

1. Atanassov, K. *Generalized Nets, World Scientific*, *Singapore, 1991*.
2. Atanassov K., *Intuitionistic Fuzzy Sets: Theory and Applications*, Springer Physica Verlag, Berlin, 1999, pp. 1-137.
3. Bondarev V.N, Troester G., Chernega V.S, *Digital signal processing: Methods and Instruments*, Ukraine, 2001.
4. Jangir, J., Singh, B., Ali, M., *Voice Identification Secure System by Statistical Model of Speech Signal Using Normalization Technique*, International Journal of Engineering Research & Technology (IJERT), ISSN: 2278-0181, Vol. 3 Issue 1, 2014.