Abstract: Modern medicine calls for new diagnostic methods. Emphasis is placed on non-invasive methods. In addition, they should be characterized by high efficiency, which is a combination difficult to predict. In this area signal processing offers the greatest potential. It is used in many branches of medicine. This article presents one of the possible uses of signal processing, focused on the pathologies of voice, resulting from brain damage caused by vascular problems. Group of 41 patients neurology branch was recorded, with indications of ischaemic stroke, or hemorrhagic. The results clearly indicate the possibility of using the selected voice signal processing algorithms.

Keywords : Signal processing, voice pathology, stroke, vocal track filter

I. INTRODUCTION

Exploiting of non-invasive method for diagnosis purpose is frequently more popular in medical environment presently. Also an increasing of diagnosis accuracy and speed of results obtainment and simplicity of evaluation has been observed. Operations of these kinds impose miscellaneous demand in relation to length and qualities of samples data. Probably, non-invasiveness is the most desirable feature for neurologists. It results from serious danger of injury during invasion operations on most important human organ – cerebrum. That’s why authors have suggested considerable expansion of existing method of patient’s condition estimate and monitor hospitalization progress, on base of voice parameters. Existing non-invasive methods of diagnosis are magnetic resonance and tomography. Nevertheless, most often they are execute only one time, during acceptance of patient on ward. Besides, analyzing of voice is fast and convenient.

II. METHODS

Presently, majority applicable method does not allow exact results getting properly, because of susceptibility of algorithm on errors in progress of recording emerged, and come of lack or scarce correction mechanisms, which could be adaptable to external conditions. Besides, it belongs to take into consideration approximation errors. The simplest methods of voice quality evaluation are based on experienced phoniatrist opinion. The subjective classification of the voice requires experience and intuition, and cannot be applied commonly, particularly in comparative investigations led through the various medical centers.

Objective acoustic analysis is perfect technique of estimate of voice quality definitely. Spectrographic, sonographic and the temporary analyses of the signal of the speech are useful in objective acoustic methods of the voice measure. Computer technology leaves across these requirements offering speed and convenience of computing.

Speech signal can be regarded as a dynamic object. Systems that track the volatility of stocks of such plants make use of the linear recursive estimation. These tools are for tests used to evaluate both the signal input and output characteristics, which are the result of actions processing functions. There are two methods here: the method of least squares and the minimal-mean-square method. Using the first one, the average signal of N samples can be estimated. It can be written as:

\[ \bar{x}_N = \frac{1}{N} \sum_{n=1}^{N} x(n) \]  

This model can be written in the form:

\[ \bar{x}_N = \frac{1}{N} x(N) + \frac{1}{N} \sum_{n=1}^{N-1} x(n) = \frac{1}{N} x(N) + \frac{1}{N} \left( \sum_{n=1}^{N-1} x(n) \right) \]  

That is, to present a new estimate of the average of N points as the sum of the old estimate, calculated on the basis of N-1 points, and its correction after taking into account the new n-th sample x(n). The adjustment is calculated as the weight value of the error between the value of a new sample, and an old estimate of a mean value. This pattern is the current standard in adaptive recursive estimation of the parameters:

new estimate = prognoses + correction  
correction = amplification  (measurement – prognoses of measurement)

where one of parameter is measured and another, related with it, is estimated.

The function of the quality of least-squares estimation is defined as:

\[ J = (\tilde{z} - H \hat{\tilde{x}})^T (\tilde{z} - H \hat{\tilde{x}}) \]
where: $\hat{x}$ is a measured vector, $\hat{\hat{x}}$ is estimate of vector generated by physical object, and $H$ is matrix of measurement system. Estimate can be appoint as:

$$\hat{x} = (H^TH)^{-1}H^T\hat{\hat{x}} \tag{4}$$

In the case of the minimal-mean-square method quality function becomes:

$$J = E[(x - \hat{x})^T(x - \hat{x})] \tag{5}$$

where $E[.]$ is the expected value in statistical terms. In this case, the model boils down to two equations: the process model and measurement model. In the case of estimation of power spectral density function of signals to process model and measurement model. In the case of speech, where:

$$P_{MM}(e^{\imath \omega}) = \frac{1}{N}\sum_{k=p+1}^{N} |v_k|^2$$

In addition, the calculations of spectroscopic estimation were carried out by Welch method. The analysis of variable frequency signals using non-time-frequency representation of signals was used. Used Fourier transform STFT (Short-Time Fourier Transform) and the Wigner-Ville transform. The first can be interpreted as non-discredited in time and frequency Gabor transform. Used a description: 

$$STFT_{\tau}^\tau(t,f) = \int_\tau^\tau x(\tau)\gamma^\gamma(\tau-t)e^{-j2\pi \tau \tau}d\tau \tag{11}$$

$$STFT_{x}^\tau(t,f) = e^{-j2\pi \tau \tau}\int_\tau^\tau X(\nu)\Gamma^\nu(\nu-f)e^{-j2\pi \nu \nu}d\nu \tag{12}$$

where $\gamma(t)$ is a function of the time window of observation, $\Gamma(f)$ is the Fourier spectrum that acts as a window.

In the case of Wigner-Ville transform should be noted that it perfectly reflects in the time-frequency linear change of frequency. By definition:

$$S_{s}^{W_{V}}(t,f) = \int_\tau^\tau x(t+\tau)\gamma^\gamma(t-\tau)e^{-j2\pi \tau \tau}d\tau \tag{13}$$

$$S_{x}^{W_{V}}(t,f) = \int_\tau^\tau X(f+y)\Gamma^\nu(f-\nu)e^{-j2\pi \nu \nu}dt \tag{14}$$

This representation is characterized by the highest concentration of energy in the time-frequency space that means that has the best resolution.

In addition, in our investigation a vocal track filter transform is determined, which maximum on the characteristics of time-frequency results from a formant diagram. The purpose of the designation of the fundamental frequency is to detect the first signal by the maximum value on the axis and cut the higher frequency harmonics. Then a maximum for each step are determined again in the given range. Used here with autocorrelation functions, which turned out to be more accurate than cepstral and timing method to determining the basic frequency. Also spectrogram of the input signal is appointed for power of the harmonics tracking.

**III. RESULTS**

Form of obtained speech sound is determined by specificity of executive voice apparatuses. Match of vocal strings generates sound and it subjects modulation during proceeding by vocal track. It depends on programming action of the central nervous system and the condition of the broadcast of stimuli in cirtial-subcortical area, in the trunk of the cerebrum, nerves and nervous-muscular
synapses. That’s why authors are of an opinion that a large capability exists to diagnose and estimate of injury of cerebrum on base of analysis of patient voice. Additionally, it is possible to get information of progress of treatment in a fast and simple manner. For our study several patients with most commonplace injury of central nervous system have been included, namely strokes hemorrhagic and ischaemic. These patients have problems with speaking out, which is defined as aphasia. However, even at patients who are good speaking out, possibility of changes detection in course of chosen characteristics has been checked. This paper contains results of present evaluations, in cooperation with neurologists of one of hospital in Lodz.

The investigation confirmed the usefulness of the signal analysis of speech in monitoring abnormalities the sound. The fundamental frequency was disturbed in almost all patients. This applies primarily to patients with ischaemic stroke of any cerebrum hemisphere. Patients with hemorrhagic stroke, and haematoma in the right hemisphere had little change in the fundamental frequency.

Fig. 1. Fundamental frequency for patient with ischaemic stroke in left hemisphere in the first (top) and last (bottom) day of hospitalization.

In some patients noted a significant increase in the fluctuations periodogram, with the progress of hospitalization. Probably this is related to the clean tone and the power generated by the larynx and depends on the vocal track in lesser extent. Interestingly, these changes were recorded for the case ischaemic stroke in left hemisphere and hematoma in the left and right hemisphere.

Fig. 2. Periodogram for patient with ischaemic stroke in the left hemisphere in the first (top) and last (bottom) day of hospitalization.

In both cases, the visibility was noted significant differences in the characteristics of vocal track filter. In the first days of the charts are very flat, and differences appear along with the progress of treatment. Each of the peaks corresponds to the formant graph, which means that they are clearer. Here was shown also another case of ischaemic stroke in the left cerebral hemisphere in right-handed person.

Fig. 3. Vocal track filter for patient with traumatic haematoma in the left and right hemisphere in the first (top) and last (bottom) day of hospitalization.
For patients with ischaemic stroke in the left hemisphere and aphasia one noted that with the progress of treatment the second formant is distinguished. This is independent of gender.

In addition, the Welch method of spectral estimation shows similar changes as the vocal track filter. In addition to this there is a more detailed, and thus signal changes can be analyzed in a more precise manner. Approximation should carry out, in order to remove the disturbance of signal and noise.

In the case of hemorrhagic stroke in the right hemisphere to increase the altitude of the peaks on the characteristics of the vocal track filter was observed. The changes are compatible with the emergence of a first and second formant. The power of the other formants remained unchanged.

In right-handed person with ischaemic stroke in right hemisphere we also noted changes in the characteristics of the vocal track filter and fundamental frequency progress. It reflects on the first formant in terms of increased clarity of this formant. And in the case of the other formants one noted a slight decrease in their gain.

At the same time, the results of other types of analysis does not give unambiguous answers to their usefulness in the diagnosis of disorders of speech. This may be due to too small amount of material inaccuracies in the test or imprecise calculation. In the future, it will be developed and tested new types of analysis on the held and the newly gained samples. Statistical functions will be designed also to computer evaluation of results.

IV. CONCLUSION

In accordance with mentioned limitations, using technique relying on more advanced signal processing for obtainment more precise results, we expect to open new capabilities in non-invasion research of patients with cerebrum injury with blood vessel problems. Authors expect, that information obtaining by using this technique will be helpful at diagnosis and for estimation of hospitalization progress for persons with mentioned above ailment. It can be useful supplement for conclusions of tomography and magnetic resonance images, as well as for monitoring condition of patient health in assigned period of time.

V. REFERENCES