UDC 534; 53.08:681.3

TIME-FREQUENCY PROCESSING OF THE MEASURING SIGNAL FOR THE ESTIMATION OF IMMUNITY OF SPEECHTHE

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Article describes results of theoretical and experimental researches on the method of processing of a LFM-signal for the estimation of speech safety. The given method expands possibilities of the existing automated monitoring systems. The effect is reached at the expense of application of more complex procedures of signal processing and hindrances in the monitoring system which use techniques of the joint time-frequency analysis of a signal.

Introduction. Wide use of transferring and processing of the semantic information causes an urgency of its protection. Transferring and processing of the speech information in the analogue form is rather actual as it is basic. Protection of speech information and video information in the analogue form is based on the energy method, i.e. noise (or noise-type) masking signals are formed. The security estimation of such signals is based on the usage of noise signals and on harmonic measuring signals in the automated systems. Noise signals do not allow to show channels of information of leakage when detecting weak signals in high level noises. As a result some channels of informational leakage do not use noise signals for the safety estimation of the dedicated object.

Harmonic signals have the advantage in comparison to noise signals. The application technology of such signals defines the usage of special measures for measuring signal decrease therefore raises the accuracy of measurements. The use of a harmonic measuring signal on mid frequencies of strips of equal legibility or on mid frequencies of third-octave strips, and even worse, on the mid frequencies of octave strips, allows considerable faults in leak channels with vividly expressed flatness of AFC in measured range of frequencies.

It is necessary to carry, first of all, reverberation hindrances, resonances, the resonant phenomena of closed spaces of dedicated premises, AFC non-uniformity, caused by local areas, attenuations non-uniformity, the presence of artificial acoustic and vibroacoustic hindrances that a referred to factors which affect the increase in security estimation errors.

In our opinion, the amendment of a number of disadvantages is teasible by using the measurement of complex signals. The usage of complex signals such as signals of linear frequency modulation (LFM-signals further), Barker code in hydrolocation, radiolocation, is widely applied, however, such signals have disadvantages too.

Namely LFM-signals possess the advantages among all these signals and LFM-signals allow to take into account the uneven frequency response. With proper selection in rate of frequency change other shortcomings may be omitted, however, the frequency-modulated signals have disadvantages due to the threshold effect.

Purpose: LFM-signal research as a measuring signal, to minimize the effect of the threshold for detecting weak signals in high noise level.

1. Problem statement

Application of LFM-signals will expand the opportunity to assess the security of speech, so in such a way the entire bandwidth of an octave (third-octave, strip of equal legibility) will be monitored, and not only of individual point on the frequencies axis.

Technical advantage in using LFM-signals is that it is possible to distribute the energy of a given duration on the frequency band and thus reduce the requirements for pulse power source.

Considering that these methods are intended for the usage in the automated monitoring channel system in the leak of speech information, such as СИА К6-6 [1], it is also necessary to solve a problem of synthesis of corresponding algorithms of digital processing of signals.

The purpose of processing is to define the degree of attenuation of measuring signals when passing through the barriers and calculation by the results of the measurements which characterize the interaction of test signals with obstacles in dedicated objects. The disadvantage of measuring LFM-signals is the presence of so-called threshold effect. That is when reducing the relation of the signal/noise to certain value (threshold) we can observe sharp decrease in "visibility" of a signal (not possibility of separation of a signal and noise).

2. Methodology of decisions

In [2] for problem solving in parametres estimation of signals with the complex time-and-frequency structure it is offered to use techniques of combined time-frequency descriptions of signals. Among set of forms of time-frequency descriptions the preference is given to the function of density distribution of alarm energy by Wigner:

$$Pw(f,t)=\int_{-\infty}^{\infty} Z_a^*(t-\tau/2)Z_a(t+\tau/2)\exp(-j2\pi f\tau)d\tau, \tag{1}$$

where $Z_a(t)=Z(t)+j\tilde{Z}(t)$ is an analytical signal; $\tilde{Z}(t)$ – Hilbert's transformation of the valid signal $Z(t)$; * is a sign of complex conjugation.

Use of synchronous accumulation method will allow to lower threshold effect when separating a signal from noise.

The measuring signal is represented by sequence of identical LFM-signals with general duration T_n:

$$T_n = CT, \tag{2}$$

where C is an amount of LFM-signals, T is a duration of one LFM-signal.

After execution of measurements the received mixture with a duration of T_n we will present, as one impulse duration T (applying a method of synchronous accumulation). We will receive the given transformation to a discrete kind:

$$Sc_n = \frac{1}{C}\sum_{i=0}^{C-1} S_{n+i\cdot N}, \tag{3}$$

where $n \in 0\ldots N-1$, Sc_n is an accumulated signal; C – accumulation count; N – Amount of digitization points of one period in the LFM-SIGNAL, S is a mixture LFM-signal and noise.

3. The received results

Experiment was carried out with the usage of classical model of the LFM-signal [3].

$$S(t) = A\sin\left(2\pi(f_0 + \mu\cdot t^2)\right), \tag{4}$$

where A is a signal amplitude, f_0 is an initial frequency of LFM-signal, μ is a speed of change of instant frequency.

Let's stop on the example of the signal analysis for the front page of equal legibility where instant frequency varies in limits from 100...420 Hz [1].

Duration of one LFM-signal T = 1 s. As the top frequency in the general controllable strip of frequencies makes 420 Hz we will believe, that the top frequency of a strip of the analysis (a strip of frequencies in which signal processing is executing) f_{max}= 512 Hz.

According to the Nyquist theorem, the sampling frequency $f_d = 2f_{max}$ = 1024 Hz. Digitization interval $\Delta t = \frac{1}{2f_{max}} = \frac{1}{1024} \approx 976{,}6$ μs. Hence, on time interval in one second the total of time keeping of a signal will be equal $N = T/\Delta t = 1024$.

In this case, for modeling the signal we will present on the computer (4) in discrete form:

$$S(\mathrm{n}) = \sin\left(2\pi\cdot\frac{n}{1024}\left((100+320\frac{n}{1024\cdot 2}\right)\right), \tag{5}$$

where $n \in 0\ldots 1024$.

In figures 1 and 2 the fragment of the received LFM-SIGNAL and its spectrum are presented.

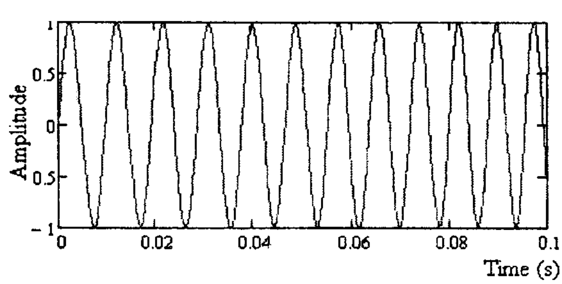

Fig. 1. Fragment signal linear frequency modulation

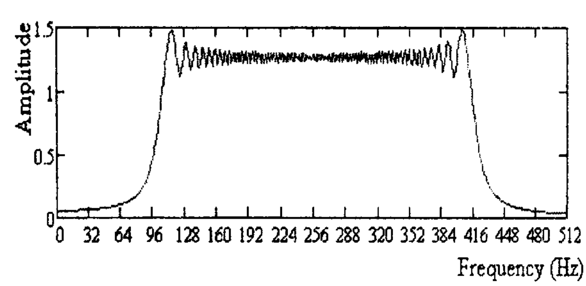

Fig. 2. Spectrum fragment LFM-signal

Amount of accumulation of LFM-signals we will establish as 10, so we will receive the following representation of a measuring signal:

$$S(n) = \sin(2\pi\cdot\frac{n\cdot ranc(n/1024)}{1024}\left(100+320\frac{n\cdot ranc(n/1024)}{1024\cdot 2}\right), \tag{6}$$

where $n \in 0\ldots 1024$, ranc – truncation of the division.

Now consider the noise model $\eta(t)$. Let's imagine that noise is a subject to a normal law of distribution. The mathematical expectation of the noise is zero, but the standard deviation of the instantaneous noise process σ changes depending on the experimental conditions. In this case, the noise will operate in the frequency band of 512 Hz.

After creation of model of a signal and noise we will pass to consideration of algorithm of synchronous accumulation. For this purpose we will use the formula (3).

In figures 3 and 4 the mixture of a measuring signal and noise before accumulation and after accumulation accordingly are represented.

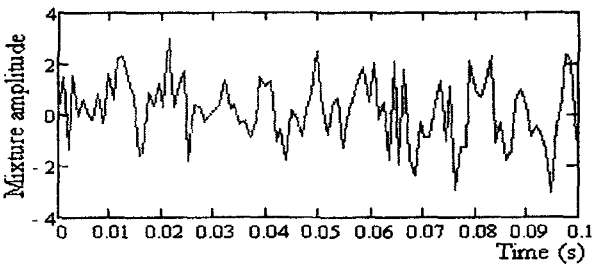

Fig. 3. Fragment of a mixture of the LFM-signal and noise before accumulation

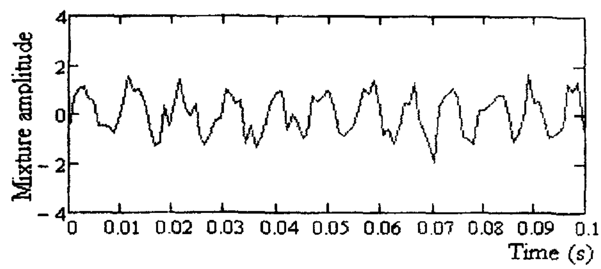

Fig. 4. Fragment of a mix after accumulation

For this purpose we will present the discrete form of Wigner distribution and we will define procedure of transformation of the valid signal in a analytical signal.

The discrete form of Wigner distribution can be presented in the expression [5]:

$$Pw(k,n) = 2\sum_{m=-N+1}^{N-1} Z_a^*(n-m)Z_a(n+m)e^{-j\frac{2\pi km}{M}}, \tag{7}$$

where $M > 2N - 1$; $k \in -N + 1 \ldots N - 1$. Note that the process of transition from analog form to the discrete Wigner distribution is not trivial. A detailed description of this procedure is given in [4]. From the above formula it is seen that, for the processing of the analog signal it must be represented in discrete form with a frequency more than twice the Nyquist sampling frequency.

The calculation procedure of an analytical signal consists of the following [5, 6].

1. Compute fast Fourier transform of real signal Z(n);
2. Spectrum for negative frequencies is cleared. The range for positive values of the frequency is multiplied by two.
3. Active inverse fast Fourier is used. In the result we have an array of complex values of the process, the real and imaginary parts of which are interconnected by Hilbert transformation.

In Figure 5 shows the frequency slices by Wigner distribution for different times (0, 500 and 1000 ms).

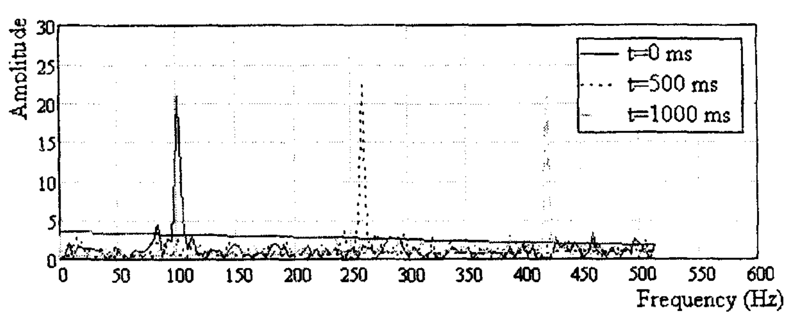

Fig. 5. Wigner distribution LFM-signal: frequency slices Wigner distribution at time t = 0, 500, 1000 ms

The legibility of speech is estimated by a technique stated in [1].

The measurement in different time intervals of a signal power and noise at a rather small values of signal power in noise causes a methodical error Measurement in one time interval (a signal + noise) and noise excludes a methodical error.

The proposed changes allow separate weak signal from noise.

In figure 6 it is displayed distribution by Wigner of the LFM-signal in noise, on it it one can track signal strength change on a range of frequencies depending on time. Figure 7 displays the distribution of the Wigner LFM-signal in the noise in the horizontal level (frequency-time).

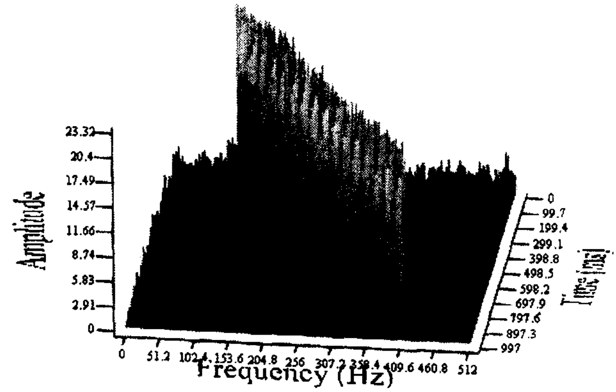

Fig. 6. Wigner distribution LFM signal in the noise

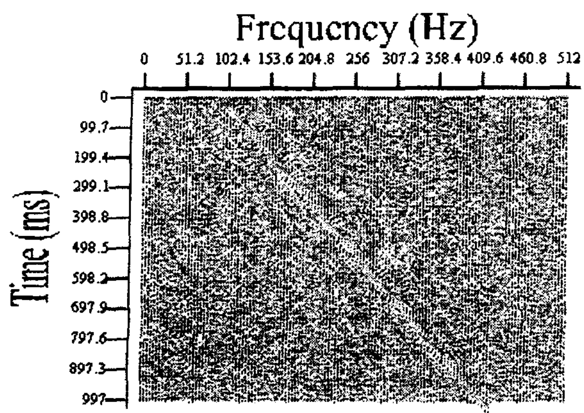

Fig. 7. Wigner distribution LFM signal in the noise

Conclusion

1. The proposed algorithm for signal analysis system for detecting a weak signal in the noise of high-level substantially expands its capabilities.
2. The usage of Wigner transformation allows, first of all to combine the time estimate of signal power and noise, and secondly, to make measurements not at individual points of the band of frequencies, and in the pre-selected octave bands (third-octave, strips of equal legibility).
3. Applications of simultaneous accumulation can significantly reduce the threshold effect for the LFM-signal.
4. Additional gains in the processing of signals in the noise will get the use of a matched filter for the speech signal.
5. The method allows to find out a signal in noise at signal-to-noise ratio more than the minus 23 дБ, thereby raises accuracy of an estimation of security of speech, in relation to other methods.
6. The proposed algorithm can be implemented in software and hardware-based system ТРАП-М. Calculations showed that compared with other methods, the processing time is not increased and is acceptable.

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