Application of Wideband Signals for Acoustic Localization

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***Abstract*—This paper describes briefly the basic ideas underlying different methods of determining the coordinates of objects by passive acoustic methods (passive acoustic location) for wideband acoustic signals, performs comparative analysis, and introduces a modified algorithm for localization of moving objects using the maximum of ambiguity function in application to passive acoustic localization of different noisy objects under various conditions**

Keywords—wideband signals; acoustic localization; difference of arrival; wideband ambiguity function

I. Introduction

The wideband technologies are very popular in the last years for the tasks of position location in radars, radio navigation and acoustic position location. This phenomenon has different reasons.

The first reason is that the use of wideband signals in the frequency domain gives us the possibility of obtaining very short signals in the time domain after the inverse Fourier transform. If the signal is really ultra-wideband, the result of signal processing can have a form close to delta-function. This gives us the opportunity of increasing the positioning accuracy and spatial resolution of the navigation, radar or acoustic positioning system.

The second reason is that very often acoustic signals which are produced by physical events or manmade equipment around us have a wide-band form by their nature. That is why if we want to process the maximum quantity of information, we have to use a wide-band or super wide-band signal processing, obtaining better results.

The third reason is in the better interference immunity of the wideband signals, especially in the case of using correlation methods for signal processing, which can be obtained as result of statistical approach.

We will consider briefly the problem of coordinate location, or the so-called passive acoustic location, of remote objects by receiving acoustic signals emitted by them. The received acoustic signal is converted into electrical one and is processed using a particular algorithm. Acoustic methods for determining the visibility of aircraft were widely used before the appearance of radar techniques in the late 30s of the previous century. Modern methods of signal processing combined with the use of high quality acoustic sensors can dramatically increase the effectiveness of passive acoustic location.

Methods for determining the coordinates of objects by passive acoustic methods (passive acoustic location) can be divided into the following categories:

– methods, based on direct solving of the inverse acoustic problem [1, 2, 3];

– methods for determining the angular coordinates of the object using acoustic radiation pattern of the acoustic antenna or the acoustic antenna array [4, 5, 6];

– methods based on the use of a sound intensity vector [7, 8];

– methods based on determining the time difference of arrival of the signal to the set of acoustic antennas [4, 9, 10,11].

In previous works we have developed ultra-wideband radars and signal processing for GPR [22], ultra-wideband noise radars in meteorology [14, 21], investigated nonparametric signal processing methods for radars and acoustic radars [14, 15, 16, 23], and suggested a new nonparametric variant of the ambiguity function – the copula ambiguity function [17], which we applied to wideband radars and sodars [18].

In this paper we will consider the basic ideas underlying different methods, mentioned above, perform comparative analysis, and introduce a modified algorithm for localization of moving objects using the maximum of ambiguity function in application to passive acoustic localization of different noisy objects under various conditions.

II. Difference of Arrival Methods

Methods, which are based on determining the time difference of arrival of the acoustic signals to a pair of microphones, are among the most successful and commonly used.

We can divide the space into areas for which the time difference of arrival of the signals to two microphones is constant. The distance from any point in space with coordinates (x, y, z) to the microphone 1 with coordinates $(x1, y1, z1)$ is determined by expression

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$$r1(x,y,z) = \sqrt{(x-x1)^2 + (y-y1)^2 + (z-z1)^2}\ . \qquad (1)$$

Distance to the microphone 2

$$r2(x,y,z) = \sqrt{(x-x2)^2 + (y-y2)^2 + (z-z2)^2}\ . \qquad (2)$$

Expression

$$r2(x,y,z) - r1(x,y,z) = const \qquad (3)$$

defines the locus of points equidistant from the two microphones. For the known sound speed, the difference of distances is determined by the time difference of arrival. For this purpose the distance difference must be divided by the speed of sound c.

These areas have the shape of hyperboloids of revolution (Fig. 1). Location of the sound source is defined as the intersection point of the set of hyperboloids with microphones situated in their focuses.

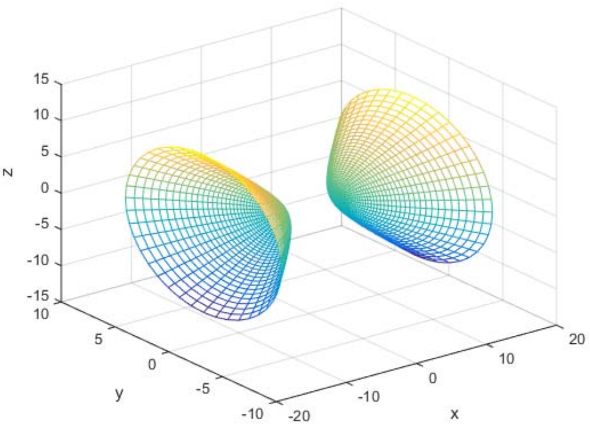

Fig. 1. Hyperboloid of revolution, which is the locus for which the time difference of arrival of the signal to two microphones is constant

This method of determining the coordinates in the theory of navigation is named by the term "Multilateration" and essentially concerns multiposition systems.

Thus, in this case the priority is to determine the time difference of arrival of signals to different microphones. This time is calculated by the correlation method. In this case, the main importance has the shape of the correlation function that describes the ambiguity function of signal position. Usually, the wider the frequency band occupied by the signal, the narrower is the correlation function. This statement is a direct consequence of the Wiener–Khinchin theorem [12 – 15], [17 – 21].

Acoustic information from the aircraft is usually sufficient broadband signal that lets us use this method to locate the position of the aircraft [9, 10, 11].

Figures 4.4 and 4.5 shows the projection of the ambiguity function to the vertical plane passing through the axis of the runway. Sound of a light plane was measured with the help of 4 microphones.

For signals at the outputs of microphones the ambiguity functions were calculated and projected to cross section of hyperboloids and the vertical plane passing through the axis of the runway (Fig. 2). At the point of intersection of the hyperboloids there is the signal from the light aircraft.

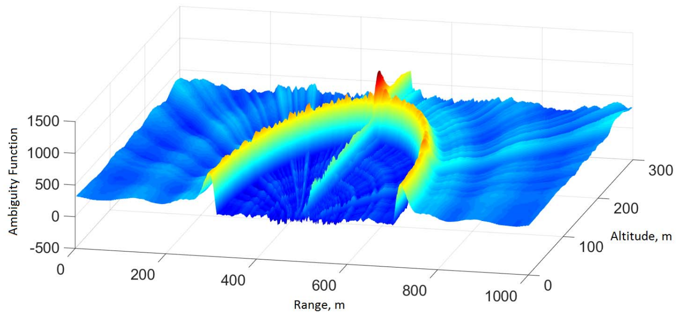

Fig. 2. The projection of the ambiguity function estimate to the family of hyperboles for different times of arrival (in 3D). Distance in m

III. Localization of Moving Objects Using the Maximum of Ambiguity Function

For localization of sound source we suggest using of a combination of signal processing methods based on determining the time difference of arrival of signals to microphones, which are elements of the array [9, 10, 11]. For unambiguous determination of the coordinates of the object in three-dimensional space the antenna array should minimally consist of four microphones located in the vertexes of the pyramid.

The time difference of arrival time Δt_{da} can be easily recalculated to the difference of distances Δr to point of location of the sound source

$$\Delta r = c \cdot \Delta t_{da}\ ,$$

where c -the speed of propagation of sound waves.

Locus for which the difference of distances to two microphones from the source of the sound is constant, forms the hyperboloid in the three-dimensional space. Its cross-section with the plane forms a hyperbola. A set of hyperboles, which corresponds to a different time differences, is shown in Fig. 3.

For obtaining the coordinates of the sound source we need to build a hyperboloid, which corresponds to this difference in time of arrival for each pair of microphones, and find the point of intersection of hyperboloids.

The problem of estimating the time difference can be solved by using correlation method for calculating the cross-correlation function between two random processes – the signals from the two microphones. It can be shown that the cross-correlation function is the maximum likelihood estimate of the time shift parameter between random processes $X(t)$ and $Y(t)$

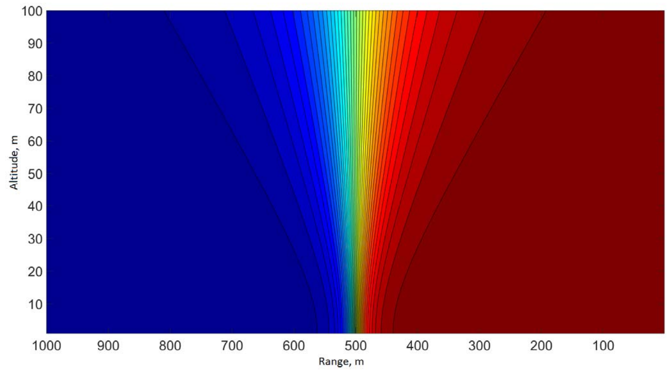

Fig. 3. Family of hyperboles for the vertical plane which passes through the axis of the runway for the various differences between the time of arrival of the signal to two microphones

$$\mathrm{cov}(\tau) = E\left\{(X(t)-m_x)(Y^*(t-\tau)-m_y)\right\}, \qquad (4)$$

where $Y^*(t)$ is the random process which is the complex conjugate with a random process $Y(t)$ m_x and m_y are mathematical expectations of these processes, τ is estimated time shift for the estimation of time difference of arrivals Δt_{da}, $E\{*\}$ is the symbol of mathematical expectations.

In the case of the great distance between the microphones it is necessary to take into account the Doppler effect, which occurs because the velocities of the sound source relative to the two microphones are different [11].

Accordingly, there is a difference of the Doppler velocities between the first and second microphones for each pair of microphones. Thus, in order to estimate the difference between the times it is necessary to find the position of maximum along the time axis of the wideband ambiguity function

$$\chi(\tau,\alpha)=\sqrt{|\alpha|}E\left\{(X(t))-m_x)(Y^*(\alpha(t-\tau))-m_y)\right\}, \qquad (5)$$

where $\alpha=\frac{c-v}{c+v}$ is a time-scale factor, which depends on the difference in velocities and the speed of sound c.

For stationary and ergodic signals expression can be rewritten in the form

$$\chi(\tau,\alpha)=\lim_{T\to\infty}\frac{\sqrt{|\alpha|}}{T}\int_0^T (x(t)-m_x)(y^*(\alpha(t-\tau))-m_y)dt. \qquad (6)$$

If the reception of the signal is carried out by a set of microphones (antenna array), from which we select a random (or not random) a pair of microphones, the ambiguity function can be represented as the expression

$$\chi(\boldsymbol{\tau},\boldsymbol{\alpha})=\frac{1}{m\cdot n\cdot(t_2-t_1)}\sum_{k=1}^{m}\sum_{j=1}^{n}\sqrt{|\alpha_{kj}|}\times \int_{t_1}^{t_2}((x_k(t))-m_{x_k})(y_j^*(\alpha_{kj}(t-\tau_{kj}))-m_{y_j})dt \qquad (7)$$

where m is the number of the first microphones in each pair, n is the number of second microphones in each pair, t_2-t_1 is the interval of observation, α_{kj} is a scaling factor for kj pair of microphones.

The discrete version of the formula (4.7) can be represented in the form

$$\chi(\mathbf{g},\boldsymbol{\alpha})=\frac{1}{m\cdot n}\sum_{k=1}^{m}\sum_{j=1}^{n}\frac{\sqrt{|\alpha_{kj}|}}{N-g_{kj}}\times \sum_{i=g_{kj}+1}^{N}((x_k(i\Delta t))-m_{x_k})((y_j^*(\alpha_{kj}(i-g_{kj})\Delta t))-m_{y_k}),$$

where Δt is a sampling interval, N is the number of time points, g_{kj} is the number of samples in time interval $\tau=g_{kj}\Delta t$ $*$ is a symbol of the complex conjugate.

For the processing the sample consisting of sample units y_j it should be resampled to the new duration of a sampling interval $\Delta t_{kj}=\alpha_{kj}\Delta t$

$$\chi(\mathbf{g},\boldsymbol{\alpha})=\frac{1}{m\cdot n}\sum_{k=1}^{m}\sum_{j=1}^{n}\frac{\sqrt{|\alpha_{kj}|}}{N-g_{kj}}\times \times\sum_{i=g_{kj}+1}^{N}((x_k(i\Delta t))-m_{x_k})((y_j^*((i-g_{kj})\Delta t_{kj}))-m_{y_k}). \qquad (8)$$

This expression can be rewritten in a more compact form

$$\chi(\mathbf{g},\boldsymbol{\alpha})=\frac{1}{m\cdot n}\sum_{k=1}^{m}\sum_{j=1}^{n}\frac{\sqrt{|\alpha_{kj}|}}{N-g_{kj}}\sum_{i=g_{kj}+1}^{N}x_{ik}y^*_{\alpha_{kj}(i-g_{kj})j}, \qquad (9)$$

where $x_{ik}=(x_k(i\Delta t))-m_{x_k})$, $y^*_{\alpha_{kj}(i-g_{kj})j}=y_j^*((i-g_{kj})\Delta t_{kj}))-m_{y_k}$.

The value of the coefficient α the corresponding maximum of the ambiguity function (4.7) for definite value τ is defined as

$$\alpha^*(\tau) = \underset{\alpha}{\operatorname{argmax}}(\chi(\tau,\alpha)) . \qquad (10)$$

Substituting the expression (4.10) (4.7) we get a spatial (time) ambiguity function that does not depend on α and thus on the speed of movement of the target

$$\chi_S(\tau) = \chi(\tau, \alpha^*(\tau)) . \qquad (11)$$

The value of a signal for a new sampling interval $\Delta t_{kj} = \alpha_{kj}\Delta t$ can be obtained using the resampling algorithm [12, 19, 20]. It includes restoring (approximation) of the output signal, for example, using the Kotelnikov series, followed by resampling with a new frequency and a new sampling interval.

This procedure is performed using a spectral method by the artificial extension of the discrete spectrum of the signal by adding zeros in discrete spectrum.

IV. Spectral Method

For the calculation of the correlation function of an acoustic signal and associated with the correlation function the ambiguity function we will apply the spectral method.

Correlation function is associated with the power spectrum – spectral power density. They are connected via the Fourier transform of (according to of the Wiener–Khinchin theorem).

$$\operatorname{cov}(\mathbf{x},\mathbf{y}) = \frac{1}{N+M-1}\sum_{n=0}^{N+M-1} e^{j\frac{2\pi}{N+M}nk} W(n) ,$$

where $W(n) = X(n)Y(n)$ is the power spectrum.

Spectral densities $X(n)$ and $Y(n)$ are calculated with the help of the fast Fourier transform (FFT) algorithm.

V. Phase Correlation

Correlation function is calculated using the special procedure of normalizing, which is called the phase transform. The idea of conversion is based on the well-known theorem of the spectral analysis on the shift of the signal. Time offset only affects the phase spectrum signal and leads to the multiplication of the spectrum by a factor $e^{-i\omega t_0}$, where t_0 is the time delay.

Thus, the amplitude spectrum does not carry information about the position of the signal in time, so it can be ignored when estimating the correlation functions and assume that the amplitude of all harmonic components of the spectral density are the same and equal to 1.

This assumption is confirmed by the known fact regarding correlation function of the ultra-wideband random process. The white noise. It has the appearance of the Dirac delta function and provides the best temporal resolution among all the signals.

For continuous signals the delta function is a mathematical abstraction, but for the discrete signals it can be realized in practice. For the implementation of the algorithm during calculation of spectral power density, it is sufficient to conduct the normalization of the power spectrum, dividing the spectral power density by the product of the amplitudes

$$W_\varphi(n) = \frac{W(n)}{|X(n)||Y(n)|} = \frac{X(n)Y(n)}{|X(n)||Y(n)|} .$$

The estimate of the phase correlation of two signals will be obtained calculating the inverse discrete Fourier transform of the $W_\varphi(n)$.

VI. Experimental Verification of the Method

A. *The research results of the acoustic localization in a closed room*

The measurements of the ambiguity function of the stationary sound source, emitting a random signal with a uniform spectral density, were made in a closed room. The resultant form of the spectrum is determined by the frequency response of the emitter (loudspeaker). The emitter is shown in Fig. 4.

Fig. 4. The sound emitter

The number of sample units in the processed sample is equal to 90000. Information processing after converting the signal into digital form with a frequency sampling rate of 50KGc was done using a program, written using the Matlab language. Fig. 5 shows the output sample, obtained from the first microphone.

In Fig. 6 sampled signals from two (second and third) microphones are accompanied by zeros for unambiguous further calculation of correlation functions. The signal of the second microphone with the time coordinates is recalculated into the distance using the speed of sound propagation. The spectral densities of the signals of these microphones are shown in Fig. 7. Spectral densities were calculated using the algorithm of fast Fourier transform (FFT).

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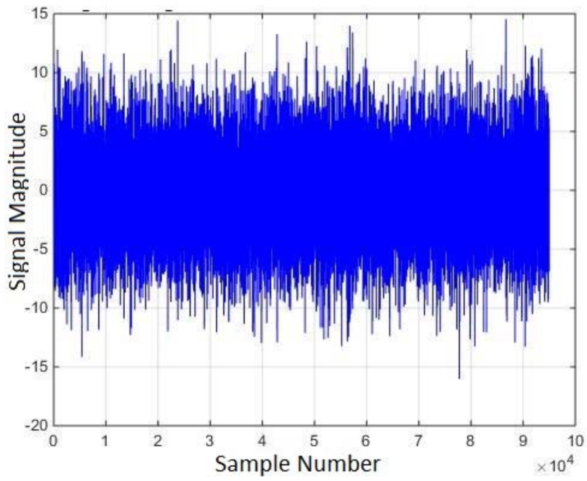

Fig. 5. Output signal sample

Two preliminary operations were done before further signal processing: the pass-band filtration in the frequency band 10 Hz – 10KHz and normalizing of amplitude spectrum for obtaining a narrow correlation function (calculation of phase correlation). The resulting spectra for converted signals are shown in Fig. 8.

After filtration of the signal the ambiguity function of an acoustic signal for a set of scaled coefficients and the time differences between the sound signals coming to the second and third microphones was calculated. The time difference was recalculated into the spatial measure (meters). The results are presented in Fig. 9.

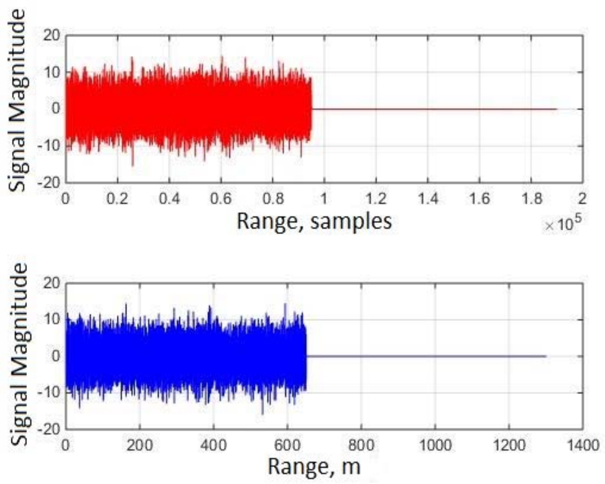

Fig. 6. Signals from the second and third microphones. Time in the sample counts of the analog-to-digital converter for the second microphone (top) and the equivalent distance (m) for the third microphone. The magnitude of the sound signal in Paskals (Pa)

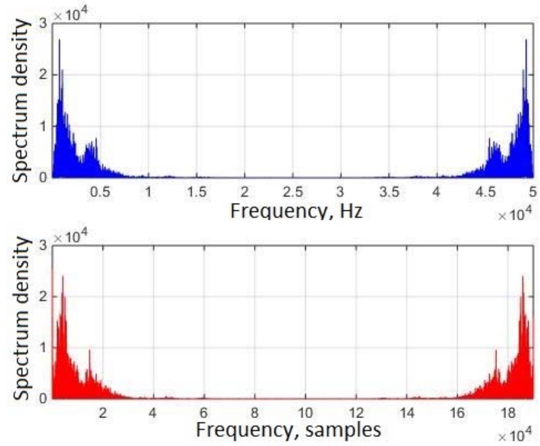

Fig. 7. The Spectral density of signals from the second and third microphones

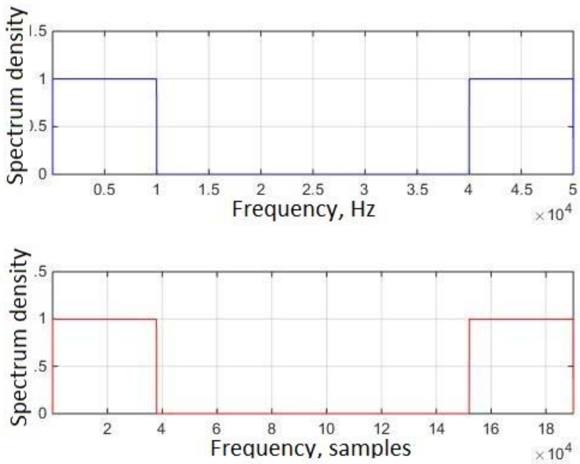

Fig. 8. The Spectral density of a signal for the second and third microphones after filtering and amplitude normalizing

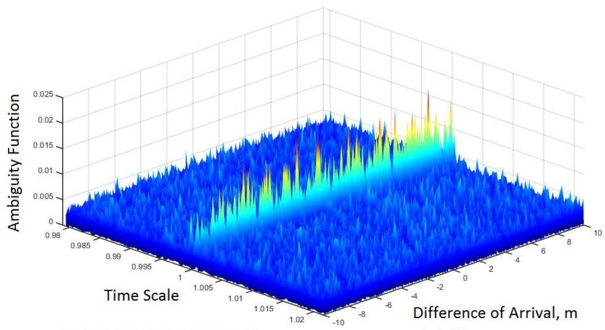

Fig. 9. The ambiguity function for the signals of the second and third microphones after filtering and amplitude normalizing

To obtain the degree of uncertainty of the difference of arrival of signals to two microphones (Fig. 10), which is independent from the velocity of the target, we chose the maximum in the set of all values of an ambiguity function for this time difference by the expression (4.10).

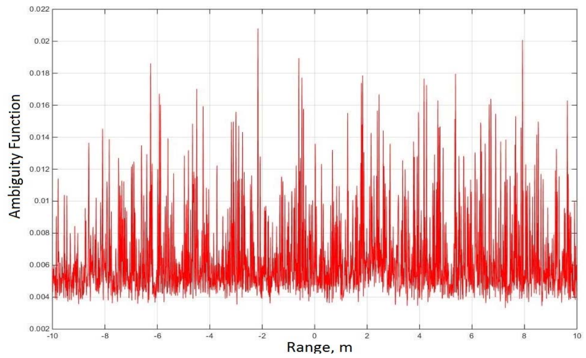

Fig. 10. The ambiguity function for the signals from the second and third microphones after selecting the maximum values for all of the calculated scale factors α

Concentration of results of measurements for α in area of zero velocities ($\alpha = 1$) as is shown in Fig. 9 confirms the correctness of the proposed algorithms, and a set of spatial peaks in Fig. 9 is explained by the reverberation of an acoustic signal.

B. The research results of the acoustic localization in a reverberation chamber

Measurement of sound in reverberation chamber is the most complicated, when it goes about solving the problem of spatial acoustic localization. Multiple rereflections from the walls of the chamber substantially complicate the localization of sound source.

In Fig. 11 the audio signal from the standard noise source model 4204 is shown. The standard source is the electromechanical device where noise is produced by the rotating rotor. The spectrum of the signal, which is shown in Fig. 11, reaches 15 kHz. The signal before the final processing was limited by the frequency band up to 3 kHz. Then the spectrum was normalized by the amplitude.

The processed signal is used for the construction of the broadband ambiguity function. We have used the sample consisting of 3000 sample units, which corresponds to a length of approximately 20 m. The processing, which was done, selects the signal of the standard noise source as the maximum of the ambiguity function. This maximum can be clearly seen at a three-dimensional diagram of ambiguity function at the Fig. 12 and the graph of the dependence of the maximum of the ambiguity function of the time difference of arrival (transferred in meters) in Fig. 13.

Presented results show that even for such a difficult case, which corresponds to the location of the object in the reverberation chamber, we were able to get a notable source localization.

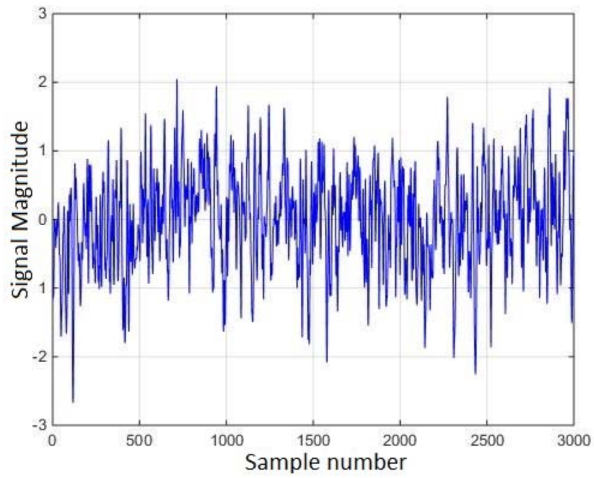

Fig. 11. The standard noise source signal

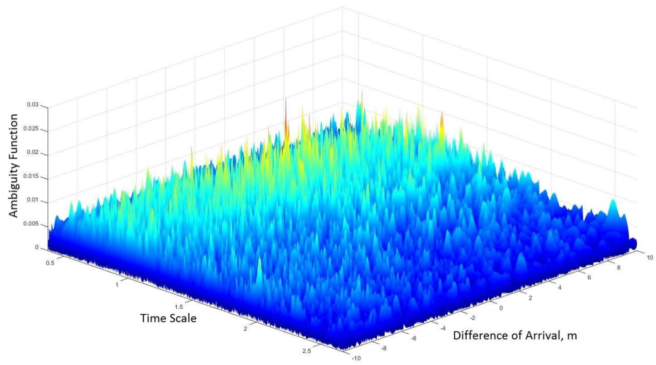

Fig. 12. The ambiguity function for the signals of the first and second microphones after filtering and amplitude

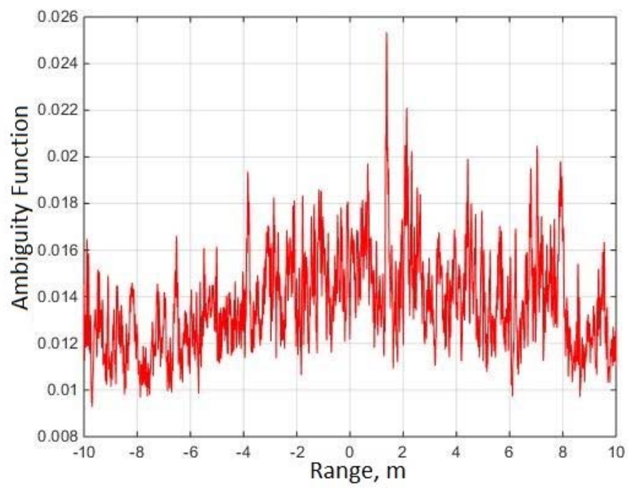

Fig. 13. The ambiguity function for the signals from the first and second microphones after selecting the maximum values for all of the calculated scale factors α

C. The acoustic localization of small multicopter in a reverberation chamber

Audio records were examined for a small unmanned aircraft in cruising mode in the reverberation chamber. The resulting signal is shown in Fig. 14. The peaks that correspond to the position of aircraft, as well as re-reflection from the walls of the acoustic chamber are visible as a result of signal processing in Fig. 15 and 16.

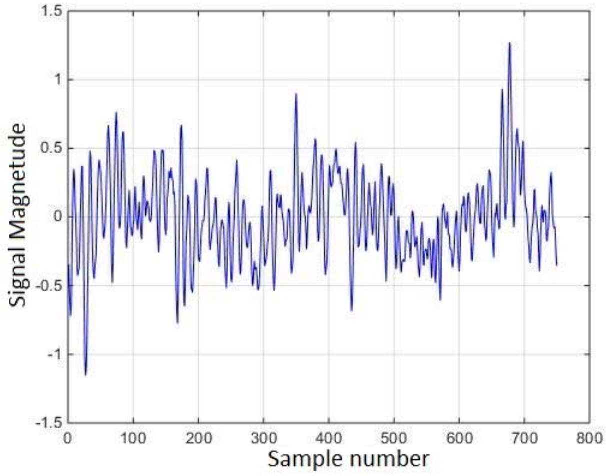

Fig. 14. The acoustic signal from the small multicopter

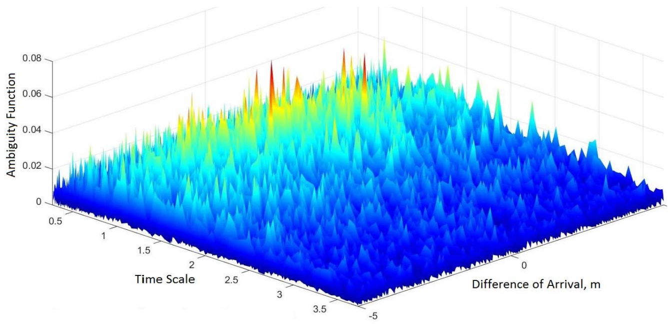

Fig. 15. The ambiguity function for the signals of the first and second microphones after filtering and amplitude normalizing

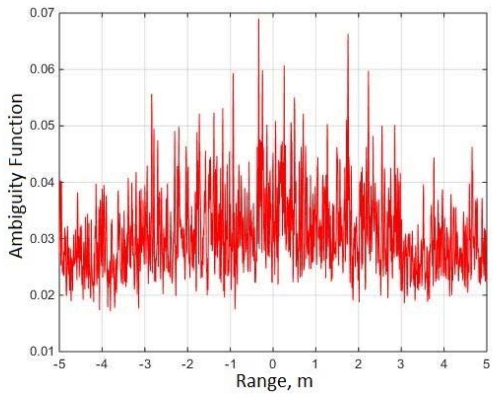

Fig. 16. The ambiguity function for the signals from the first and second microphones after selecting the maximum values for all of the calculated scale factors α

D. The acoustic localization of light sport plane

We also have investigated the acoustic signal from the light sport plane. The position of microphones relative to the runway is shown in Fig. 17.

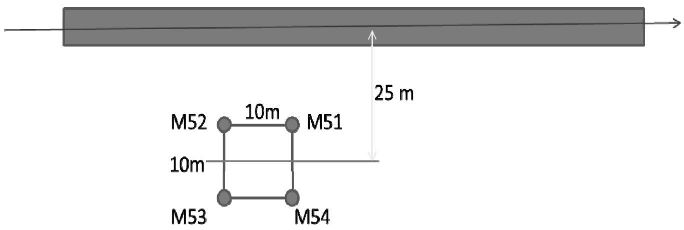

Fig. 17. -Runway and antenna array of four microphones (M51, M52, M53, M54), which are located at a distance of 10 m from one another and at a distance of 25 m from the runway

The antenna array consisting of 4 microphones was located at a distance of 25 m from the runway. Recorded signals of 2 microphones are shown in Fig. 18. The maximum signal corresponds to the passing of aircraft near the antenna array.

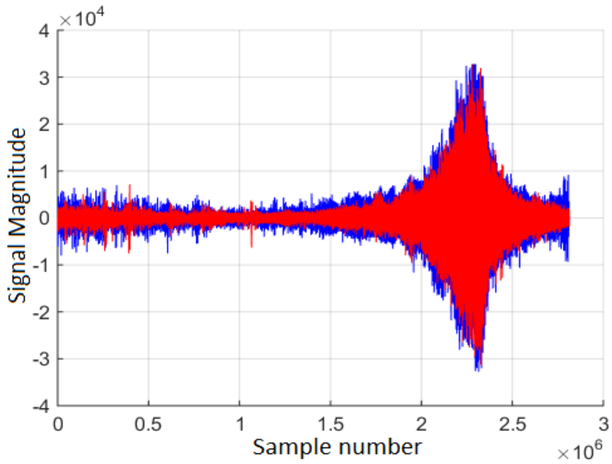

Fig. 18. Signals from microphone M-51 and M-54 in the process of the aircraft landing

The visual analysis of microphone signal, which is presented in Fig. 19, shows the spectral lines associated with the engine work.

Previous signal processing with preliminary filtering, phase converting (amplitude normalization) and calculation of the broadband ambiguity function, provides an opportunity to obtain the form of a spatial ambiguity function, which is presented in Fig. 20 and Fig. 21.

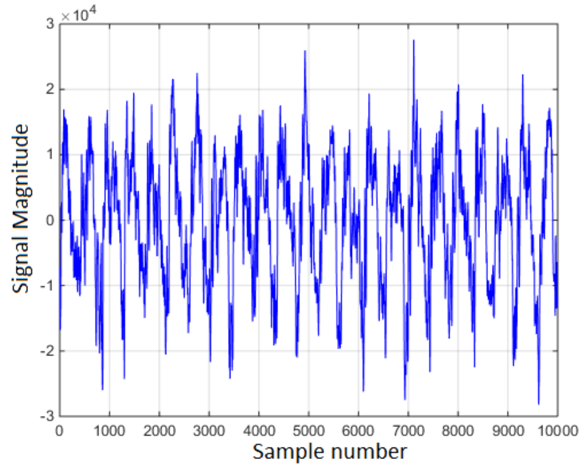

Fig. 19. Signal from the microphone M51 in time processing window with duration of 10000 samplle units

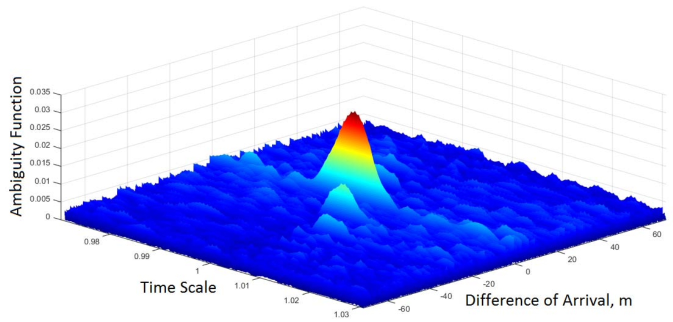

Fig. 20. The mutual ambiguity function for signals from microphones M51 and the M54 in time processing window with the duration of 10000 time sample units

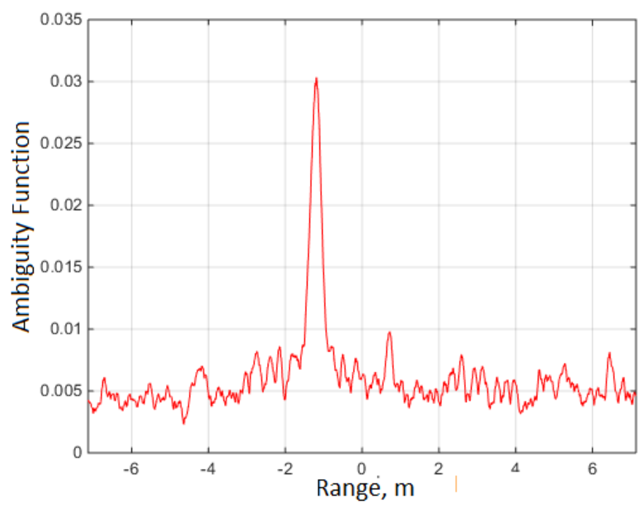

Fig. 21. The spatial ambiguity function for signals of microphones M51 and M54 after selecting the maximum values for all of the calculated scale factors α

VII. Conclusions

In this paper, the basic ideas underlying different methods of determining the coordinates of objects by passive acoustic methods have been discussed.

The modified method of localization of moving objects based on receiving and processing their own wideband noisy acoustic signal has been developed. This method traditionally uses measurements of the time difference arrival by the set of receivers but signal processing is performed using the maximum of ambiguity function for making a decision on the object coordinates.

The developed approach and the modified algorithm has been applied to passive acoustic localization of different noisy objects under various conditions.

Analysis of experiment results has confirmed that the proposed method and algorithms can successfully localize a source of the wideband acoustic noise, in particular, indoor, even in the reverberation chamber, and outdoor in case of a light airplane as a source of noise during approaching and landing.

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