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ACOUSTIC ANTENNA ARRAY FOR LOCATION OF THE SONIC SIGNAL SOURCES

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The article offers an analysis of methods for location of sonic signal source, and a pattern for their use in building speech information security systems. The methods for speech information security have been formulated. A model of acoustic antenna array was developed for Fraunhofer zone and Fresnel zone. Optimization of weight factors for a 25-element antenna array was made according to minimum of side-lobes and minimum width of direction pattern.

Keywords: acoustic antenna array, directional diagram, weight factors, side lobe, location of the sonic signal sources, information Protection from leakage on the acoustic channels (защита информации от утечки по акустическим каналам).

Introduction

Protection of speech information in a dedicated room includes research to determine possible acoustic and vibration voice data leakage channels. Voice data leakage channels may appear because of low acoustic and vibration insulation between adjacent rooms. A technique for determination of areas in buildings with low acoustic and vibration isolation technique was developed. This technique is based on the creation of acoustic signals in the protected room, with sound pressure levels generated by loud speech in the relevant frequency range and measuring the level of acoustic signals outside the premises in the areas of possible eavesdropping. Difficulties of using this technique are the need to control the extended elements of the walling constructions discretely (in a large number of points) and the preparation of an acoustic portrait of the investigated element constructs [1]. In this case, there is a probability that the points with maximum level of acoustic signals might be missed due to their discrete choice. The use of laser scanning vibrometer or other such device allows getting a picture of the vibrating object. However, localization of voice data leakage channels which may appear because of holes and cracks in enclosing structures is not possible. Most often, the voice data leakage channels may appear near water pipes, power supply network, gas pipes and other structural elements with low acoustic insulation. To improve the performance, we offer searching for leakage channels (sites with a low sound insulation), using an acoustic antenna array. This enables us to determine the sources of acoustic signals with maximum amplitude. The same principle can be used to improve sound insulation in residential premises.

Acoustic antenna array model within the Fraunhofer region

To solve the above-stated problem we should limit the source data variation range and formulate the requirements on check conditions of spoken information leakage security. Receivable acoustic signals frequency range varies from 150 Hz to 5600 Hz with the possibility of a sub-range division. Acoustic signal sources may be located in both near-field and far-field regions (Fraunhofer region & Fresnel region). The beam width is required to be not greater than 15°, and the side lobe level to be -20 dB. Planar discrete equispaced array was chosen as a model.

Linear array is the example of an elementary aerial array; its Fraunhofer region model is shown at fig. 1.

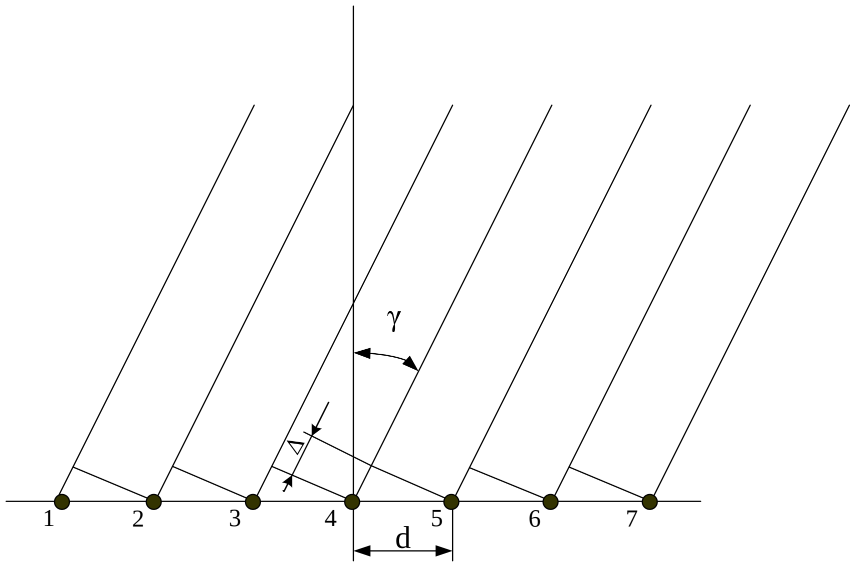

Fig. 1. Linear equispaced array

Let a plane acoustic wave grazes onto a linear equispaced array at the γ angle. The distance between aerial array microphones equals d. Then the wave arrival difference between two adjacent microphones would be equal to

$$\Delta = d \cdot \sin\gamma\,, \tag{1}$$

where γ is an angle between an acoustic wave direction of arrival and an aerial array incidence normal.

Acoustic wave phase difference between the adjacent microphones at the f_s – frequency is defined as follows

$$\Delta\varphi = \frac{d \cdot \sin\gamma \cdot f_s \cdot 2 \cdot \pi}{c}\,, \tag{2}$$

where c is a sound velocity (in the atmosphere) and π is a constant equal to 3,14.

Simply summing up all the microphone output signals, grazing at different γ – angles, we can obtain linear equispaced array directional diagram

$$R\ \gamma\ = \sum_{i=1}^{n} A_i \cdot \cos\left(2 \cdot \pi \cdot f_s \cdot t + \frac{i \cdot d \cdot f_s \cdot 2 \cdot \pi \cdot \sin\gamma}{c}\right), \tag{3}$$

where n is a microphone quantity i is a microphone sequence number A is an i^{th} – microphone signal amplitude and t is time.

Shifting a phase relative to the central microphone permits us to obtain a handy view of the directional diagram for the odd number of microphones in the aerial array

$$R\ \gamma\ = \sum_{i=1}^{n} A_i \cdot \cos\left(2 \cdot \pi \cdot f_s \cdot t + \frac{(i-(n+1)/2) \cdot d \cdot f_s \cdot 2 \cdot \pi \cdot \sin\gamma}{c}\right). \tag{4}$$

In order to exclude sidelobes (with the amplitude equal to the main lobe's) & add a possibility of diagram rocking in the directional diagram, it is necessary to fulfill the condition $d<\lambda$, wherein $\lambda = f_s/c$. Directional diagram rocking can be performed by inserting the rocking angle parameter α into the eq. (4). Thus, we have

$$R\ \gamma\ = \sum_{i=1}^{n} A_i \cdot \cos\left(2 \cdot \pi \cdot f_s \cdot t + \frac{(i-(n+1)/2) \cdot d \cdot f_s \cdot 2 \cdot \pi \cdot\ \sin\gamma - \sin\alpha}{c}\right). \tag{5}$$

where α is a current angle value of array directional diagram rocking.

A brief way of rewriting the normalized directional diagram of an equispaced linear aerial array is presented in the paper [2].

$$\mathrm{R}_1(\gamma)=\frac{\sin\left[\frac{n\cdot\pi\cdot d}{\lambda}(\sin\gamma-\sin\alpha)\right]}{n\cdot\sin\left[\frac{\pi\cdot d}{\lambda}(\sin\gamma-\sin\alpha)\right]}. \tag{6}$$

where λ is a wavelength.

Performing an aerial array optimization according to the parameters of minimal beamwidth & minimal level of sidelobes is possible whether an optimization problem is mathematically formulated, i.e. target function and its requirements are mathematically notated. Varying A_i – coefficients and $\boldsymbol{\gamma_i}$ – angles in the eq. (5) we can evaluate them, whereby

$$\min\to\int_{-\gamma_1}^{\gamma_1} R(\gamma)\, d\gamma \ \ \&\ \ \max\to R(0). \tag{7}$$

where γ_1 and γ_1 are the angles, whereby $R(\gamma)$ equals 0.7 of the maximum value.

Moreover, target function may be written as a maximum of an expression

$$\max\to\frac{\int_{-\gamma_1}^{\gamma_1} R(\gamma)\cdot d\gamma}{\int_{-\frac{\pi}{2}}^{-\gamma_1}\left|R(\gamma)\right|\cdot d\gamma+\int_{\gamma_1}^{\frac{\pi}{2}}\left|R(\gamma)\right|\cdot d\gamma}. \tag{8}$$

Normalized directional diagram of an equispaced linear discrete aerial array, containing seven microphones, without considering the directional diagram of individual microphones, is presented on fig. 2.

Allowing for the fact that certain microphones directional diagrams are identical and close to a cardioids, the directional diagram of a linear equispaced discrete aerial array may be determined by the expression:

$$R(\gamma)=R_m(\gamma)\cdot\sum_{i=1}^{n} A_i\cdot\cos\left(2\cdot\pi\cdot f_s\cdot t+\frac{(i-(n+1)/2)\cdot d\cdot f_s\cdot 2\cdot\pi\cdot(\sin\gamma-\sin\alpha)}{c}\right), \tag{9}$$

where $R_m(\gamma)$ is a individual microphone directional diagram.

The above-mentioned antenna array directivity property may be enhanced by means of finding the optimum A_i – coefficients in the expression (4). In the case with the symmetrical equispaced linear aerial array, containing seven microphones, coefficient values are: A_1=A_7=3, A_2=A_6=2, A_3=A_5=1,5, A_4=1. Also, a pairwise multiplication of signals from symmetrically located elements might be applied with a successive summation.

Microphones space tapering in the linear aerial array allows of forming the directional diagram with high resolution properties and a minimum of sidelobes.

Space-tapered aerial array synthesis might be performed using various methods for the adjusted target function [3, 4].

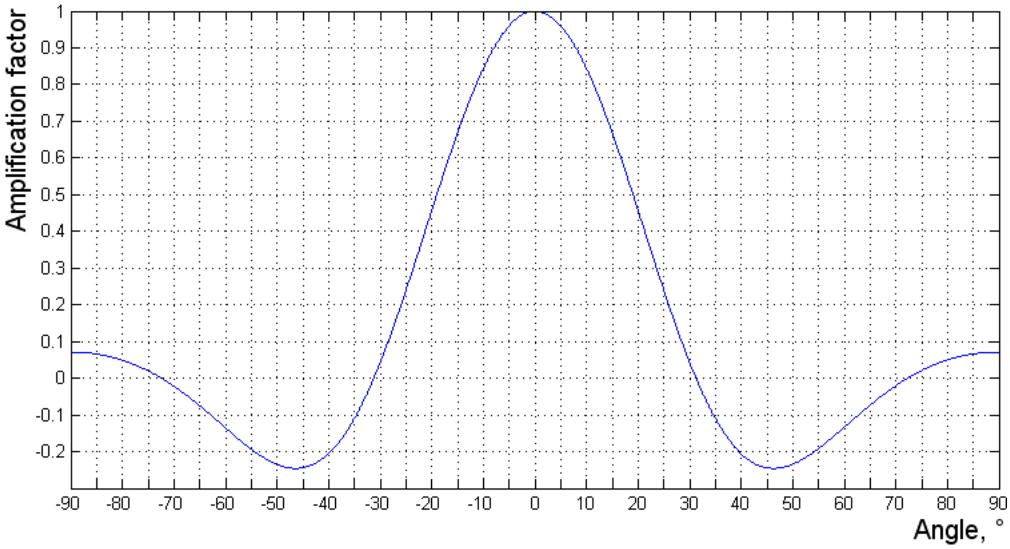

Fig. 2. Directional diagram of a linear equispaced array, containing seven microphones, spaced 100 mm at frequency of 1000 Hz within the Fraunhofer region

Acoustic antenna array model within the Fresnel region

The directional diagram for the Fresnel region was calculated on the assumption that a linear aerial array was located at a distance of 3 meters from the acoustic source movement line. Moreover, a linear array and a sound source movement line are situated in the same plane. Aerial array model within the Fresnel region is shown at fig. 3.

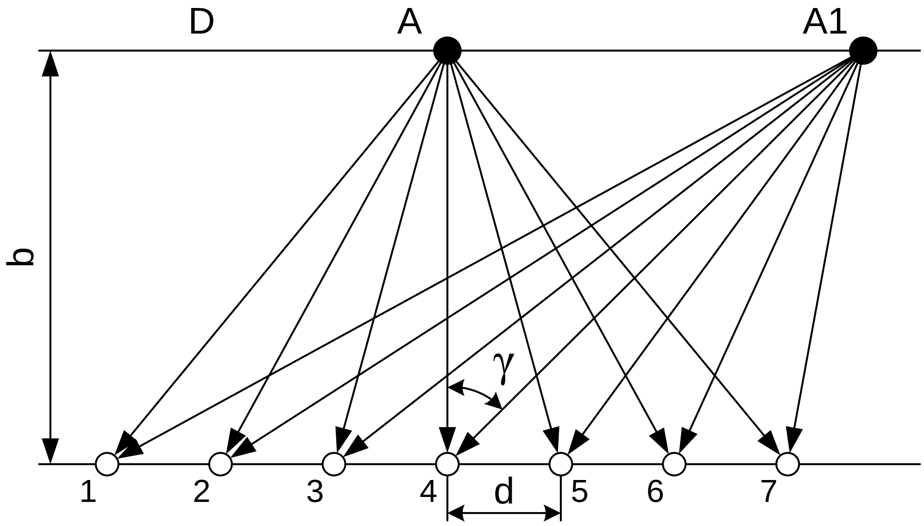

Fig. 3. Fresnel region aerial array model

Acoustic source *A* may be located at any point on the *D*-line, which is at a distance of *b* from the aerial array.

Acoustic wave phase difference between central microphone and the rest of them at f_s – frequency is determined as follows

$$\Delta\varphi = \left(\sqrt{b^2 + (\left|i-(n+1)/2\right| + b \cdot tg\gamma)^2} - b \cdot \cos\gamma \right) \frac{f_s \cdot 2 \cdot \pi}{c}, \quad (10)$$

where *b* is a distance between the aerial array and the acoustic source movement line γ is an angle between the acoustic source movement line perpendicular and the direction from the central microphone to the acoustic source *n* is an odd number of microphones in the aerial array ***c*** is a sound velocity (in the atmosphere) and f_s is frequency.

Aerial array directional diagram within the Fresnel region may be determined as follows

$$R(\gamma) = \sum_{i=1}^{n} A_i \cdot \cos\left(2\cdot\pi\cdot f_s \cdot t + \left(\sqrt{b^2 + \left(\left|i-(n+1)/2\right| + b\cdot tg\gamma\right)^2} - b\cdot\cos\gamma\right)\frac{f_s\cdot 2\cdot\pi}{c} \right). \qquad (11)$$

Fig. 4 reveals the directional diagram of a linear equispaced aerial array, containing seven microphones, within the Fresnel region.

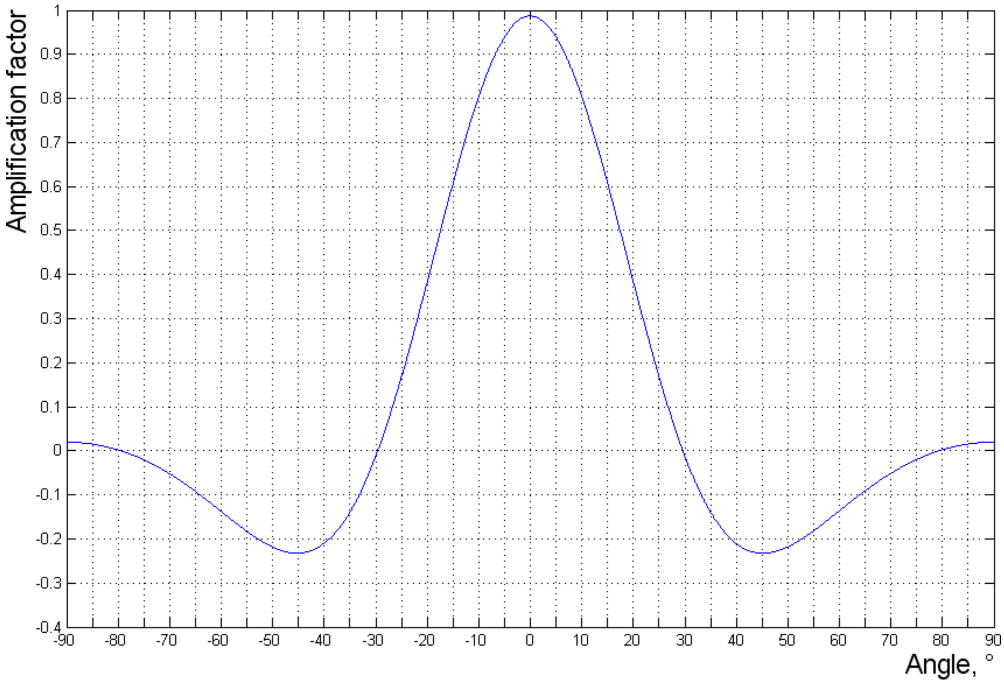

Fig. 4. Directional diagram of a linear equispaced array, containing seven microphones, spaced 100 mm at frequency of 1000 Hz within the Fresnel region

Comparing Fraunhofer and Fresnel region directional diagrams of linear equispaced arrays revealed the difference of $\approx 2°$ at 0,7 – level in beamwidth. The Fresnel region beamwidth makes up $\approx 26°$.

The model of the acoustic antenna array in a diamond

The flat antenna array in a diamond lattice was proposed to increase the base both vertically and horizontally for the acoustic location of sound sources. Location of microphones in the array is shown in fig. 5.

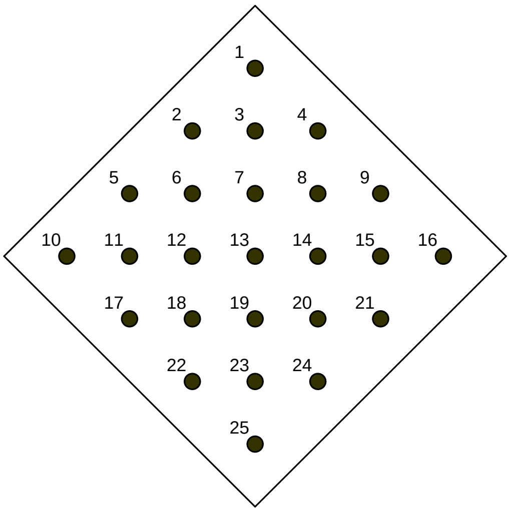

Fig. 5. Location of microphones in the array

Distance from sound source to each point of the antenna array can be calculated as

$$D_i = |S - A_i| = \sqrt{(x_s - x_a)^2 + (y_s - y_a)^2 + (z_s - z_a)^2}\,, \tag{12}$$

where S is the coordinates (x_s, y_s, z_s) of the source; $A_i = \{A_1, A_2, \ldots, A_N\}$ is the set of coordinates (x_a, y_a, z_a) of microphones in the array.

Thus, we can assume that at each point of the antenna array signal comes from the sound source, which (not taking into account the attenuation of the signal propagation) can be written as

$$u_i = G_i \times e^{D_i \times k \times j}\,, \tag{13}$$

where G_i is the gain in signal i-th microphone array; j is imaginary unit; $k = \frac{2\pi}{\lambda}$ is wave number; $\lambda = \frac{c}{f}$ is the wavelength of the sound source; f is the frequency at which the analysis of the diagram of the antenna array (the frequency of the sound source, which should capture array) c is the speed of sound in air (≈ 331,46 m/s).

Thus, if for the output signal values microphone array is added, the resulting signal can be written as

$$R(S) = \frac{1}{N}\sum_{i=1}^{N} u_i = \frac{1}{N}\sum_{i=1}^{N} G_i \cdot e^{D_i \cdot k \cdot j} = \frac{1}{N}\sum_{i=1}^{N} G_i \cdot e^{|S-A_i| \cdot \frac{2 \cdot \pi \cdot f}{c} \cdot j}\,, \tag{14}$$

where N is number of microphone array; I is number of microphones in the array.

The value $R(S)$ is complex, the module of this quantity describes the directional pattern of antenna array, i.e. gain of the acoustic propagation from its point S, and the argument describes the change in phase with respect to the phase of the emitted signal.

Depending on the algorithm for combining signals from different microphones array is the last formula can be changed.

Fig. 6 shows the directional characteristic of a plane diamond-shaped antenna array with 25 microphones.

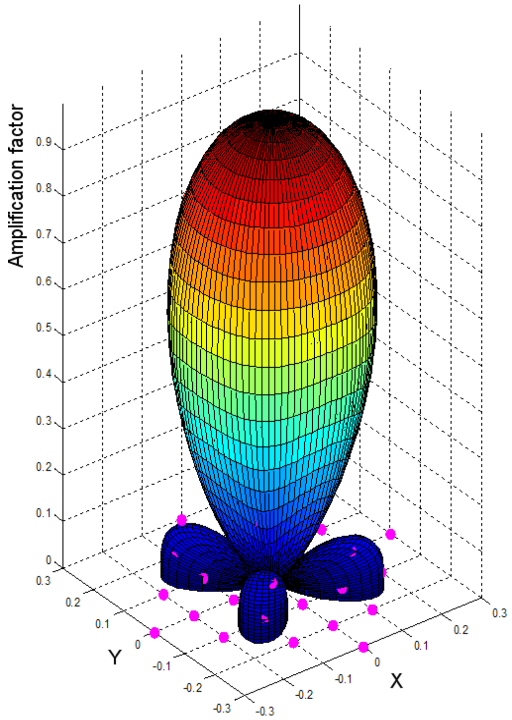

Fig. 6. Diagram of a planar rhombic antenna array

To reduce the width of the directional characteristic can be applied pairwise multiplication of the signals symmetrically placed microphones on the center of the orthorhombic lattice with subsequent summation of the result. This procedure is used in radio systems to determine more accurately the source of the microwave signal. However, this leads to the appearance of additional sidelobes. In order to identify areas in the designs of buildings with low acoustic and vibration isolation, such processing of signals from the microphones is permissible.

The phenomenon of reverberation in a room can lead to distortion of the overall pattern of sound pressure distribution on the surface scan. To eliminate this disadvantage, it is recommended to use multi-frequency scanning with non-multiple frequencies. This scan can be performed in a consistent manner. The scanning results should be treated in such way as to identify areas in which a signal most often comes out. However, you can use to exclude the effect of reverberation instead of microphones intensimeter heads.

Conclusion

The time of electron scanning will be determined by directional characteristic of the lattice and the frequency range of the sources of acoustic signals. It is proposed that the minimum time required for signal processing is 20 periods at least, for a frequency of 1000 Hz is 20 ms. When directional characteristic width is 10 degrees during the time of electron scanning of the aperture angle horizontally in 120 degrees and vertically 120 degrees, with step scan of 15 degrees is not less than 12.

During electron scanning also the resolution of directional characteristic will be changed. Maximum angle of electron scanning from – 60 degrees to plus 60 degrees. The effective base of the antenna array will change depending on the angle. At angles of – 60 and 60 degrees effective base of array is reduced by half compared with the angle of 0 degrees. Differences in the directional characteristics of the Fraunhofer zone and Fresnel zone are no more than 2 degrees; the model of antenna array for the Fraunhofer zone can be used to localize sound sources in the Fresnel zone.

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