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Restoring the distribution of pulse signal in space using methods of near-field acoustic holography

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Abstract

The development of acoustic measurement technology gives opportunity to measure noise levels at a convenient distance from the object, with respect to its the geometry and signal-to-noise ratio, and then to calculate signal levels at the desired distance. The methods of acoustic holography is widely used here. To do so, the amplitude and phase of the signal must be measured in a sufficient number of points. Of great interest is the measurement of the characteristics of broadband signal sources is of great interest. When solving such a problem, two parameters are most important: the form of temporary realization and the average level of the pressure field. The calculated waveform as a function of time is best illustrated by the example of pulse signals. In the present work, describe the corresponding laboratory experiment conducted in an anechoic chamber. The technique for reconstructing the far field is given, followed by the results of calculations and comparison with experimental data.

Key words: nearfield acoustic holography, far field, pulse signal.

Восстановление распределения импульсных сигналов в пространстве методами ближнепольной акустической голографии

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Аннотация

Одно из направлений развития методов акустических измерений - это переход к измерению уровней шума на том расстоянии от объекта, где это удобно исходя из геометрии и соотношения сигнал-шум, с последующим расчетом уровня сигнала на интересующей дистанции, с применением методов акустической голографии. При этом амплитуда и фаза сигнала должна быть измерена в достаточном количестве точек. Большой интерес представляет измерение характеристик источников широкополосных сигналов. При решении подобной задачи, если речь идет об акустике, наиболее важными являются два параметра: форма временной реализации и средний уровень поля давления. Форму временной реализации нагляднее всего исследовать на примере импульсных сигналов. В настоящей работе описан соответствующий лабораторный эксперимент, проведенный в безэховой камере, методика восстановления дальнего поля, а так же результаты расчетов и сравнение с экспериментальными данными.

Ключевые слова: *ближнепольная акустическая голография, дальнее поле, импульсные сигналы.*

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Introduction

The use of near-field holographic methods opens up possibilities for measuring characteristics of sources under complex external conditions. For example, near-field methods are used if it is necessary to measure the source field in the far zone, and this is not possible, due to the limited size of the laboratory setup on which the experiment is being conducted. For example, one can place a large-aperture sonar can in an anechoic pool to measure the parameters of its signal in the near zone. Then a specific procedure is available to calculate the radiation pattern in the Fraunhofer zone [1], while the size of the basin does not allow direct measurements of the far field.

In the previous work [4], we described a number of methods for converting the results of near-field measurements to the far zone. Continuous signals were considered in that work, but the methods described above can be generalized to the case of broadband signals, which is going to be done in this work. The application of near-field methods to the analysis of sources of broadband signals is of particular interest to researchers [2, 3] in view of the great practical importance of this issue.

When reconstructing the field of a broadband source in space, two aspects are of interest: the average signal level and its waveform. Proper reconstruction of a waveform is best illustrated by the example of pulsed signals. Methods for calculating the field in the far zone are verified in the present work with the use of the results of near-field measurements for pulsed signals.

1. Experimental installation

Similar to the previous work [4], the experiment was carried out in an anechoic chamber. The installation diagram is shown in Fig. 1. The installation consisted of a loudspeaker without an enclosure and a microphone. The measurements were carried out in a plane where a system of thin metal cables was stretched, allowing the microphone to be fixed at the nodes of a rectangular grid. The microphone was alternately installed in the grid nodes, the speaker remained stationary. At each point, the same realization of the pulse signal was played, the start time of the playback and the start time of the recording were synchronized. After combining all the records, a general picture of the field in a plane was obtained. Records were made at two heights of the speaker anchorage, which corresponded to the near and far zone. In the first series of measurements, the speaker was located at a distance of $z_{near} = 18$ cm from the measurement zone above its center and emitted a sequence of pulses, each of which had a frequency of 1 kHz, a duration of 4 ms, and the time between pulses was 5 s. The estimate of the Fresnel parameter is: $F_r = 1{,}53$ ($F_r = \lambda z_{near}/D^2$, here: λ is the wavelength, D is the size of the aperture of the loudspeaker). The measurement zone was a rectangle with sides of 192 cm and 131 cm. The distance between the nodes in both coordinates was variable and averaged 12 cm. In the second case, the source was located at a distance $z_{far} = 215$ cm from the measurement zone, which corresponds to the Fraunhofer zone ($F_r = 18{,}27$). In Fig. 2 a wave front is constructed along the line of sensors $y = -84$ cm, $z = z_{near}$.

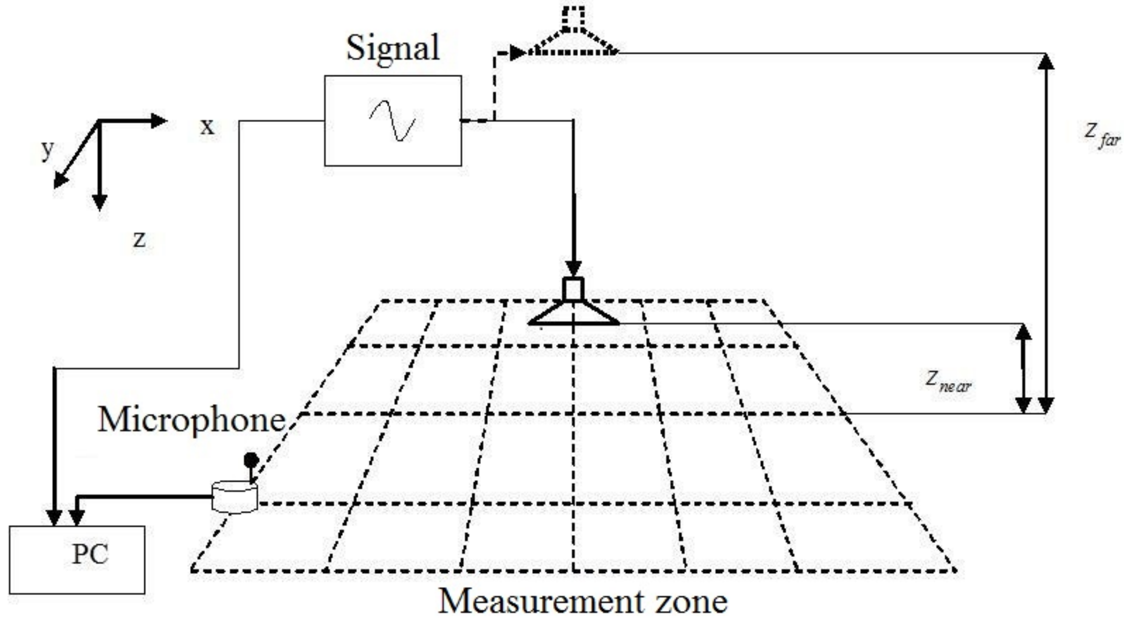

Fig. 1. Experimental installation

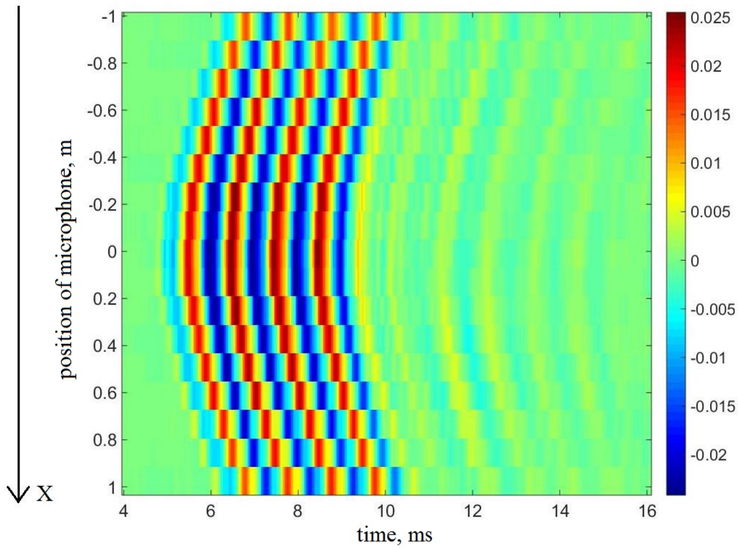

Fig. 2. Temporal scan of the signal at several points along the line at the boundary of the measuring zone

2. Analysis of the results

2.1. Signal processing of the synthesized planar antenna

To calculate the field, we use the reasoning given in [4,5]. Write the Kirchhoff – Helmholtz integral, for this use the green function for the soft boundary and the fact that the size of the measuring section of the plane in the near zone was taken large enough so that the source field would fall along the edges and no stationary phase points appeared on the areas outside it [6].

$$p(\mathbf{R}) = -\frac{1}{2\pi}\iint\limits_S p(x,y)\frac{\partial}{\partial n}\left(\frac{e^{ikR}}{R}\right)dS \tag{1}$$

where i – imaginary unit, p is the complex pressure amplitude [usually Pa, but conventional units are used in this article - V], single frequency is considered here, the transition to a wide spectrum will be made below, k is the wave vector [m^{-1}], S – zone in the plane ($z = z_{near}$) where pressure was measured, $\mathbf{R}$ set of vectors from nodes in near zone (x, y, z_{near}) to the required observation point [m], n is the normal to the measurement plane. Formula (1) is called Huygens second integral formula. It determines the value of sound pressure in a half-space based on the known value of sound pressure on the plane. If the distance to the observation point is much longer than the wavelength, then formula (1) can be converted as follows:

$$p(\mathbf{R}) \approx -\frac{ik\cos(\alpha)}{2\pi R}\iint\limits_S p(x,y)e^{ikRr}dS \tag{2}$$

where r - set of vectors from the origin to the nodes of the measuring zone (x, y, z_{near}) [m], α is the angle [rad] between the normal to the measurement plane and $\mathbf{R}$. Since we made an assumption the $R \gg r$, now $\mathbf{R}$ can be understood as the radius vector from the center of the plane to the required observation point in the far zone. This approach for calculating the far field was called FPK (Far Plane Kirchhoff). Expression (2) can be generalized to the case of wideband signals, and in particular for pulsed signals, by carrying out the standard procedure of transition from the spectral to the temporal representation [7]. Then (2) takes the form:

$$p(\mathbf{R},t)R = \frac{\Delta s\cos\alpha}{2\pi c}\cdot\frac{\partial}{\partial t}\sum_j p\left(\mathbf{r}_j,t-\frac{\mathbf{R}\mathbf{r}_j}{Rc}\right) \tag{3}$$

where c is the speed of sound [m/s], Δs is the area of the cell [м^2], $\mathbf{r}_j$ is the vector at the j-th measurement point in the near zone [m].

The values obtained by formula (3) were compared with the signals recorded at this point in the far zone. Fig. 3 shows the results for several positions of the receiving microphone. The calculated signals on these graphs are represented by a blue curve, the red line corresponds to the measured signal at this point.

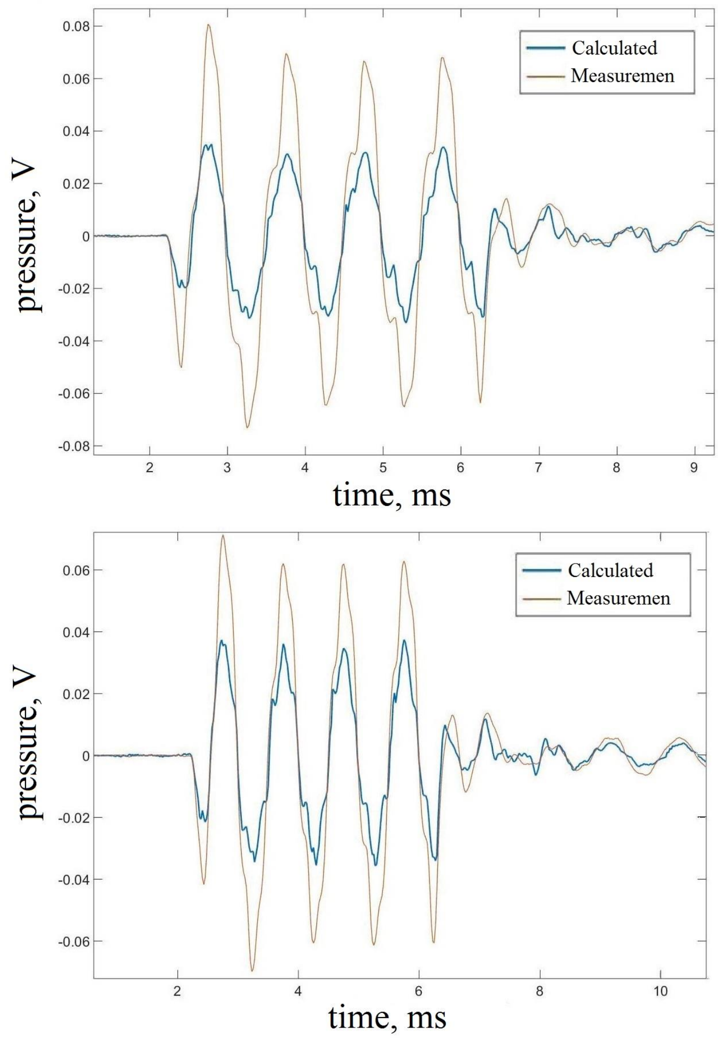

Fig. 3. Comparison of the results of direct measurements in the far zone and calculations based on near-field measurements. Microphone position: top plot: $x = 0$ cm, $y = 0$ cm, $z = z_{far} = 215$ cm, bottom plot $x = 24$ cm, $y = 24$ cm, $z = z_{far}$. FPK method. Acoustic pressure is plotted in relative units over the vertical axis.

It can be seen from the graphs that this method, although quite simple, allows for sufficient accuracy in restoring the signal in the far zone from near-field measurements, which was demonstrated by comparison with the control signals recorded in the far zone.

The obtained results demonstrate that this method, with its simplicity, allows one to restore the waveform and phase center of a signal in the far zone with sufficient accuracy. The difference in amplitude is associated with the instability of the method, because the integral in formula (2) can diverge in the general case. To show this, let us assume that the section S is infinitely large and substitute the spherical wave as p. The origin of the spherical wave is $(0, 0, z_0)$. To reduce the calculations, assume $\alpha = 0$, i.e. $\mathbf{R}$ is aligned with the z axis. After the substitutions made and also assuming $r \gg z_0$, we can rewrite expression (2) in the following form:

$$p(\mathbf{R}) \approx -\frac{ik}{R}\left(\int_0^r \cos(2kr)dr + i\int_0^r \sin(2kr)dr\right)$$

where λ – wavelength [m].

Considering the integration zone as a set of rings of $\lambda/4$ width, one can notice that both integrals turn into alternating series whose members do not decrease by absolute values:

$$p(\mathbf{R}) \approx -\frac{ik\cos(\alpha)}{R}\left(\int_0^{\lambda/8} \cos(2kr)dr + \sum_n \int_{\lambda/8+n\lambda/4}^{3\lambda/8+n\lambda/4} \cos(2kr)dr + i\sum_n \int_{n\lambda/4}^{\lambda/4+n\lambda/4} \sin(2kr)dr\right)$$

Therefore, equation (2) contains a diverging integral.

However the discussed integral is convergent for sources with a higher multipole order. Coming back to the practical task of processing the measurement results, we note that the calculation by formula (3) will result in a finite value due to the finite area of S and due to the introduced apodizing factor. But the increase of the measuring section of the plane may not lead to the expected result of increasing the accuracy. Therefore, the method is workable, and in a number of tasks the achieved accuracy is sufficient.

2.2. Signal processing of the synthesized linear antenna

Next, consider the technique described in [5], and tested in [4], where it was called Far Line Transfer (FLT). This technique allows to restore the field in the far zone, according to the results of a linear antenna [7]. Here, this is simulated by sampling for processing data from sensors located on one straight line in the plane of the source and the point at which the field is restored, in this case the straight line is $y = 0$, $z = 0$.

Introduce a cylindrical coordinate system with the x axis, φ is the angle between the z axis and the axis of the receiving antenna [rad], radius $\rho = \sqrt{y^2 + z^2}$ [m]. The acoustic field can be expanded in cylindrical waves:

$$p(x,\varphi,\rho) = \sum_l \sum_m \left[b_{lm} H_l(\kappa_m \rho)\cos(l\varphi) + c_{lm} H_l(\kappa_m \rho)\sin(l\varphi)\right] e^{(ik_{xm}x)} \tag{4}$$

where b_{lm}, c_{lm} - coefficient, H_l – Hankel function of the l-th order, 1-st kind (sets the diverging wave in the time dependence $e^{-i\omega t}$), wave number $\kappa_m = \sqrt{k^2 - k_{xm}^2}$ [m^{-1}], if a $k^2 \geqslant k_x^2$, otherwise $\kappa_m = i\sqrt{k_{xm}^2 - k^2}$ and $H_l(\kappa_m \rho)$ attenuates with increasing ρ.

The main idea of the proposed method is to eliminate the summation over l in formula (4). To do this, specify the following limitation of the measurement method.

1. the length of the antenna must exceed the length of the source plus two radii of the 1-st Fresnel zone,

2. it is required to calculate the field in the floor of the plane formed by the axis of the source and the axis of the antenna, here is the half-plane $y = y_0 = 0$¸ $z > 0 (\varphi = 0)$.

3. the source should not have an overly complex radiation pattern in the y, z plane. The expansion (4) of the source field can be limited in azimuthal indices to $l \leqslant l_{max}$, and the restored distribution in x in the far zone can be limited to harmonics $k_{xm} \leqslant k_{x,max} = \sqrt{k^2 + \kappa_{min}^2}$, where $\kappa_{min} z_{near} \gg \sqrt{l_{max} + 1}$

The FLT method was described in [5]. According to the indicated work, it is possible to calculate the field in the far zone at the point $(R\sin\alpha,\ y_0,\ R\cos\alpha)$ in the following way:

$$p(\alpha,R) = \frac{b_{m^*} N \Delta x}{R} \sqrt{\frac{\omega z_{near} \cos\alpha}{2\pi c}} \tag{5}$$

where α – the angle between the direction perpendicular to the axis of the antenna and the vector to a point in the far zone [rad], N is the number of measuring points, the coefficient b_{m^*} be according to the formulas (6):

$$b_{m^*} = \int b_m e^{ik\omega} d\omega$$

$$b_m = \frac{1}{N} \sum_{n=-N/2}^{N/2} h_1(x_n) p(x_n, y_0, z_{near}) e^{(-ik_{xm} x_n)} \tag{6}$$

where m^* (coefficient index b_{m^*}) is chosen so that $k_{xm^*}/k = \sin\alpha$.

h_1 – Hanning function, defined by the following formula (7):

$$h_1(x_n) = \frac{1}{W}\left(1 - \cos\frac{2\pi(x_n - x_1 + \Delta x/2)}{x_N - x_1 + \Delta x}\right). \tag{7}$$

W be out of the normalization condition $\sum\limits_{n=1}^{N} h_1^2(x_n) = N$.

The signal calculated using equation (5) is presented на Fig. 4

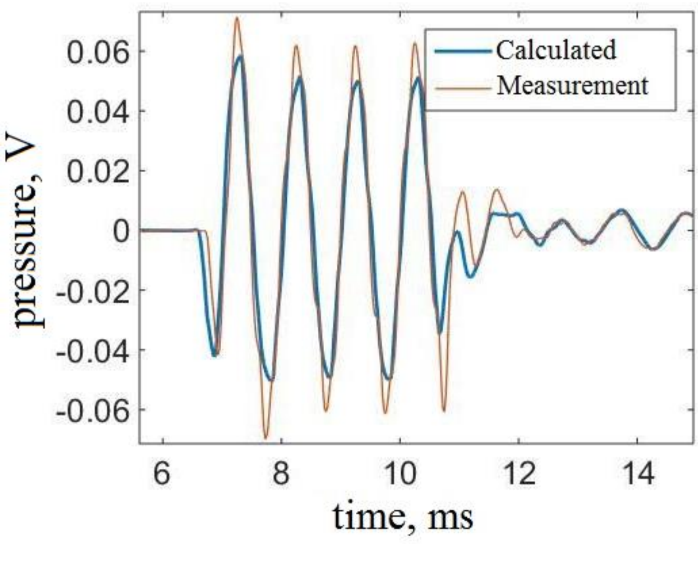

Fig. 4. Restore a field at a point $(0,0,z_{far})$ using a linear antenna method «FLT». The ordinate axis represents acoustic pressure in relative units

This method with good accuracy allows you to calculate the field in the far zone, using a single measurement of a linear antenna.

Conclusion

The paper is devoted to comparing methods of near-field acoustic holography, applied to pulsed signals. The restoration of the field in the far zone from the nearfield measurements is in good agreement with the control measurements by the phase and the waveform. Using the "linear antenna" method, the amplitude is also restored with high accuracy. The proposed methods allow, for example, to measure the pulse transfer function of the speaker system without using an anechoic chamber, provided that the duration of the generated pulse allows one to suppress the reflection from the walls by the gating method.

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