Paper No.334/2021

DETERMINATION OF MONOPOLE AND DIPOLE SOURCES OF FLOW AROUND A CYLINDER USING A VIRTUAL MICROPHONE ARRAY

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***Abstract**: In this paper, the problem of localization of noise sources of flow around a cylinder was considered on the basis of a computational experiment using a virtual 54-channel microphone array. Numerical simulation was performed using the computational fluid dynamics software ANSYS Fluent. Several spatial orientations of the cylinder were considered for the generation of dipoles with different directions. Simulation of a simplified two-microphone azimuthal decomposition technique (ADT) is performed to determine the sound pressure level of the generated dipoles at a vortex shedding frequency of 1450 Hz. A procedure of localization of the noise of the flow around a virtual cylinder was performed using monopole and dipole beamforming algorithms. It was found that the numerical simulation results are in good agreement with the data obtained by other researchers, both in terms of the sound pressure level and the results of the localization of dipoles in space.*

***Keywords:** CFD, ANSYS Fluent, Ffowcs Williams and Hawkings acoustic analogy, beamforming, acoustic dipole*

***DOI:** 10.36336/akustika20213976*

1. INTRODUCTION

World experience in the development of aircraft shows that a lot of attention is paid to the tools and techniques used that contribute to an expanded and deep understanding of the mechanisms of noise and acoustic load generation. The practical and economic efficiency of these methods is extremely high. The creation of the newest domestic passenger aircraft that meet the ICAO certification requirements for the permissible level of noise generated is impossible without the use of advanced technologies in the field of noise suppression. In this regard, today the development of methods and means to deal with aircraft noise is a particularly urgent task.

To solve the problems associated with undesirable acoustic impact, it is necessary to have information about radiation sources: about the mechanisms of generation of this noise, its spectral characteristics, as well as about the distribution of sound sources in space. There are various ways to achieve this goal. One such method is the use of multi-microphone arrays.

A microphone array is a set of acoustic sensors, arranged among themselves in a certain way, to register the sound pressure at a known distance from the plane of the estimated sources. After carrying out the post-processing procedure for the set of received sound signals, it becomes possible to reconstruct the sound field based on model ideas about the types and distribution of the considered sources. Post-processing is performed using mathematical algorithms that allow obtaining information about the distribution of acoustic sources in the investigated area of space, spectral characteristics at various points in this area, as well as sound pressure levels at frequencies of interest. Thus, beamforming is a contactless method of obtaining information about sound sources created by various objects, which, together with the subsequent application of various noise suppression methods, makes it a powerful tool for solving problems associated with unwanted acoustic impact.

The efficiency of localization is directly influenced by two factors: the choice of the microphone array and its parameters, as well as the algorithm for post-processing of the recorded sound pressure by microphones to create localization maps and obtain the spectral characteristics of the investigated sources. The most widely used is the classic post-processing method called conventional (or «Delay-and-sum») beamforming [1, 2]. However, it has a relatively low resolution, which reduces the overall localization accuracy [3]. Currently, research is actively developing in the search for new, more efficient methods of post-processing. A large number of algorithms have been developed that make it possible to achieve a significant improvement in localization maps by overcoming the limit of spatial resolution of conventional beamforming. For example, in the DAMAS, CLEAN, NNLS methods, the results of conventional beamforming are the initial data for constructing cleaned localization maps using iterative post-processing methods and deconvolution methods [3].

The vast majority of post-processing methods are based on representing sound signals as uncorrelated monopole sources. This assumption can lead to distortion of the results obtained depending on the objects of the sound field generation. This fact is especially evident in the problems of aeroacoustics. For example, one of the main sources of noise generated by a modern passenger aircraft in flight is the aerodynamic noise of the airframe flow, which in the landing mode is comparable to the noise of the aircraft power plant [4, 5]. The occurrence of such noise is caused by pulsations of aerodynamic forces on the wing, a turbulent boundary layer, vortices generated from the flow around the airframe surfaces, as well as turbulent wakes behind poorly streamlined protrusions. For an airplane in cruise mode, the main source of airframe noise is airflow noise. During the landing approach, with the landing gear extended and flaps deflected, the main noise sources are caused by the stalling flow around the structural elements protruding into the flow. When the flow around the landing gear struts in a certain range of Reynolds numbers, there is a noticeable dipole sound radiation into the environment due to the force effect of the flow [6]. A dimensional analysis of the general equation of sound generation by a flow in the presence of solid boundaries shows that aerodynamic noise is generated by dipole-type sources, which are fluctuating forces arising from the interaction of the flow with a streamlined body [7].

Obviously, traditional post-processing algorithms are inapplicable for studying these problems, since they can give distorted results. Consequently, it becomes necessary to apply modified dipole algorithms to carry out the post-processing procedure. There are a number of scientific articles devoted to the localization of dipole sound sources using microphone arrays. In [8], the «D-Beam» method was used with an algorithm correction based on the reference model dipole to determine its most probable orientation. Using this approach, experimental studies have been carried out in a wind tunnel with a flow around a cylinder in various spatial orientations relative to the incident flow to generate dipole sound sources. TsAGI specialists performed a similar approach of applying their own dipole algorithm using the azimuthal decomposition technique [9, 10] in order to evaluate the contribution of individual azimuthal modes and determine the level of their intensity in the overall level of noise generated during an experiment with a jet flow around a cylinder. In [11], a dipole algorithm is described in detail, which is based on the use of a pseudoinverse matrix. A thorough comparison of the efficiency of localization of this method with other well-known algorithms (Conventional Beamforming [1, 2], CLEAN-SC [12], Adaptive Beamforming [13], DAMAS [14], etc.) has been carried out. The efficiency of localization of various types of sound sources using this algorithm was also tested in detail in [15] on model and experimental sound signals.

In addition to the development of new post-processing algorithms, array optimization methods are also actively developed to improve the quality of localization in order to find the optimal distribution of microphones in the array. The choice of microphone positions directly affects the clarity of the final localization map by influencing the levels of unwanted side lobes, which in certain cases can be mistaken for suspected acoustic sources. In addition, it is necessary to provide a high array resolution for high-quality separation of closely spaced sound sources from each other. Thus, the main goal of optimization is to achieve an optimal combination of high dynamic range values in conjunction with a high array resolution. To achieve this goal, various researchers apply various optimization algorithms (taking into account the optimal costs of computing resources, the convergence rate, the ability of the algorithm to effectively overcome local minima of the objective function, etc.). Methods such as Minimax [2], Differential Evolution [16], Genetic Algorithm [17] and others are used.

After optimization of the microphone positions, there are several ways to check the efficiency of localization using this array: using an analytical model, conducting a natural experiment, or by conducting a computational experiment. It should be remembered that in practice, when carrying out a full-scale experiment, the real efficiency of the microphone array will differ from the optimal one to varying degrees due to the influence of many different factors. For example, due to an error in the installation of microphones, a discrepancy between the real sound field specified in the mathematical model, non-fulfillment of the free field conditions, the presence of curvature of the sound wave front, and also due to the influence of other factors depending on the measurement conditions.

To carry out natural experiments, it is necessary to mount an array, install microphones in it, calibrate them, connect power cables, spectrum analyzers and a webcam, prepare a source for generating a sound field, perform acoustic measurements and post-process the results. These actions take a significant amount of time, and if there are several optimized array sets, and the array does not have the ability to quickly reposition the microphones [18], then the time costs will increase even more. Therefore, to save time, preliminary confirmation of the declared efficiency of the optimized arrays can be carried out on analytical models or using a computational experiment.

Taking into account the relevance of the localization of dipoles for the problems of aeroacoustics, in this work it was decided to perform a numerical simulation of the noise of the flow around a cylinder to generate dipole sound sources. The numerical simulation results will be the initial data as the sound pressure recorded by the virtual microphones. In this work, to create a model acoustic dipole, a numerical experiment was performed on the flow around a cylinder in several spatial orientations, followed by localization of the generated dipole using a virtual microphone array using a modified dipole algorithm for post-processing.

2. DESCRIPTION OF BEAMFORMING ALGORITHMS

Conventional beamforming is an effective method of post--processing acoustic signals that can determine the position and sound pressure level of a source at various frequencies. This method is widely applicable and allows acoustic research to be carried out in a wide range of tasks. However, it should be remembered that one of the most important assumptions of any algorithm is the choice of a sound source model. Conventional beamforming generally assumes that at each point of the discrete grid there is a monopole source with an unknown amplitude. A monopole source in the frequency domain can be described by the expression [19]:

$$\boldsymbol{p(\vec{x},f)} = \frac{a(f)e^{-2\pi i f \Delta t_e}}{4\pi\|\vec{x}-\vec{\xi}\|} \tag{1}$$

where

a(f) is the pressure amplitude,

i is the imaginary unit,

$\boldsymbol{\Delta t_e}$ is the frequency,

$\vec{\boldsymbol{x}}$ is the emission time delay,

$\vec{\boldsymbol{\xi}}$ is the microphone position and is the monopole position.

More generally, this expression can be written as:

$$\boldsymbol{p(\vec{x},f)=a(f)g(\vec{x},f)} \tag{2}$$

where

$\boldsymbol{g(\vec{x},f)}$ is a steering function linking the coordinates of the point grid of and the positions of microphones in the array at the frequencies of interest with a certain time delay.

The array of sound pressures recorded by each microphone can be written as an N-dimensional vector $\boldsymbol{p}$ (where N is the total number of microphones):

$$\boldsymbol{p}=\begin{pmatrix}\boldsymbol{p_1(f)}\\ \vdots \\ \boldsymbol{p_N(f)}\end{pmatrix} \tag{3}$$

Based on this vector, you can write the cross-power matrix $\boldsymbol{C}$:

$$\boldsymbol{C=\frac{1}{2}pp^*} \tag{4}$$

The asterisk means complex conjugate transpose.

Next, the model pressure vector is approximated to the experimental one by choosing the amplitudes. The error function (discrepancies between the experimentally obtained data with their theoretical representation) can be written as:

$$\boldsymbol{J=\|p-\bar{a}g\|^2} \tag{5}$$

where

$\boldsymbol{g}$ is the steering vector containing the components $\boldsymbol{g_n=g(\vec{x},f)}$. By minimizing this norm by the least squares method, you can get a solution for finding a set of complex amplitudes:

$$\boldsymbol{\bar{a}=\frac{g^*p}{\|g\|^2}} \tag{6}$$

The final values for constructing a beamforming map are estimated source auto-powers, which are expressed in terms of complex amplitudes as:

$$\boldsymbol{\bar{A}=\frac{1}{2}|\bar{a}|^2=\frac{1}{2}\bar{a}\bar{a}^*=\frac{1}{2}\left(\frac{g^*p}{\|g\|^2}\right)\left(\frac{g^*p}{\|g\|^2}\right)^*=\frac{1}{2}\frac{g^*pp^*g}{\|g\|^4}=\frac{g^*Cg}{\|g\|^4}} \tag{7}$$

This expression is an analytical solution for determining the estimated source auto-powers at each point of the original grid.

The source model (2) can be used not only for monopoles, but also for dipoles, quadrupoles, and multipoles by modifying the steering function $\boldsymbol{g(\vec{x},f)}$. Therefore, to estimate the source auto-powers of dipole sources, the steering vector $\boldsymbol{g}$ must be corrected taking into account the dipole strength $\boldsymbol{a(\omega)l=1}$. Thus, the single component of the steering vector will be equal to [8]:

$$\boldsymbol{g_n=\frac{-e^{-i\omega\Delta t_e}\cdot DPL}{4\pi\vec{r}l}}, \qquad \boldsymbol{DPL=-i\omega l\cdot\nabla(\Delta t_e)} \tag{8}$$

where

$\boldsymbol{l}$ is the distance between closely spaced mono poles with an opposite phase,

$\boldsymbol{\omega}$ is the angular frequency,

$\boldsymbol{\vec{r}=\|\vec{x}-\vec{\xi}\|}$ is the propagation vector from source to receiver.

3. NUMERICAL SIMULATION OF THE NOISE OF THE FLOW AROUND A CYLINDER

In this paper, the problem of determining the flow noise around a cylinder was considered. The experimental conditions and the spectrum of the experimentally measured sound pressure level were taken from [20]. In the experiment, the uniform upstream velocity U=72m/s and the cylinder diameter is D=0.01m corresponding to Reynolds number is 46000. The vortex shedding frequency is about 1500 Hz at this condition. To find unsteady pressure, velocity and density fields around a cylinder, the ANSYS Fluent CFD software was used. Detached-eddy simulation is performed to compute the flow field and the far-field noise is calculated with the Ffowcs Williams and Hawkings acoustic analogy [21].

Fig. 1 shows the computational mesh. The computational domain is a rectangle built around a cylinder with dimensions: 3d wide; 6d in front of the cylinder; 14d after the cylinder; 5d above and below the cylinder. The mesh is composed of about 2.4×10^5 grid points: 120 points around circumference, 100 radial and 20 along span. The first grid distance is 2×10^{-5} m, according to y+ almost 0.8. Due to limitations in computational resources, the mesh spanwise length is 3D, which is similar to other works on flow around a cylinder [20, 22, 23].

When carrying out the gas-dynamic calculation, the following boundary conditions were set: at the inlet (Fig. 1) – mass flow inlet 0.319 kg/s, which corresponds to an incoming flow velocity of 72 m/s, a temperature of 290 K; at the outlet and upper and lower boundaries – an open boundary with zero gauge pressure and a temperature of 290 K. Periodic boundary conditions were set on the side surfaces. The calculation used the implicit control-volume method with the second order of accuracy in space and time and the DDES approach for modeling turbulence. The calculation was carried out with a time step of 2×10^{-5} s. The number of time steps in the calculation was 16000. For example, Fig. 1 shows the instantaneous velocity field obtained in the calculation in the cross section averaged over the length of the cylinder.

The acoustic analogy approach means that the flow simulation and the noise calculation separately, and the far-field noise can be obtained from the near-field flow. To determine the noise in the far field, non-stationary fields of pressure, density and velocity on the Kirchhoff surface were recorded into files. The selected Kirchhoff surface is a cylindrical surface with a radius equal to 1 cylinder diameter and an axis coinciding with the cylinder axis (see Fig. 1).

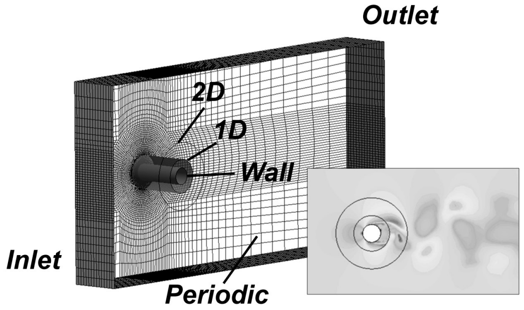

Fig. 1: CFD mesh and instantaneous flow visualization

Non-stationary fields on Kirchhoff surfaces were consistently used for calculations according to Ffowcs Williams and Hawkings acoustic analogy at the point where the sound level was measured in the experiment. This point is 0.15 meters upstream and 1.85 meters above the cylinder.

For the received signals, a Fourier transform was performed and the PSD was determined. Due to the fact that a region of length 3d was used in the calculation, and in the experiment the length of the cylinder was 30d, a noise level correction was performed. According to the work [20], the correction value was 15 dB. A comparison of the noise spectra at this point is shown in Fig. 2. As you can see, the calculation results are good at predicting the level of the peak in the spectrum and the level of the broadband signal. The peak frequency in the calculation is 1450 Hz, which also correlates well with the frequency in the experimental spectrum (about 1500 Hz). In this case, the Strouhal number is about 0.2, which is in good agreement with the results for this type of flows.

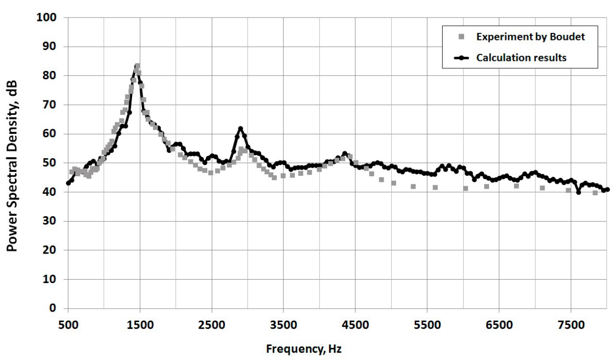

Fig. 2: Comparison of the spectrum obtained as a result of the numerical simulation at point (0.15, -1.85, 0.00) with the experimental data obtained in [20]

To determine the directivity of the noise radiation, the virtual microphone installation points were added to the acoustic calculation. The observation points were located at a radius of 1 m every 10 degrees in a plane perpendicular to the axis of the cylinder passing through its middle. The dependence of the sound pressure level on the angle for the vortex shedding frequency (1450 Hz) is shown in Fig. 3. As can be seen, the directivity has the shape of a figure eight, which is typical for dipole sound sources.

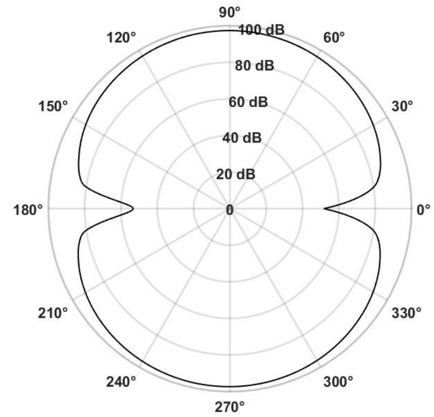

Fig. 3: Directivity of noise radiation at the vortex shedding frequency of 1450 Hz

At the next stage, to construct the frequency spectrum of the components of the longitudinal and transverse dipoles, it was decided to measure the sound pressure additionally at two more points equidistant symmetrically relative to the flow axis at a distance of 1 m in the nozzle exit plane to simulate the application of a simplified two-microphone azimuthal decomposition technique, as was carried out by TsAGI specialists [10]. The signals corresponding to the longitudinal (mode a0) and transverse (mode a1) dipoles can be calculated by the formulas:

$$\begin{aligned} a_0(x,t) &\approx (p_1(x,t) + p_2(x,t))/2 \\ a_1(x,t) &\approx (p_1(x,t) - p_2(x,t))/2 \end{aligned} \tag{9}$$

The corresponding signal spectra are shown in Fig. 4. The maximum sound pressure level for the longitudinal dipole was 94.2 dB at a frequency of 1450 Hz.

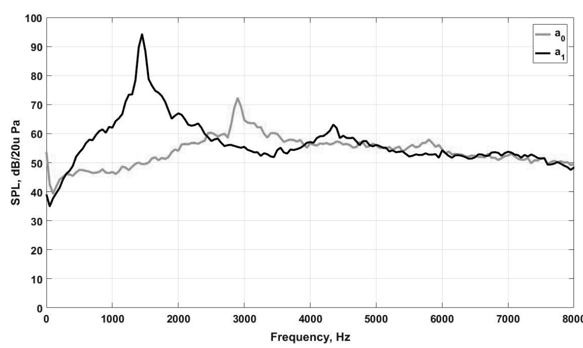

Fig. 4: The spectrum of longitudinal (mode a0) and transverse (mode a1) dipoles obtained by simulating the application of a simplified two-microphone azimuthal decomposition technique

Further, the noise sources of the flow around a cylinder were localized using a virtual microphone array. For this, a computational experiment was performed for two spatial orientations of the cylinder: in the first case, the longitudinal axis of the cylinder crossed the center of the microphone array; in the second case, it was parallel to the plane of the microphone array and the nozzle exit plane. If in the first case the classical generation of an acoustic dipole in a plane parallel to the array was considered, then the second case was performed to check the correctness of determining the radiation intensity of the generated dipole by the modified dipole localization algorithm. In this spatial orientation, the region of the phase jump of the signal is directed straight to the array, and the sound pressure level at the radiation peak localized by the monopole algorithm is comparable to the sound pressure level of the initial acoustic dipole [10]. Thus, using the monopole-based Conventional Beamforming algorithm, it is possible to debug the created algorithm for the localization of acoustic dipoles at various stages of its development.

A set of coordinates corresponding to the location of microphones in the 54-channel Bruel & Kjaer Type WA-1676-W-003 array available at the PNRPU Acoustic Research Center was used as points for recording the sound pressure. The variants of the location of the cylinder relative to the plane of the microphone array are shown in Fig. 5. The results of localization of the noise sources of the flow around a cylinder for these locations are shown in Fig. 6.

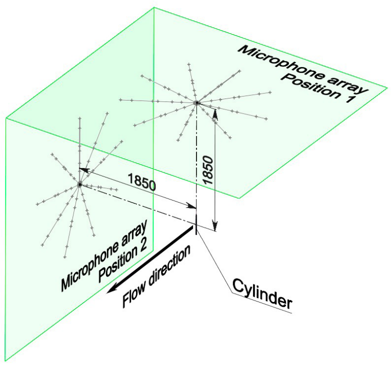

Fig. 5: Positions of the virtual microphone array relative to the cylinder

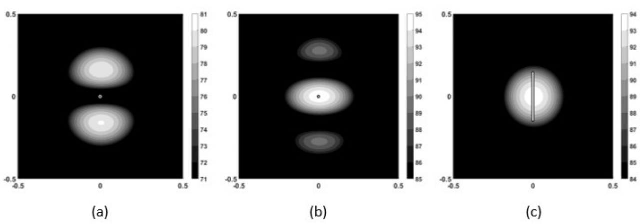

Fig. 6: Localization of the noise sources of the flow around a cylinder using a virtual microphone array at a vortex shedding frequency of 1450 Hz. (a) – monopole-based conventional beamforming algorithm, array position 1; (b) – dipole-based beamforming algorithm, array position 1; (c) – monopole-based conventional beamforming algorithm, array position 2. A schematic representation of the cylinder orientation is shown in the center of the map.

In Fig. 6a and 6b, the cylinder axis is perpendicular to the plane of the microphone array (Position 1), in Fig. 6c is parallel to it (Position 2). Fig. 6a shows a conventional beamforming algorithm, Fig. 6b shows a modified dipole-based beamforming algorithm, and Fig. 6c again shows a conventional beamforming algorithm for representing the sound pressure level in the region of the phase jump of the signal directed straight to the array, compared to the sound pressure level of the localized dipole in Fig. 6b. As you can see, these levels differ by no more than 1 dB, which indicates the correct operation of the dipole noise source localization algorithm. Also, an additional comparison of the sound pressure level of the longitudinal dipole at a frequency of 1450 Hz, obtained from the spectrum using the azimuthal decomposition technique, with the peak level on the acoustic dipole localization map obtained using the developed algorithm (Fig.6b), showed comparable results differing by no more than by 1 dB. Thus, we can conclude that the computational experiment was performed correctly and is in good agreement with the data obtained by other researchers.

4. CONCLUSION

In ANSYS Fluent, a computational experiment was carried out on the flow around a cylinder to generate a virtual dipole source. The sound pressure at the points corresponding to the microphone mounting points in the virtual microphone array was obtained using an acoustic analogy to calculate the sound in the far field using the Ffowcs Williams and Hawkings equation. A comparison of the spectral characteristics of the noise of flow around the cylinder with the data obtained by other researchers was carried out. Localization of the generated dipole using a virtual microphone array in several spatial orientations using a modified dipole algorithm for post-processing was carried out. The dipole source localized by the virtual array is in good agreement in terms of the sound pressure level with the values obtained using the virtual application of the simplified azimuthal decomposition technique at the vortex shedding frequency (1450 Hz).

ACKNOWLEDGEMENT

The reported study was funded by RFBR, project number 19-32-90071.

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