



# Sound source localization method in an environment with flow based on Amiet–IMACS



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## ABSTRACT

A sound source localization method is proposed to localize and analyze the sound source in an environment with airflow. It combines the improved mapping of acoustic correlated sources (IMACS) method and Amiet's method, and is called Amiet–IMACS. It can localize uncorrelated and correlated sound sources with airflow. To implement this approach, Amiet's method is used to correct the sound propagation path in 3D, which improves the accuracy of the array manifold matrix and decreases the position error of the localized source. Then, the mapping of acoustic correlated sources (MACS) method, which is as a high-resolution sound source localization algorithm, is improved by self-adjusting the constraint parameter at each iteration process to increase convergence speed. A sound source localization experiment using a pair of loud speakers in an anechoic wind tunnel under different flow speeds is conducted. The experiment exhibits the advantage of Amiet–IMACS in localizing a more accurate sound source position compared with implementing IMACS alone in an environment with flow. Moreover, the aerodynamic noise produced by a NASA EPPLER 862 STRUT airfoil model in airflow with a velocity of 80 m/s is localized using the proposed method, which further proves its effectiveness in a flow environment. Finally, the relationship between the source position of this airfoil model and its frequency, along with its generation mechanism, is determined and interpreted.

## 1. Introduction

With the rapid development of the aviation industry, attention focused on the aerodynamic noise problem faced by large civil aircraft and high-speed air vehicles during flight, is gradually increasing [1]. This problem is not only harmful to the environment, cabin crew and passengers, and aeronautic equipment, but it also causes acoustic fatigue in the structure of air vehicles [2,3]. Numerous works that measure and evaluate aerodynamic noise is conducted to address this potential safety hazard [3–10]. These works offer scientific proofs and data supports for the noise reduction design and acoustic fatigue analysis of aircraft.

Experimental studies on the sound sources of air vehicles or those of their components are generally conducted in wind tunnels, where the microphone array technique is implemented to analyze the sound source information of the airframe surface and the structure placed in airflow. As one of the widely used array signal processing algorithms, beamforming [11–15] can be performed to localize the sound

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source on the surface of the observed object. During the beamforming process, an observation area is divided into a group of points, called reconstructed points, in which the sound pressure levels are computed via beamforming to identify the main sound sources in the area. In the past, conventional delay and sum (DAS) beamforming has been primarily proposed to identify sound sources. Each channel of a signal is time-delayed and summed using this algorithm, which enhances the output in the observed direction and suppresses those in the other directions. Therefore, the position of the sound source is determined. DAS beamforming was introduced as early as the 1970s in aeroacoustic research, such as in detecting the source of aircraft engine noise or airframe noise [16,17].

DAS beamforming exhibits the advantages of good robustness and low computational burden. However, it suffers from poor spatial definition caused by the Rayleigh limit, particularly for low-frequency sound sources. The size of the reconstructed source using DAS beamforming is considerably larger than that of a real sound source which is even point like. Therefore, precisely localizing the sound source is difficult via DAS beamforming. In recent years, beamforming algorithms combined with deconvolution methods have been proposed, such as DAMAS [18] and CLEAN [19]. These algorithms reduce the negative influence of the Rayleigh limit on source localization to improve spatial definition. However, the aforementioned high-definition methods assume that acoustic sources are uncorrelated, and thus, they do not exhibit ideal performance when the sources are correlated. Afterward, a few beamforming algorithms that consider source correlation have been proposed, such as DAMAS-C [20] and CMF-C [21]. However, impractical computational burden limits the application of these algorithms in aeroacoustic measurement. In 2010, Tarik Yardibi [22] from the University of Florida proposed a new beamforming algorithm, called mapping of acoustic correlated sources (MACS), which considered source correlation based on CMF. In this method, the relationship between the cross-spectrum of the microphone array signal and the covariance matrix computed using the sound pressure at each reconstructed point is established. Then, convex optimization is implemented based on a simplified calculation model to obtain the final sound pressure of each reconstructed point with a remarkably reduced computational burden.

The performance of MACS in an environment with flow will be improved if the following disadvantages can be overcome. First, a constant constraint parameter is used during the convex optimization process of the original MACS, which negatively affects convergence speed. Then, as the propagation path of the sound waves deflects in an environment with airflow, array manifold correction must be conducted in advance to realize a more accurate sound source localization result using MACS.

To achieve sound source localization with high spatial definition in an environment with airflow, an updated version of MACS method (i.e. IMACS) with faster convergence speed than MACS, is combined with Amiet's method [23] in this study. During the implementation of this approach, the shear flow correction model proposed by Amiet is introduced to correct the sound propagation path in an environment with airflow to achieve a more accurate array manifold matrix. Then, the corrected array manifold matrix and the microphone array signals are fed into the IMACS algorithm to identify the position of the sound source with airflow. The combined Amiet–IMACS method exhibits the following characteristics: (1) the 3D sound propagation path is corrected using Amiet's method in an environment with flow, which improves the accuracy of the array manifold matrix and reduces the estimated position error of the sound source; (2) the IMACS algorithm exhibits the advantage of self-adjusting the constraint parameter during the convex optimization process compared with the original MACS, which increases convergence speed; and (3) the Amiet–IMACS method presents the advantage of accurately identifying the aerodynamic noise source.

Sound field measurement in an anechoic wind tunnel with a maximum flow velocity of 80 m/s is conducted to prove the effectiveness of the proposed method. An Archimedes spiral microphone array with 28 microphones is placed in the wind tunnel to measure a pair of loud speakers that produces correlated sound sources at different flow velocities. Then, the aerodynamic noise produced by a NASA EPPLER 862 STRUT airfoil model in a flow with a velocity of 80 m/s is measured using the same microphone array. The sound source analysis results of the experiments validate the advantage and effectiveness of Amiet–IMACS in localizing the sound source in an environment with flow.

The rest of this paper is presented as follows. The principle of the proposed method is introduced in Section 2. Then, the experiment that localizes the correlated sound sources produced by a pair of loud speakers is discussed with the data analysis in Section 3. In Section 4, the result of the analysis of the aerodynamic sound source distribution of the NASA EPPLER 862 STRUT airfoil model in airflow is presented. Finally, the conclusions of the study are provided in Section 5.

## 2. Amiet–IMACS theory

### 2.1. Effect of airflow on sound source localization

A microphone array that consists of  $M$  microphones and a reconstructed plane that consists of  $L$  discrete points are shown in Fig. 1. Each reconstructed point can emit the sound wave signal to the microphone array. If the flow speed  $V_0$  is 0, then the sound wave propagation path from a reconstructed point to a microphone will be a straight line, such as the dashed line shown in Fig. 1. Consequently, the signals received by the microphone array are expressed as Eq. (1):

$$\mathbf{y}(b) = \mathbf{A}\mathbf{s}(b) + \mathbf{e}(b), \quad b = 1, 2, \dots, B, \quad (1)$$

where  $\mathbf{y}(b) = [y_1(b), \dots, y_m(b), \dots, y_M(b)]^T$ ,  $\mathbf{A} = [\mathbf{a}_1, \dots, \mathbf{a}_l, \dots, \mathbf{a}_L]$ ,  $\mathbf{s}(b) = [s_1(b), \dots, s_l(b), \dots, s_L(b)]^T$ , and  $\mathbf{e}(b) = [e_1(b), \dots, e_m(b), \dots, e_M(b)]^T$ .  $y_m(b)$  represents the frequency domain sound pressure data of the  $m$ -th microphone at block  $b$ ,  $B$  is the number of data blocks used to implement fast Fourier transform,  $\mathbf{A}$  is the array manifold matrix with a  $M \times L$  dimension,  $s_l(b)$  is the frequency domain source signal of the  $l$ -th reconstructed point at block  $b$ , and  $e_m(b)$  is the frequency domain noise signal of the  $m$ -th microphone at block  $b$ . Each array manifold vector  $\mathbf{a}_l$  in the array manifold matrix  $\mathbf{A}$  is expressed as Eq. (2):

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