



Improving acoustic beamforming maps in a reverberant environment by modifying the cross-correlation matrix[☆]



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ABSTRACT

A method to improve the quality of acoustic beamforming in reverberant environments is proposed in this paper. The processing is based on a filtering of the cross-correlation matrix of the microphone signals obtained using a microphone array. The main advantage of the proposed method is that it does not require information about the geometry of the reverberant environment and thus it can be applied to any configuration. The method is applied to the particular example of aeroacoustic testing in a hard-walled low-speed wind tunnel; however, the technique can be used in any reverberant environment. Two test cases demonstrate the technique. The first uses a speaker placed in the hard-walled working section with no wind tunnel flow. In the second test case, an airfoil is placed in a flow and acoustic beamforming maps are obtained. The acoustic maps have been improved, as the reflections observed in the conventional maps have been removed after application of the proposed method.

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1. Introduction

Recently, microphone array methods have gained popularity in the field of sound source localization, and more specifically when combined with a Beamforming (BF) algorithm [1–3]. The main idea behind beamforming is to delay and sum the microphone signals with respect to several positions where the source, normally (but not always) assumed to be a monopole, is sought. The so-called delay-and-sum beamforming is preferably used in the frequency-domain and is then known as Conventional Beamforming (CBF). However, some recent works have made use of the time-domain formulation of the algorithm [4,5] which can be useful especially when investigating intermittent noise sources [6–8].

The first limitation of the CBF algorithm is known as its frequency-dependent resolution, which increases beamwidth at lower frequencies. In the case of a line array of microphones, the resolution, defined as the width of the main lobe at -3 dB, is proportional to the inverse of the frequency of interest [9]. To overcome this issue, some deconvolution algorithms have been proposed, such as the popular and effective Clean-SC method [10] and the Deconvolution Approach for the Mapping of Acoustic Sources (DAMAS) [11]. In addition to providing a reduced beamwidth over the frequency range of interest, these algorithms also reduce the intensity of the sidelobes, which are a consequence of the limited number of sensors. However, these algorithms, especially DAMAS, require a large amount of computational resources. Thus, Dougherty [12] proposed an improvement known as DAMAS2 which is an optimized (in terms of computation time) version of the initial DAMAS. These methods, as well as other deconvolution algorithms, were compared on several numerical and experimental cases by Chu et al. [13]. More recently, a hybrid method, based on inverse problem and eigenvalue decomposition of the Cross-Spectral Matrix (CSM) of the microphone

[☆] Fully documented templates are available in the elsarticle package on CTAN.

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signals was proposed by Padois et al. [14]. However, although the resolution of CBF, Clean-SC and DAMAS maps is improved, this technique increases the amount of computational time.

Another issue that affects the CBF algorithm occurs when the environment is reverberant, which is usually the case in hard-walled, closed-return wind tunnels. Indeed, CBF is based on the assumption of a free-field propagation of a monopolar source and thus may not be adapted in some cases such as highly reverberant test sections for instance. The problem of reflections in acoustic maps has been first investigated by Guidati et al. [15] in the case of a rectangular test section. The positions of the image sources due to the wall were estimated theoretically and their effect was incorporated into the beamforming process. When applying this so-called reflection canceller on experimental trailing edge noise, the beamforming map was better resolved. Using the same idea, Fenech et al. [16] proposed a de-reverberation method called the Image Source Model (ISM) that used a modeled Green's function. This method was also used by Fischer et al. [17] who have compared empirical de-reverberation beamforming maps by using numerical and experimental Green's functions as inputs for the beamforming algorithm. They showed that the resolution of the main peak was improved when using the experimental steering vectors [18]. Another approach was proposed by Sijtsma et al. [19] where the beamforming algorithm was modified to take into account the influence of a mirror source coherent with the main peak. The spectra of the focused beamformer on the source position was better recovered using this method but the localization accuracy was not improved. Blacodon et al. [20] proposed a cepstral method to remove the influence of reflections in array measurements. They observed that the quefrequencies of echoes affect microphones differently, while those of the actual source do not. Thus, the echoes could be successfully removed for numerical and experimental test cases.

The numerous image sources caused by a reverberant environment not only affect the resolution and location of the beamforming output but can also deteriorate the level of the main source. One way to estimate the level difference due to a reverberant test section is to compare the acoustic maps of a model placed in free-field and reverberant environments. Fleury et al. [21] compared the spectra of airfoil slat noise measured in a reverberant closed test section and an open anechoic wind tunnel. The spectra are quite comparable for low angles of attack even though slight differences can be noticed in some frequency bands. Pagani et al. [22] have also measured the slat noise in a closed-section wind tunnel at low to moderate angles of attack. However, they compared their results with a numerical simulation conducted by a lattice-Boltzmann solver. The level on the spectra agreed, in particular regarding the dominant parts of the spectra. However, in a closed-section wind tunnel the slat-wall junction region is potentially affected by both spurious and image sources, which reduces the length of the wing span that can be safely used for noise sources integration [23,24].

Previous methods to remove the effects of wall reverberation use either simplistic models or complex empirical ones that are time consuming to develop and are limited to a specific wall geometry. This paper presents a new approach that is able to remove the effects of wall reflections without the need for an additional wall-reflection model. The aim of the proposed method is to drastically improve the location and resolution of the main source on the beamforming output. While this is achieved at the cost of a deterioration in the accuracy of the level estimation, some solutions are presented that aim to correct this issue. In this paper, the method is demonstrated in a hard-walled wind tunnel environment, both for a generic acoustic source without flow and for the challenging test case of airfoil self noise during a wind tunnel test.

2. Experimental set-up

2.1. Speaker source with no flow

All experiments were conducted in the Large Wind-Tunnel (LWT) of the aerospace laboratory at UNSW Sydney. For the first test case, no flow was considered and a speaker was used as a controlled source. The aim of this test case is to isolate the effect of the walls so as to demonstrate the de-reverberation technique. The dimensions of the rectangular closed test section are 1.28 m $\times$ 0.92 m. The experimental setup in the wind tunnel is shown in Fig. 1. Fig. 1(a) is a photograph showing the wall mounted array and an automated traverse that is used to hold the speaker and accurately position it in space. Fig. 1(b) shows the dimensions of the test section as well as the microphone array and scanning grid locations.

A 31-channel acoustic array, presented in Fig. 2, was set on one side of the wind tunnel with flush mounted microphones. The array has a multi-arm logarithmic design and is composed of 5 arms with 6 microphones in each plus one additional microphone in the center. The inner and outer radii of the spiral are respectively 0.15 m and 0.22 m. Each 1/4" GRAS 40PH phase matched microphone (frequency range [50 Hz; 10 kHz]) was connected to a PXIe-4499 24 bit simultaneous sample computer. Actually, the microphones are sensitive to 20 kHz but beyond 10 kHz the gain of the frequency response is not constant. A GRAS RA0067 transmitter combined with a prepolarized 1/2" GRAS 40AD microphone and driven by a Type 14AA electrostatic actuator was used as a noise source (the speaker) and uses a white noise signal as an input. The noise source was fixed on a two-dimensional DANTEC traverse system so that it can be moved in a plane parallel to the array in the middle of the test section. The source-array distance is $L = 0.64$ m. The center microphone of the array, which is set as the reference of the domain, is located at $(x; y; z) = (0; 0; 0)$ m while the center point of the scanning grid is positioned at $(x; y; z) = (0; 0; -0.64)$ m. The transmitter microphone was moved using a traverse system over 33 positions in a domain delimited by $[-0.2; 0.2]$ m along x and y . These positions are denoted by the square symbols shown in Fig. 2. For each source position, the acoustic array pressure signals were recorded simultaneously.

All data presented in this work have been measured with a sampling frequency $f_s = 65,536$ Hz over a duration $T = 10$ s. The block-averaging procedure was performed using Welch's periodogram [25] over 160 blocks of $N_{\text{FFT}} = 8192$ discrete points

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