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Beamforming in a reverberant environment using numerical and experimental steering vector formulations ☆

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ABSTRACT

The effect of acoustic reflections on beamforming maps and their correction is investigated in this paper. By replacing the usual steering vector expression in the beamforming algorithm with an adapted one, the effect of reflections can be reduced. Two formulations of the steering vectors are considered. The first makes use of an experimental Green's function, which is obtained by measuring simultaneously the signal of a speaker and of a 31-channel acoustic array in a hard-walled test-section. The second formulation is based on the assumption that the reflections can be modeled by a set of monopoles located at the image source positions. This numerical model is first validated by comparing the obtained Green's function with the experimental one. Then, the beamforming algorithm is modified by using the different steering vector formulations. In addition, the deconvolution algorithm Clean-SC has been used and implemented with the different formulations. The best results in terms of location and resolution accuracy were obtained when using the experimental Green's function formulation.

1. Introduction

Beamforming (BF) is a very popular tool used in aeroacoustics to locate sound sources during wind tunnel testing [1,2]. The idea of locating sound sources by using several focused sensors was initially used during World War II. At the time, soldiers aimed to detect the arrival of enemy aircraft using radar antenna. The first published example of an acoustic array successfully demonstrated noise localisation on an Olympus engine [3]. Since then, acoustic imaging techniques based on phased microphone arrays have gained popularity and effectiveness [1,4]. These are usually computed in the frequency-domain even though some recent work have made use of the time-domain formulation of the algorithm to investigate intermittent noise sources [5–7].

When beamforming makes the assumption of a monopolar source, it is known as Conventional Beamforming (CBF). The principle is to use the acoustic signals recorded by microphones in an acoustic array to recover the source location. The microphone signals are delayed (or phase-shifted in the frequency domain) and summed assuming they are coming from a fictional source located at a so-called focusing point. The procedure is repeated for many focusing point locations in order to create a two-dimensional map of sound, on which the maximum corresponds to the source location.

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

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Even though CBF can be very helpful in a majority of aeroacoustic situations, it suffers from some drawbacks. The most well-known is the frequency dependency of the source resolution, leading to large beamwidth at low frequencies. This issue has been overcome using deconvolution techniques. The most popular are Clean-SC [8] and the Deconvolution Approach for the Mapping of Acoustic Sources (DAMAS) [9]. Clean-SC enables the removal of the side lobes, by using their coherency with the main source, and also reduces the main lobe size. DAMAS provides a constant resolution, set by the user, over all the frequency bands but is much more expensive in terms of computational time. More efficient techniques were introduced by Dougherty [10] who proposed an improvement called DAMAS2. Padois et al. [11] have also proposed a hybrid method, based on inverse problem and eigenvalue decomposition of the Cross-Spectral Matrix (CSM) of the microphone signals. Aside from improving the resolution of CBF, Clean-SC and DAMAS maps, that hybrid method significantly reduces the amount of computational time.

The issue of locating dipole sources has been recently investigated by several authors [12,13] by implementing dipole source directivity characteristics in their beamforming routines. In this situation, the work achieved by Suzuki [14] must be highlighted. He has developed an algorithm that can solve many benchmark problems such as multipole source localization or distributed sources, even though the computational time remains high.

Aerocaoustic tests are sometimes conducted in closed-section wind tunnels which represent a highly reverberant environment due to the reflection of sound from the hard walls. Only a few authors have previously investigated this problem such as Mimani et al. [15] and Guidati et al. [16], who incorporated the image sources in the beamforming process. In Guidati et al., the image source position was estimated theoretically using the rectangular shape of the test section. When applying the so-called reflection canceller on experimental trailing edge noise, the beamforming map was better resolved. Using the same idea, Fenech and Takeda [17] proposed a de-reverberation method called the Image Source Model (ISM) using a modeled Green's function that was compared with an experimental one. He concluded that the best agreement is obtained when considering only first order reflections in the numerical model. Another approach was proposed by Sijtsma and Holthusen [18] 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. A cepstral method has been proposed by Blacodon and Bulte [19] who observed that the quefrencies of echoes are different to those of the source. Thus, the echoes could be successfully removed. More recently, Fischer and Doolan [20] 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 measured steering vectors. However, the scanning grid used for measuring the Green's function was very small (6×6).

In this paper, empirical de-reverberation beamforming methods based on numerical and experimental formulations for the steering vector are proposed, which can be thought of an extension of the work of Fischer and Doolan [20]. Also, this is the first time that a Green's function is measured in a reverberant test-section with a 31-channel acoustic array and using a speaker located over 41×41 different positions. The paper is structured as follows. In Section 2, a brief review of the frequency-domain CBF algorithm is given. The numerical and experimental formulations for the steering vector are also presented and the deconvolution technique Clean-SC is introduced in the end of this section. Section 3 presents the experimental set-up that was used to acquire the acoustic data in the reverberant test-section. The results obtained using the conventional, numerical and experimental formulations for BF and Clean-SC are then compared in Section 4. An analysis of the sound source localization is also provided in terms of location accuracy and resolution. Finally conclusions are given in the closing section of the paper.

2. Beamforming theory

Beamforming is a common tool used in aeroacoustic testing to locate sound sources. The sound radiated by an acoustic source is recorded using a set of microphones, called an acoustic array, usually placed in a two-dimensional plane. The array, which can be seen as an acoustic camera, steers the microphone signals to several positions in a focusing plane where the source is sought. In the time-domain, this process is achieved by delaying each microphone signal with respect to its position and focusing point location. This section will summarize the main equations of CBF and of the deconvolution algorithm Clean-SC. For more detailed information about the algorithms, please refer to the following papers [1,4,8].

2.1. Conventional beamforming

An acoustic array composed of M microphones where the mth microphone is located at $\mathbf{x_m}$ is used to measure the sound of a given source. Each microphone provides a time signal which is projected in the frequency domain using a Fourier transform. The CSM of the frequency vectors $\mathbf{P}(f) = [p_1, \ldots, p_M]^T$ is constructed using the following convention:

$$\mathbf{C}(f) = \overline{\mathbf{P}(f)\mathbf{P}(f)^H} \tag{1}$$

where superscript ^{H} represents the hermitian transpose and $\overline{X}$ denotes the averaging of the quantity X in a number of discrete time blocks using Welch's periodogram. All the data presented in this work have been measured with a sampling