ARTICLE IN PRESS

Journal of Sound and Vibration ∎ (∎∎∎∎) ∎∎∎–∎∎∎

Contents lists available at ScienceDirect

Journal of Sound and Vibration

journal homepage: www.elsevier.com/locate/jsvi

DAMAS with compression computational grid for acoustic source mapping

Wei Ma [a,*], Xun Liu [b]

[a] School of Aeronautics and Astronautics, Shanghai Jiao Tong University, Shanghai, PR China
[b] Shanghai KeyGo Technology Company Limited, Shanghai, PR China

ARTICLE INFO

Article history:
Received 12 January 2017
Received in revised form
10 March 2017
Accepted 26 March 2017
Handling editor: L.G. Tham

Keywords:
Microphone arrays
Beamforming
DAMAS
Acoustic

ABSTRACT

Nowadays phased microphone arrays have become a standard technique for acoustic source mapping. Compared with the conventional delay-and-sum beamforming method, deconvolution approaches such as DAMAS successfully improve the spatial resolution, however require high computational effort. Without optimizing DAMAS algorithm, recently DAMAS with wavelet compression computational grid (denoted by DAMAS-CG1) has reduced significantly computational run time of DAMAS in applications (Ma and Liu (2017) [17]), and Liu (2017) [16]), however DAMAS-CG1 has an inevitable deficiency that the occurrence probability of aliasing increases slightly for complicated sound source. This paper proposes a novel algorithm that DAMAS with computational grid compressed by discarding grid points with non-positive beamforming (denoted by DAMAS-CG2). Application simulations and an airfoil trailing edge noise experiment show that DAMAS-CG2 not only reduces significantly the computational run time but also retains the spatial resolution of DAMAS on original grid. Moreover DAMAS-CG2 overcomes the inevitable deficiency of DAMAS-CG1, and is simpler and more practical than DAMAS-CG1, although DAMAS-CG2 is usually less effective than DAMAS-CG1.

1. Introduction

Nowadays phased microphone arrays have become a standard technique for acoustic source mapping [1]. The conventional delay-and-sum (DAS) beamforming algorithm constructs a dirty map of source distributions from array microphone out pressure signals [2]. Although conventional DAS beamforming is simple and robust, its main disadvantages include poor spatial resolution particularly at low frequencies [3] and appearance of ghost sources due to side-lobe effects [4,5].

Deconvolution approaches have been developed to overcome the disadvantages of beamforming, through reconstructing a clear map of source distributions from dirty map by iteratively deconvolution. Some well-known deconvolution approaches used in acoustic field include Non-Negative Least-Squares (NNLS) [6], CLEAN [7], DAMAS [8], CLEAN-SC [9] as well as compressive sensing algorithm [10–13]. In these deconvolution approaches, DAMAS is the breakthrough due to it makes deconvolution become well established in the acoustic field. Unfortunately, deconvolution approaches require a relatively high computational run time compared with the conventional beamforming [14]. With less computational run time of

* Corresponding author.
E-mail address: mawei@sjtu.edu.cn (W. Ma).

http://dx.doi.org/10.1016/j.jsv.2017.03.027
0022-460X/

deconvolution, it is possible to reduce significantly the cost of measurements, and to improve the ability of real-time display and online analysis. A lot of effects of researches have thus been devoted to reducing computational run time of deconvolution.

For reducing computational run time of deconvolution, one strategy is applying more efficient deconvolution algorithms such as Fourier-based algorithms (e.g. DAMAS2 [15], FFT-NNLS [16]). Spectral procedures are applied in these Fourier-based algorithms, under an assumption that PSF is shift-invariant, tantamount to assuming that the source field consists of plane waves. However this assumption is invalid in most aeroacoustic applications, especially when the distance between the observation plane and the microphone array is not large compared with the extension of the region of interest.

An alternative strategy to reduce the computational run time of deconvolution is using compression computational grid that only contains the significant grid points and does not contain the redundant grid points. This strategy is based on the fact that computational run time of deconvolution decreases with the decrease of the number of computational grid. In previous work [17], DAMAS with wavelet compression computational grid (denoted by DAMAS-CG1) has reduced significantly computational run time of DAMAS in applications, particularly when sound sources are just located in a small extent compared with scanning plane and a band of angular frequency needs to be calculated. However DAMAS-CG1 has an inevitable deficiency that the occurrence probability of aliasing increasing slightly for complicated sound source.

In order to not only reduce the computational run time of DAMAS but also overcome the deficiency of DAMAS-CG1, in this paper we propose a novel algorithm that DAMAS with computational grid compressed by a new method (denoted by DAMAS-CG2). The rest of this paper is organized as follows. Conventional beamforming and DAMAS are introduced in Section 2. The new algorithm DAMAS-CG2 is illustrated in Section 3. Some application simulations are examined in Section 4. An experimental application for aeroacoustic noise is examined in Section 5. A discussion is presented in Section 6. Finally, conclusions are given in Section 7.

2. Conventional beamforming and DAMAS

Fig. 1 illustrates a setup with a planar microphone array that contains M microphones and has a diameter of D, as well as a 2-D region of interest. Stationary noise sources are located in a x-y plane at a distance of z_0 from the centre of the microphone array. The length of the scanning plane is $L=2z_0\tan(\alpha/2)$, where α is the opening angle. The source plane is divided into $S=N\times N$ equidistant points.

In conventional beamforming, cross-spectral matrix (CSM) for each test case data set is firstly calculated using simultaneously acquired data from the microphone array. The acquired data of each microphone are divided into I frames. Each frame is then converted into frequency bins by Fast Fourier Transform (FFT). For a given angular frequency ω, CSM is averaged over I blocks

$$\mathbf{C}(\omega) = \overline{\mathbf{p}(\omega)\mathbf{p}(\omega)^H} = \frac{1}{I}\sum_{i=1}^{I}\mathbf{p}_i(\omega)\mathbf{p}_i(\omega)^H \tag{1}$$

where $\mathbf{p}(\omega) = \left[p_1(\omega), p_2(\omega), \ldots, p_M(\omega)\right]^T$, $(\cdot)^H$ denotes complex conjugate transpose. For the sake of brevity, ω is omitted in the following. Notice that $\mathbf{C}\in\mathbb{C}^{M\times M}$. The conventional mean-square DAS beamforming output can then be written as

$$b(\mathbf{r}) = \frac{1}{M^2}\mathbf{e}(\mathbf{r})^H\mathbf{C}\mathbf{e}(\mathbf{r}) \tag{2}$$

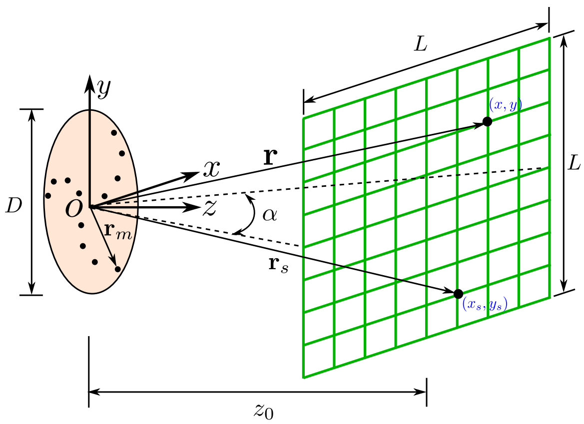

Fig. 1. Sketch of setup with a planar microphone array and a 2-D region of interest. Origin of the coordinate system is placed in the centre of the microphone array.