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Compression computational grid based on functional beamforming for acoustic source localization

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ABSTRACT

Phased microphone arrays have become a standard technique for acoustic source localization. Compared with beamforming algorithms such as the conventional beamforming, deconvolution approaches such as DAMAS successfully improve the spatial resolution. However deconvolution approaches usually require high computational effort compared to beamforming algorithms. Without optimizing deconvolution algorithm, recently DAMAS with compression computational grid based on the conventional beamforming (denoted by DAMAS-CG2) has reduced computational run time of DAMAS in applications (Ma and Liu, 2017). This paper proposes a novel algorithm that DAMAS with a novel compression computational grid based on an advance beamforming algorithm functional beamforming (denoted by DAMAS-CG3). This new algorithm takes advantages of functional beamforming to obtain large compression ratio. Simulated applications and experimental applications of benchmark test DLR1 show that DAMAS-CG3 is one order of magnitude faster than DAMAS-CG2 in most cases. In addition, the advantage of DAMAS-CG3 compared to DAMAS-CG2 is particularly more obvious with the threshold decreasing. However for some extreme situations that very complicated sources distribute to a larger extent relative to the scanning plane, the advantage of DAMAS-CG3 compared to DAMAS-CG2 may disappear. In order to get a large compression ratio in any application, the authors highly recommend compressing computational grid based on not only conventional beamforming but also functional beamforming, and then choosing the compression grid with larger compression ratio.

1. Introduction

In recent years with the development of society, the awareness of the impact of noise on health has increased significantly, environmental comfort has been becoming more and more important, and consequently acoustic source localization has been increasingly critical in noise diagnosis. Nowadays phased microphone arrays have become a standard technique for acoustic source localization. The conventional beamforming (CB) algorithm constructs a dirty map of source distributions from array microphone pressure signals [1]. CB is simple and robust, however its main disadvantages include poor spatial resolution particularly at low frequencies and poor dynamic range due to side-lobe effects [2].

Deconvolution algorithms reconstruct a clean map of source distributions from a dirty map via iterative deconvolution, and thus can significantly improve the spatial resolution. Many deconvolution algorithms that have developed in many fields of imaging, such as optical and radio astronomy or optical microscopy, have gradually applied in acoustic-array measurements. Dougherty and Stoker [3] first applied the CLEAN algorithm in acoustic array measurements. DAMAS Sijtsma [4] extended CLEAN to CLEAN-SC for coherent sources. Brooks and Humphreys [5,6] proposed DAMAS algorithm for sound source localization and extended it to three-dimensional acoustic image [7] and for coherent acoustic sources [8]. DAMAS makes researchers fully aware of the advantages and feasibility of deconvolution, and thus is considered as the breakthrough for deconvolution in aeroacoustic. Unfortunately, DAMAS requires high computational effort such as run time and computer memory compared with CB.

In order to reduce computational run time, Dougherty [9] proposed a more efficient deconvolution algorithm DAMAS2 by applying spectral procedure into DAMAS for the first time in the literature. This is under an assumption that PSF is shift-invariant, tantamount to assuming that the source field consists of plane waves. However the accuracy of DAMAS2 is strongly constrained by this assumption, owing to the fact that this assumption is not valid in most aeroacoustic applications, especially when the distance between the observation plane and the microphone array is not large compared to the extension of the region of interest. About this limitation some examples and more discussions

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can be found in Refs. [10,11]. A lot of researches have been devoted to applying more efficient deconvolution algorithms, such as generalized inverse beamforming [12,13], linear programming [14,15], compressive sensing algorithm [16–18], and FISTA [19,20]. Deconvolution algorithms still require a relatively high computational effort compared to CB due to the inevitable iterations used in the deconvolution algorithms. Additionally deconvolution algorithms introduce new problems by replacing continuous source distributions with misleading spots.

For algorithms with better performances, many researchers have drawn attentions from deconvolution algorithms back to beamforming algorithms and proposed some advance beamforming algorithms. Sarradj [21] proposed orthogonal beamforming, a more efficient beamforming with an improved spatial resolution, based on an eigenvalue decomposition of the cross spectral matrix. Huang et al. [22] successfully applied robust adaptive beamforming, an advance beamforming algorithm with improved spatial resolution and high efficiency, in aeroacoustic measurements by systematically choosing a small Tikhonov factor that give stable results. Dougherty [23] recently proposed functional beamforming (FB), a novel nonlinear beamforming algorithms, which could obtain dramatic dynamic range, improved spatial resolution, and identical speed compared to CB. Concerning spatial resolution, these advance beamforming algorithms have obvious superiority compared to CB, however they are not as good as deconvolution algorithms especially DAMAS.

Recently the authors [24,25] successfully improved the efficiency of DAMAS via compression computational grids that only contains significant grid points and does not contain redundant grid points, based on computational run time of deconvolution decreases with the decrease of the number of computational grid. Computational grid is compressed via wavelet compression based on CB in Ref. [24], while by discarding grid points with non-positive CB in Ref. [25]. The algorithms are denoted by DAMAS-CG1 and DAMAS-CG2 respectively. Although DAMAS-CG2 is usually less effective than DAMAS-CG1, DAMAS-CG2 is simpler and more practical than DAMAS-CG1, and overcomes the inevitable deficiency of DAMAS-CG1 that the occurrence probability of aliasing increases slightly for complicated sound source. In order to improve further the efficiency of DAMAS via compression computational grid, compression ratio should be as large as possible, and thus more effective compression method is needed to investigate.

This paper presents a novel algorithm that DAMAS with computational grid compressed based on FB (denoted by DAMAS-CG3). In terms of run time, this new algorithm takes full advantage of high dynamic range of FB to obtain a large compression ratio, and thus is faster than DAMAS-CG2. In terms of spatial resolution, this new algorithm has the same spatial resolution as DAMAS and thus is better than FB. The combination of the advantage of DAMAS in spatial resolution and the advantage of FB in dynamic range could provide a more powerful investigation method for acoustic source localization. The rest of this paper is organized as follows. Algorithms are presented in Section 2. Three simulated applications are examined in Section 3. Experimental application for benchmark test DLR1 is examined in Section 4. A discussion is presented in Section 5. Finally, conclusions are given in Section 6.

2. Algorithms

2.1. Conventional beamforming

Fig. 1 illustrates a setup with a planar microphone array that contains M microphones and has a diameter of D, as well as a two-dimensional region of interest. Stationary noise sources are located in an 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 region of interest is divided into $S = N \times N$ equidistant points.

The conventional delay-and-sum (DAS) beamforming

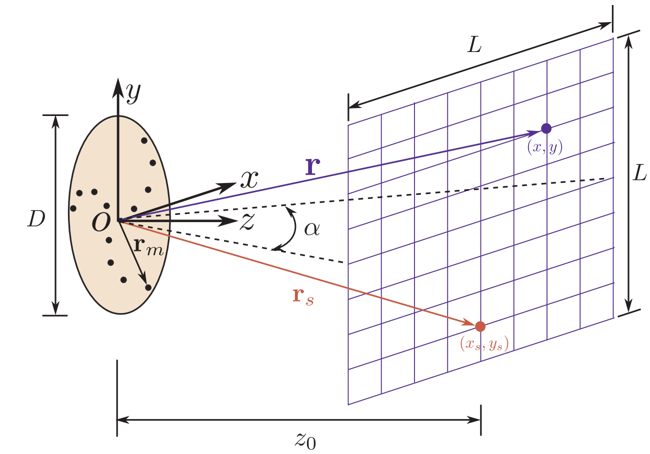

Fig. 1. Sketch of setup with a planar microphone array and a two-dimensional region of interest. Origin of the coordinate system is placed in the centre of the microphone array.

$$b(\mathbf{r}) = \frac{\mathbf{e}(\mathbf{r})^H\mathbf{C}\mathbf{e}(\mathbf{r})}{\|\mathbf{e}(\mathbf{r})\|^4} \tag{1}$$

where the superscript H denotes complex conjugate transpose of vector. The matrix $\mathbf{C} \in \mathbb{C}^{M\times M}$ is cross spectral matrix (CSM), and $\mathbf{C} = \overline{\mathbf{pp}^\mathbf{H}}$ where $\mathbf{p}$ is sound pressure contribution at microphones in frequency domain. The vector $\mathbf{e}(\mathbf{r}) \in \mathbb{C}^{M\times 1}$ is the steering vector at $\mathbf{r}$ and

$$\mathbf{e}(\mathbf{r}) = [e_1(\mathbf{r}),...,e_m(\mathbf{r}),...,e_m(\mathbf{r})]^T \tag{2}$$

In simulated applications discussed in Section 3, the notation of steering vector of Brooks and Humphreys [6] is used.

$$e_m(\mathbf{r}) = \frac{\|\mathbf{r}-\mathbf{r}_m\|}{\|\mathbf{r}\|}\exp\{-j2\pi f/c_0\|\mathbf{r}-\mathbf{r}_m\|\} \tag{3}$$

where $\|\mathbf{r}\|$ is the distance from the beamformer focus position to the centre of the array, $\|\mathbf{r}-\mathbf{r}_m\|$ is the distance from the beamformer focus position to the mth microphone (see in Fig. 1), and c_0 is speed of sound.

In experimental applications discussed in Section 4, the notation of steering vector of Sijtsma [26,27] under monopole point source assumption and in a medium with a uniform flow is used.

$$e_m(\mathbf{r}) = \frac{-\exp\{-2\pi j f\Delta t_e(\mathbf{r},\mathbf{r_m})\}}{4\pi\sqrt{[\overrightarrow{Ma}\cdot(\mathbf{r}-\mathbf{r_m})]^2 + \beta^2\|\mathbf{r}-\mathbf{r_m}\|^2}} \tag{4}$$

where $\overrightarrow{Ma}$ is a vector of Mach numbers and Δt_e is the emission time delay

$$\Delta t_e(\mathbf{r},\mathbf{r_m}) = \frac{1}{c_0\beta^2}(\overrightarrow{Ma}\cdot(\mathbf{r}-\mathbf{r_m}) + \sqrt{[\overrightarrow{Ma}\cdot(\mathbf{r}-\mathbf{r_m})]^2 + \beta^2\|\mathbf{r}-\mathbf{r_m}\|^2}) \tag{5}$$

and

$$\beta^2 = 1-\|\overrightarrow{Ma}\|^2 \tag{6}$$

2.2. DAMAS

The sound pressure contribution at microphones can be written as

$$\mathbf{p} = \sum_{s=1}^{S}\mathbf{e}(\mathbf{r_s})q_s \tag{7}$$

where q_s is source amplitude in terms of the pressure produced at source point s. For incoherent acoustic sources, CSM thus becomes

$$\mathbf{C} = \sum_{s=1}^{S}|q_s|^2\mathbf{e}(\mathbf{r_s})\mathbf{e}(\mathbf{r_s})^H \tag{8}$$

The conventional DAS beamforming output can then be written as