Arrangements of phased microphone arrays for acoustic source localization based on deconvolution algorithms

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Abstract. Nowadays phased microphone arrays have become a standard technique for acoustic source localization. Until now microphone arrangement is generally based on the conventional beamforming. Compared to the conventional beamforming, deconvolution algorithms such as DAMAS can achieve significantly improved spatial resolution. Recently the computing speeds of deconvolution algorithms have become faster and faster due to the developments of acceleration algorithms and computer ability. In this paper, phased microphone arrays based on deconvolution algorithms is preliminarily investigated in this paper. Three common microphone arrangements (circle array, grid array and spiral array) with same microphone channels are designed firstly. Subsequently the dynamic ranges with deconvolution algorithms are compared. At low frequency, these three arrays have nearly the same dynamic range. However at high frequency, circle array nearly has the same dynamic range with spiral array, while grid array has smallest dynamic range. Considering the experimental costs, circle array could replace spiral array as the first choice of microphone arrangement based on deconvolution algorithms.

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

Deconvolution algorithm reconstructs a clean map of source distributions from dirty map via iteratively deconvolution, and thus significantly improves the spatial resolution. Many deconvolution algorithms developed in fields of imaging, optical and radio astronomy and optical microscopy, have gradually applied in acoustic-array measurements. Brooks and Humphreys [5] proposed DAMAS for sound source localization, and subsequently extended it to three-dimensional acoustic imagine and for coherent acoustic sources. 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 a significant high computational effort such as run time and computer memory compared with conventional beamforming.

In order to reduce computational run time, Dougherty [6] proposed a more efficient deconvolution algorithm DAMAS2 by first applying spectral procedure into DAMAS, 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 with the extension of the region of interest. Recently Ma and Liu [7, 8, 9] successfully improved the efficiency of DAMAS via compression computational grids that only contains the significant grid points and does not contain the redundant grid points, based on computational run time of deconvolution decreases with the decrease of the number of computational grid.

Microphone array arrangement is a key factor that determine the performance of microphone array [10]. Until now microphone array arrangement is generally based on conventional beamforming [11, 12], due to conventional beamforming is widely used in the acoustic source localization society. As introduced above with acceleration algorithms, the speeds of deconvolution algorithms have been significantly improved. Coupled with the development of computer ability, deconvolution algorithms have become faster and faster. As a consequence, deconvolution algorithms may replace conventional beamforming as the most used algorithms in the near future. In this paper, phased microphone arrays based on deconvolution algorithms is preliminarily investigated.

2. Three microphone arrangements and their performances
Three microphone arrangements are designed firstly, as shown in Fig. 1. Each microphone arrangement has 64 channels and 1.00 m aperture. The first arrangement is a circular array with 64 microphone equally spaced around the perimeter. This arrangement is widely used to measure rotation noise due to the characteristics of rotation noise. The second arrangement is an 8-by-8 square grid array. The third arrangement is an Underbrink multi-arm spiral array, designed according to the instruction of Underbrink [13]. Although the first and the second arrangements are more convenient to build than the third arrangement, the third arrangement is the first choice in the current array design due to its superiority based on the conventional beamforming.

![Array arrangements](image)

*Figure 1. Array arrangements, 64 channels, 1.00 m aperture. (a) circle array; (b) grid array; (c) spiral array.*
Figure 2. f=2.0 kHz. White cross symbols, positions of synthetic point sources. First line, conventional beamforming; second line, DAMAS. First column, Array 1; second column, Array 2; third column, Array 3.

Figure 3. f=9.5 kHz. White cross symbols, positions of synthetic point sources. First line, conventional beamforming; second line, DAMAS. First column, Array 1; second column, Array 2; third column, Array 3.
The performances of these three arrays are shown by numerical applications in this section. In the geometrical setup, the observation plane is parallel to the array plane, and the region of interest is right in front of the array. The distance between array plane and observation plane is 5 m. The opening angle is 60°. The original computational grid is 30×30 with 900 grid points. No diagonal removal is applied on the PSF used for DAMAS. The DAMAS algorithm is run with 1000 iterations, which appeared to be more than enough for convergence. The number of iterations is 1000. The plotting scale is 30 dB. A single synthetic point source with maximum sound pressure is placed at the centre of the scanning plane. Other six synthetic point sources with -3 dB, -6 dB, -9 dB, -12 dB, -15 dB and -18 dB equally space around a perimeter with the diameter of 3.85 m. Conventional beamforming and DAMAS results are shown in Fig. 2 and Fig. 3, for frequency of 2.0 kHz and 9.5 kHz, respectively.

In Fig. 2 for conventional beamforming results in the first line, array 3 has the best performance (in Fig. 2c), while array 1 has worst spatial resolution (in Fig. 2a) and array 2 has obvious sidelobes (in Fig. 2b). For DAMAS results in the second line, all these three arrays constructs perfectly the source locations except the source with -18 dB. In Fig. 3 for conventional beamforming results in the first line, none of these arrays gives satisfactory result. For DAMAS results in the second line, array 1 and array 3 nearly have the same dynamic range and is better then array 2, although both array 1 and array 3 do not find the source with -18 dB. For keeping this paper more concise, the results of other deconvolution algorithms, such as NNLS and CLEAN, are not shown in this paper. Although dynamic range changes with deconvolution algorithms for each arrangement, the comparison conclusions of these three arrangements are consistent with the same deconvolution algorithm.

3. Conclusions
In this paper, phased microphone arrays based on deconvolution algorithms is initially investigated in this paper. For three common microphone arrangements (circle array, grid array and spiral array) with same microphone channels, at low frequency, they have nearly the same dynamic range. However at high frequency, circle array nearly has the same dynamic range with spiral array, while grid array has smallest dynamic range. Compared to spiral array, circle array is more convenient to build in the experiment, and could save experimental costs. And thus circle array could replace spiral array as the first microphone arrangement choice based on deconvolution algorithms.

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