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Functional delay and sum beamforming for three-dimensional acoustic source identification with solid spherical arrays

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ABSTRACT

Solid spherical arrays have become particularly attractive tools for doing acoustic sources identification in cabin environments. Spherical harmonics beamforming (SHB) is the popular conventional algorithm. Regrettably, its results suffer from severe sidelobe contaminations and the existing solutions are incapable of removing these contaminations both significantly and efficiently. This paper focuses on conquering these problems by creating a novel functional delay and sum (FDAS) algorithm. First and foremost, a new delay and sum (DAS) algorithm is established, and for which, the point spread function (PSF) is derived, the determination principle of the truncated upper limit of the spherical harmonics degree is explored, and the performance is examined as well as compared with that of SHB. Next, the FDAS algorithm is created by combining DAS and the functional beamforming (FB) approach initially suggested for planar arrays, and its merits are demonstrated. Additionally, performances of DAS and FDAS are probed into under the situation that the source is not at the focus point. Several interesting results have emerged: (1) the truncated upper limit of the spherical harmonics degree, capable of making DAS meet FB's requirement, exists and its minimum value depends only on the wave number and the array radius. (2) DAS can accurately locate and quantify the single source and the incoherent or coherent sources, and its comprehensive performance is not inferior to that of SHB. (3) For single source or incoherent sources, FDAS can not only accurately locate and quantify the source, but also significantly and efficiently attenuate sidelobes, effectively detect weak sources and acquire somewhat better spatial resolution. In contrast to that, for coherent sources, FDAS is not available. (4) DAS can invariably quantify the source accurately, irrespectively of the focus distance, whereas FDAS is burdened with a quantification deviation growing with the increase of the exponent parameter, when the focus distance is unequal to the distance from the source to the array center or the focus directions do not embrace the source direction. Fortunately, the deviation can be commendably compensated for by the introduced scale-and-integrate method. This study will be of great significance to the accurate and quick localization and quantification of acoustic sources in cabin environments.

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1. Introduction

Beamforming with microphone arrays has become fairly popular in the context of acoustic source identification [1–5]. One-dimensional linear, two-dimensional (2D) plane and three-dimensional (3D) open or solid sphere are all typical array geometries. Solid spherical arrays cannot only record panoramic sound scenes by virtue of their rotation symmetry but also acquire high signal-noise ratio due to their diffraction affects, which facilitates the omnidirectional and robust visualization of acoustic sources in 3D cabin environments. The solid spherical array also therefore becomes an attractive tool to identify sources for interior noise, like in a car or an airplane [6–8].

SHB [8–11] is a commonly utilized algorithm for solid spherical arrays. Theoretically, it outputs an excellent Dirac delta function based on the orthogonality of spherical harmonics. What appear in the non-source positions are all zero. Unfortunately, the reality is never so perfect. Owing to the truncation in spherical harmonics degree, as well as the finiteness and discreteness of microphones, plenty of high-level sidelobes occur, which can drown the real weak sources, and at the same time can be mistaken for sources, thus burdening the identification results with severe uncertainty. To attenuate these sidelobes, Filter-And-Sum [12] and deconvolution [13] have been particularly developed. The former was introduced by Hald in 2013. Its efficiency is high, but effect is not significant. The average attenuation is only about 5 dB, and the maximum sidelobe levels of some typical arrays are still higher than -10 dB at high frequencies. The latter was suggested by the authors of this paper in 2014. It possesses significant effect (more than 20 dB average attenuation), but suffers from heavy computational cost in practical applications. In a nutshell, neither can attenuate sidelobes both significantly and efficiently.

In 2014, based on the decline characteristic of the exponential function with a base less than 1, Dougherty [14,15] proposed a novel FB approach for the acoustic source identification with 2D planar arrays to attenuate its sidelobes, whose process is convenient and can be summarized as follows. First, an exponent parameter is introduced, next the array cross-spectral matrix (CSM) is raised to power of the reciprocal of this exponent in the functional sense, then the existing algorithm is applied to the modified CSM, and finally the resulting values are raised to power of the non-reciprocal exponent. Strikingly, the approach enjoys the dual advantages of amazing effect and wonderful efficiency [14–16]. Interestingly, it is expected that a clear and unambiguous map will be quickly achieved for acoustic sources in 3D cabin environments if the FB approach could be extended to suit for solid spherical arrays, which will be of great significance to the accurate and quick identification of sources. Intrinsically, FB requires the PSF, defined as the response of the existing algorithm to a monopole source with unity average pressure contribution, outputs 1 in the source position and values less than 1 in others [14,15]. Regrettably, the above SHB algorithm seriously fails to meet the requirement, which means that an alternative algorithm must be created. This paper is motivated to address this issue. Main highlights include: (1) a new DAS algorithm that meets FB's requirement is established for the first time, and for this algorithm, the determination principle of the truncated upper limit of the spherical harmonics degree is explored, as well as the performance is examined and compared with that of SHB. Two points are worth mentioning. The first is that the algorithm does not depend on the orthogonality of spherical harmonics and therefore is not affected by the orthogonality error of array, which is distinctly different in the established DAS from the existing SHB as well as some other algorithms like constrained optimization beamforming [17], Dolph–Chebyshev beamforming [18] and minimum variance distortionless response beamforming [19,20]. The second one is that even though DAS has been commonly used and widely studied for the planar or open spherical arrays [21–27], the literature concerning its application in solid spherical arrays is still rather sparse. Rafaely [28] and Roig et al. [29] presented some opinions or work. In [28] Rafaely only mentioned that DAS's realization is possible for solid spherical arrays under the spherical harmonics framework. In [29] Roig et al. constructed a DAS output for solid spherical arrays based on plane wave assumption and enhanced its maps at low frequencies using acoustic holography. Just like SHB, their DAS also does not meet FB's requirement. (2) The FB approach is extended to the 3D acoustic source identification with solid spherical arrays successfully. The established FDAS algorithm is capable of attenuating sidelobes from DAS both significantly and efficiently. (3) To provide guidance for practical applications, properties of DAS and FDAS are explored under the situation that the focus distance is unequal to the distance from the source to the array center and the focus directions do not embrace the source direction.

The remainder of this paper is organized as follows. In Section 2, the forward theory of acoustic signal propagation is presented. Thereafter, in Section 3, the backward DAS acoustic source identification algorithm is established and studied thoroughly: first, the output function is defined, then the determination principle of the truncated upper limit of the spherical harmonics degree is explored with simulations of the derived PSF, and finally the simulation conclusions are validated as well as the performance of the algorithm is examined and compared with that of SHB with experiments. Subsequently, in Section 4, the FDAS algorithm is established and studied with both simulations and experiments. Finally, in Section 5, conclusions and perspectives are summarized.

2. Forward theory of acoustic signal propagation

Fig. 1 depicts the coordinate system of the beamforming with a solid spherical array, where the origin is located at the array center. An arbitrary position in 3D space can be described by (r, Ω), where r is the distance from the position to the origin and $\Omega = (\theta, \varphi)$ indicates the direction with θ and φ being the elevation and azimuth angles respectively. The symbols "•" represent array microphones embedded in the solid sphere, which are employed to pick up sound signals. Assuming a is