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Theoretical and experimental comparative analysis of beamforming methods for loudspeaker arrays under given performance constraints ☆, ☆☆

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ABSTRACT

Methods for beamforming are available that provide the signals used to drive an array of sources for the implementation of systems for the so-called personal audio. In this work, performance of the delay-and-sum (DAS) method and of three widely used methods for optimal beamforming are compared by means of computer simulations and experiments in an anechoic environment using a linear array of sources with given constraints on quality of the reproduced field at the listener's position and limit to input energy to the array. Using the DAS method as a benchmark for performance, the frequency domain responses of the loudspeaker filters can be characterized in three regions. In the first region, at low frequencies, input signals designed with the optimal methods are identical and provide higher directivity performance than that of the DAS. In the second region, performance of the optimal methods are similar to the DAS method. The third region starts above the limit due to spatial aliasing. A method is presented to estimate the boundaries of these regions.

1. Introduction

Purposefully designed input signals can be used to control the directivity pattern of an array of loudspeakers of compact form factor for the implementation of personal audio systems [2,3]. This is usually achieved by focusing sound in a given region of the space, hereafter also referred to as acoustically bright area, while minimizing sound radiation elsewhere. The acoustically bright area may consist of a number of so-called control points [3] or a single control point [25]. In the first case, we refer to it as the process of generating private sound zones, whilst in the second case, which is the one considered in this work, we aim at the generation of private sound beams, hereafter also succinctly referred to as sound beams.

Input signals that allow for the creation of private sound beams can be designed using methods for beamforming for loudspeaker arrays, such as the delay-and-sum (DAS) method or through the application of Time-Reversal signal processing [4], to cite a few. From 1990 onwards, research has focused on increasing directivity performance of loudspeaker arrays by

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means of optimal beamforming methods which can be broadly divided into three categories. (1) Methods that aim at the reproduction (or synthesis) of a target field, such as the Pressure-Matching (PM) method [5]. (2) Methods for the control of the acoustic energy density in the control zone, hereafter also referred to as energy based methods, such as the Acoustic Contrast Maximization [3,6] (ACM), and the Energy Difference Maximization [7,8] (EDM) methods. (3) Hybrid methods that combine the synthesis of a target field and the control of the acoustic energy density [9–11]. In general, for a given configuration of loudspeaker array and control area, input signals designed with a given method for beamforming may depend on a number of tunable parameters, such as the regularization parameters for the PM and ACM methods. Due to these differences, a comparison between strategies for beamforming is not straightforward. Methods for beamforming for loudspeaker arrays have been previously compared for various geometrical arrangements and applications, such as in personalized audio systems for vehicles [12], generation of warning signals for the safety of pedestrians [13], and for loudspeaker arrays operating in a reflective environments [14] or under free-field conditions [3,15–17]. Trade-offs between concurring performance factors, such as directivity performance [3,15,17,28], input energy required to control sound field [3,17,28], robustness of the system against mismatch of the responses between sources and control points [3,17], and frequency response of the reproduced field [12,15] have been reported in the literature as well as the fact that system performance is highly dependent on the choice of tunable parameters of a given beamforming method [8,15–17].

The increasing demand for high-performance portable devices for communications, such as tablet or mobile phones, suggests that systems for personalized audio should be able to provide listeners with a suitable sound pressure level over a broad frequency range (hereafter succinctly referred to as audio quality) and limited energy consumption. These two concurring performance constraints form the basis of the comparison between beamforming methods proposed in this work. More specifically, input signals generated under the above-mentioned performance constraints with four widely used beamforming methods for loudspeaker arrays, i.e., the delay-and-sum (DAS) beamformer, the PM, the ACM, and the EDM methods, are compared using a linear array prototype by means of numerical simulations and experiments in an anechoic environment. A strategy is presented to calculate the parameters of each method (such as the regularization parameter) in order to ensure that input signals designed with the methods considered satisfy the same constraints in terms of audio quality and required input energy at each frequency. The main result of this work is that, based on the given constraints, performance of the optimal methods considered relate to each other and to that of the DAS as a function of frequency. More specifically, they are identical at low frequencies and they resemble that of the DAS in the mid-frequency range. A method is presented to estimate the frequencies at which the boundaries of these regions occur. Moreover, the results of the experiments with the real loudspeaker array show the effects of the mismatches between transducers of the array on system robustness, the latter being discussed and analyzed by using the measured transfer functions of the array in the anechoic chamber. The remainder of the paper is organized as follows. The theoretical preliminaries are introduced in Section 2. A review of the beamforming methods considered in this work, their theoretical model and their differences are reported in Section 3. The strategy for the beamforming method comparison is described in Section 4. The details of the linear array, the analysis of the designed filters, the numerical and experimental validations are described in Section 5. The conclusions are drawn in Section 6.

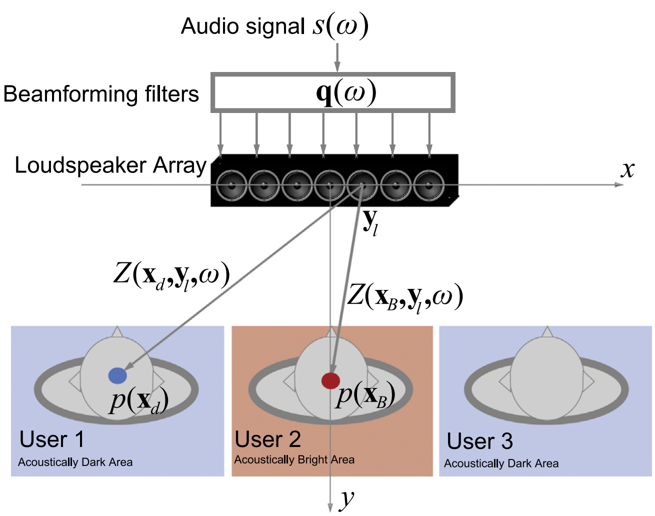

Fig. 1. Example of a beamforming problem for personalized audio applications. A loudspeaker array is driven by purposefully designed input signals to focus sound where User 2 is located while minimizing sound pressure where User 1 and User 3 are located.