A Virtual Endfire Loudspeaker Array for the Generation of Sound Beams

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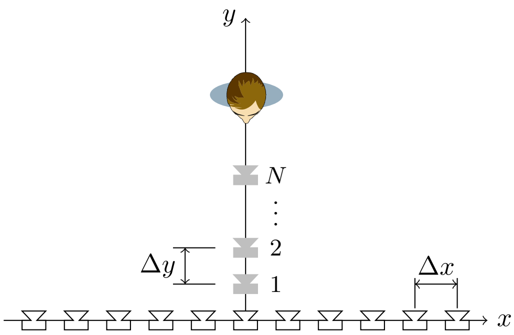

Figure 1: Geometry of the (virtual) endfire array.

Introduction

Sound beams are, similar as light beams, confined linear regions in space were the sound pressure level is significantly higher than besides the beam. The synthesis of sound beams by loudspeaker arrays has been considered in the literature for applications like personal audio, binaural synthesis and listening room equalization. Representatives of such approaches are for instance Acoustic Contrast Control [1], modal techniques [2, 3] and endfire loudspeaker arrays [4, 5]. In this paper we will consider the use of virtual endfire arrays for the generation of sound beams synthesized by a broadside loudspeaker array.

Beamforming with Endfire Loudspeaker Arrays

In contrast to linear loudspeaker arrays used in broadside operation, endfire arrays consider that the main direction of interest lies on the axis the loudspeakers are arranged on. Figure 1 illustrates the geometry considered in this paper. The gray loudspeaker symbols denote N acoustic sources placed in endfire configuration on the y-axis. The spacing between the individual sources is denoted by Δy. In sound field synthesis theses sources are termed as secondary sources. Under the idealized assumption that the secondary sources radiate like an acoustic monopole, the synthesized sound field is given as

$$P(\mathbf{x},\omega) = \sum_{n=1}^{N} D(n,\omega)\, \frac{e^{-j\frac{\omega}{c}\|\mathbf{x}-\mathbf{x}_n\|}}{\|\mathbf{x}-\mathbf{x}_n\|}\ , \qquad (1)$$

where c denotes the speed of sound, $\mathbf{x} = [x\ y\ z]^T$ a field point, $\mathbf{x}_n = [0\ n\Delta y\ 0]^T$ the position of the n-th secondary source and $D(n,\omega)$ its frequency dependent weight (driving function). In the simplest case, delay-and-sum beamforming, $D(n,\omega) = e^{-j\omega\tau_n}$ would only realize delays $\tau_n = \frac{n\Delta y}{c}$ which account for the propagation delays between the secondary sources. This way, by constructive interference of the individual sound fields of the secondary sources, the sound pressure is maximized in endfire direction. However, delay-and-sum techniques result in a limited directivity for low frequencies. In order to overcome this issue, superdirective techniques known from microphone array processing have been applied to endfire loudspeaker arrays [4, 5]. Here differential mechanisms are used for improved directivity at low frequencies.

As one representative of such techniques we consider the Minimum Variance Distortionless Response (MVDR) design [6]. Here the driving signal is given by

$$\mathbf{D}^T(\omega) = \frac{\mathbf{W}^H(\omega)\left(\mathbf{S}(\omega)+\beta(\omega)\mathbf{I}\right)^{-1}}{\mathbf{W}^H(\omega)\left(\mathbf{S}(\omega)+\beta(\omega)\mathbf{I}\right)^{-1}\mathbf{W}(\omega)}, \qquad (2)$$

where

$$\mathbf{D} = [D(1,\omega), D(2,\omega), \dots, D(N,\omega)]^T, \qquad (3)$$

$$\mathbf{W} = [e^{j\frac{\omega}{c}\Delta y}, e^{j\frac{\omega}{c}2\Delta y}, \dots, e^{j\frac{\omega}{c}N\Delta y}]^T. \qquad (4)$$

The $N \times N$ identity matrix is denoted by $\mathbf{I}$ and β denotes a regularization parameter. For monopoles as secondary sources, the elements $S_{nm}(\omega)$ of the $N \times N$ covariance matrix $\mathbf{S}(\omega)$ are given by

$$S_{nm}(\omega) = \frac{\sin(\frac{\omega}{c}(y_m - y_n))}{\frac{\omega}{c}(y_m - y_n)}. \qquad (5)$$

Figure 2 shows the level of the sound field synthesized by a superdirective endfire loudspeaker array consisting of $N = 8$ monopole sources placed on the y-axis at a distance of $\Delta y = 10\,\mathrm{cm}$ when synthesizing a monofrequent source with $f = 1000\,\mathrm{Hz}$. Further simulations not shown here revealed a similar directivity for lower frequencies. Due to the MVDR design, the directivity is maintained constant by differential mechanisms for low frequencies. Above approximately $f_h = \frac{c}{2\Delta y}$, spatial aliasing is becoming prominent in the synthesized sound field limiting the achievable directivity.

This illustrative example shows that endfire arrays together with superdirective beamforming techniques allow to synthesize sound fields which are confined to a narrow beam in direction of the array. The practical performance of endfire beamforming techniques is limited by the differences (mismatch) between the individual secondary sources [4, 5]. Additionally, the direction of the

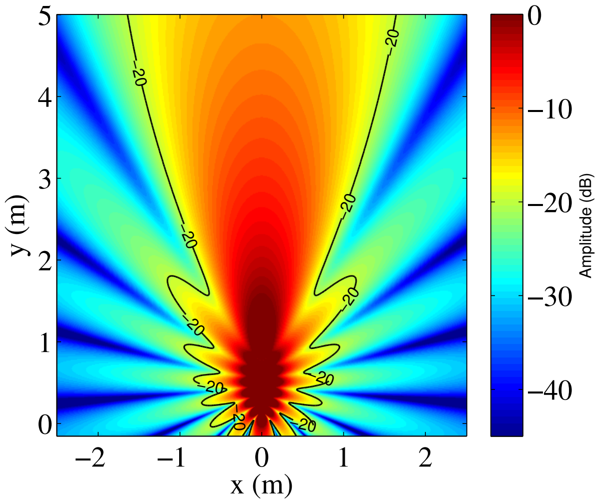

Figure 2: Level of monofrequent ($f = 1000\,\mathrm{Hz}$) sound field synthesized by a superdirective endfire beamformer using point sources ($\Delta y = 10\,\mathrm{cm}$, $N = 8$, $\beta = 10^{-6}$).

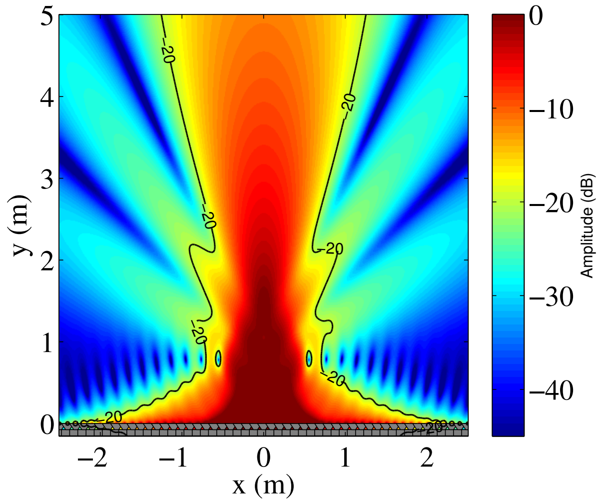

Figure 4: Level of monofrequent ($f = 1000\,\mathrm{Hz}$) sound field synthesized by a virtual superdirective endfire beamformer using focused sources ($\Delta y = 2.5\,\mathrm{cm}$, $N = 32$, $\Delta x = 10\,\mathrm{cm}$ $\beta = 10^{-6}$).

sound beam can only be steered to a very limited extend. In order to cope for off-axis or moving listeners, it is typically suggested to mechanically rotate the array. In order to overcome the limitations of endfire loudspeaker arrays the concept of a virtual endfire loudspeaker array is proposed in this paper.

A Virtual Endfire Loudspeaker Array

Sound Field Synthesis techniques, like Wave Field Synthesis (WFS) [7], higher-order Ambisonics (HOA) [8] or the Spectral Division Method (SDM) [9], allow synthesizing focused sources in front of broadside loudspeaker arrays. We stick to 2.5-dimensional WFS in the remainder of this paper for a first proof of concept. For the synthesis of a focused source, a converging wave front is synthesized focusing at the focus point which diverts like a point source behind. The basic properties of focused sources as synthesized by WFS have been reviewed e.g. in [10]. An interesting property of focused sources is, that compared to non-focused virtual sources, the synthesis accuracy is significantly higher in the vicinity of the focus point for higher frequencies. The amplitude decay is in between a line and point source. Hence, as first approximation focused sources can be used as virtual secondary sources.

This contribution investigates the usage of focused sources as virtual secondary sources in endfire configuration. This is realized by a two stage approach: (i) N focused sources are synthesized in endfire configuration in front of the loudspeaker array (see Fig. 1) which are (ii) driven by the driving function (2). Figure 3 shows a block diagram of the proposed approach. A number of numerical simulations have been conducted as a first proof of concept. For this purpose N focused sources (virtual secondary sources) have been synthesized by a linear loudspeaker array located symmetrically on the x-axis. The spacing between the secondary sources was chosen as $\Delta x = 10\,\mathrm{cm}$ for the sake of comparison. The total length of the array was chosen to $L = 10\,\mathrm{m}$ in order to avoid truncation artifacts.

Figure 4 shows the level of the sound field synthesized by a virtual endfire loudspeaker array consisting of $N = 32$ monopole sources placed on the y-axis at a distance of $\Delta x = 2.5\,\mathrm{cm}$ when synthesizing a monofrequent source with $f = 1000\,\mathrm{Hz}$. The parameters have been chosen in accordance with the properties of focused sources. Although the distance between the virtual secondary sources has been chosen smaller than in Figure 2, the overall length of the virtual endfire array is equal. Comparison of Figures 4 and 2 reveals that a narrower sound beam is generated by the virtual array. This is due to the smaller distance Δy of the virtual endfire sources which could not be realized by real loudspeakers in endfire configuration.

We keep the design of the superdirective beamformer fixed for a sound beam in direction of the endfire array. Steering of the beam can be realized by tilting the axis on which the virtual secondary sources are located on. No redesign of the superdirective beamformer is required. Figure 5 shows the synthesized field when the axis of the virtual secondary sources is tilted 30 degrees to the right. The synthesized sound beam is also tilted as a consequence. This example shows that the proposed approach allows the steer the beam effectively by signal processing means.

Conclusions

The proposed technique for the synthesis of sound beams in front of a loudspeaker array bases on the concept of synthesizing virtual secondary sources in endfire configuration, which are driven by superdirective beamforming techniques. This two stage approach features a fixed superdirective design for the endfire array without considering steering of the beam. Steering is realized by tilting

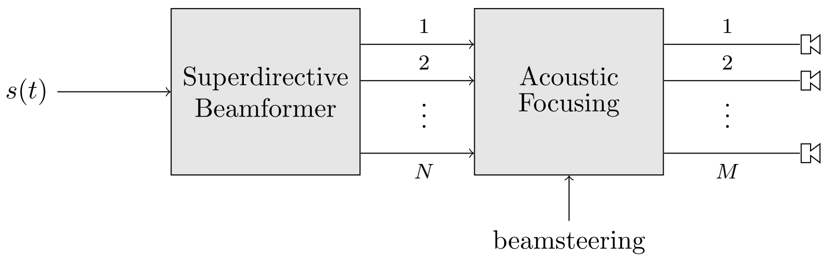

Figure 3: Block diagram of the two-stage approach to the generation of sound beams by a virtual endfire loudspeaker array.

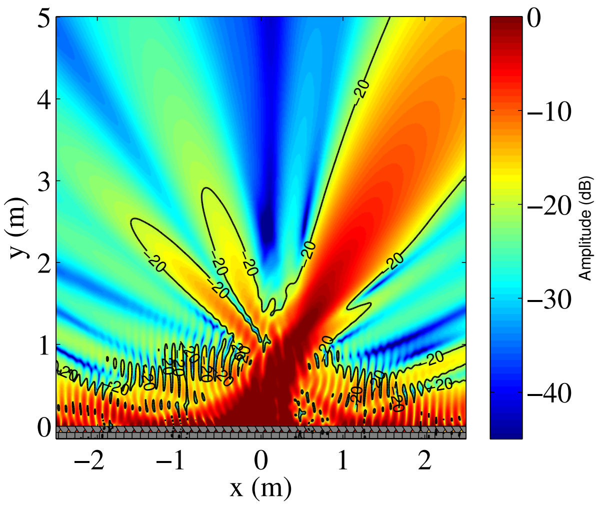

Figure 5: Level of monofrequent ($f = 2000\,\mathrm{Hz}$) sound field synthesized by a virtual superdirective endfire beamformer using focused sources ($\Delta y = 2.5\,\mathrm{cm}$, $N = 32$, $\Delta x = 10\,\mathrm{cm}$, $\beta = 10^{-6}$).

the axis the virtual secondary sources are located on. A major benefit of the proposed approach is its flexibility with respect to the placement of the virtual endfire sources. The sources can be placed closer together as this would be possible with real loudspeakers. The distance can even be chosen to be frequency dependent in order to optimize the broadband performance. The approach is furthermore not limited to linear loudspeaker arrays. Rectangular or circular loudspeaker arrays can be used as well. This paper presents a first proof of the concept. The simulation results show its feasibility. Future research should investigate on the robustness of the approach to cope with variations between individual loudspeakers. Compared to microphone arrays, where the superdirective beamforming techniques have initially been developed for, individual loudspeakers vary to much larger degree than microphones. As discussed briefly above, the properties of focused sources differ in comparison to ideal point sources. It is therefore of interest to incorporate the properties of focused sources into the design of the beamformer.

The numerical simulations in this paper are based on the Sound Field Synthesis Toolbox [11], a MATLAB/Octave implementation of various sound field synthesis techniques. The scripts used to compute the results, as well as the slides from the talk are available at `http://spatialaudio.net/virtual-endfire-array/`.

Acknowledgments

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