High-Directional Beamforming with a Miniature Loudspeaker Array

Christoph Sladeczek, Daniel Beer, Jakob Bergner, Albert Zhykhar, Maximilian Wolf, Andreas Franck
Fraunhofer Institut for Digital Media Technology IDMT, 98693 Ilmenau, Deutschland
Email: christoph.sladeczek@idmt.fraunhofer.de

Introduction

Practical electroacoustic applications often demand a directional behavior of the dedicated loudspeaker setup. A common goal in sound reproduction applications is a uniform distribution of sound pressure over the listening area. A desired application in sound reinforcement is to focus the emitted sound to a defined region, to locally enhance speech intelligibility in reverberant environments or to keep high sound pressure levels on certain areas. The use of loudspeaker arrays with dedicated signal processing is a well-known technique to create directional radiation characteristics. This technology, often denoted as loudspeaker beamforming, e.g., [6, 8], allows controlling the radiation of sound to a distinct direction. For decades directional sound radiation is used in public address systems. In recent years similar requests came up for other markets like home entertainment. Due to the size of conventional speaker systems, their usage in this field is still not feasible. This paper presents the design of a miniature beamforming array. The presented concept shows an array that consists of a small number of loudspeakers, while providing a high directional radiation behavior.

Electroacoustic System

For multiple industrial applications the usage of an loudspeaker array is often limited due to its large dimensions. For such applications the following requirements are defined:

- Size: 40 cm x 10 cm x 5 cm
- Number of channels: ≤ 8
- Frequency bandwidth: speech band
- Channel separation: ≥ 20 dB (beam to no-beam)

A small possible box volume V_b requires a driver with a high force factor Bl, a low driver resonance frequency f_s and a small equivalent air volume V_as. For further information see [3] or [4]. In order to build an array that fulfills these criteria a broadband speaker, available on the market was used. The frequency response of this micro–speaker, who has a diameter of 3.6 cm is depicted in Figure 1. This speaker is characterized by a sensitivity of ≈ 77 dB/1 W/1 m, a resonance frequency of ≈ 180 Hz and 8 % THD at 100 Hz/1 W. The array that was build out of these speakers is shown in Figure 2. With a force factor of $Bl = 3.17$ N/A, a total Q-factor $Q_\mathrm{ts} = 0.25$ and an equivalent air volume $V_\mathrm{as} = 0.07$ l a box volume $V_\mathrm{b} \approx 20$ ml can be achieved for a critical damped tuning with $Q_\mathrm{tc} = 0.5$. This results in a sealed enclosered

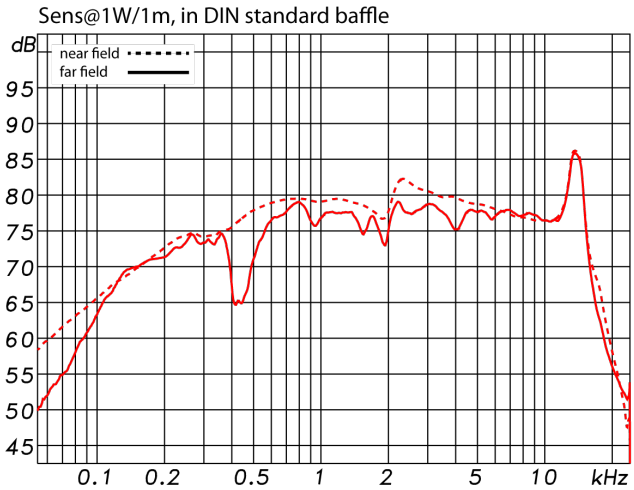


Figure 1: Measured frequency response in the near- and far-field of the micro-speaker mounted in a DIN standard baffle.

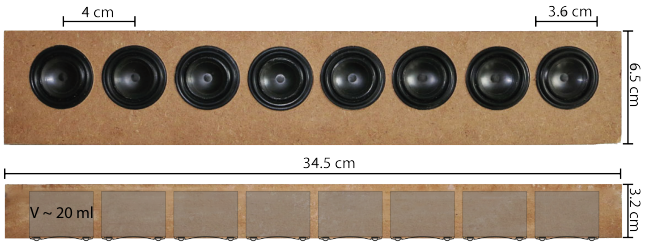


Figure 2: Loudspeaker array consisting of 8 micro-speaker used for high-directional beamforming.

resonance frequency $f_\mathrm{s} \approx 290$ Hz. An additional preprocessing enables a flat frequency response over a broad frequency area.

Beamforming Approach

Generally, beamforming techniques approximate a desired directivity pattern by applying individual amplitude and/or phase changes to the transducer signals. Because the presented loudspeaker beamforming technique targets wideband audio applications, these changes are frequency-dependent, Consequently, nonlinear-phase FIR filters are used to generate the loudspeaker signals. While these filters are static for a given beampattern, variable directivities can be achieved by switching between different filter sets. Likewise, multiple signals with different radiation patterns can be reproduced simultaneously by superimposing multiple driving signals. The resulting filter set can then consequently be used with regular audio DSPs or other suitable audio systems.

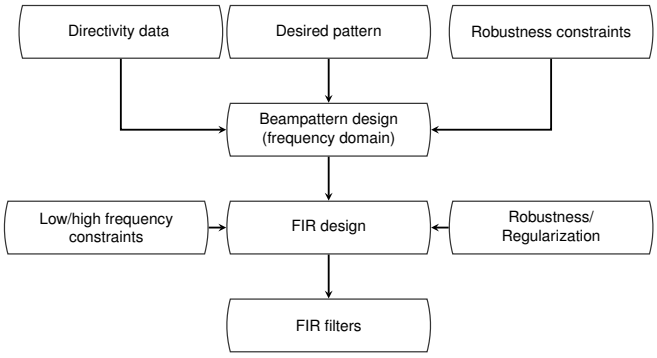


Figure 3: Flowchart filter design.

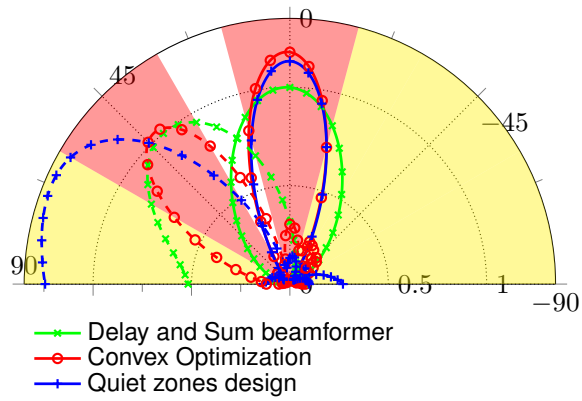


Figure 4: Beampattern design.

The FIR filters are designed in a two-stage process, with both stages based on convex optimization techniques. The general design process is depicted in Figure 3. In the first stage, complex-valued loudspeaker gains are computed for dense grid of discrete frequencies. These complex gains form the specification of the desired frequency responses of the FIR filters. In the second stage, the coefficients of each FIR filters are optimized separately to optimally approximate the corresponding desired frequency response. The main motivation for this two-stage process is complexity, because it enables the design of relatively long FIR filters with good approximation capabilities, while existing one-stage techniques are typically limited to very low filter orders, e.g., [7].

The frequency-domain beampattern design of the first stage determines a set complex-valued loudspeaker weights such that the resulting pattern optimally approximates a desired characteristic with respect to a prescribed error norm. This approach is referred to as pressure matching, e.g., [8]. Here it is solved as a convex optimization problem

$$\begin{aligned} \underset{\mathbf{h}}{\operatorname{argmin}} \quad & \|\mathbf{P}_{\text{des}} - \mathbf{D} \cdot \mathbf{h}\|_p \\ \text{subject to} \quad & (\text{optional constraints}) \,, \end{aligned} \tag{1}$$

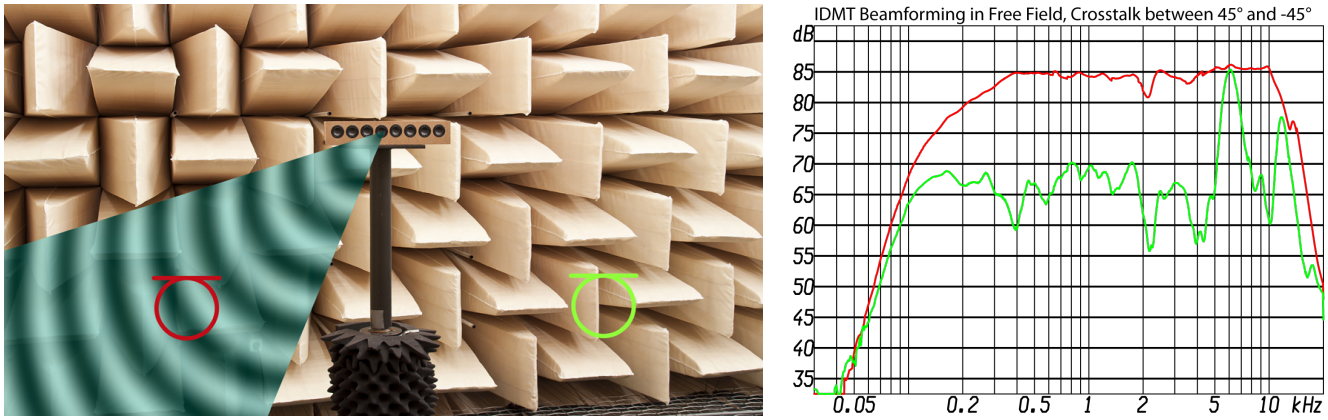
where $\mathbf{h}$ is the vector of complex driving gains for all transducers, P_{des} the desired sound pressure at a number of control points, and $\mathbf{D}$ is the transfer function matrix from the loudspeakers to the control points. Common choices for the error norm p are $p = 2$ (least-squares) and $p = \infty$ (Chebyshev or minimax optimization). While P_{des} could be set to arbitrary patterns, it is advisable to use only specifications that are within the physical limits set by the loudspeakers and the array geometry. Figure 4 shows a typical pattern specification consisting of the desired beam directions or bright zones (red), stopbands or dark zones (yellow), where the radiation is to be minimized, and "don't care" or transition bands, where the radiation is not restricted. As the directivity of the individual loudspeakers has a significant impact on the array response, it can be incorporated in the optimization process by forming the transfer function matrix $\mathbf{D}$ from measured directivity data. In particular, it has been observed that the resulting beampatterns improve if the position and the layout of the transducer in the array (baffled/not baffled) are taken into account. Robustness of the array response with respect to transducer positioning and directivity imperfections, the effects of the listening room, and amplifier noise is another important aspect in this design stage. This is typically done by adding a regularization term to (1), e.g., [8]. In the proposed approach, a norm constraint on the gain vector $\mathbf{h}$ is used instead, thus gaining better control and avoiding an iterative design.

In the design second stage, the filter coefficients are computed separately for each FIR filters by minimizing the approximation error to the complex frequency response determined by the first design stage. It is based on optimization-based filter design techniques for arbitrary-phase FIR filters, e.g., [5]. However, domain-specific knowledge, such as control of the filter implementation delay, smoothing of the desired responses, the handling of very low and high frequencies, and robustness constraints, is necessary to create wellformed and robust FIR filters. This is achieved either by adjusting the design specification, or by adding additional constraints to the filter design optimization problem.

Beamforming Analysis

In order to analyze the performance of the miniature beamforming array an acoustic measurement was conducted. For this the array was installed in an anechoic chamber according to DIN EN ISO 3745:12[2]. The array was configured to radiate an audiobeam in 45°where a single microphone was placed. The second microphone was placed outside the audiobeam at -45°direction to the array. The setup is depicted in Figure 5(a). Using a logarithmic sweep measurement method the frequency responses on both positions was acquired. Figure 5(b) shows these results.

To support the beamforming design process and to cope with large numbers of model parameters and big amounts of measurement data a special GUI-Tool was developed. Figure 6 shows the main window providing different data representation views: directivity diagram for a particular frequency, frequency response plot for a chosen direction, and isobar plot. The tool allows to define and load mul-



(a) (b)

Figure 5: Setup (a) and measurement results (b) of the beamforming approch using the miniature loudspeaker array under free-field like conditions. The red line shows the frequency response measured by the microphone placed in beam direction while the green line shows the frequency response outside of beam direction.

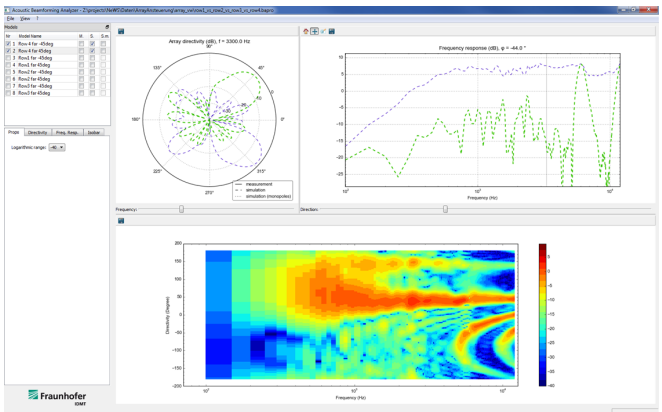


Figure 6: Screenshot of the beamforming analyzer user interface that was developed for pre-simulation of loudspeaker array behavior.

tiple beamforming models, simulate loudspeaker array beamforming using the designed FIR filters, as well as to select and visualize several models simultaneously to effectively analyze and compare them.

Beamforming Performance

The performance evaluation of the proposed beamforming approach was conducted with physical measurements under anechoic conditions. Figure 5(b) shows the frequency response in the desired beam direction (bright zone) as well as in the stop directions (dark zone).

It can be seen that in beam direction between 400 Hz and 10000 Hz a sufficiently flat frequency response of ±2 dB (with an exceptional dip at around 2100 Hz) can be obtained. The amplitude response in the dark zone can be supressed by at least 15 dB relatively to the bright zone in the range between 300 Hz and 5000 Hz. At around 6000 Hz a prominent side lobe due to the loudspeaker spacing gets apparent with almost equal amplitude.

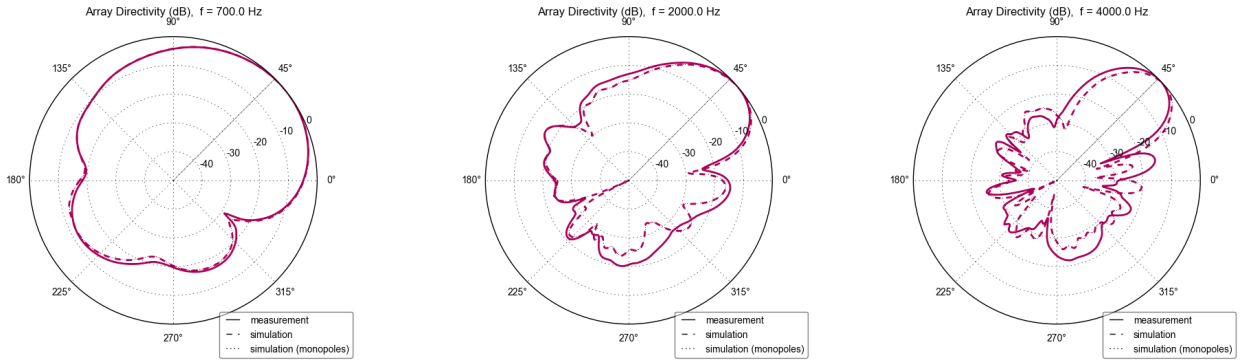
Beside the pure frequency response of two separated directions the frequency-dependent polar pattern holds

Figure 8: Loudspeaker array installed in an ITU–R BS.1116 room, for validation of local loudness approach under "standard" living room conditions.

necessary information about the directional behavior. Figure 7 shows the directivity pattern of the array for selected frequencies. It is apparent that the directivity gets stronger with increasing frequency. Due to the array's relatively small overall dimension the beamforming effect gets becomes relevant at frequencies above 1000 Hz.

Application: Local Loudness Control

In several listening situations hearing impaired people need to listen to audio content that is presented to normal hearing people. One example is watching TV at home. If the electro-acoustic reproduction system does not have a possibility to present audio in a perceptual-adapted way, often the hearing impaired listener would like to turn up the volume. Depending on the level this could bother the normal hearing listener who is feeling disturbed by a too high volume. In such a situation, a device that is able to increase the volume only on a small listening spot could be very helpful. In order to evaluate the capabilities of the miniature beamforming-array for such an application the measurement setup depicted in Figure 8 was used. The beamforming array was installed on top of a TV that was placed in a special acoustical laboratory that was build according to recommendation ITU–



(a) (b) (c)

Figure 7: Polar pattern of beamforming array for frequencies 700 Hz (a), 2000 Hz (b) and 4000 Hz (c).

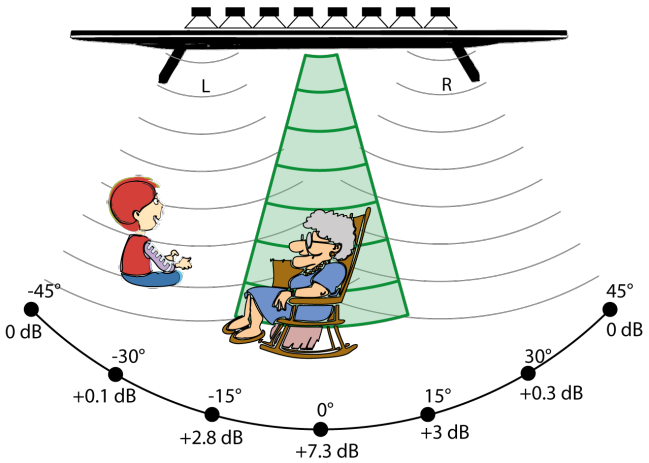


Figure 9: Measurement results of the local loudness experiment. The SPL level of sound reproduction is increased up to 7 dB in the region ±15°while the SPL increase is negligible for all other positions.

R BS.1116-1 to simulate the room acoustic properties of a "standard" living room [1]. The two outer speakers of the array were used as stereo speakers. On top, all speakers of the array were used to reproduce an audio beam perpendicular the loudspeaker array (0°). On a circular path with a radius of 2 m 7 omnidirectional microphones of type MK360 by Microtech Gefell were installed with a spacing of 15°. White noise was used as the measurement signal. Figure 9 shows the results of the measurement. For the stereo reproduction a level of 72.1 dB SPL was used. The levels depicted in Figure 9 represent relative dB values. It can be seen, that the sound pressure level can be increased up to 7 dB at the 0°measurement position while the SPL increase is negligible for all positions outside of ± 15°.

Conclusion

In this paper a loudspeaker array system for high-directional radiation of sound was presented. To build the loudspeaker array a small broadband speaker was used. To control the array which consisted of 8 loudspeakers a beamforming approach based on the concept of pressure matching was developed. The algorithm allows to control the array in different ways, depending on the target use–case. To validate the approach, the array system was evaluated using acoustic measurements under free-field like conditions. Furthermore, the paper presented a practical application where the array is used to increase the reproduction volume only within a defined area.

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