PROCEEDINGS of the
23rd International Congress on Acoustics

9 to 13 September 2019 in Aachen, Germany

A field-matching method for sound field synthesis for large scale sound reinforcement systems

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ABSTRACT

There have been ongoing attempts to adapt existing Sound Field Synthesis methods for large scale sound reinforcement systems. In comparison to smaller indoor setups large scale systems have significant distances between secondary sources and often irregular loudspeaker layouts due to rigging conditions. These restrictions make the use of WFS-related methods unfeasible. In addition, prominent and frequency-dependent directivity patterns of concerts loudspeakers obstruct the use of usual level and delay algorithm as it assumes omnidirectional secondary sources.

The paper proposes an alternative method based on a Field Matching technique that bypasses the above-mentioned restrictions. Despite the typically low aliasing frequency of a concert sound system layout the Field Matching approach was shown to provide an accurate localisation and a stable virtual source across the audience area including front seats. The paper presents a theoretical background and a Matlab simulation along with listening tests results. Limitation and challenges of the method for live sound applications are discussed.

Keywords: Sound reinforcement, field-matching, sound field synthesis.

1. INTRODUCTION

There are several popular methods to reproduce a sound field of a combination of virtual sources through an array of loudspeakers: WaveFieldSynthesis (1,2), Higher Order Ambisonics (HOA) (3), Vector Based Amplitude Panning (3) to name just a few. While being well established, each of them poses restrictions on the array geometry, loudspeakers characteristics or the audience area. Loudspeakers are often assumed to be monopoles or dipoles and have to be located right next to each other. This works well for small rooms but not so well for larger areas. Systems like HOA only work for a "sweet spot" and not for an audience area.

Modifications of WFS have been developed to account for loudspeaker directivity, however, loudspeakers are required to be identical and located on a straight line or in a circle (4,5).

The main objective of this work is to find a method that doesn't have any basic limitations on the loudspeakers layout and directivity.

The goal is to formulate a matrix equation for the driving functions and check if the driving functions provide the desired result. The practicalities of numerically solving the equation are left for the future work.

The approach described below presents a version of the field matching method described (6).

2. PROBLEM LAYOUT

Ideally, a concert or theatre sound system designed for spatial sound reproduction should be able to provide stable virtual source localization in the entire audience area. The loudspeakers or loudspeaker arrays are mainly located above the stage, along the sides and sometimes at the back. There can be a separate row of smaller loudspeakers to cover the front row (so called "front fills") and additional loudspeakers to cover the back of the audience (so called "delays"). Due to rigging restrictions, costs and logistics, especially in touring, the amount of loudspeakers in a setup is limited, and therefore the distances between single loudspeakers can be up to several metres. The aliasing frequency for such a setup is typically very low.

Concert loudspeakers have carefully designed directivity that tends towards cardioid at low

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frequency and becomes narrower at higher frequencies. The directivity patterns don't introduce a large error to a synthesized sound field if the loudspeakers are located close to each other and the audience area is small, so assuming monopoles or dipoles works fine for these applications. However, at large distances to the listener and between the speakers the directivities don't overlap enough and create significant error.

The problem layout is shown in the fig. 1. For simplicity only one virtual source is shown, but their number is not limited.

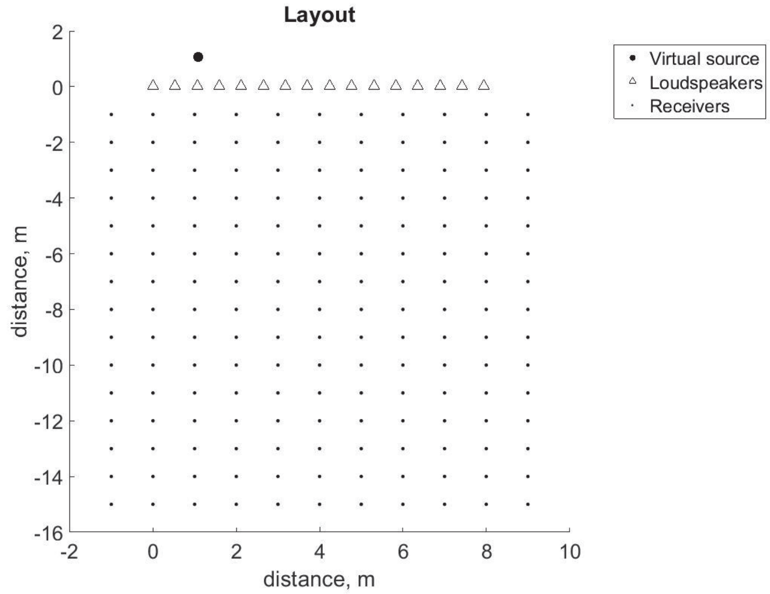

Figure 1 – Problem layout

3. ANALYTICAL SOLUTION

Our goal is to reconstruct the desired sound field at all the receivers. The desired field is the one that the virtual source/sources would create in the absence of loudspeakers. Knowing the positions of virtual sources and the receivers, the desired field can be calculated using eq. (1):

$$p^d(r_r, f) = \sum_s^S \frac{e^{-ikr_{rs}}}{r_{rs}} \tag{1}$$

where S is the number of virtual sources.

If needed, directivity and frequency response of the virtual source can be included:

$$p^d(r_r, f) = \sum_s^S A_s(f) D_s(f) \frac{e^{-ikr_{rs}}}{r_{rs}} \tag{1.1}$$

For simplicity, we assume an omnidirectional virtual source with flat frequency response.

The reconstructed field is a combination of the fields of all the loudspeakers with their respective unknown driving functions:

$$p^{rec}(r_r, f) = \sum_l^L q_l(f) D_l(f) \frac{e^{-ikr_{rl}}}{r_{rl}} \tag{2}$$

where $q_l(f)$ is the unknown frequency dependent driving function for each loudspeaker, $D_l(f)$ is its directivity and L is the number of loudspeakers.

Ideally, reconstructed field should be made equal to the desired field by finding appropriate driving functions:

$$p^d(r_r, f) = p^{rec}(r_r, f) \tag{3}$$

$$\sum_s^S \frac{e^{-ikr_{rs}}}{r_{rs}} = \sum_l^L q_l(f) D_l(f) \frac{e^{-ikr_{rl}}}{r_{rl}} \tag{4}$$

In order to solve Eq.4 for $q_l(f)$, we can write it in a matrix form for each frequency:

$$\widehat{H} \times \vec{q} = \overrightarrow{V}, \tag{5}$$

$$\text{where } \widehat{H} = H_{rl} = D_l(f)\frac{e^{-ikr_{rl}}}{r_r} \text{ is an } R \times L \text{ matrix,} \tag{6}$$

$\vec{q}$ is a vector of driving function values and

$$\overrightarrow{V} = V_r = \sum_{s}^{S} \frac{e^{-ikr_{rs}}}{r_{rs}} \tag{7}$$

The unknown vector of driving functions can be found by the matrix inversion:

$$\vec{q} = \widehat{H}^{-1} \times \overrightarrow{V}, \tag{8}$$

Eq. 8 is a typical inverse problem.

Depending on the number of loudspeakers L and receiver points R, the system of equation can be under defined, defined or over defined. [kim choi] Since receiver points are just virtual microphones and their number can be easily adjusted, we don't need to consider the first case. In case of R=L, matrix inversion is straight forward. In case of R>L, the Eq. (8) is solved using a built in Least Mean Squares method in Matlab.

4. SIMULATIONS

Eq. (8) was solved using Matlab. The setup consisted of 16 identical loudspeakers at the distance of 0.53 m between their centers. The simulation used measured loudspeaker's directivity with 5° precision and interpolation in between. The setup is too small for a live sound system, but is representative for the first tests. The same setup was later used for an indoor listening test.

Figure 2 shows the normalized sound pressure level distribution of the desired and reproducted fields. At low frequencies the reconstrusted field looks very similar to the desired field. However, at higher frequencies interference patterns are clearly visible.

5. LISTENING TESTS

5.1 Setup

Listening tests were conducted to evaluate the performance of the method. The setup shown in Figure 3 consisted of 16 loudspeakers located with spacing of 53cm, same as the simulation in 4. The loudspeakers were hidden behind an acoustically transparent curtain to exclude visual cues. Four listeners positions were defined to represent the entire listening area. Virtual sources are located in the "stage" area behind the loudspeakers; one virtual source is placed far off the stage on the side. The test signals were three pulses of broadband pink noise. Each listener position was provided with a disk marked in degrees on the floor. Participants were asked to write down the perceived direction of the source which was later compared with the geometrical direction.

On day one, 32 respondents took part in the test. Every participant was tested at two listeners positions, so that every position has 16 independent results. On the first day only locations of virtual sources were tested (Table 1).

On day two, 14 respondents were available and provided 7 independent results for each listener position. Three more virtual sources were added as well as two real sources (loudspeakers) that were located behind an acoustically transparent curtain (Table 2).

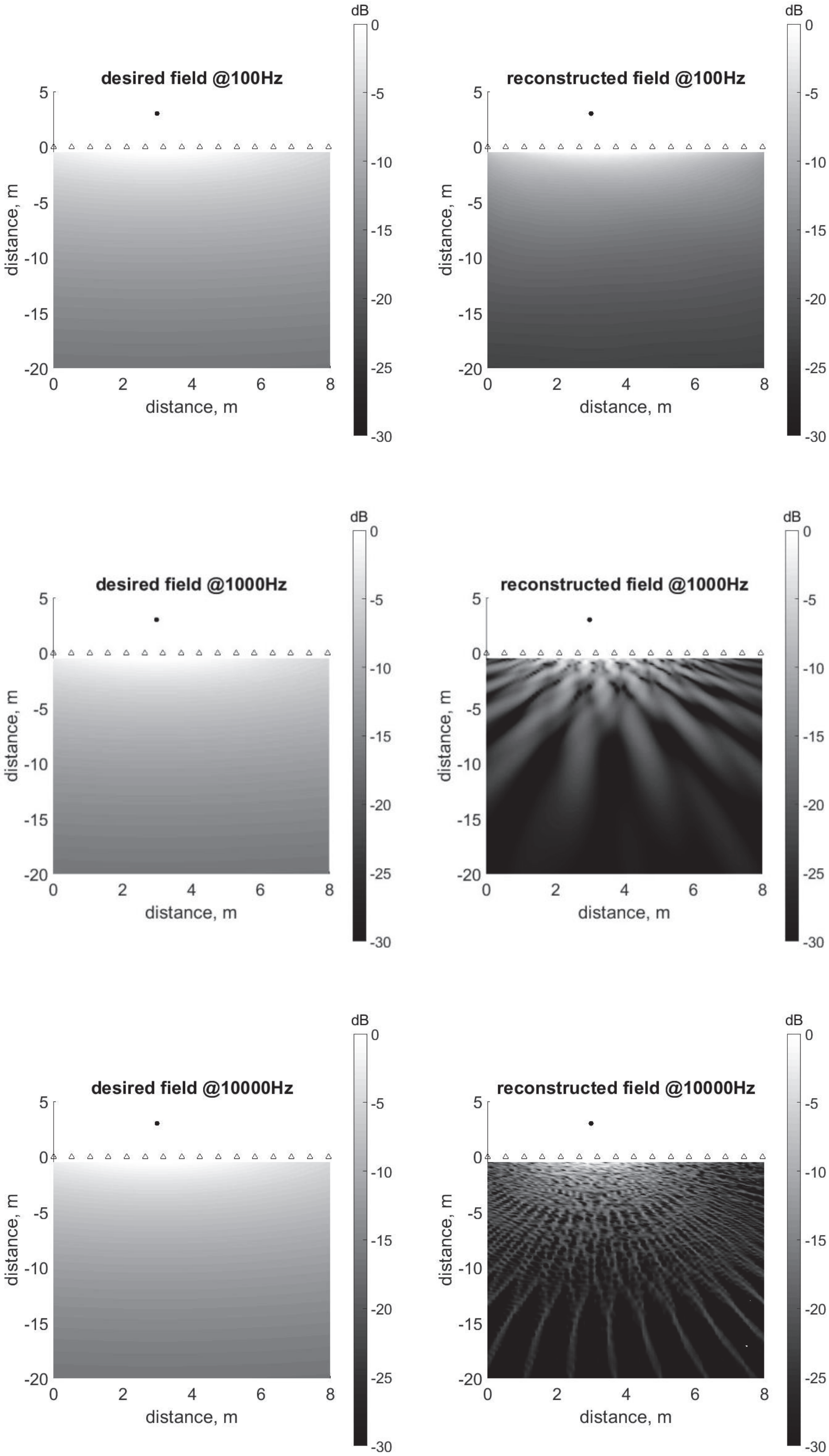

Figure 2 – Simulated normalized sound pressure level distribution of desired and reconstructed fields

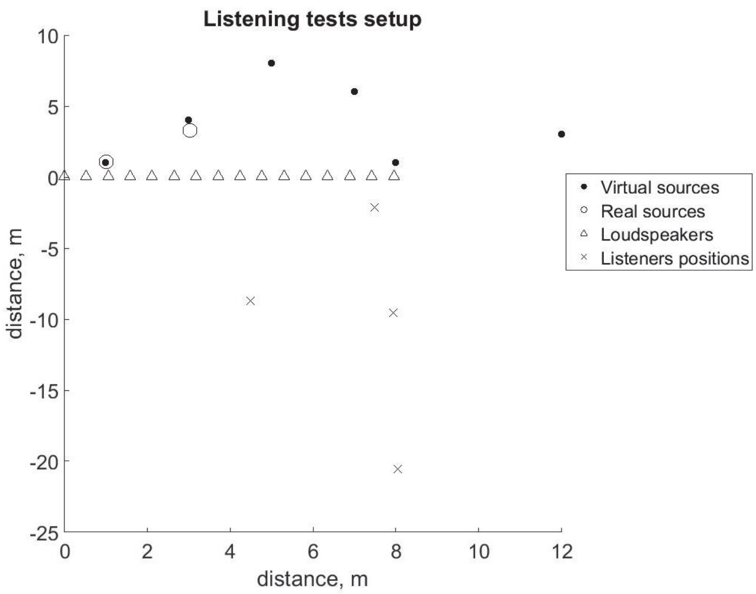

Figure 3 – Listening test setup

5.2 Results

Table 1 – Listening test results, day1

	Listener positions				
Virtual sources coordinates	1	2	3	4	average
(1, 1)	1.67	1.6	5.40	4.00	3.17
(5, 8)	2.33	2.20	4.67	4.00	3.30
(12, 3)	34.33	13.33	9.47	4.00	15.28

Table 1 presents the average deviation of the perceived direction of the virtual source from the geometrical direction in degrees.

Table 2 – Listening test results, day2

	Listener positions				
Virtual sources coordinates	1	2	3	4	average
(1, 1)	2.39	4.34	10.73	3.57	5.26
(5, 8)	2.39	1.71	4.34	1.09	2.38
(12, 3)	34.24	16.17	13.34	14.38	19.53
(3, 4)	2.41	3.71	7.75	9.73	5.90
(7, 6)	3.76	3.22	6.92	0.26	3.54
(8,10)	2.03	0.87	3.67	6.26	3.21
Real sources coordinates					
(1, 1)	7.87	5.00	7.79	5.88	6.14
(3, 3.3)	1.91	3.52	6.67	6.58	6.25

Table 2 presents the average deviation of the perceived direction of the virtual source from the geometrical direction in degrees.

5.3 Discussion

The virtual source with coordinates (12, 3), located far outside the "stage" area was localized by far the worst, with average deviation of 17.4°. Virtual sources within the stage were localized with the average deviation of 3.8°. The overall average deviation is 6.8°.

For real sources, the average was about 6.2° in average for both sources and all listener positions.

With all virtual source positions taken into account, localization of real sources was slightly better than virtual sources. However, if the virtual source outside the stage area is discarded as an unrealistic scenario, the localization of remaining virtual sources was (surprisingly) better than the localization of the real sources.

All in all, listening tests show that the field-matching method provides sufficiently accurate localization despite aliasing.

6. CONCLUSIONS

The paper shows how to derive an explicit expression for driving functions for a sound field synthesis setup without posing initial restrictions on loudspeakers' layout or directivity. Preliminary tests show that the method provides reliable localization with precision of about 7°, despite visible aliasing effects at higher frequencies.

Future work may include analyzing numerical methods to solve Eq. (8), possibly introducing regularization to obtain smooth filters as driving functions.

ACKNOWLEDGEMENTS

The author would like to thank the team of d&b audiotechnik for participating the listening tests, and specially Julian Ebert for his help with the setup.

REFERENCES

1. Berkhout A.J., de Vries D., Vogel P., Acoustic Control by WaveFieldSynthesis, The Journal of the Acoustical Society of America 93, 2764 (1993)
2. Ahrens J., Analytical Methods of Sound Field Synthesis, Springer, 2012
3. Roginska A., Geluso P., Immersive Sound, Routledge, 2018
4. de Vries D., Sound Reinforcement by WFS: Adaptation of the Synthesis Operator to Loudspeaker Directivity Characteristic, J. Audio Eng. Soc., Vol.44, No. 12, 1996.
5. Ahrens J. and Spors S., Sound field reproduction employing non-omnidirectional loudspeakers, 126th Conv. Audio Eng. Soc. (2009), preprint 7741.
6. Kim Y.-H., Choi J.-W., Sound Visualization and Manipulation, Wiley, 2013