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INFLUENCE OF LOUDSPEAKER DISPLACEMENT ON THE REPRODUCTION QUALITY OF WAVE FIELD SYNTHESIS SYSTEMS

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ABSTRACT

The concept of Wave Field Synthesis (WFS) nowadays is a well known audio reproduction method. Based on the Huygens principle numerous secondary sound sources (loudspeakers) are used to create a replica of a primary sources wave fronts. Due to the fact that there is no standardised speaker placement when using WFS the positioning of the secondary sources can be individually designed. For rendering a virtual audio scene to real loudspeaker signals it is necessary to know the real speaker positions. Doing the position measurement of the speakers with low accuracy or changing the loudspeaker positions by accident leads to differences between the real and the rendered loudspeaker position. During a present study the effect of loudspeaker displacement has been investigated. This paper gives a general survey about current acoustic simulations and measurements.

INTRODUCTION

By applying the holographic approach to acoustics [1,2] a new sound reproduction method called Wave Field Synthesis (WFS) was introduced during the late 1980'ies. As holophonic audio systems aim for the reconstruction of the original sound wave fronts over a wide listening area, WFS enables an accurate representation of the original wave field with its natural temporal and spatial properties in the entire listening space and therefore offers a sophisticated listening experience.
The theoretical concept of WFS has been well studied so far. But when implementing a WFS sound system it is necessary to overcome practical limitations and one has to deal with several kinds of artefacts. In practice it is not possible to deal with an endless distribution of infinitesimal dense secondary sources. Truncation artefacts at the end of the loudspeaker arrays lead to slight diffraction phenomena and the discretization can be seen as a process of spatial sampling that causes aliasing artefacts which are figuratively comparable to time domain sampling.
A remarkable feature of WFS for many areas of applications is the reproduction of moving sound sources. Although several current WFS systems are able to reproduce moving sound sources, the synthesis causes a number of distinct artefacts [3].
In this paper the focus is set on errors caused by deficiencies that are associated with the temporal and spatial validity of the rendered audio signals. These deficiencies will affect the physically correct superposition of the sound field components which furthermore could lead to a lack in perceived audio quality. On look for the causes for this physically imperfect reconstruction one can, for instance, identify temporal fluctuations on the signal path (jitter) or discrepancies within the geometrical data (loudspeaker coordinates) used for rendering an audio scene (Fig. 2).

WAVE FIELD SYNTHESIS

The 2 ½ D – Operator

The underlying physical principle for WFS is the Huygen's Principle (Fig.1a). It states that every point on a wave curvature can be seen as the origin of another wavefront. A superposition of these secondary wave fronts reproduces the wave field of the original (primary) source.

Arrays of closely spaced loudspeakers are used for the reproduction of the targeted (or primary) sound field. The audio signal for each loudspeaker is individually adjusted with well balanced gains and time delays, the WFS parameters, depending on the position of the primary and the secondary sources. For the calculation of these parameters an operator has been developed [4]. The so called 2½D-Operator (Eq.1) is usable for two dimensional loudspeaker setups, which means that all loudspeakers are positioned in a plane defining the listening area (Fig.1b).

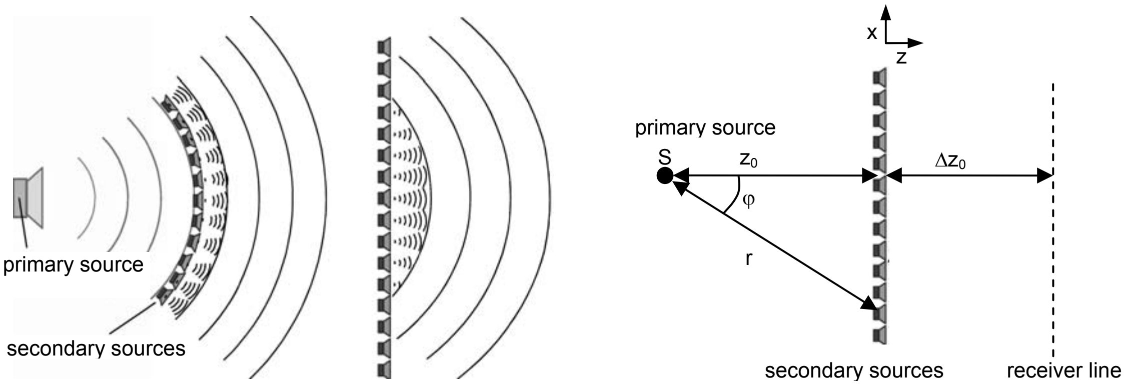

Figure 1: A) Huygen's Principle   B) Development of the 2½D Operator

In (Eq.1) the 2½D-Operator in frequency domain can be subdivided into a frequency dependent term H(w) and the term Y(r) which is related to geometrical data only.

$$Q_m(r,\omega) = S(\omega)\underbrace{\sqrt{\frac{jk}{2\pi}}}_{H(\omega)}\underbrace{\sqrt{\frac{\Delta z_0}{z_0+\Delta z_0}}\cos\varphi\frac{\exp(-jkr)}{\sqrt{r}}}_{Y(r)} \qquad \text{(Eq. 1)}$$

After Fourier transformation into the time domain,

$$Q_n(\omega) = Y_n H(\omega) S(\omega) \qquad FT \Rightarrow \qquad q_n(t) = Y_n(t) * \left[h(t) * s(t)\right] \qquad \text{(Eq. 2)}$$

where * denotes the time domain convolution operator, one can separate the geometrical term

$$Y_n(t) = a_n \delta(t - \tau_n) \qquad \text{(Eq. 3)}$$

$$\text{with} \quad a_n = \sqrt{\frac{\Delta z_0}{z_0+\Delta z_0}}\frac{\cos\varphi}{\sqrt{r_n}} \quad \text{and} \quad \tau_n = \tau_0 - \frac{r_n}{c} \qquad \text{(Eq. 4a, 4b)}$$

Where a_n is the gain for the n^{th} loudspeaker and τ_n is the according time delay.

From (Eq.4) one can see that the degree of precision of the geometric information directly affects the calculation of WFS parameters and, hence, makes an impact on the quality level of the synthesized wave field.

Practical Aspects

The schematic view of a typical WFS implementation following an object oriented approach for audio reproduction is depicted in (Fig.2). Temporal inconsistencies due to jitter e.g. caused by the digital-to-analog converters (DAC's) were neglected for this report so that only inaccuracies in the geometric information of the speakers are taken into account.

Today's WFS systems are equipped with different loudspeaker technologies. Single speaker systems, speaker panels and flat panel speakers with multiple exciters are in use. For this reason single speaker and also grouped speaker displacements are investigated.

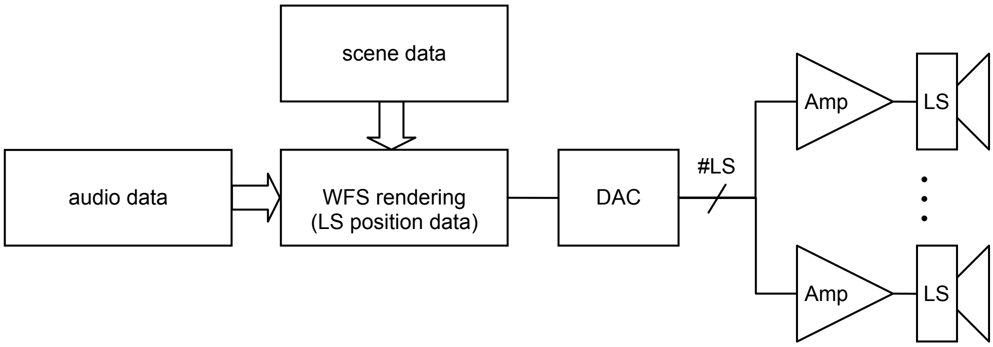

Figure 2: Schematic signal and data paths of a typical WFS system

As an effect of imprecise speaker position data the WFS algorithm delivers incorrect signal gains and delays for the calculation of the loudspeaker driving signal (Eq. 4). This virtual displacement due to invalid position data of the speaker is in diametrical opposition to a real change of its position. Shifting a speaker towards a virtual source decreases the calculated delay and the speaker will start earlier to play the audio signal and vice versa.

METHODOLOGY

As a start it is useful to get more knowledge about the dimensions of the displacements that could cause any synthesis errors and to recognize when they turn out to become apparent to the synthesized wave fronts. An acoustic computer simulation is used for the visualization of the generated sound field. These findings serve as a basis for the preparation of the real measurements and also for listening tests. Suitable configuration files for the loudspeaker positioning are to be selected that enable a direct comparison of the results made in simulations, measurements and listening tests.

Simulation

For the acoustic simulation a frequency domain model is used under free field conditions. Primary and secondary sources are represented as omnidirectional point sources. The loudspeaker array consists of 32 elements with a spatial interval of $\Delta x = 0.175m$ and therefore has an overall aperture of approximately $5.6m$. This means that the aliasing frequency f_{al} (Eq. 5) is as low as 1000Hz. With respect to the real measurements the sound field is calculated inside a quadratic area with 8 meters side length.

$$f_{al} = \frac{c}{2\Delta x} = \frac{340\, m/s}{2 \cdot 0.175m} = 971Hz \qquad \text{(Eq. 5)}$$

During preliminary test cycles it was found that the ideal test signal is a pure sine wave with a frequency right below the aliasing frequency of the loudspeaker array. This can be seen as a worst case scenario because the shorter the wavelength in comparison to the loudspeaker displacement the stronger the impact will be. Increasing the signal frequency beyond f_{al} would cause artefacts due to spatial aliasing which is not an object of this study.
For the description of the irregularities it is helpful to quantify the deviations of the inaccurate rendered wave field with the help of a logarithmic synthesis error (Eq. 6).

$$L = 10\log_{10}\left[\left(\frac{P}{P_{ref}}\right)^2\right] \quad [dB] \qquad \text{(Eq. 6)}$$

This synthesis error is calculated for every point in the simulation area as the pressure ratio between the original (P_{ref}) and the disturbed (P) wave field. As will be shown in the next chapter, this gives good insight to the spatial distribution of the deviations [4].
The displacement of the loudspeakers was systematically investigated during numerous simulation runs. From the simulation kit it is possible to add a shift in direction of the Cartesian coordinates or to turn a panel around an arbitrary axis (Fig. 4). Also combinations of both and randomly calculated displacements can be applied to single speakers as well as speaker panels.

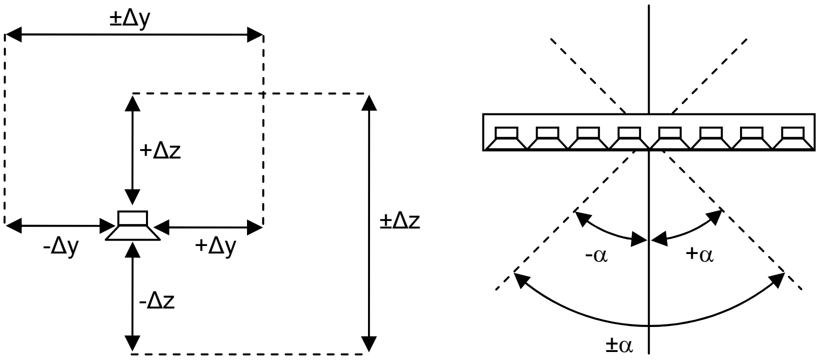

Figure 4: A) Displacement in the range of ±Δy/Δz of single loudspeakers
B) Rotation of loudspeaker panels

Measurements

Wave Field Synthesis is a holophonic audio reproduction method. Hence, if one would gather relevant information about the quality of the synthesized wave field, it is necessary to extend the measurements to a representative area in the listening plane. For this purpose the measurement tool RAWES (Room Acoustical mEasurement System) has been developed at Fraunhofer IDMT [5]. Using a graphical user interface the measurement procedure can be easily set up. Automatically the impulse responses on a predefined grid are acquired using a swept-sine technique [6]. It is also possible to only collect the impulse responses along a line through the reproduction area. This data can be used for time domain analysis and for plane wave decomposition [7]. In order to achieve comparable results the real loudspeakers were arranged in the same way as in the simulation model and the measurement was done in the same room where the listening test took place. Also the virtual source setup and the loudspeaker displacements have been kept identically to the simulation.

RESULTS AND DISCUSSION

Due to the rich variety of investigated setups some examples are picked up now for discussion to get an idea of essential connections between specific loudspeaker displacements and their synthesis artefacts. In the simulation results, the speakers are placed along the x-axis and the wave fronts are travelling in direction of the z-axis from the left to the right side of the diagrams.
In the first example virtual sources behind the loudspeakers and focused sources are compared (Fig. 5). Every single loudspeaker has a random position error in the range of $\Delta x = \Delta z = +/-1.5cm$
It is clearly to see that for focused sources (Fig. 5 c,d) the resulting artefacts are much more stronger than for the non focused soured. Therefore the synthesis error in the latter case keeps below 3dB in contrast to high error levels up to 10dB at the lateral areas of the reproduction area of the focused source.
For displacement caused by rotated panels, as simulated in the second example, one can state that it is important that the panels meet at their ends. In this case (Fig. 6 a,b) there occurs no abrupt phase difference caused by e.g. a gap between panels as depicted in (Fig. 6 c,d). The panels are rotated with an angle of $\alpha = 10°$.
From further simulations it can also be established that for speaker displacements perpendicular to the direction of the wave propagation (y-axis) the defects in the synthesized wave fronts are less intense than for displacements within the direction of propagation (z-axis).

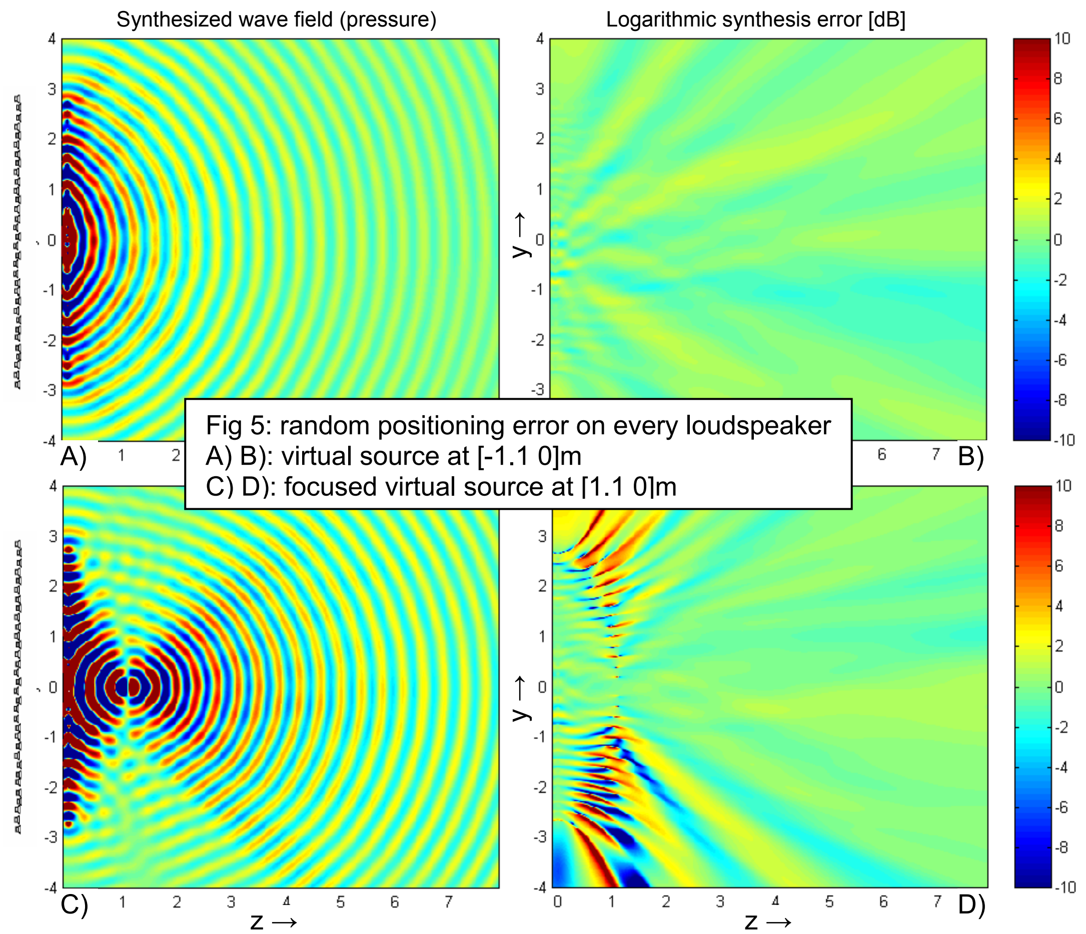

Fig 5: random positioning error on every loudspeaker
A) B): virtual source at [-1.1 0]m
C) D): focused virtual source at [1.1 0]m

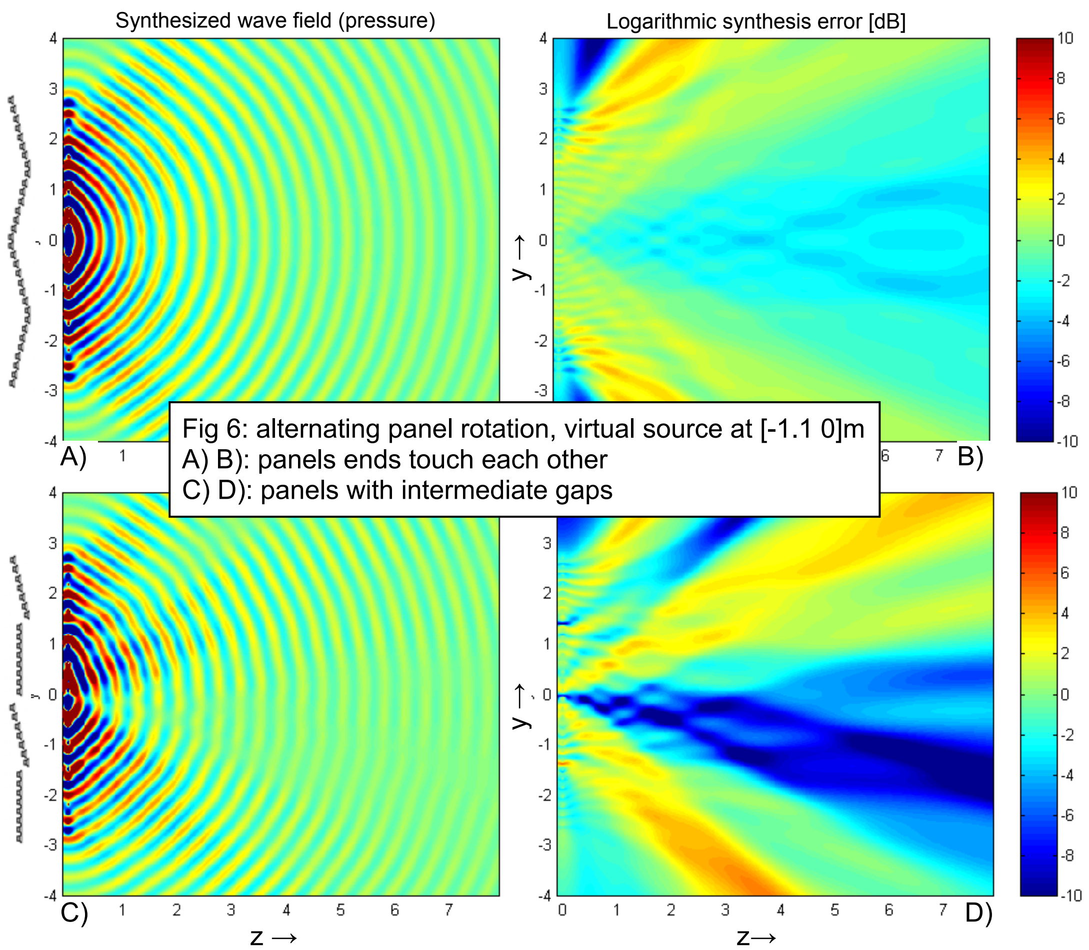

Fig 6: alternating panel rotation, virtual source at [-1.1 0]m
A) B): panels ends touch each other
C) D): panels with intermediate gaps

Another example (Fig. 7) shows some results from the measurements with RAWES. The speaker setup is as described at an earlier stage in this paper. In this special case the impulse responses are collected along a line parallel to the y-axis at z=1.5m. The position of the focused source is at z=1.1m. The impulse responses are depicted as x-t plot. According to the simulation results one can recognize increasing deformations of the wave front, even for small speaker displacements.

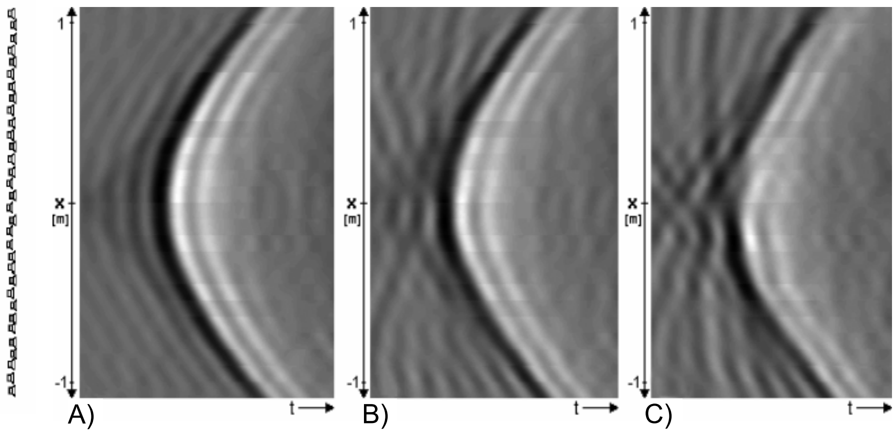

Figure 7: Impulse responses x-t plot, random single speaker positioning error, focused virtual source. A) original wave field B) error range Δy = Δz = ± 1cm: C) error range Δy = Δz = ± 3cm

Sets of impulse responses as from the measurement described above could build up the basis for a more substantially sound field analysis by plane wave decomposition.

CONCLUSIONS

The effect of loudspeaker displacement in Wave Field Synthesis audio reproduction systems has been investigated during a current study. This papers reports about the variety and the dimensions of the displacements that could cause any synthesis errors. In view of the presented examples from simulations and measurements, it may be shown when these geometrical deficiencies turn out to become apparent to the synthesized wave fronts. It can be stated that the influence on focused sources seems to be higher than on other virtual source types.
In addition to the objective part of the investigation a listening test was conducted. It is of great importance to establish valid relations between the objective and subjective evaluation. The publication of detailed results is planned for the future.

REFERENCES

[1] A.J. Berkhout. A holographic approach to acoustic control. Journal of the Audio Engineering Society, 36(12):977–995, December 1988.
[2] A. J.Berkhout, D. de Vries, and P. Vogel. Acoustic control by wave field synthesis. Journal of the acoustical society of America, 95(5):2764–2778, May 1993.
[3] A. Franck, A. Gräfe, T. Korn, M. Strauß. Moving Sound Sources and Wave Field Synthesis – Aspects of Reproduction Artifacts. accepted for the AES 32nd Conference, Hillerød, Denmark, 2007
[4] E. Start. Direct Sound Enhancement by Wave Field Synthesis. PhD thesis, Delft University of Technology, 1997.
[5] H. Kutschbach. Verification for Spatial Sound Systems. AES 24th Conference, Banff, 2003.
[6] A. Farina. Simultaneous Measurements of Impulse Response and Distortion with a Swept-Sine-Technique. AES Preprint, 108th Convention, February 19-22, 2000.
[7] E. Hulsebos. Auralization using Wave Field Synthesis. PhD thesis, Delft University of Technology, 2004.
[8] M. Munderloh. Untersuchung zum Einfluss der Genauigkeit der Lautsprecherpositionierung bei Schallwiedergabe mittels Wellenfeldsynthese. Master Thesis, Technical University Ilmenau, Germany, 2005.