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IMPLEMENTING AMBISONICS ON A 48 CHANNEL CIRCULAR LOUDSPEAKER ARRAY

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Abstract: *This contribution describes an implementation of near field coding higher order Ambisonics on an existing circular array of 48 loudspeakers. Static and moving sources can be reproduced with a variable Ambisonics order and variable number of loudspeakers. Informal listening tests were performed to evaluate the influence of Ambisonics order and loudspeaker number on the perception quality and localization accuracy.*

Key words: Ambisonics, Near Field Coding, Circular Array, Moving Sources, Doppler Effect, Listening Tests

1 INTRODUCTION

The usual way to implement an Ambisonics system proceeds in two steps. In the first step the system is designed with respect to the Ambisonics order, the number of loudspeakers and other choices, e.g. near field coding etc. In the second step, the system is implemented by setting up the required hard- and software. Once the system is in operation, any changes to the Ambisonic order or the number and placement of loudspeakers requires considerable effort.

Here the procedure started from an existing loudspeaker setup in the form of a circular array with 48 loudspeakers. Suitable software and audio interfaces allow to address each single loudspeaker or an arbitrary subset individually. This way the choice of the setup of active loudspeakers and the corresponding order and type of the Ambisonics system becomes a software issue. This contribution describes such a variable Ambisonics implementation and reports about some listening tests.

The existing circular array with 48 loudspeakers belongs to the audio facilities of the Telecommunications Laboratory at the University of Erlangen-Nürnberg and is shown in Figure 1. Each loudspeaker can be individually driven via a control computer with two multichannel soundcards and by corresponding multichannel digital input analog output amplifiers. Although this setup has been designed as a wave field synthesis (WFS) system it is suitable for any massive multichannel audio reproduction method.

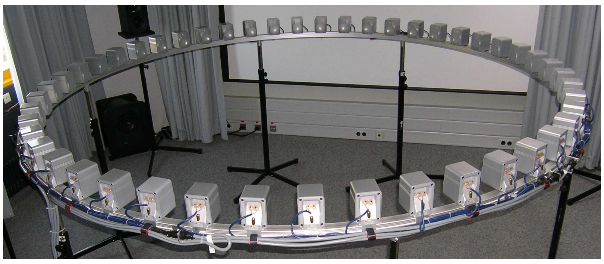

Figure 1: 48 channel circular loudspeaker array.

With the recent developments of Ambisonics (for a literature overview see [2–7] and the references cited there) it was straightforward to implement a two-dimensional Ambisonics system on the given circular loudspeaker array. It calculates the appropriate loudspeaker driving signals for different reproduction scenarios without the need for any hardware manipulations. Especially the Ambisonics order and the number of used loudspeakers are variable and at the disposal of the user via a graphical interface.

The realized MATLAB implementation shows successfully the competetive properties of Ambisonics in comparison to the already existing WFS implementation.

2 OVERVIEW ON THE IMPLEMENTED ALGORITHM

2.1. Hardware Constraints

The array shown in Figure 1 consists of 48 loudspeakers placed on a circle of 1.5 m radius at equal angles. It allows to choose a subset of L loudspeakers at equidistant directions with

$$L \in \{3, 4, 6, 8, 12, 24, 48\} \tag{1}$$

The directions of the selected loudspeakers are given by the angles

$$\theta_l = \frac{2\pi}{L}\, l\ . \tag{2}$$

This setup features the possibility to realize Ambisonics orders N from 1 to 23. There are a total of 53 possible and meaningful combinations of Ambisonics order and number of loudspeakers, which satisfy the equation

$$L \geq 2N + 1. \tag{3}$$

2.2. General Procedure

The general procedure for the Ambisonics implementation on this specific circular array follows the descriptions given in the literature [2–7]. A model-based approach has been implemented:

- A total of K individual plane wave sources with source direction θ_k (static or moving).
- A total of K individual point sources with source direction θ_k and distance ρ_k (static or moving).

The trajectories of variable sources are restricted to movements either in radial or in angular direction. First the so-called Ambisonics signals are encoded, which are also known as Ambisonics components for one single virtual sound source [3]. Then these signals are converted into loudspeaker driving signals for the circular array.

2.3. Coding of Ambisonics Signals

Regarding the coding of Ambisonics signals we now distinguish between static plane waves, static point sources and moving sources, whereas the calculation procedure is respectively very similar. In the following the time domain signal of the k-th source at angular position θ_k relative to the sweet spot is denoted by $g(t, \theta_k)$. Ambisonics signals are denoted by $g^\circ(t, \nu)$, where $\nu \leq |N|$ is an integer.

2.3.1 Plane Waves

For K plane waves from the directions θ_k the Ambisonics signals are calculated as follows

$$g^\circ(t, \nu) = \sum_{k=0}^{K-1} g(t, \theta_k) \cdot X_\nu^*(\theta_k). \tag{4}$$

Each source signal $g(t, \theta_k)$ is weighted by the conjugate complex circular harmonic $X_\nu^*(\theta_k)$ with

$$X_\nu(\theta_k) = e^{j\nu\theta_k}. \tag{5}$$

Then all weighted source signals are summed up to a two-dimensional matrix of Ambisonics signals. For three dimensions the circular harmonics are replaced by the well known spherical harmonics.

2.3.2 Point Sources

To realize acoustical point sources the Near Field Coding (NFC) approach of [3] was implemented. With equation 4 and the NFC-Filters with the impulse responses $h_\nu^{\mathrm{NFC}}(t, \rho_k, R)$, which depend on the source distance ρ_k and the array radius R (here 1.5 m), this leads to the new equation

$$g^\circ(t, \nu) = \sum_{k=0}^{K-1} \left[g(t, \theta_k) * h_\nu^{\mathrm{NFC}}(t, \rho_k, R)\right] \cdot X_\nu^*(\theta_k). \tag{6}$$

Figure 2 illustrates the generation of Ambisonics signals for point sources. First each source signal $g(t, \theta_k)$ is filtered by the NFC-Filters $h_\nu(t, \rho_k, R)$, where $\nu \leq |N|$ is an integer and

$$h_0(t, \rho_k, R) = 1 \tag{7}$$
$$h_\nu(t, \rho_k, R) = h_{-\nu}(t, \rho_k, R). \tag{8}$$

Second these signals are weighted by the circular harmonics $X_\nu^*(\theta_k)$. After all source signals $g(t, \theta_k)$ with the individual input parameters (ρ_k, θ_k) are processed in the described way, they are finally summed up to the output matrix $g^\circ(t, \nu)$.

2.3.3 Moving Sources

To realize moving sources with arbitrary trajectories in two dimensions, basic motions along the unit vectors in polar coordinates were implemented. K plane wave sources moving along the angular coordinate φ with a constant rotational angular velocity $\omega_{\text{rot,k}}$ and a start angle θ_k result in the following expanded equation for the Ambisonics signals

$$g^\circ(t, \nu) = \sum_{k=0}^{K-1} g(t, \theta_k) \cdot X_\nu^*(\theta_k + \omega_{\text{rot},k} \cdot t). \tag{9}$$

For rotating point sources we obtain

$$g^\circ(t, \nu) = \sum_{k=0}^{K-1} \left[g(t, \theta_k) * h_\nu^{\mathrm{NFC}}(t, \rho_k, R)\right] \cdot X_\nu^*(\theta_k + \omega_{\text{rot},k} \cdot t). \tag{10}$$

Motions along the radial coordinate ρ are divided into short sections (one millimeter or smaller) with approximately constant NFC-Filters. Then the calculation matches equation 6. This procedure is illustrated in Figure 3 in a simplified way. Furthermore the Doppler effect was incorporated with signal dilations and compressions using an appropriate sample rate conversion before filtering. Finally it should be mentioned that in this first test system complex trajectories are not possible yet, but for an upcoming realtime implemention this should be no problem.

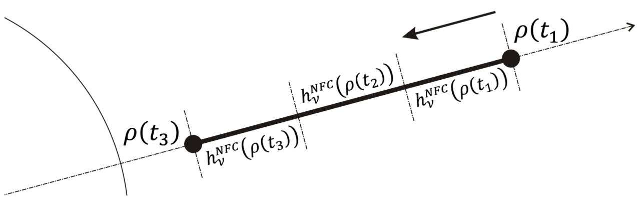

Figure 3: Radial source motion.

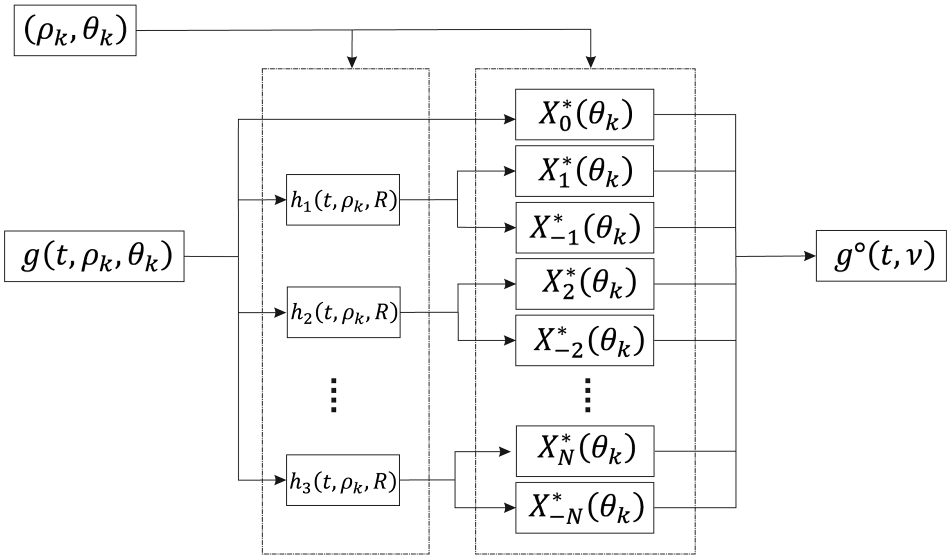

Figure 2: Generation of Ambisonics signals for point sources (similar to [3]).

2.4. Calculation of Loudspeaker Driving Signals

Having computed the Ambisonics signals they are converted to L loudspeaker driving signals at the positions θ_l by a Discrete Fourier Transform (DFT)

$$g(t, \theta_l) = \mathrm{DFT}_L^{-1}\left\{g^\circ(t, \nu)\right\} = \frac{1}{L}\sum_{\nu=0}^{L} g^\circ(t, \nu) \cdot e^{j\nu l \frac{2\pi}{L}}. \tag{11}$$

The DFT length is determined by the number of loudspeaker channels. For a number of loudspeakers smaller than 48 the computed channels are finally placed at equidistant positions in the circular array.

3 MATLAB REALIZATION

In Figure 4 the implemented MATLAB graphical user interface (GUI) is depicted. In the present implementation one can declare a maximum of 10 sources, which can be static or moving. The sample rate is 44100 Hz. In the standard configuration the sources are calculated as plane waves. Optionally near field coding (NFC) can be enabled [3]. For moving sources there are start and end positions required and various other adjustable parameters. After configuration of all desired parameters the algorithm produces a 49 channel wave file, which can be played back via an existing reproduction software. The 49th channel is for an optional subwoofer. Because of the necessary amount of data for 48 channels and memory restrictions in MATLAB one can only calculate output files up to a length of about one minute.

4 RESULTS OF INFORMAL LISTENING TESTS

After implementation listening tests were performed to evaluate the influence of Ambisonics order and loudspeaker number on the localization accuracy and perception quality. For this purpose 11 of the 53 possible combinations of Ambisonics order and number of loudspeakers were chosen, mainly with small orders up to five and one reference file with 48 loudspeakers and an order of 23 (see Table 1).

1st order	3 loudspeakers
1st order	6 loudspeakers
1st order	12 loudspeakers
1st order	24 loudspeakers
1st order	48 loudspeakers
3rd order	8 loudspeakers
3rd order	12 loudspeakers
3rd order	24 loudspeakers
3rd order	48 loudspeakers
5th order	12 loudspeakers
23rd order	48 loudspeakers

Table 1: Test scenarios for the listening tests: Ambisonics order and number of loudspeakers.

For each of these test scenarios, there were three testfiles, male and female speech and white noise. They were played back at arbitrary chosen positions θ_k in the circular array. The choice of these source positions θ_k did not include the loudspeaker positions θ_l, such that there was always an interaction between at least two loudspeakers. The testfiles were about 20 seconds long and were played back only once.

The task of the test person was to detect the encoded directions of virtual sources as exact as possible. Then the deviation from the correct angle was calculated. The users's self confidence in his subjective estimation was the measure for the perception quality already mentioned above. In some test cases the person had to turn several times in the sweet spot to estimate the angle, whereas in some other cases the estimate was clear in the first few seconds of the test file. This circumstance was evaluated in confidence grades from

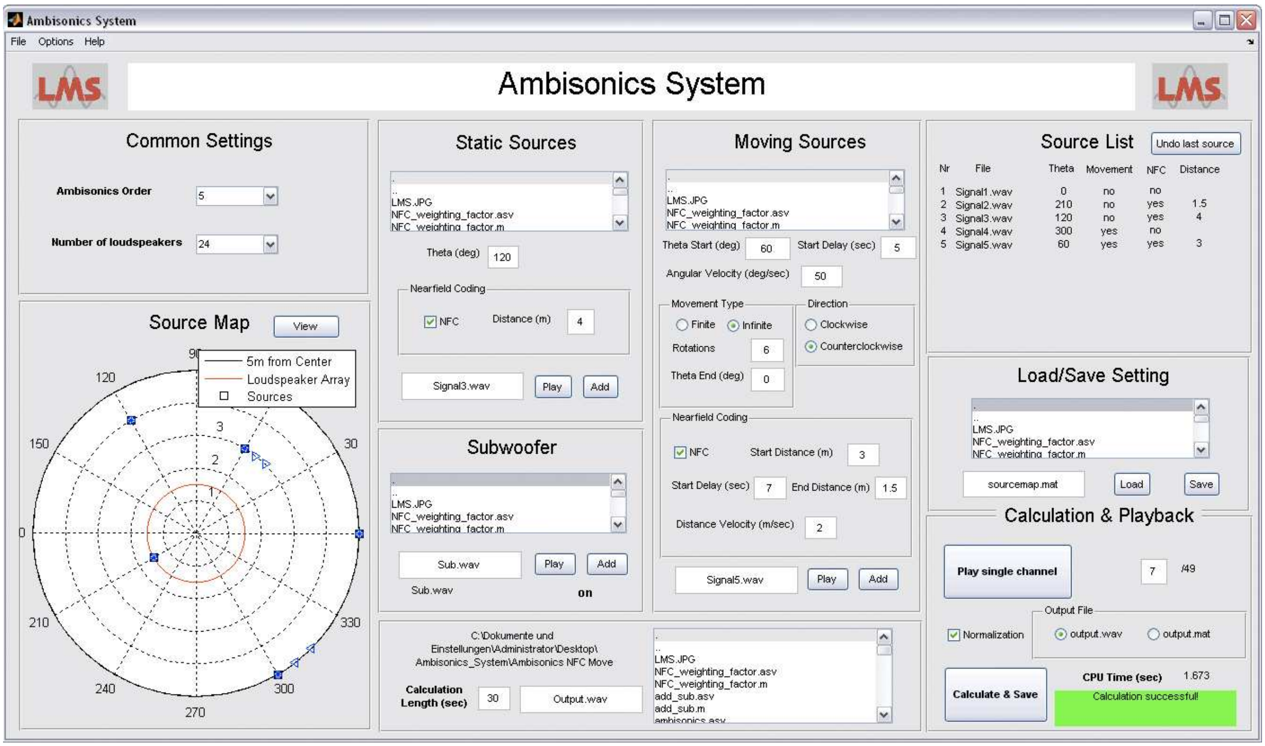

Figure 4: Implemented MATLAB Graphical User Inferface.

one to three, where one was the best. At the beginning two reference files of the best and worst perception quality were played back for calibration puposes.

Although these tests were rather informal, they showed very consistent results for the localization accuracy. Because the results for male and female speech and for noise were very similar, we now focus on the overall results, which are the mean values of the three test cases. These values are depicted in Figure 5 and 6 (results were a little better for the noise signal).

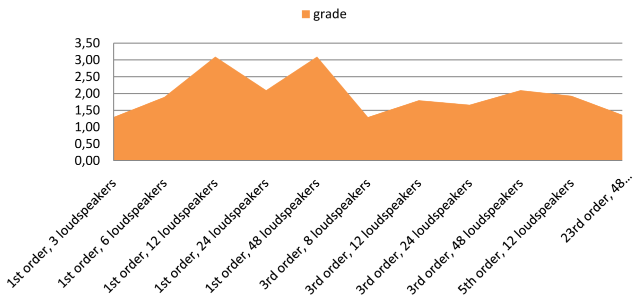

Figure 6: Perception quality shown as confidence grade between 1 (high) and 3 (low).

Figure 5 shows the deviation of the perceived source angles from the correct ones. It's easy to understand that for an Ambisonics order of one and three loudspeakers, it's very difficult to detect an exact source direction. On the other hand the test case with Ambisonics order 23 and 48 loudspeakers has the smallest deviation from the exact angle. Furthermore the large deviations for Ambisonics order one and 12 and 48 loudspeakers respectively stand out. We can summarize that the Ambisonics order and the number of loudspeakers have a strong influence on the perceived source angle for orders up to three and channel numbers up to eight. It is also apparent that for small orders and an excessive number of loudspeakers the result is getting worse. Best results are obtained for all combinations where the equality in equation 3 is satisfied as close as possible. For high orders and high channel numbers the deviation from the calculated angle was about five degrees but it did not get much better for the highest possible Ambisonics order and the maximum of possible loudspeakers. Therefore excessive orders and loudspeaker numbers should be avoided at least for static sources unless the much higher expense in calculation effort does not matter.

The evaluation of perception quality according to Figure 6 shows similar results. The detection of source direction does not get easier for orders above three.

5 CONCLUSION

In this contribution we reported on an Ambisonics realization on an existing circular loudspeaker array with 48 loudspeakers, which was originally designed for wave field synthesis. This implementation of a model-based Ambisonics system features the possibility of choosing freely between Ambisonics orders from 1 to 23 and a number of loudspeakers from 3 to 48 only by software control. A total of 10 plane waves and point sources (static or moving) can be placed at different angular positions and distances. The implementation showed successfully the competetive proper-

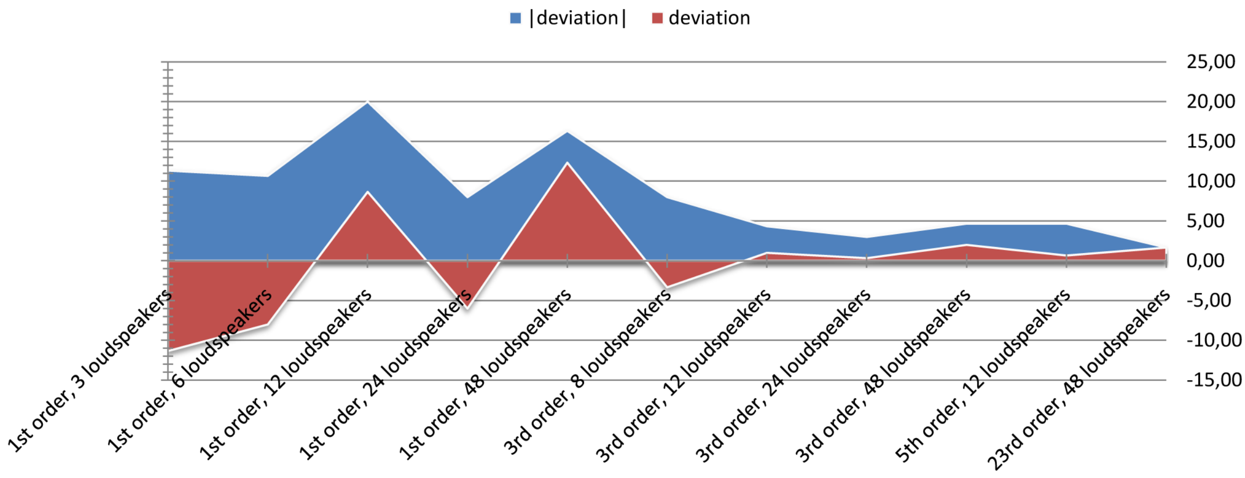

Figure 5: Directional accuracy as absolute value and magnitude of the deviation of the perceived source angles from the correct ones in degrees.

ties of Ambisonics in comparison to the already existing WFS implementation.

In an informal listening test the influence of Ambisonics order and the number of loudspeakers on localization accuracy and perception quality was investigated. They have strong influence on the percepted source angle for orders up to three and channel numbers up to eight. At least for static sources the much higher expense of Ambisonics order 23 and 48 loudspeakers is not justified.

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