FURTHER INVESTIGATIONS ON 3D SOUND FIELDS USING DISTANCE CODING

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ABSTRACT

This investigation proposes a possibility to synthesise a true 3D sound field over loudspeakers. A new approach concerning the distance coding is presented. We tried to combine the benefits both using the Wave Field Synthesis (WFS) approach and Higher Order Ambisonics (HOA). Therefore the proposed system can be divided into two main parts. Firstly the determination of the driving functions of the sound sources using the WFS approach. Secondly the coding for transmission or storage whereby the scheme is based on the Ambisonics approach using higher orders.

The paper is organised in three sections. The first section gives a brief introduction about the WFS and the HOA approaches. In the second section the derivation of the driving functions is presented and the coding scheme of the derived source signals is explained. Finally the paper is concluded and further possible research directions are identified.

1. INTRODUCTION

In the following a brief introduction about Wave Field Synthesis and Higher Order Ambisonics is given. Further extended information can be found concerning WFS in [1], [2], [3] and in the case of HOA in [3], [4], [5].

1.1. Wave Field Synthesis

The WFS approach is based on the Huygens Principle that is mathematically described by the Kirchhoff-Helmholtz-Integral (see Eq. 1). The Kirchhoff-Helmholtz-Integral implies that the wave field of a source free volume V can be described by the knowledge of the pressure along the enclosure surface S and the gradient of the pressure normal to the surface S.

$$P(\vec{r}_R) = \frac{1}{4\pi}\oint_S \left[P(\vec{r}_S)\cdot\nabla_S G(\vec{r}_R|\vec{r}_S) - G(\vec{r}_R|\vec{r}_S)\cdot\nabla_S P(\vec{r}_S)\right]\cdot\vec{n}\,dS \qquad (1)$$

whereby $G(\vec{r}_R|\vec{r}_S) = \frac{e^{-jk|\vec{r}_R-\vec{r}_S|}}{|\vec{r}_R-\vec{r}_S|}$ is known as the Green's function.

Therefore each arbitrary sound field inside a source free volume can be reproduced with distributed monopole and/or dipole sources along the surrounding surface. This leads to a technique called „holographic audio" [1] or also known as „holophone systems".

1.2. Higher Order Ambisonics

In general the Ambisonic approach is presented with the well-known B-format with the signals W, X, Y and Z. The system approach is based on the sound field's spatial decomposition in spherical harmonics of 0^{th} and 1^{st} order. This approach can be extended to higher order systems (HOA) [5], [6], resulting in better localisation properties and a wider listening area. However increasing the system order will also increase the required transmission channels and also the amount of necessary loudspeakers. In the 3D case, the mathematical description of the arriving planar wave is extended in a series leading to an infinite sum of cosine and sine weighted Legendre-functions P_l^m, leading to the real spherical harmonics. Using higher order signals will reduce the reconstruction errors and also energy spread over the loudspeakers. There are compromises necessary to overcome the problem of finite system orders, and even new complex microphone characteristics are required in case of recording real sound fields.

1.3. Motivation and System Design

The basic idea is to build up a system based on the advantages of HOA (e.g. channel coding, independence of the sound space and the location of the real loudspeakers) and WFS (possibility to mime a realistic transmission path). We use the curvature of the direct arriving sound field to mime the distance perception of a virtual source. The motivation of this system design is based on coding the curvature with the WFS approach and rendering the sound field and coding the transmission channels with the HOA approach. The resulting system can therefore be divided in two main parts (see figure 1).

Each source $S_i(\vec{p})$ is defined by a signal or sound file, and a time dependent position $\vec{p}(t) = p(\varphi(t), \vartheta(t), r(t))$ (azimuth, elevation and distance) related to the system origin, which is identical to the ideal listening position.

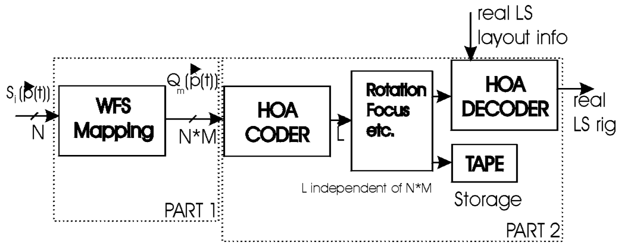

Figure 1: *System model.*

In the first part each source signal is mapped under defined conditions to virtual loudspeaker feeds $Q_i(\vec{p})$. These loudspeakers are located at a defined fixed distance r_0 that depends on the average distance of the real loudspeaker array. According to the Ambisonics assumptions the loudspeakers should be placed symmetrically on a circuit in the 2D case, and arranged symmetrically on the surface of a sphere in the 3D case [7].

In the second part these loudspeaker feeds are coded to the HOA domain. In the HOA domain different sound field manipulations can be performed (e.g. sound field rotation, acoustic focusing, etc.). Afterwards the Ambisonics signals are decoded to the existing real loudspeaker rig.

2. CODING SCHEME

In the following we consider the 2 dimensional case, reducing additional mathematically complexity. We assume that our virtual loudspeaker rig is positioned on a circuit around the ideal listening position at a defined fixed distance r_0. The spacing between each virtual loudspeaker can be regarded as infinitely small. Therefore each real source position at distance r with $(r_0 - \delta < r < r_0 + \varepsilon)$, whereby δ and ε define the possible projection area, can be associated with a specific unique virtual loudspeaker position. This position is defined by the intersection of the virtual circuit with the connecting line of the origin to the source position. First the source is mapped to the virtual loudspeaker rig in the sense of WFS. Therefore a specific amount of adjacent virtual loudspeakers around this defined location is used to reproduce the sound field caused by the source.

2.1. Calculation of the driving functions

In the following the involved loudspeaker feeds are derived. In figure 2 the scheme of the virtual loudspeaker placement is given. The driving functions of the required virtual loudspeakers are obtained by using the WSF approach. Starting from the "Kirchhoff-Helmholtz-Integral" given in equation 1 and equation 2, describing the propagation of sound, we obtain by using the Rayleigh I integral and the stationary phase condition the driving functions Q_m (a detailed derivation is given in [3]).

$$P(\vec{r},\varpi) = S(\varpi)\cdot D(\varphi,\vartheta,\varpi)\cdot \frac{e^{-jk\vec{r}}}{|\vec{r}|} \qquad (2)$$

Whereby $S(\varpi)$ is the Fourier transformed source signal, D is the directivity characteristic of the source (whereby in the following it is set to 1), and k is the wave number.

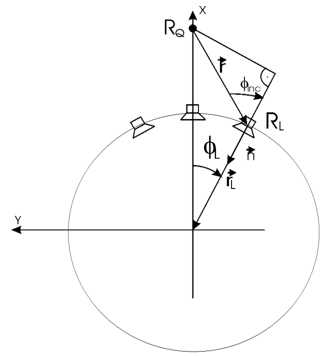

Figure 2: *Scheme of virtual loudspeaker placement.*

$$Q_m(\vec{r},\varpi) = S(\varpi)\cdot\sqrt{\frac{jk}{2\pi}}\cdot\sqrt{\frac{|\Delta\vec{r}|}{|\vec{r}|+|\Delta\vec{r}|}}\cdot D\cdot\cos\varphi_{inc}\cdot\frac{e^{-jk\vec{r}}}{\sqrt{|\vec{r}|}} \qquad (3)$$

After some modifications using the geometrical relations $r_0 = |\vec{r}_L|$, $r_Q = |\vec{r}_Q|$, and $|\Delta\vec{r}| = |\vec{r}_L| = r_0$ we obtain:

$$r = |\vec{r}| = \sqrt{{r_0}^2 + {r_Q}^2 - 2\cdot {r_0}^2\cdot {r_Q}^2\cdot\cos\phi_L} \qquad (4)$$

$$\cos\varphi_{inc} = \frac{r_Q\cdot\cos\phi_L - r_0}{r} \qquad (6)$$

In the case $r_Q > r_0$, whereby r is related to position r_Q:

$$Q_m(r,\varpi) = S(\varpi)\cdot\sqrt{\frac{jk}{2\pi}}\cdot\sqrt{\frac{r_0}{r_0+r}}\cdot\frac{r_Q\cdot\cos\phi_L - r_0}{r}\cdot\frac{e^{-jkr}}{\sqrt{r}} \qquad (7)$$

For $r_Q < r_0$:

$$Q_m(r,\varpi) = S(\varpi)\cdot\sqrt{\frac{k}{j2\pi}}\cdot\sqrt{\frac{r_0}{r_0-r}}\cdot\frac{r_Q\cdot\cos\phi_L - r_0}{r}\cdot\frac{e^{-jkr}}{\sqrt{r}} \qquad (8)$$

And for $r_Q = r_0$:

$$Q_m(\varpi) = S(\varpi) \qquad (9)$$

The number of the adjacent virtual loudspeakers and their spacing, defined by ϕ_L, are important design parameters and have to be further investigated.

In the case $r_Q = r_0$, where the real source is placed exactly at the distance of the virtual loudspeaker rig, a unique virtual loudspeaker with the driving function given in equation 9 mimes the source.

Neglecting the propagation property $\frac{1}{\sqrt{r}}$, just focused on the curvature of the sound field, the coding of the distance can be realised by simple gain scaled and phase delayed $\sqrt{\varpi}$ proportional filters (depending on distance ratio r_Q / r_0). Before the filtering process is introduced, the source signal must be pre emphasised with an inverse $\sqrt{\varpi}$ proportional filter.

2.2. Channel coding and audio rendering

In the following coding section, to the Ambisonics domain, each virtual loudspeaker is coded depending on its position and driving function. Out of this data the Ambisonics signals of the virtual loudspeaker rig is calculated. The advantage of this strategy is the upper bounded amount of transmission channels independent of the introduced number of sources. Furthermore in the Ambisonics domain the sound field transformations features (e.g. rotation, acoustical focusing) can be realised easily. Afterwards the Ambisonics signals are decoded to the existing real loudspeaker rig or stored to recording tapes.

3. CONCLUSIONS

The proposed system design enables us to control sound sources around a large auditorium. The reproduced directions of the sources are almost independent of the listening position inside the auditorium. This is achieved by dividing the reproduction of synthesised sound fields into two parts: the describing (virtual) and the rendering part.

First of all reliable objective and subjective measurements on the proposed scheme have to be done. We will investigate the behaviour of the reconstruction error if the number of loudspeakers is not increased respectively to the used system order and also the fade out of higher order signals to avoid spatial aliasing. Further investigations have to be done concerning the reproduction of the real sound field e.g. the microphone arrays.

4. REFERENCES

[1] Berkhout, A.J., ”A Holographic Approach to Acoustic Control“, J. Audio Eng. Soc., 36(12), pp. 977-995, 1988.

[2] Verheijen, E., ”Sound Reproduction by Wave Field Synthesis”, Thesis, TU Delft, 1998.

[3] Start, E.W., ”Direct sound enhancement by wave field synthesis”, Thesis, TU Delft, 1997.

[4] Bamford, J., ”An Analysis of Ambisonic Sound Systems of First and Second Order”, Thesis presented to the University of Waterloo, Waterloo, Ontario, Canada, 1995.

[5] Daniel, Jérôme, ”Acoustic field representation, application to the transmission and the reproduction of complex sound environments in the multimedia context” (Engl. translation), PhD Thesis, 2000.

[6] Malham, D.G. ”Higher order Ambisonic systems for the spatialisation of sound” Proceedings, ICMC99, Beijing, October 1999.

[7] Sontacchi, A: Höldrich R., ”Enhanced 3d sound field synthesis and reproduction system by compensating interfering reflections“, DAFX’00 Verona, Italy, December 2000.