AMBISONICS SYMPOSIUM 2009
June 25-27, Graz

THE B-FORMAT MICROPHONE REVISED

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Abstract: *The B-Format microphone was invented by* PETER CRAVEN *and* MICHAEL GERZON *in the 1970th. Their goal was to have recordings of natural sound providing a spatial impression to the listener with full 3-dimensional information. More recently, 3-dimensional sound fields are described in terms of a Multipole Expansion, also denominated as Higher Order Ambisonics (HOA). Especially in the literature on microphone arrays the Spherical Fourier Transform is found, which is another representation of the same mathematical background.*

This paper revises the B-Format microphone in terms of current microphone array technologies. E. g., the B-Format signal can be seen as an Ambisonics representation of order one. The processing of the B-Format signal was initially described with regard to analogous signal processing and is real-valued, whereas Ambisonics or Multipole computations are usually carried out digitally and are described using complex valued mathematics. Differences of these approaches are explored and verified by simulation examples.

Key words: Soundfield microphone, Higher Order Ambisonics, A-Format, B-Format

1 INTRODUCTION

The B-Format microphone was invented by PETER CRAVEN and MICHAEL GERZON [5] in the 1970th. Their goal was to have recordings of natural sound providing a spatial impression to the listener with full 3-dimensional information. The original idea to obtain such information was to have a coincident setup of microphones as it is used for stereo and surround sound recordings [11]. As the perfect coincidence cannot be achieved due to the finite size of practical microphone, the basic idea is compensate the spatial mismatch by means of signal processing. After that, the output of the four microphones can be seen as signals of one omnidirectional and three figure-of-eight microphones that are perfectly coincident. If only 2-dimensional sound field representations are required, the so called native B-Format microphone can be employed [4], which is also referred as "double mid-side" arrangement [20].

Nowadays, 3-dimensional sound fields are often described in terms of the spherical harmonics decomposition [15, 19], also known as Higher Order Ambisonics [2, 6, 13]. In this context, the B-Format signal holds an Ambisonics signal of first order. Since natural sound fields have order infinity, some approximation errors are to expect. These are already discussed in [11], some investigations to overcome such problems for the B-Format microphone are found in [3].

This paper revises the B-Format microphone in terms of Ambisonics. In section 2 the microphone array setup is described. Section 3 gives an overview to the basic Ambisonics theory and reformulates all signal processing steps in modern terminology. Finally section 4 discusses some shortcomings of the B-Format microphone with regard to Ambisonics.

2 MICROPHONE ARRAY SETUP

The microphone array setup described here refers to the commercially available B-Format microphone DSF-1 of Soundfield [16]. For other implementations the reader is referred to [1].

The capsules of the B-Format microphone are mounted on the edges of a regular tetrahedron [18]. A regular tetrahedron is composed of four triangular faces which meet at each vertex. Using spherical coordinates $\boldsymbol{r} = (r, \theta, \phi)$, the edges $\boldsymbol{r}_l, l = 1 \dots 4$ of the tetrahedron are given by

$$\begin{aligned} \boldsymbol{r}_1 &= (R, \tfrac{\pi}{2} + \theta_{\text{tilt}}, 0), & \boldsymbol{r}_2 &= (R, \tfrac{\pi}{2} - \theta_{\text{tilt}}, \tfrac{\pi}{2}), \\ \boldsymbol{r}_3 &= (R, \tfrac{\pi}{2} + \theta_{\text{tilt}}, \pi), & \boldsymbol{r}_4 &= (R, \tfrac{\pi}{2} - \theta_{\text{tilt}}, \tfrac{3\pi}{2}). \end{aligned} \tag{1}$$

The radius is $R = 1.47\,\text{cm}$ for the B-Format microphone, the tilt of the capsules is $\theta_{\text{tilt}} = \arctan \frac{1}{\sqrt{2}} = 35.26°$ [9].

The choice of the capsules' positions is done with regard to a good spatial coverage on the surface on a sphere [11]. This can also be expressed as spatial sampling scheme. Various spatial sampling schemes with regard to the Ambisonics representation are discussed by RAFAELY [15] or ZOTTER [21].

2.1. A-Format

The A-Format containing the capsules' outputs is determined by the order of their position. Soundfield provides this order using the notation "left front" (LF), "right front" (RF), "left back" (LB), and "right back" (RB) [16]. The output vector containing the A-Format signal is given by

$$\boldsymbol{s}_{\mathrm{A}}(k) = [s_{\mathrm{LF}}(k), s_{\mathrm{RF}}(k), s_{\mathrm{LB}}(k), s_{\mathrm{RB}}(k)]^T. \tag{2}$$

The wave number $k = \frac{\omega}{c}$ indicates the frequency dependency, with c being the velocity of sound. Note that other signal orders are also in use [8]. Next, some characteristics of the microphone capsules themselves are considered.

2.2. Microphone capsules

The polar pattern of the capsules used in the B-Format microphone is cardioid, hence the output signal is written

$$s(\boldsymbol{r},k) = \alpha\, p(\boldsymbol{r},k) - (1-\alpha)\,\rho_0 c\, v_{\mathrm{R}}(\boldsymbol{r},k)), \tag{3}$$

with pressure $p(\boldsymbol{r},k)$ at point $\boldsymbol{r}$ [13]. Constant ρ_0 is the specific density of air, $v_{\mathrm{R}}(\boldsymbol{r},k)$ denotes the radial velocity. The first-order parameter α for the B-Format microphone is found in [9] as $\alpha = \frac{2}{3}$. Capsules with such an α are also termed “sub-cardioid”. This directivity pattern is shown in figure 1 (top) for four capsules at their regarding positions as given in equation (1).

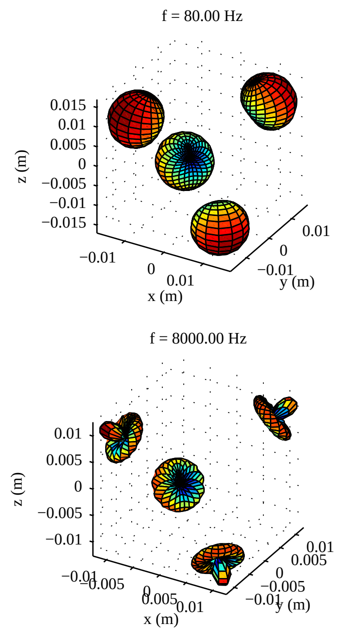

Figure 1: Directivity patterns of the capsules arranged at the positions as given in equation (1).

Besides the polar pattern another important design parameter concerning the capsule is the diameter of the diaphragm. Smaller microphone capsules cause less distortion of the sound field, but also show typically a decreased signal to noise ratio. The size of the capsule causes a decay of level when its size gets in the same dimension as the wavelength and the diaphragm is hit sideways from the impinging wave. This is especially the case for the tetrahedral arrangement used here. Figure 1 (bottom) illustrates this for a frequency of 8 kHz. Deviations from the typical sub-cardioid pattern which arises at sufficient low frequencies (80 Hz in the example on top) are clearly visible.

3 AMBISONICS THEORY

The Ambisonics representation is a sound field description method employing a mathematical approximation of the sound field in one location. The pressure at point $\boldsymbol{r}$ in space is described by

$$p(\boldsymbol{r},k) = \sum_{n=0}^{\infty} \sum_{m=-n}^{n} A_n^m(k)\, j_n(kr)\, Y_n^m(\theta,\phi). \tag{4}$$

Note that this series is sometimes regarded as “Fourier-Bessel-Series” [6] or “Multipole Expansion” [12]. In the literature on microphone arrays equation (4) is often regarded as inverse “Spherical Fourier Transform” [14]. Normally n runs to a finite order N. In the special case of the B-Format the order is $N = 1$. The coefficients $A_n^m(k)$ of the series describe the sound field (assuming sources outside the region of validity [19]), $j_n(kr)$ is the spherical Bessel function of first kind and $Y_n^m(\theta,\phi)$ denote the spherical harmonics. Coefficients $A_n^m(k)$ are regarded as Ambisonics *coefficients* in this context.

3.1. Spherical Harmonics

The spherical harmonics $Y_n^m(\theta,\phi)$ only depend on the angles and describe a function on the unity sphere [19]. They are defined as

$$Y_n^m(\theta,\phi) = \sqrt{\frac{(2n+1)}{4\pi}\frac{(n-m)!}{(n+m)!}}\, P_n^m(\cos\theta)\, \mathrm{e}^{\mathrm{i}m\phi}. \tag{5}$$

The term $P_n^m(\cos\theta)$ denotes the Legendre functions [19], also known as *elevation function*, and finally $\mathrm{e}^{\mathrm{i}m\phi}$ is the azimuth function. The complex-valued spherical harmonics form a complete orthogonal system in the functional space $L^2(S_u)$, where S_u is the surface on the unit sphere [12]. Any set of linear combinations from Y_n^m forms a orthogonal system again. Since the spherical harmonics have the property

$$Y_n^{-|m|}(\theta,\phi) = (-1)^m Y_n^{|m|*}(\theta,\phi). \tag{6}$$

a real-valued function basis can be obtained defining

$$S_n^m = \begin{cases} \frac{(-1)^m}{\sqrt{2}}(Y_n^m + Y_n^{m*}) & m > 0 \\ Y_n^m & m = 0 \\ \frac{-1}{\mathrm{i}\sqrt{2}}(Y_n^m - Y_n^{m*}) & m < 0 \end{cases} \tag{7}$$

In this case the azimuth function of equation (5) is exchanged by

$$\mathrm{trg}_m(\phi) = \begin{cases} \sqrt{2}\cos(m\phi) & m > 0 \\ 1 & m = 0 \\ -\sqrt{2}\sin(m\phi) & m < 0 \end{cases} \tag{8}$$

which leads to the real-valued definition given by equation (7). In the context of the B-Format microphone as well as for microphone arrays often real-valued spherical harmonics are used because some signal processing can be carried out in the time domain.

3.2. Correction filters

In terms of Ambisonics, the tetrahedral microphone setup as shown in section 2 is doing a free-field sphere decomposition [13]. The coefficients $A_n^m(k)$ from such an array are obtained using

$$A_n^m(k) = V_{n,\alpha}(kr) \int_0^{2\pi}\int_0^{\pi} s(\boldsymbol{r},k)\, Y_n^m(\theta,\phi)^* \sin(\theta)\, \mathrm{d}\theta\, \mathrm{d}\phi, \quad (9)$$

where $s(\boldsymbol{r},k)$ denotes the capsules signal as described in equation (3). The function

$$V_{n,\alpha}(kR) = \frac{1}{\alpha\, j_n(kR) - \mathrm{i}(1-\alpha)\, j_n'(kR)} \quad (10)$$

contains the array response in its denominator [17]. These filters are referred as equalisation or non-coincidence correction filters [7, 9]. The B-Format counterparts of the filters $V_{0,\alpha}(kR)$ and $V_{1,\alpha}(kR)$ for analogue filter design are denoted as W and X in [5]. Figure 2 (top) shows the frequency responses of both variants. Note that all gains are set to unity. The filter functions show good accordance up to 10 kHz. Above this frequency, in praxis a diffuse field compensation is added to filter functions W and X of the soundfield microphone. The design of such filters is described in [11] and further discussed in [4]. Here, the diffuse field compensation is left out for the sake of comprehensibility.

Looking at the phase response of both filter implementations in figure 2 (bottom), a difference between filter X and $V_{1,\alpha}(kR)$ becomes visible. The latter adds a phase shift of 90° to the signal, whereas the B-Format filter does not. This is discussed later in section 3.4 of this paper. Further investigations of the correction filters are found in [7, 8].

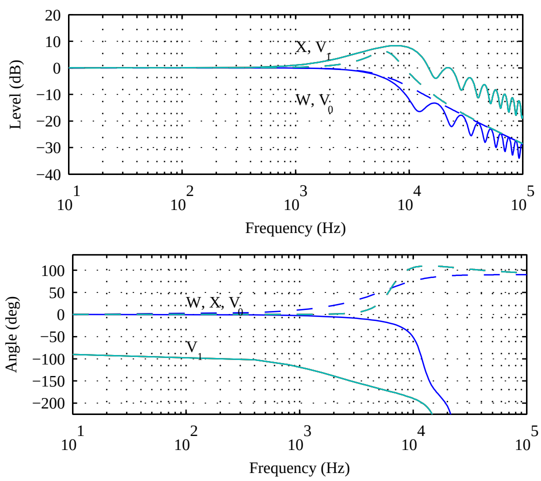

Figure 2: The filter functions for post filtering, from equation (10) "—" and from [5] "– –". All gain factors discussed in the text are set to unity for better comparability.

3.3. Calculation of Ambisonics coefficients

Using L discrete microphones for spatial sampling on a sphere, the integral becomes a sum written as

$$A_n^m(k) = V_{n,\alpha}(kR) \sum_{l=1}^{L} g_l\, s_l(\boldsymbol{r}_l,k)\, Y_n^m(\theta_l,\phi_l)^*. \quad (11)$$

The constant factor g_l is introduced by changing the integral to a sum [10]. The functions $Y_n^m(\theta,\phi)^*$ can be combined to a mode matrix $\boldsymbol{\Psi}$. This matrix holds mode vectors $[\boldsymbol{Y}_1^*, \boldsymbol{Y}_2^*, \boldsymbol{Y}_3^*, \boldsymbol{Y}_4^*]$, where $\boldsymbol{Y}_l(\theta_l,\phi_l) = \left[Y_0^0, Y_1^{-1}, Y_1^0, Y_1^1\right]^T$ holds the respective spherical harmonic values for the series of order $N = 1$. If the real valued version of Y_n^m is chosen and angles are taken from equation (1) the mode matrix can be written

$$\boldsymbol{\Psi} = \frac{1}{\sqrt{4\pi}} \begin{pmatrix} 1 & 1 & 1 & 1 \\ 1 & -1 & 1 & -1 \\ 1 & -1 & -1 & 1 \\ 1 & 1 & -1 & -1 \end{pmatrix} \quad (12)$$

as it is found in [5] omitting the constant factor, taking into account the specific order of the A-Format signals as well as the order of spherical harmonics.

Using the A-Format signal $\boldsymbol{s}_{\mathrm{A}}(k)$ of equation (2) the computation of the Ambisonics coefficients is carried out

$$\boldsymbol{A}(k) = \mathrm{diag}\{\boldsymbol{V}_n(kr)\}\, \boldsymbol{\Psi}\, \boldsymbol{s}_{\mathrm{A}}(k), \quad (13)$$

where vectors $\boldsymbol{V}$ and $\boldsymbol{A}$ are arranged in the same manner as the vectors $\boldsymbol{Y}_l$ in $\boldsymbol{\Psi}$. Note that $V_{0,\alpha}(kr)$ introduces a gain of $\frac{3}{2}$ and $V_{1,\alpha}(kR)$ causes a level gain of 9 when $\alpha = \frac{2}{3}$ is chosen. This is different from the gain factors given in [5] and has to be taken into account when converting a B-Format signal to a Ambisonics representation. More precisely, the related functions of filters $V_{0,\alpha}(kR)$ and $V_{1,\alpha}(kR)$, which are W and X, exhibit gain factors of unity for W and $\sqrt{12}$ for X. An additional gain factor $\sqrt{\frac{3}{2}}$ for the directional signals finally leads to the overall amplification of $\sqrt{18}$ ($\hat{=}$12.6 dB) as mentioned by FARRAR [9]. To obtain an Ambisonics representation from a B-Format signal, therefore a gain of $\sqrt{2}$ ($\hat{=}$3 dB) has to applied to the directional components x, y, z.

The components w, x, y, and z of the B-Format signal are found in the Ambisonics vector $\boldsymbol{A}(k)$ of equation (13) as A_0^0, A_1^1, A_1^{-1}, and A_1^0.

3.4. B-Format signals

The B-Format microphone assumes plane waves, amd therefore the theoretically expected values of Ambisonics coefficients describing a plane wave are derived now. A plane wave impinging from direction $\boldsymbol{k}_i$ is written as

$$p_i(\boldsymbol{r},k_i) = \mathrm{e}^{\mathrm{i}\boldsymbol{k}_i^T \boldsymbol{r}} \quad (14)$$

in the frequency domain [19]. The Ambisonics coefficients describing such a plane wave are [13]

$$A_{n,\mathrm{plane}}^m(\theta_i,\phi_i) = 4\pi\, \mathrm{i}^n\, Y_n^m(\theta_i,\phi_i)^*, \quad (15)$$

and it is easily seen that they are constant in frequency. As equation (4) is a monochromatic description of the sound field, the only varying parameter is $Y_n^m(\theta_i, \phi_i)^*$ which is denoted as Ambisonics *signal* here. The Ambisonics *coefficients* $A_n^k(k)$ as defined earlier in this section carry a 90° phase shift which is introduced by filter $V_{1,\alpha}(kR)$ as shown above. This phase shift in turn is reflected by the factor of i, if order $N = 1$ is assumed.

4 PROPERTIES OF THE B-FORMAT MICROPHONE

As outlined in the introduction, the B-Format signals refer to the directivity patterns of omnidirectional and figure-of-eight microphones. For traditional microphones as well as for microphone arrays these become disturbed when the physical size of the microphone (array) gets similar to half of the wavelength of the sound field [3, 11]. In this section some simulation results illustrate this problem.

In figure 3 the directivity pattern of A_0^0 (or W component of the B-Format signal) is shown for 80 Hz (top) and 8 kHz (bottom). Obviously the error is especially large in regions of the sphere where the capsules of the array have largest distance.

Also for the directivity pattern of the A_1^0 coefficient (or Z component of the B-Format signal) and similarly the A_1^1 and A_1^{-1} components deviate from the ideal figure-of-eight for higher frequencies. For A_1^0 the highest error is found at 45° of incidence in the *xy*-plane. This in turn means a reduced quality of the spatial information carried by the Ambisonics signal. Some aspects to overcome this problems are discussed in [3]. Comparisons of the B-Format microphone directivity patterns with native B-Format microphone implementations are carried out by BENJAMIN [4].

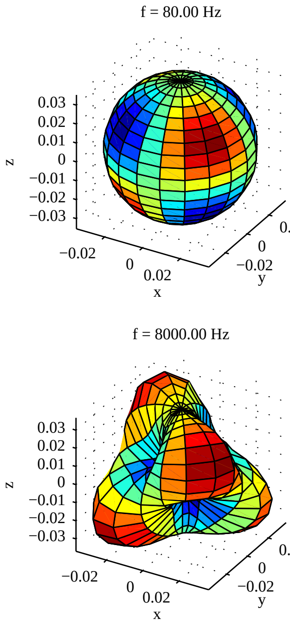

Figure 3: Directivity pattern of the A_0^0 or W component of the B-Format signal. Where the pattern is shaped ideally for 80 Hz (top), strong deviations occur for 8 kHz (bottom).

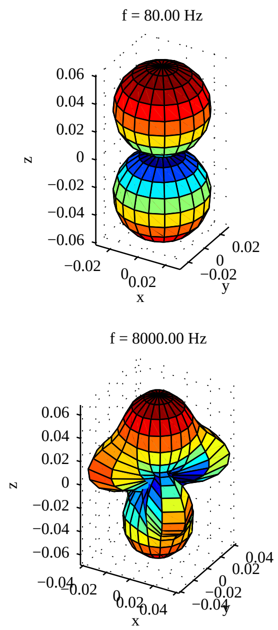

Figure 4: Directivity pattern of the A_1^0 or Z compoent of the B-Format signal. The two other components A_1^1, A_1^{-1} or X, Y are similarly shaped, but with different spatial direction.

5 CONCLUSION

The B-Format microphone was revised in modern terms of terminology of Ambisonics. A detailed investigation of all processing steps gave links from the traditional B-Format processing to current fields of research. Namely the use of new spatial sampling schemes or modified correction filters lets expect further developments for this type of microphone.

REFERENCES

[1] Ambisonia. `http://www.ambisonia.com`. An ambisonic community project by Etienne Deleflie. Last access 2009-05-29.

[2] Jens Ahrens and Sascha Spors. Analytical driving functions for higher order ambisonics. In *Proc. of the ICASSP*, pages 373–376, 2008.

[3] Johann-Markus Batke and Hans-Hermann Hake. Design aspects for an improved b-format microphone. In *European Signal Processing Conference*, 2009. accepted.

[4] Eric Benjamin and Thomas Chen. The native b-format microphone: Part i. In *Audio Engineering Society Preprints*, October 2005. Paper 6621 presented at the 119th Convention.

[5] Peter Graham Craven and Michael Anthony Gerzon. Coincident microphone simulation covering three dimensional space and yielding various directional outputs. United States Patent, 1977. US 4,042,779.

[6] Jérôme Daniel, Rozenn Nicol, and Sébastien Moreau. Further investigations of high order ambisonics and wavefield synthesis for holophonic sound imaging. In *Audio Engineering Society Preprints*, March 2003. Paper 4795 presented at the 114th Convention.

[7] Christof Faller and Mihailo Kolundzija. Design and limitations of non-coincidence correction filters for soundfield microphones. In *Audio Engineering Society Preprints*, May 2009. Convention Paper 7766 presented at the 126th Convention.

[8] Angelo Farina. A-format to b-format conversion. `http://pcfarina.eng.unipr.it/Public/B-format/A2B-conversion/A2B.htm`. last visit 2008-08-08.

[9] Ken Farrar. Soundfield microphone — design and developement of microphone and control unit. *Wireless World*, pages 48–50, 1979.

[10] Jörg Fliege and Ulrike Maier. A two-stage approach for computing cubarure formulae for the sphere. Technical report, Fachbereich Mathematik, Universität Dortmund, 1999. Node numbers are found at `http://www.mathematik.uni-dortmund.de/lsx/research/projects/fliege/nodes/nodes.html`.

[11] Michael A. Gerzon. The design of precisely coincident microphone arrays for stereo and surround sound. In *Audio Engineering Society Preprints*, February 1975. Paper L-20 presented at the 50th Convention.

[12] Nail A. Gumerov and Ramani Duraiswami. *Fast Multipole Methods for the Helmholtz Equation in Three Dimensions*. Elsevier, first edition edition, 2004.

[13] M. A. Poletti. Three-dimensional surround sound systems based on spherical harmonics. *J. Audio Eng. Soc.*, 53(11):1004–1025, November 2005.

[14] Boaz Rafaely. Analysis and design of spherical microphone arrays. *IEEE Transactions on Speech and Audio Processing*, 13(1):135–143, January 2005.

[15] Boaz Rafaely. Spatial aliasing in spherical microphone arrays. *IEEE Transactions on Signal Processing*, 55(3):1003–1010, March 2007.

[16] *DSF-1 Performance Microphone System – User Guide*. Soundfield Ltd., West Yorkshire, England, 1.0 edition.

[17] Heinz Teutsch. *Modal Array Signal Processing: Principles and Applications of Acoustic Wavefield Decomposition*. Springer-Verlag, Berlin, 2007.

[18] Wikipedia. Tetrahedron. `http://en.wikipedia.org/wiki/Tetrahedron`, last visit 2008-09-03.

[19] Earl G. Williams. *Fourier Acoustics*. Academic Press, 1999.

[20] Jörg Wuttke. Surround recording of music: Problems and solutions. In *Audio Engineering Society Preprints*, October 2005. Paper 6556 presented at the 119th Convention.

[21] Franz Zotter. Sampling strategies for acoustic holography/holophony on the sphere. In *NAG/DAGA*, March 2009.