

# DESIGN ASPECTS FOR AN IMPROVED B-FORMAT MICROPHONE

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## ABSTRACT

The B-Format microphone has been used since more than 20 years for recordings of natural 3-dimensional sound fields. It outputs one pressure signal and three directional signals. Acoustical measurements carried out in this contribution show some systematic quality limitations of directional signals. A simulation environment is verified against these real-world measurements. It is used then to vary the design parameters of the B-Format microphone. Technical possibilities for quality improvements of the directional signals are identified and discussed.

## 1. INTRODUCTION

The B-Format microphone was invented by PETER CRAVEN and MICHAEL GERZON [2] 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 the spherical harmonics decomposition [16, 12], also known as Higher Order Ambisonics [3, 11]. In this context, the B-Format holds an Ambisonics signal of first order.

The B-Format microphone is formed by a tetrahedral microphone array. It delivers a pressure signal  $w$  and directional signals  $x, y, z$  proportional to the velocity of air in direction of the respective Cartesian coordinates. As an example, a wave front impinging in direction of the  $x$ -axis should result in a pressure component  $w$  and an  $x$ -component, whereas the  $y$ - and  $z$ -components should vanish. Measurements quickly show that this is not the case. Theoretically this problem becomes clear when the finite size of the microphone array is considered. Spatial aliasing occurs when dimensions of the array are similar to half of the wavelength of the sound field, which is discussed by RAFAELY especially with regard to higher orders in [12]. In contrast, this paper focusses on the enhancement of the output of the B-Format microphone, i. e. staying at a first order representation.

For our evaluation a modified DSF-1 system of Soundfield was chosen as a professional design. It comprises the microphone array and a digital control unit that acts as an A-Format to B-Format converter. The A-Format denotes the unprocessed output of the microphone capsules. An extra switch enables the user to select this A-Format or the B-Format as output of the control unit.

The A-Format to B-Format conversion is described with regard to analogue signal processing in the literature of the 1970th [2, 7, 5]. Also the current literature on microphone describes such a conversion [12]. The B-Format signals can be easily converted to the Ambisonics representation by a gain adaption of the directional signals. Since this is often misunderstood, this paper summarises the processing chain of the B-Format microphone in terms of Ambisonics in detail. This also covers the complex valued spherical harmonic

functions occurring in the Ambisonics theory in opposite to the real valued B-Format processing and filter design aspects.

In the following section 2, the setup of the microphone array and the properties of the microphone capsules are outlined. After that, an elaborated theoretical description of the A-Format to B-Format conversion will be given covering the analogue roots of this technology. The expected results for a simulation model of the microphone array are considered. Section 3 compares the actual acoustic measurements using the DSF-1 B-Format microphone with this simulation model. To do so, the output of the Soundfield B-Format signal is compared with the results of an own A-Format to B-Format conversion. After a thorough analysis of the limitations of the currently employed microphone setup, section 4 highlights possible improvements of the B-Format microphone using the simulation environment.

## 2. THEORETICAL BACKGROUND

### 2.1 Array setup

The capsules of the typical B-Format microphone are mounted on the edges of a regular tetrahedron [15]. A regular tetrahedron is composed of four triangular faces which meet at each vertex. Using spherical coordinates  $\mathbf{r} = (r, \theta, \phi)$ , the edges  $\mathbf{r}_l, l = 1 \dots 4$  of the tetrahedron are given by

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

The radius is  $R = 1.47$  cm for the B-Format microphone, the tilt of the capsules is  $\theta_{\text{tilt}} = \arctan \frac{1}{\sqrt{2}} = 35.26^\circ$  [5]. Figure 1 shows the resulting positions.

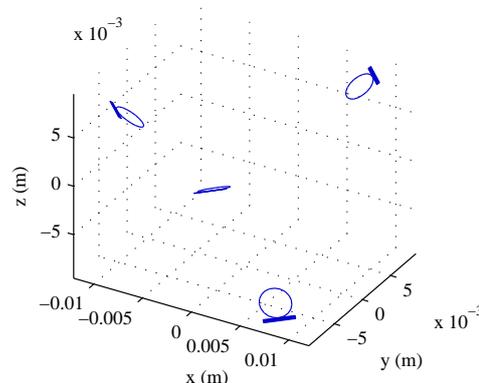


Figure 1: The tetrahedral setup of the capsules in the Soundfield microphone.

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) [13]. The output vector containing the A-Format signal is given by

$$\mathbf{s}_A(k) = [s_{LF}(k), s_{RF}(k), s_{LB}(k), s_{RB}(k)]^T. \quad (2)$$

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 [4]. Next, some characteristics of the microphone capsules themselves are considered.

## 2.2 Microphone capsules

The polar pattern of the capsules used in the DSF-1 is cardioid, hence the output signal is written

$$s(\mathbf{r}, k) = \alpha p(\mathbf{r}, k) - (1 - \alpha) \rho_0 c v_R(\mathbf{r}, k), \quad (3)$$

with pressure  $p(\mathbf{r}, k)$  at point  $\mathbf{r}$  [11]. Constant  $\rho_0$  is the specific density of air,  $v_R(\mathbf{r}, k)$  denotes the radial velocity. The first-order parameter  $\alpha$  for the B-Format microphone is found in [5] as  $\alpha = \frac{2}{3}$ . Capsules with such an  $\alpha$  are also termed “sub-cardioid”.

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.

## 2.3 B-Format in terms of Ambisonics

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

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

Note that this series is sometimes regarded as “Fourier-Bessel-Series” [3] or “Multipole Expansion” [8]. 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 [16]),  $j_n(kr)$  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. The spherical harmonics  $Y_n^m(\theta, \phi)$  only depend on the angles and describe a function on the unity sphere [16]. They are defined as

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

The term  $P_n^{|m|}(\cos \theta)$  denotes the Legendre functions [16], also known as *elevation function*, and finally  $e^{im\phi}$  is the azimuth function. For real valued spherical harmonics the latter is exchanged with function

$$\text{trg}_m(\phi) = \begin{cases} \sqrt{2} \cos(m\phi) & m > 0 \\ 1 & m = 0 \\ \sqrt{2} \sin(m\phi) & m < 0 \end{cases} \quad (6)$$

as used in [9].

In terms of Ambisonics, the tetrahedral microphone setup as shown in section 2.1 is doing a free-field sphere decomposition [11]. The coefficients  $A_n^m(k)$  from such an array are obtained using

$$A_n^m(k) = V_{n,\alpha}(kr) \int_0^{2\pi} \int_0^{\pi} s(\mathbf{r}, k) Y_n^m(\theta, \phi)^* \sin(\theta) d\theta d\phi, \quad (7)$$

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

$$V_{n,\alpha}(kR) = \frac{1}{\alpha j_n(kR) - i(1-\alpha)j_n'(kR)} \quad (8)$$

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

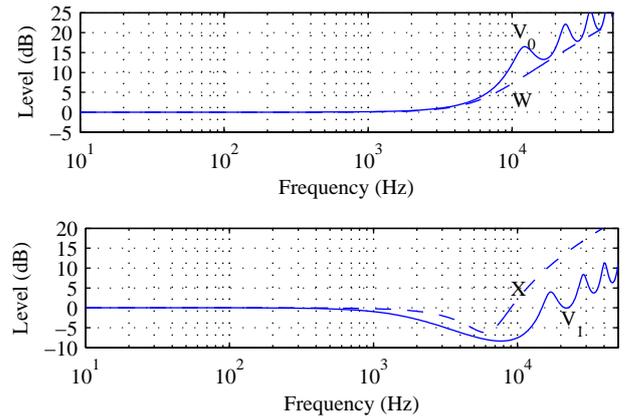


Figure 2: The filter functions for post filtering, from equation (8) – – and from [2] —. All gain factors are set to unity for better comparability.

Using  $L = 4$  microphones for spatial sampling on a sphere, the integral becomes a sum written as

$$A_n^m(k) = V_{n,\alpha}(kR) \sum_{l=1}^L g_l s_l(\mathbf{r}_l, k) Y_n^m(\theta_l, \phi_l)^*. \quad (9)$$

The constant factor  $g_l$  is introduced by changing the integral to a sum [6]. The functions  $Y_n^m(\theta, \phi)^*$  can be combined to a mode matrix  $\Psi$ . This matrix holds mode vectors  $[\mathbf{Y}_1^*, \mathbf{Y}_2^*, \mathbf{Y}_3^*, \mathbf{Y}_4^*]$ , where  $\mathbf{Y}_l(\theta_l, \phi_l) = [Y_0^0, Y_1^{-1}, Y_1^0, Y_1^1]^T$  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

$$\Psi = \frac{1}{\sqrt{4\pi}} \begin{pmatrix} 1 & 1 & 1 & 1 \\ 1 & -1 & 1 & -1 \\ 1 & -1 & -1 & 1 \\ 1 & 1 & -1 & -1 \end{pmatrix} \quad (10)$$

as it is found in [2] omitting the constant factor.

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

$$\mathbf{A}(k) = \text{diag}\{\mathbf{V}_n(kR)\} \mathbf{\Psi} \mathbf{s}_A(k), \quad (11)$$

where vectors  $\mathbf{V}$  and  $\mathbf{A}$  are arranged in the same manner as the vectors  $\mathbf{Y}_j$  in  $\mathbf{\Psi}$ . 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 [2] and has to be taken into account when converting a B-Format signal to an 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  $\sqrt{18}$  ( $\cong 12.6$  dB) as mentioned by FARRAR [5]. To obtain an Ambisonics representation from a B-Format signal, therefore a gain of  $\sqrt{2}$  ( $\cong 3$  dB) has to be 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  $\mathbf{A}(k)$  of equation (11) as  $A_0^0, A_1^1, A_1^{-1},$  and  $A_1^0$ .

## 2.4 Free field response

To describe the characteristics of the B-Format microphone, the free field response is evaluated here. The free field response of a microphone is the output signal when the microphone is exposed to a plane wave. The theoretically expected values of Ambisonics coefficients describing a plane wave are derived firstly, since the output of the B-Format microphone should contain this information.

A plane wave impinging from direction  $\mathbf{k}_i$  is written as

$$p_i(\mathbf{r}, \mathbf{k}_i) = e^{i\mathbf{k}_i^T \mathbf{r}} \quad (12)$$

in the frequency domain [16]. The Ambisonics coefficients describing such a plane wave are [11]

$$A_{n,\text{plane}}^m(\theta_i, \phi_i) = 4\pi i^n Y_n^m(\theta_i, \phi_i)^*, \quad (13)$$

and it is easily seen that they are constant in frequency. A perfect B-Format microphone would output signals matching such coefficients. The simulation model of the B-Format microphone uses equation (11) assuming a plane wave as given by equation (12). The capsules' signals are easily calculated using equation (3).

## 3. MEASUREMENTS

All measurements described here were carried out using a loudspeaker system (JBL LSR 6325), a measurement microphone (Neumann KM 100) and the reference device, a Soundfield DSF-1 B-Format microphone with digital control unit. The distance between microphone and loudspeaker is about 1 m. The measurement setup is depicted in figure 3. Above the Soundfield microphone the measurement microphone is positioned. Using the output of this microphone, the frequency responses of the Soundfield microphone are corrected to compensate the influence of the loudspeaker system (not shown) used for stimulus playback.

All measurements are done twice, the first time capturing the A-Format signal from the modified control unit of



Figure 3: The measurement setup. Above the Soundfield microphone (DSF-1) a measurement microphone (Neumann KM100) is positioned.

the DSF-1, and a second time capturing the B-Format signal. The Soundfield microphone was mounted on a turn table and was rotated in steps of  $\phi_t = n 45^\circ, n = 0 \dots 7$ , whereas the inclination is  $\theta_t = 90^\circ$  in all cases. As mentioned in section 1, the theoretically expected signals for  $\phi = 0^\circ$  are a pressure component  $A_0^0$ , and some  $A_1^1$  component belonging to the  $x$ -axis,  $A_{11}^{-1}$  and  $A_{11}^1$  should vanish. In case of  $\phi = 45^\circ$  the directional components of  $A_1^1$  and  $A_1^{-1}$  should have the same size,  $A_1^0$  should be zero again.

First, the B-Format output of the A-Format to B-Format conversion using Ambisonics is evaluated against the output of the Soundfield system. The plots in figure 4 show the frequency responses of the Ambisonics signal  $\mathbf{A}(k)$  for both conversions with directions  $\phi_t = 0^\circ$  (top) and  $\phi_t = 45^\circ$  (bottom). An  $A_1^0$  component appears in both cases that interestingly depends on the direction. In case of  $\phi_t = 45^\circ$  the  $A_1^0$  component is larger for wider frequency range than for  $\phi = 0^\circ$ . Due to the symmetry of the tetrahedral microphone setup this is the worst case for the spatial quality of the B-Format signal. Moreover for the  $\phi_t = 45^\circ$  direction of incidence it is clearly visible that  $A_1^1$  and  $A_1^{-1}$  are not at the same value. This problem is covered below in the calibration of the simulation system against the measured data. Besides that both outputs, Soundfield conversion and our conversion, show full accordance up to 10 kHz. Above this frequency due to the missing free field compensation in our conversion a significantly higher signal level for the  $A_0^0$  signal is observed. Comb filtering effects occur for the  $A_1^0$  component since the sound field is impaired by reflections of the microphone housing below the tetrahedral setup and the measurement microphone above.

Next, a simulated plane wave recording using equation (3) and (12) is compared with the measured sound field. The Ambisonics coefficients of the simulated sound field are also referred here as output of a simulated microphone. Figure 5 depicts the output of this simulated microphone together with the output of the Ambisonics converter using the recorded A-Format signal. As the B-Format signal contains already the diffuse field correction, deviations in the higher frequency range from the simulated microphone output become visible again.

Strong deviations of synthesised and measured coeffi-

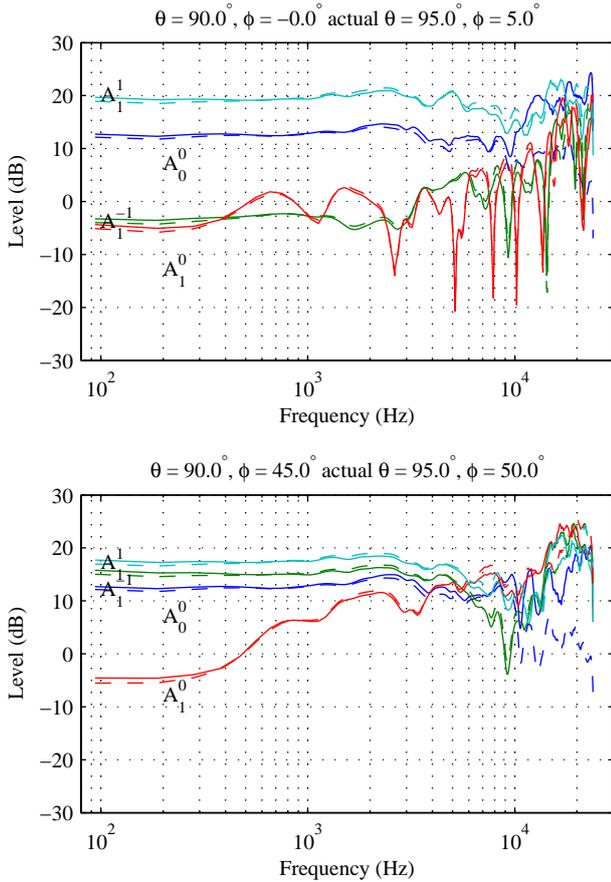


Figure 4: Comparison of Soundfield (– –) and Ambisonics (—) A-Format to B-Format conversion,  $\phi_t = 0^\circ$  (top) and  $\phi_t = 45^\circ$  (bottom).

coefficients occur when the angles of incidence are taken as ( $\theta = 90^\circ, \phi = 0^\circ$ ), since the measured sound field was impinging from a slightly different direction. This is due to the fact that the mechanical calibration of the tetrahedral microphone was done only by sight. Therefore the sound intensity was used to determine the angles of incidence from the measured signals [10]. The resulting angles ( $\theta = 95^\circ, \phi = 5^\circ$ ) fit the measured sound field with reasonable results. In turn, this illustrates the high quality of the DSF-1 output.

#### 4. EVALUATION

The measurements and simulations carried out in section 3 show a height component  $A_1^0$  that should ideally not appear. The reason for this error is spatial aliasing mainly due to the size of the tetrahedral array, but also the size of the capsules. We aim now on get a better suppression of the  $A_1^0$  component which should ideally disappear for all directions of incidence with  $\theta = 90^\circ$ . Since the finite size of the microphone array is the reason for spatial aliasing, first the variation of the array radius  $R$  will be considered. In fact the choice of a smaller radius  $R$  gives better results, especially with respect to higher frequencies. This in turn implies that the usefull frequency range of the coefficients is extended. As an example the coefficients of an impinging wave with  $\phi = 45^\circ$  as the worst case for the  $A_1^0$  component is shown in figure 6. The radius is var-

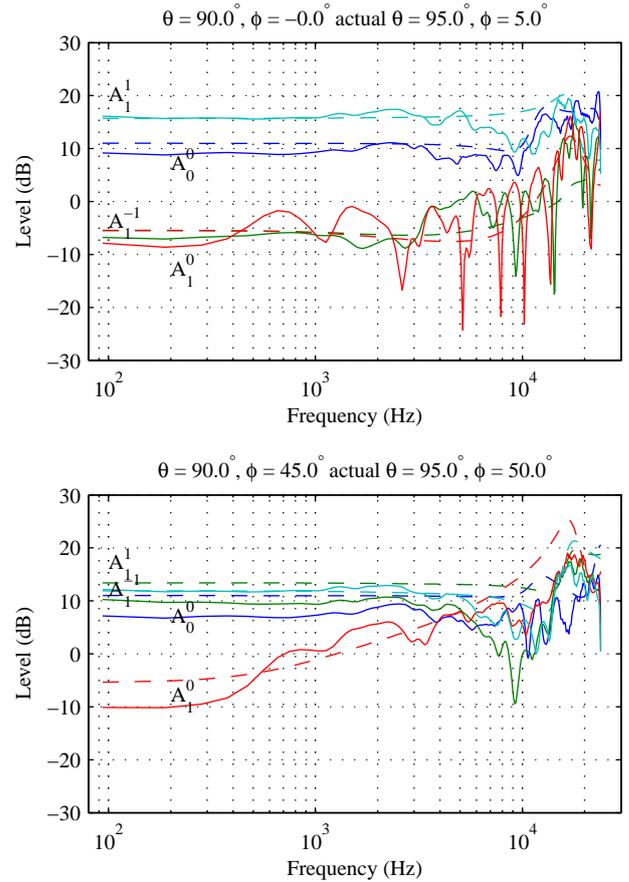


Figure 5: The output of the simulated microphone (– –) vs. real-world signals (—),  $\phi_t = 0^\circ$  (top) and  $\phi_t = 45^\circ$  (bottom).

ied from  $R, \frac{R}{2}, \frac{R}{4}$ , starting with  $R = 1.47$  cm (see section 2.1). The smaller the radius becomes, the lower the unwanted  $A_1^0$  component is for a wider frequency range.

Another less obvious possibility to improve the spatial quality of B-Format signals is to use more capsules, i. e. the microphone array is over-determined with respect to the order of the Ambisonics signal. Over-determined microphone arrays are often found in arrangements for HOA (e.g. [9]), but they are even useful in this first order scenario. Examples are shown in figure 7 using regular arrangements with 4, 9, and 16 microphones [6]. Other arrangements using numbers enumerated from five are found in [14]. A much better behaviour of the  $A_1^0$  component is obtained using an over-determined array. The reason for this improvement is the higher spatial density of capsules on the surface of the sphere. This in turn leads to a better angular resolution of the microphone array resulting in lower approximation errors with regard to the actual sound field.

#### 5. SUMMARY AND CONCLUSIONS

In this contribution the analogue roots of the B-Format microphone were referred to current microphone array literature. More specifically, the A-Format to B-Format conversion using the terms of an Ambisonics representation has been presented. Measurements carried out using a DSF-1 Soundfield microphone have shown good accordance be-

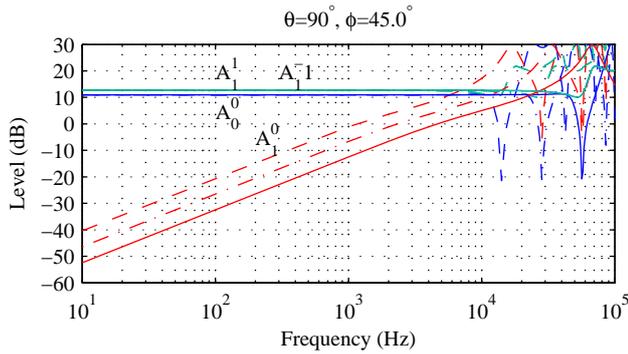


Figure 6: Smaller values for the array radius  $R$  yield better results for  $A_1^0$ . In this example, values of  $R$  (—),  $\frac{R}{2}$  (- ·), and  $\frac{R}{4}$  (—) are used.

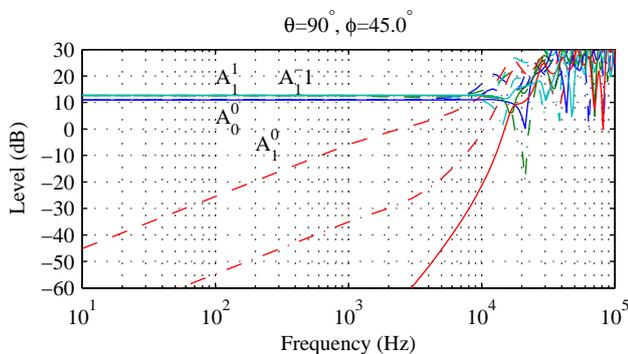


Figure 7: A much better behaviour of the  $A_1^0$  component is obtained using an overdetermined array. In this example, regular arrangements with 4 (—), 9 (- ·), and 16 (—) positions are shown.

tween the B-Format output of the DSF-1 and the conversion as described in this paper. The quality of spatial information of the B-Format signal depends on the direction of incidence which has been illustrated by the presence of a height component in the B-Format signal caused by a plane wave without any height component.

Coefficients from simulations of the B-Format microphone exposed to a plane wave have been compared to measurements. Design parameters have been varied in the simulation environment to obtain a output signal of better quality. As a result, the radius of the tetrahedral array should be chosen as small as possible with respect to spatial aliasing. Spatial aliasing can be further suppressed using an overdetermined microphone array, meaning to use a higher number of microphones than 4 which is the minimum required for the computation of Ambisonics signals with order  $N = 1$ .

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