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ANALYSIS OF EFFECTS OF SPHERICAL MICROPHONE ARRAY PHYSICAL PARAMETERS USING SIMULATIONS*

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Abstract. *Spherical microphone arrays are used for spatial sound field analysis. Although there are commercially available products, they are not the most suitable for research due to their price and working limits of the embedded software. In those cases it is more convenient to build an own prototype in a lab. In this paper, the analysis of the effects of the physical parameters of a spherical microphone array is presented. The observed parameters are radius of the sphere, distance from the sound source and distribution of the microphone elements points over the sphere. The obtained results provide useful inputs for building a spherical microphone array for the desired applications.*

Key words: *spherical microphone array, spherical harmonic decomposition, simulations*

1. INTRODUCTION

Spatial characteristics of sound field have become an important topic in much scientific research, and numerous techniques for their analysis have been developed. One of them is to employ spherical microphone array for spatial recording of the sound field [1]. Spherical microphone array represents an array of microphone elements distributed in a sphere. It is new and still not completely explored concept, so the analysis of its physical and working parameters is an interesting subject for future study.

In order to obtain spatial characteristics of the sound field, a spherical harmonic decomposition has to be performed [1], [2]. Instead of the time domain sampling, here a spatial sampling method has been introduced, where each microphone represents one spatial sample. Features of spherical decomposition depend on the chosen sampling technique characterized by the number and geometrical allocation of samples [1]-[3].

In this paper, the effects of physical parameters of a spherical microphone array on its characteristics have been analyzed with a goal to provide a useful input for the development of a real model, that is, prototype. The considered parameters are radius of a sphere, distance from the sound source and number of microphones (samples). Two different

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types of sphere have been analyzed - an open sphere and closed sphere. The term open sphere refers to an acoustically transparent microphone casing, while the term closed sphere refers to a hard (reflective) microphone enclosure. The mentioned effects are analyzed using the simulations developed in Matlab.

2. SPHERICAL MICROPHONE ARRAYS

As its name dictates, a spherical microphone array consists of a number of transducers allocated over a spherical surface according to some, often geometrically based distribution. Their acoustical properties are affected by number of microphone elements and geometry of their positions.

Spherical microphone arrays have various applications. Spatial analysis and recording of sound field [3], noise cancellation, source localization [4] and beamforming [5] have been the most popular among scientists. An advantage of a spherical microphone array over a conventional directional microphone (e.g. shotgun) lies in a fact that this array is more flexible due to typically large number of transducers and their distribution, but also built in software for generation of a particular directivity pattern. This allows a wide range of different directivity patterns without physical changes to the microphone.

One of important applications of a spherical microphone array is spherical harmonics decomposition [1], [2]. This decomposition depends on spatial coordinates of source and receiver. It is possible to locate a position of the sound source and apply beamforming to capture sound in the desired direction. This feature is applied in video conferencing. In combination with optical cameras, spherical microphone array becomes an acoustic camera, which has wide-spread applications nowadays in industry.

The most popular spherical microphone array model is "Eigenmike" by Mh acoustic (Fig. 1, left). It is composed of 32 microphone elements (distributed over the hard spherical surface) and spatial signal processing software "Eigenbeams". Brüel&Kjær company offers the acoustical camera also made of 32 microphone elements and 12 optical cameras (Fig. 1, right). This model is widely used in the source localization applications. Although there are commercially available microphone arrays, typically they are not suitable for research purposes because of limits in their software, so many researchers develop their own models in laboratory [6].

Fig. 1 Commercially available spherical microphone arrays: "Eigenmike" (left) and Brüel&Kjær's acoustic camera (right)

3. ACOUSTICS IN SPHERICAL COORDINATES

Spherical coordinate system consists of three coordinates denoted as (ρ, θ, φ), where θ is the elevation, φ is the azimuth and ρ represents the radius from origin. Standard Cartesian, defined by the coordinates (x, y, z), and spherical coordinate system are inter-related by the following equation:

$$\begin{bmatrix} x \\ y \\ z \end{bmatrix} = \rho \cdot \begin{bmatrix} \cos(\phi)\sin(\theta) \\ \sin(\phi)\sin(\theta) \\ \cos(\theta) \end{bmatrix}. \quad (1)$$

Sound field can be described by the wave equation written in the spherical coordinate system as [7]-[10]:

$$\frac{1}{\rho}\frac{\partial}{\partial\rho}(\rho^2\frac{\partial p}{\partial\rho}) + \frac{1}{\rho^2\sin\theta}\frac{\partial}{\partial\theta}(\sin\theta\frac{\partial p}{\partial\theta}) + \frac{1}{\rho^2\sin^2\theta}\frac{\partial^2 p}{\partial\phi^2} - \frac{1}{c^2}\frac{\partial^2\Phi}{\partial t^2} = 0\,, \quad (2)$$

where p is the sound pressure, c is the sound velocity and ρ, θ and φ are the spherical coordinates. In this paper, a solution to the wave equation is only briefly discussed, while more detailed discussion can be found in [7].

Equation (2) is a partial differential equation that can be solved by separation of the variables:

$$p(\rho,\theta,\phi,t) = R(\rho)\Theta(\theta)\Phi(\phi)T(t)\,. \quad (3)$$

In the last equation, the function $T(t)$ represents time dependence of the sound pressure, function $R(\rho)$ depends solely on radius and solution can be described as:

$$\begin{aligned} R(\rho) &= R_1 j_n(k\rho) + R_2 y_n(k\rho) \\ R(\rho) &= R_2 h_n^{(1)}(k\rho) + R_2 h_n^{(2)}(k\rho) \end{aligned}, \quad (4)$$

where j_n and y_n are spherical Bessel functions of the first and second kinds, respectively, while $h_n^{(1)}$ and $h_n^{(2)}$ are spherical Hankel functions of the first and second kinds, respectively. Depending on a source position and type of wave (standing or traveling), one of the functions from Eq. (4) can be chosen as a solution for $R(\rho)$. Solution to angular part of Eq. (3) (regarding angle functions - $\Theta(\theta)$, $\Phi(\phi)$) can be combined into a single function called spherical harmonics defined as:

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

where P_n^m is the Legendre polynomial, n is the order of spherical harmonic (n can be any positive integer) and m is the degree taking the values from $-n$ to n including 0 [7]-[10]. Graphical representations of spherical harmonics of certain orders and degrees are given in Fig. 2.

Solutions to Eq. (2) for standing and traveling waves are given by the next two expressions, respectively:

$$p(\rho,\theta,\phi,\omega)=\sum_{n=0}^{\infty}\sum_{n=-m}^{n}(A_{nm}j_n(k\rho)+B_{nm}y_n(k\rho))Y_n^m(\theta,\phi)\ , \tag{6}$$

$$p(\rho,\theta,\phi,\omega)=\sum_{n=0}^{\infty}\sum_{n=-m}^{n}(A_{nm}h_n^{(1)}(k\rho)+B_{nm}h_n^{(2)}(k\rho))Y_n^m(\theta,\phi)\ . \tag{7}$$

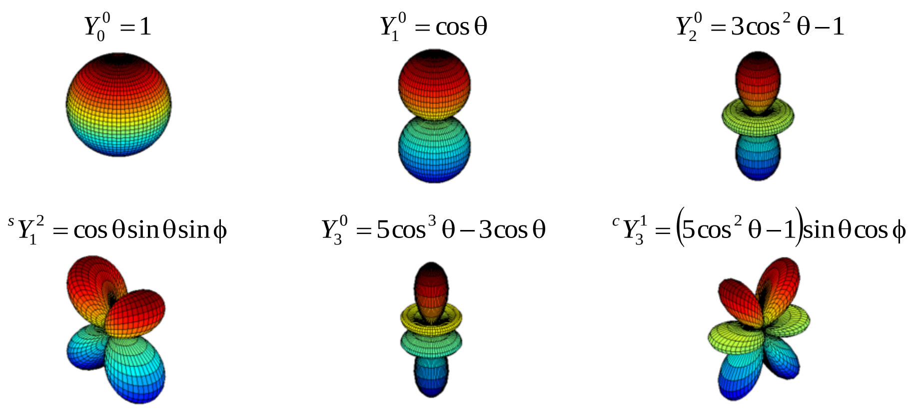

Fig. 2 Spherical harmonics of certain orders (0 to 3) and degrees (0 and 1)

The sound pressure generated on a sphere of radius *r* by the point source positioned at distance *d* from the surface of a sphere can be determined by the equation [2]:

$$p=4\pi\sum_{n=0}^{\infty}i^n b_n(kr,kd)\sum_{m=n}^{n}Y_n^m(\theta,\varphi)\bar{Y}_n^m(\theta_m,\varphi_m)\ , \tag{8}$$

where (θ, φ) represents the direction of an incoming plane wave, that is, the position of the point source in reference to a particular sample point of the array, and (θ_m, φ_m) describes the observation point position. Parameter b_n represents the mode strength and it depends on acoustical properties of the sphere [7]. For acoustically transparent sphere, this parameter is:

$$b_n(kd)=i^n j_n(kd)\ , \tag{9}$$

while for a hard sphere (sphere with reflective surface) it is [7]:

$$b_n(kr,kd)=j_n(kd)-\frac{h_n^{(2)'}(kr)}{j_n^{'}(kr)}h_n^{(2)}(kd)\ . \tag{10}$$

In the first case, Eq. (9), the mode strength only depends on the distance from the sound source (d), while in the second case, it is also related to the radius of sphere (r). This type of representation of sound field is called spherical harmonic decomposition or spherical Fourier transform.

4. SPATIAL SAMPLING

Spherical microphone array performs spatial sampling of wave functions (e.g. sound pressure) defined over the surface of the sphere. Each microphone element represents a sample point. Similar to the time domain sampling, in order to avoid aliasing, there has to be a sufficient number of harmonics (harmonic orders) for the desired frequency range. The higher harmonic order is used, the larger number of sample points (microphone elements) is needed to be considered [1], [6]. There are several spatial sampling techniques that differ in number of samples and sample positions on a sphere [1].

When sample points lie at the intersection of each of latitude and longitude of a grid with the same number of azimuth and elevation lines, sampling is equiangular (see Fig. 3) [1]. Number of microphones needed for this scheme can be calculated based on the desired harmonic order (N) as $4(N+1)^2$. The advantage of equiangular sampling scheme is the simplicity of implementation. Equal number of azimuth and elevation lines provides proper distribution of coordinates, which is useful if the array is replaced with a single microphone that rotates over the sphere. The number of microphone elements needed for this configuration is large, which raises the cost of practical implementation and represents a drawback of the sampling scheme.

Distribution of the samples can be formed based on geometry of some regular polyhedron, such as tetrahedron, dodecahedron or icosahedron [1]. This gives a higher degree of freedom, so different configurations could be implemented. For this scheme, distance between adjacent points on a sphere has to be equal or nearly equal, so samples are usually placed on vertices of polyhedron.

Corresponding schemes are called uniform and nearly uniform sampling (see Fig. 3), and they have similar properties. The same as in the equiangular sampling, number of needed samples is in relation with the maximum desired harmonic order, $1.5(N+1)^2$ [1]. The truncated dodecahedron is a regular polyhedron formed as a combination of dodecahedron and icosahedron containing 32 vertices. If this geometry is used, the harmonics of order 0, 1, 2 and 3 can be observed.

5. SIMULATION METHOD

The simulations presented here have been performed using Matlab. Virtual models for both transparent and hard sphere with equiangular, uniform and nearly uniform sample configurations have been developed and observed.

The equiangular coordinate system where sphere is defined as a latitude-longitude grid is the easiest to implement. Corresponding Matlab function calculates the positions of the sample points for the desired radius of the sphere r and different number of microphones. Each of the positions is determined by the azimuth θ and elevation φ that change from 0 to 2π and 0 to π, respectively. The equiangular sampling scheme configuration is not limed by geometry. Accordingly, huge number of sample points can be placed on a sphere. This is exactly what is done here in order to assume that sample points are so close to each other that the entire surface of the sphere can be considered as pressure sensitive.

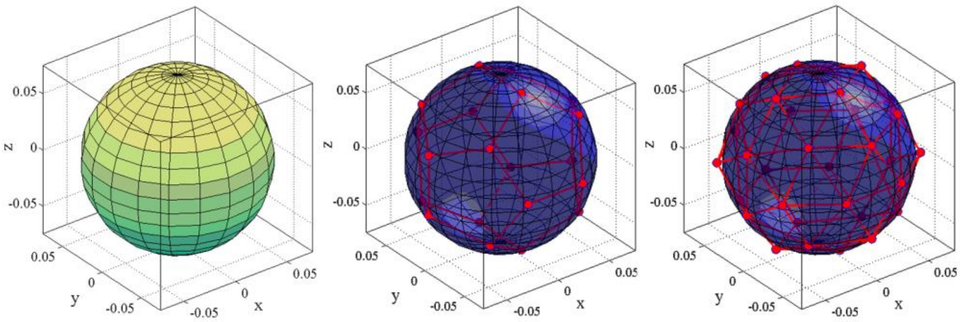

Fig. 3 Equiangular (left), nearly uniform (middle) and uniform (right) sampling configurations

For uniform and nearly uniform sampling configurations, geometries of dodecahedron and truncated dodecahedron have been used, respectively. For these configurations, change of azimuth and elevation angles is not linear as in the case of equiangular scheme, so the coordinate system must be built in a different way. Dodecahedron and truncated dodecahedron are regular polyhedrons, which are very useful since a sphere can be circumscribed around them touching all of their vertices. For the desired radius of sphere r as an input, the Cartesian's coordinates of sample points have been determined by geometry of the platonic solid. The spherical coordinates are then derived from the Cartesian's ones.

From Eq. (8), it can be observed that the sound pressure on a spherical microphone array depends on radius of the sphere r, distance from the sound source d and spherical harmonic order, that is, the number of microphones (related to the spherical harmonic order). This research analyzes the effects of these parameters on the properties of a spherical microphone array. In that regard, sound pressure magnitude (representing the mode strength) of n-th harmonic, b_n, is determined at each microphone position (sample point) for particular sampling scheme. The mean values of the mode strength for different harmonic orders n are observed. The mode strengths for different sampling configurations are mutually compared. The coordinates of the sound source are determined based on the position of one (reference) microphone element (taken as the boundary of the sphere) since the microphone element coordinates are already predetermined. Once they are set, the distance of every other microphone from the source is calculated (see Fig. 4). For positioning of the sound source (and calculation of distances between the sound source and microphones as well as their coordinates), any microphone can be used as a reference one as long as the source lies normal to the surface of the sphere where this microphone element is located. Having this in mind, one microphone is randomly chosen for determination of the source position. This is repeated for several positions of the source as well as for different sphere radii.

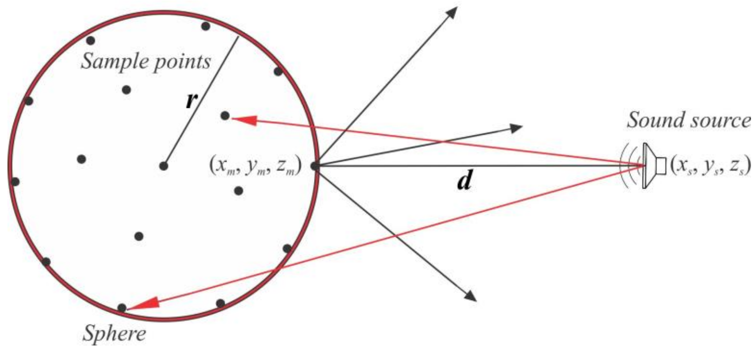

Fig. 4 Method of establishing the microphone and source positions (their coordinates)

6. Open Sphere Model Analysis

As mentioned previously, the term "open" refers to the acoustically transparent sphere. This means that there is no physical presence of the reflective sphere surface (thus, there is no scattered sound field). In this case, only the incident (plane) sound waves create sound field around the microphone array, and mode strength (b_n) only depends on the distance from the source d (see Eq. (9)). Nevertheless, the radius of sphere (r) needs to be defined for determination of coordinates of the sample points.

Equiangular sampling scheme is analyzed using sphere with large number (400) of sampling points. Here, the entire sphere surface can be considered pressure sensitive. Mean values of the mode strength (b_n) are obtained up to the harmonic order 5 for sphere radius of 0.01 m, 0.075 m, 0.15 m and 0.5 m, and they are presented in Fig. 5. The distance from the sound source is set to be 0.1 m.

By increasing the size of the sphere at lower frequencies, the mode strength of all harmonic orders is increased (amplified). Amplification of mode strength for the harmonic order 4 between the spheres of radius 0.01 m (Fig. 5, up-left) and 0.5 m (Fig. 5, down-right) is from-35 dB to -2 dB at 200 Hz, which is a significant value. Spheres with larger radius are useful when the desired frequency band is lower. If a small sphere is used in this case, harmonics of higher orders need to be amplified by an excessive amount. In this way (for example by rising the signal by 33 dB), self-noise of the microphones will be amplified by the same value, which will cause the reduction of the Signal-to-Noise Ratio (SNR). On the other hand, by increasing the size of the sphere, sensitivity of the array (the presented magnitude of the mode strength) for the observed harmonic orders (n) looking towards high frequencies is reduced. As an example, by observing the 1st order harmonic (green dashed curve in Fig 5), it is possible to see that maximum of the pressure magnitude moves towards lower frequencies with increasing the sphere radius, also rising the fluctuations from mid to high frequency range.

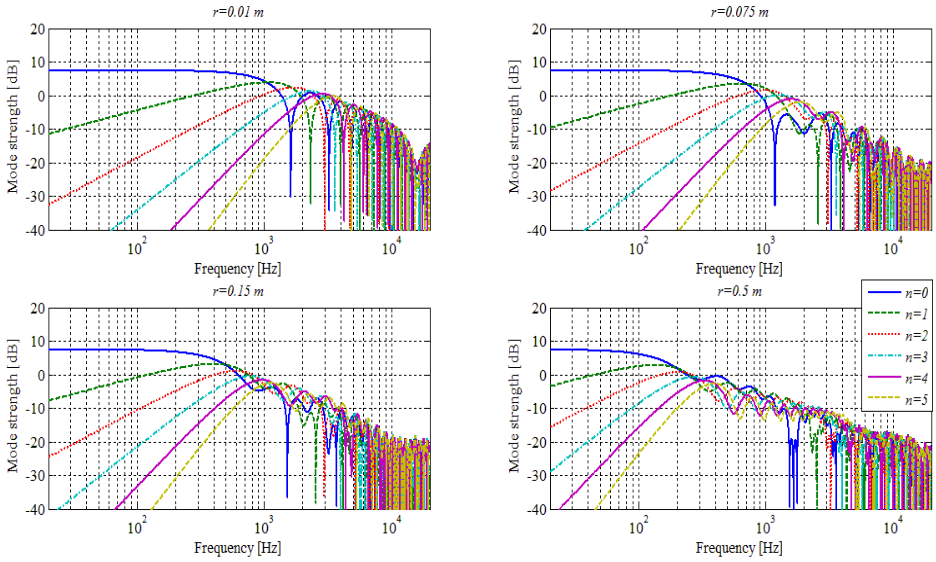

Fig. 5 Mean values of mode strengths (b_n) for different harmonic orders (n) and radius of the sphere (r) in case of open sphere and equiangular sampling configuration

For size of the sphere of 0.075 m (Fig. 5, up-right), the harmonic orders from 3 to 5 (n=3 to 5) are considered to be useful, since their mode strengths have values above 0 dB and they do not develop notches within 1 kHz to 2 kHz frequency band (taken as a representative band where important features are easily visible). For the sphere of radius 0.15 m, only 5th of the observed harmonics is useful, since distortion of previous four starts between 1 and 2 kHz, while for larger sphere (r= 0.5 m, Fig. 5, down-right), distortion is dominant within the entire band from 1 kHz to 2 kHz for all five harmonics. This feature is very important for beamforming. Let us consider an example - if one wants to use 4th order beamformer in the observed frequency band (from 1 kHz to 2 kHz), radius of the sphere should not be larger than 7.5 cm. This implies that for larger spheres, harmonics of higher orders need to be used, and thus a large number of microphone elements has to be included.

In the mentioned band from 1 kHz to 2 kHz, for the sphere size of 0.01 m (Fig. 5, up-left), the harmonics of orders from 0 to 4 have high values of the mode strength (above 0 dB). The 1st harmonic has deep notch at about 1.7 kHz and it is not the most suitable option for this case.

The mean values of the mode strength for the uniform sampling scheme are shown in Fig. 6 in a similar way as in Fig. 5. Here, the truncated dodecahedron (as explained above) is used with 32 sampling points. There are no significant differences in the results from Figs. 5 and 6, especially for smaller spheres of radius 0.01 m (up-left) and 0.075 m (up-right). Equiangular distribution requires large number of microphones, while truncated dodecahedron has 32 sampling points.

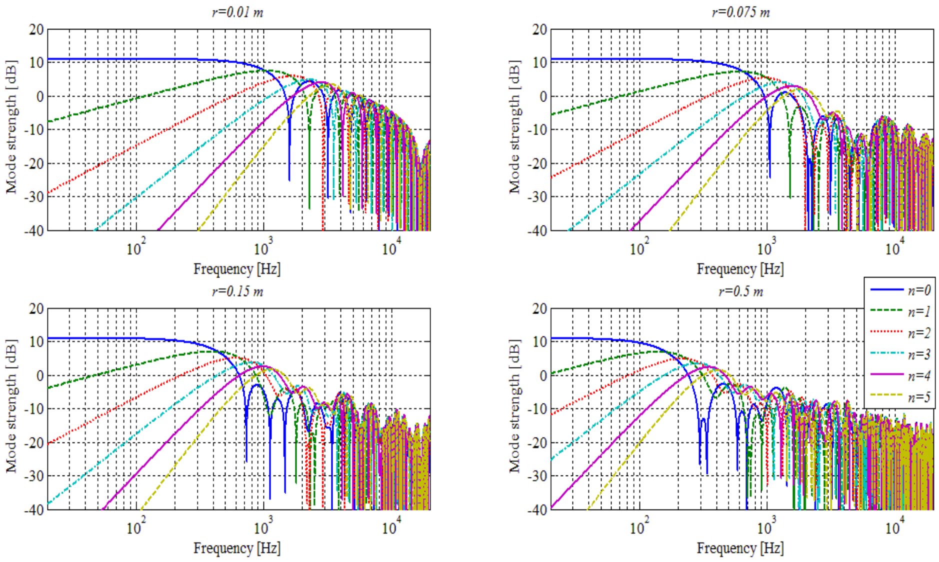

Fig. 6 Mean values of mode strengths (b_n) for different harmonic orders (n) and radius of the sphere (r) in case of open sphere and uniform sampling configuration

It is observed that huge number of microphones distributed on a small sphere in the equiangular configuration does not make a difference when compared to 32 or 20 microphones in the uniform and nearly uniform sampling configurations, respectively. On the other hand, for larger spheres, r=0.15 m (Figs. 5 and 6, down-left) and r=0.5 m (Figs. 5 and 6, down-right), the effect of rising the number of sampling points becomes visible. The mode strength for n=0 and n=1 has notches at lower frequencies in case of the uniform sampling scheme than in case of the equiangular distribution. For n=0 and r=0.15 m, the first notch appears at about 700 Hz in the uniform scheme and 1.6 kHz in the equiangular scheme, while for the same harmonic order and sphere of radius r=0.5 m, the first notch is located at about 300 Hz in the uniform sampling and 1.7 kHz in the equiangular sampling. Consequently, comparing Figs. 5 and 6 (down-right), for a larger sphere, it can be concluded that the frequency range of lower order harmonics widens for equiangular sampling.

The same analysis is also performed for nearly uniform sampling with dodecahedron configuration, and the results (not shown here) are very similar to those from Fig. 6.

7. Hard (Closed) Sphere Model Analysis

Presence of a reflective enclosure leads to the appearance of sound scattered from the reflective surface affecting the overall sound field. The parameter r now represents the radius of rigid surface (sphere). From Eq. (10), it can be concluded that both sphere radius (r) and distance from the source (d) have a certain influence on the mode strength

(b_n), and this is why the results here are presented with regards to those parameters. Again three different sampling schemes are analyzed.

Some results for the hard sphere are the same or very similar to those for the open sphere. Thus, increasing the sample points to a large number (example with pressure sensitive sphere in equiangular sampling) for a smaller sphere does not necessarily improve, that is, increase the mode strength. On the other hand, for a sphere of larger radius, the frequency range of the lower order harmonics widens in equiangular sampling.

By increasing the distance from the sound source (*d*), as a result of the scattered field, a stronger comb filter effect appears, as shown in Fig. 7. Change of the sphere radius has the most prominent effect for shorter distances from the source, where the scattering effect (comb filter effect in the mode strength curves) becomes more observable for larger sphere radius. In this case, the sphere itself becomes large enough in comparison to the sound wavelength at mid and high frequencies.

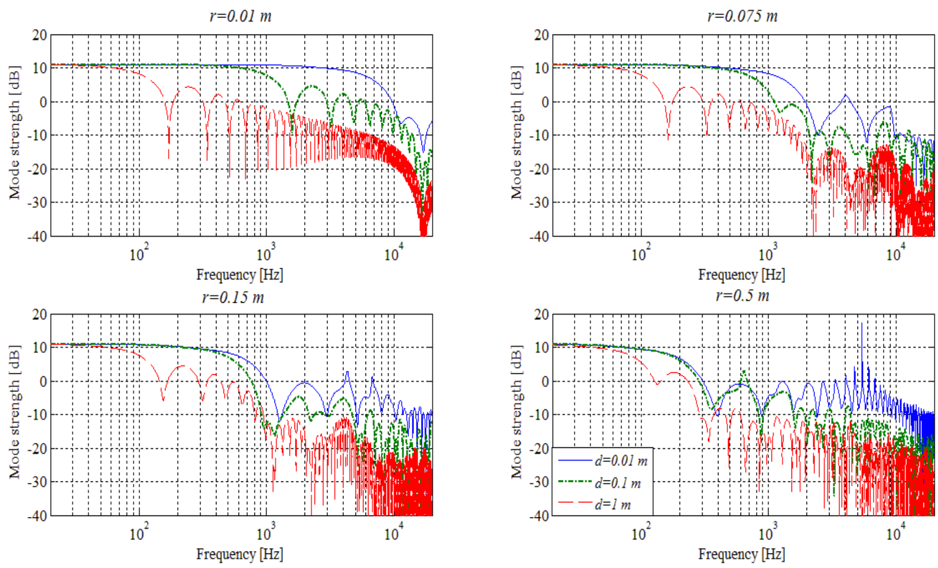

Fig. 7 Mean values of mode strengths (b_n) for harmonic of order 0 and different radii of the sphere (*r*) in case of hard sphere and uniform configuration

For the 3rd order harmonic (*n*=3), the effect of the scattered field becomes clearly visible with an increase of the distance from the sound source, especially for smaller spheres, as shown in Fig. 8 (up-left and up-right). So, by increasing the distance (*d*), maximum of the mode strength is shifted towards lower frequencies and comb filter effect is more prominent. At 100 Hz, the difference between the mode strengths for distances *d*=0.01 m and *d*=1 m is almost 45dB (Fig. 8, up-left). For higher distances from the source (1 m and higher), change in radius of the sphere affects the mode strength by a few dBs translating its max value from 250 Hz (Fig. 8, up-left, red curve) to 150 Hz (Fig. 8, down-right, red curve).

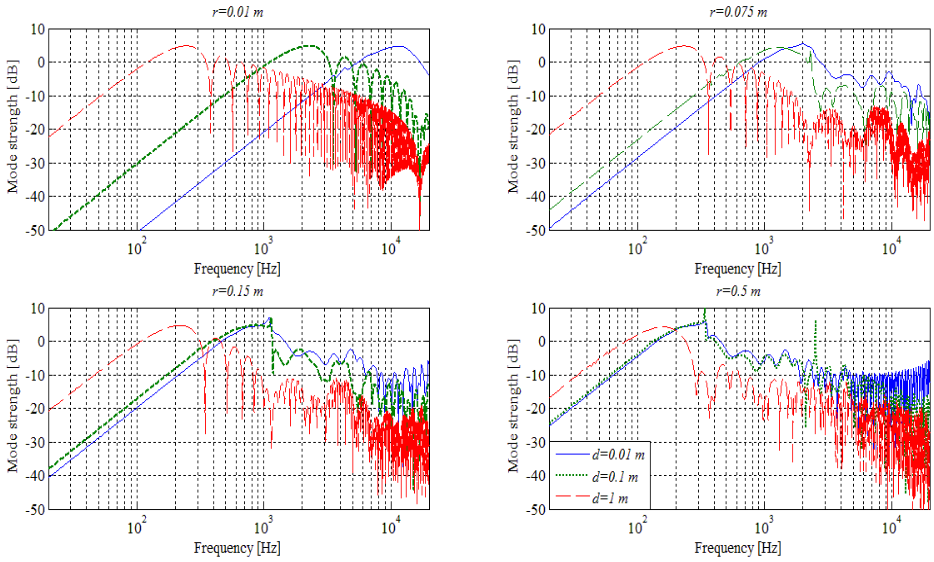

Fig. 8 Mean values of mode strengths (b_n) for 3rd order harmonic and different radius of the sphere (*r*) in case of hard (closed) sphere and uniform configuration

By observing the open (Fig. 6, up-right, cyan curve) and closed sphere results (Fig. 8, up-right, green curve) for radius *r*=0.075 m, distance from the source *d*=0.1 m and harmonic order *n*=3, it can be seen that the mode strength is about 10 dB higher in hard sphere case. This is a big advantage of the hard sphere model, since less amplification of the mode strengths is needed. Also, the hard sphere model possesses less deep notches when compared to the acoustically transparent sphere, which results in a better response at high frequencies.

Similarly as in case of the open sphere, the results for nearly uniform sampling and hard sphere with dodecahedron configuration of the sample points are very similar to the results for uniform sampling presented in Figs. 7 and 8.

8. Practical Implementation of Prototype in Progress

Based on the results obtained in the simulations, the sphere of radius of approximately 7.8 cm with 32 elements and uniform sampling is chosen for a prototype of spherical microphone array. Enclosure of the array will be removable, thus one microphone array can be used as an open and hard sphere model. So far, the frame holding 32 small (electret) microphones for open model (Fig. 9, up-left) has been built, and also microphone amplifier with 32 channels. Each channel has an operational amplifier circuit, shown in Fig. 9, down-left. Microphone amplifier box with input and output connectors, but also layout of the amplifier circuit board are presented in Fig. 9, up-right and low-right, respectively. Signal from each of the microphones is amplified by 35 dB in order to reach a recording sound card with high enough level in comparison to noise. A special setup is being developed for calibration of the array, where relative differences between responses of the microphones will be equalized. Both microphones and amplification circuits will be powered by two Li-Ion batteries providing a symmetric power of ± 7.2 V.

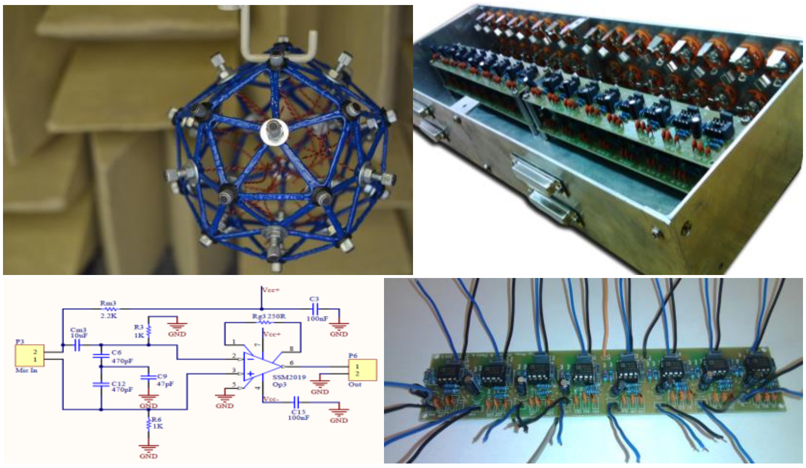

Fig. 9 Prototype of the spherical microphone array: frame holding 32 microphones (up-left), microphone amplifier circuit (down-left), microphone amplifier box (up-right) and amplifier circuit board (down-right)

9. Conclusion

The simulations presented in this paper aim to investigate the effects of physical parameters of the spherical microphone array on its characteristics and to facilitate design of such an array. For designing of an array, a compromise between certain physical parameters is necessary to be made. If too small sphere is used, there is a risk of insufficient mode strength, while too large sphere requires high order harmonics that increases the number of microphone elements needed, and in this way the cost. In case of hard sphere, scattering from the surface creates strong comb filter effects, which, for large spheres, become dominant in mid and high frequency range. The decision regarding microphone parameters should be made as a tradeoff between different requirements that also depends on the desired frequency range and harmonic order. Well defined coordinate system of the sampling configurations builds up a foundation for deeper sound field analysis, beamforming and source localization, which are planned to be the main topics for the future research.

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