Sound recording with the application of microphone arrays

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In this article the issues concerning sound recording with the use of three-dimensional systems of several microphones were considered. The issues under study concern the so called beamforming, which is modeling three-dimensional directivity patterns of microphone arrays, as well as modern technologies of multichannel recording production with the purpose of reproduction in surround sound systems.

1. Introduction

Modern technology of sound recording is, in most cases, based on multi-microphone technology. For this purpose the microphones with specific directivity patterns are used: omnidirectional (donut shaped), figure–8 and cardioid (including the following subgroups: sub-, super-, hyper- cardioid). Application of a single microphone - with a definite directivity pattern - to sound recording enables full control of the recording: in all probability we can predict the magnitude and geometry of an environment in which the sound emitted by particular sources will be successfully recorded. However, while applying the multi-microphone technology the final result is often difficult to predict. In such cases, in sound engineering, the most popular method consists in conducting several test recordings while changing the geometry of the microphone system. It is still more complicated when a produced recording is supposed to be reproduced on a surround sound system, e.g. 5.1. In such a case, apart from fulfilling the requirement of recording the sound from particular sources with high quality, the possibility of apparent sound source planar localization during multichannel reproduction should also be ensured. Bearing in mind all these considerations, it can be claimed that an analytic device enabling simulative microphone system (microphone array) configuration would be extremely useful. In the literature this process is often referred to as “beamforming” [1, 2].

2. Directivity patterns of microphone

The most important properties of microphones are determined by two characteristics: the sensitivity and the directivity pattern. The sensitivity at a specific frequency is the ratio of the voltage at the terminals (output) of a microphone loaded with nominal impedance to the acoustic pressure in the place of location of the microphone [3]:

$$S_u = \frac{U}{p}, \quad \left[\frac{V}{Pa}\right] \tag{1}$$

The directional properties of a microphone are determined by the ratio its of sensitivity at any direction of incidence of a sound wave on the microphone to the sensitivity at a perpendicular direction of incidence on the element receiving the acoustic energy. The range of this ratio in the function of incidence angle of wave is called the directivity pattern [3].

In simulation testing the most frequently modelled pattern is the directivity pattern of a single microphone as a difference of the characteristics of an omnidirectional (pressure) microphone and a figure-8 pattern (pressure – gradient) microphone, located from one another at a distance of l. If τ is the delay between signals of the two microphones, f determines the frequency and θ direction (azimuth angle) from which the wave plane reaches both microphones, then the directivity pattern of the microphone modelled is expressed by the following equation [4]:

$$U(f,\theta) = 1 - A \cdot \exp\left(-j \cdot 2 \cdot \Pi \cdot f \cdot \left(\tau + \frac{l \cdot \cos\theta}{c}\right)\right) \tag{2}$$

where: $A \in < 0, 1 >$, c – speed at which sound travels in the air is equal to 340 m/s. Depending on coefficient A and delay time τ it is possible to simulate a directivity pattern from an omnidirectional to a figure-8 pattern. Figure 1 shows an example pattern of a microphone with a cardioidal pattern.

In Figure 1(b) the axes are dimensionless. Every point of the surface presented represents the sensitivity of the microphone along direction of a surface point to the centre of the co-ordinate system.

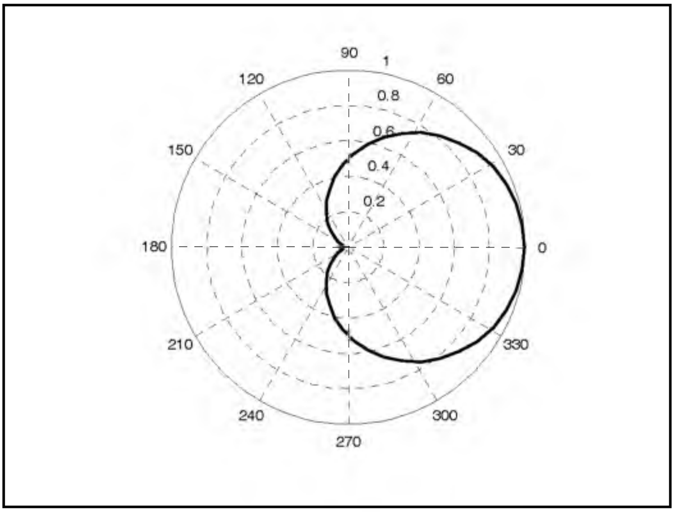
Modelling of directivity patterns of arrays (matrices) of microphones in general can be performed for a near – field or for a far – field.

In case of a near – field it is necessary to take into account the distance of a sound source from individual microphones during calculations. For the far – field it is assumed that the distances between microphones in the array are much smaller than the distance of geometrical centre point of the microphone array from the sound source; the front of acoustic wave is flat.

Furthermore, it is assumed that the far – field case is considered. For an array of microphones located on a horizontal surface, the directivity pattern of such an array can be described using the following equation:

$$U_M(f,\theta) = \sum_m U_m(f, \theta - \alpha_m) \cdot \\ \cdot \exp\left(-j \cdot 2 \cdot \Pi \cdot f \cdot \frac{l_m}{c} \cdot \cos(\theta_m - \theta) \right) \tag{3}$$

a)



b)

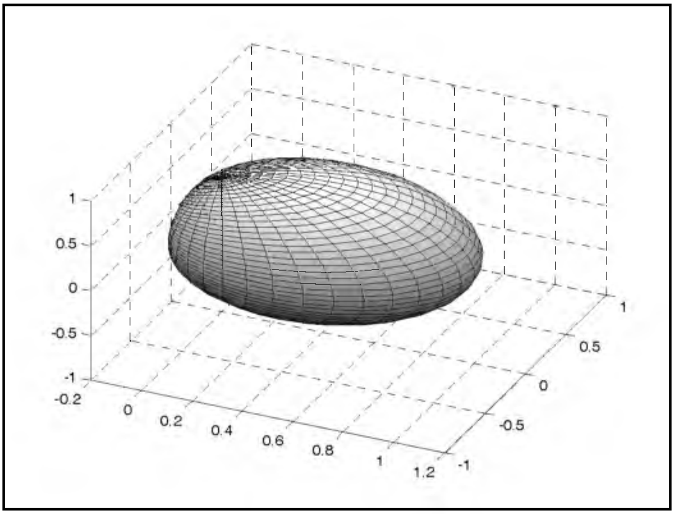


Fig. 1. Directional characteristics of cardioidal microphone for one frequency of 1000Hz: (a) polar plot, (b) 3-D directivity pattern, as a function of elevation and azimuth angle

Formula (3) enables the determination of a directivity pattern U_M for a planar array of microphones, taking into account the directivity patterns U_m of the microphones forming the pattern. The position of each microphone is described by the distance l_m from the centre of the XY co-ordinate grid and the angle θ_m of direction from the microphone to the sound source. Angles α_m are the angles of rotation around the axis of each microphone and are measured in relation to the positive part of axis X in a counter-clockwise direction. The formula (3) can be generalized for the case of a 3-D array including a number of microphones "m" , then:

$$U_M(f,\theta,\varphi) = \sum_m U_m(f,\theta-\alpha_m,\varphi-\beta_m)\cdot \cdot \exp\left(-j\cdot 2\cdot \Pi\cdot f\cdot \frac{l_m}{c}\cdot \cos(\theta_m-\theta)\cdot \cos(\varphi_m-\varphi)\right) \quad (4)$$

where: α_m, β_m – angles of azimuth and elevation of microphone "m", θ_m, φ_m – angles of azimuth and elevation of direction microphone "m" – sound source.

3. Experimental and simulation tests

Tests were performed using the "Atmos" microphone system comprising Brauner's five microphones VM1 mounted on five-armed planar stand ASM5.

The "Atmos" system is designed to make recordings dedicated to surround sound systems 5.0 (5.1). The microphones, all having cardioidal pattern, record the following signals: C – front center (center), LF and RF – front left and front right, LR and RR – rear left and rear right respectively. Each microphone can be rotated around its axis within the range +/– 90^0 in relation to the axis of a matching stand arm. The construction of the array is symmetrical to the front-rear axis and its geometry is shown in Fig. 2.

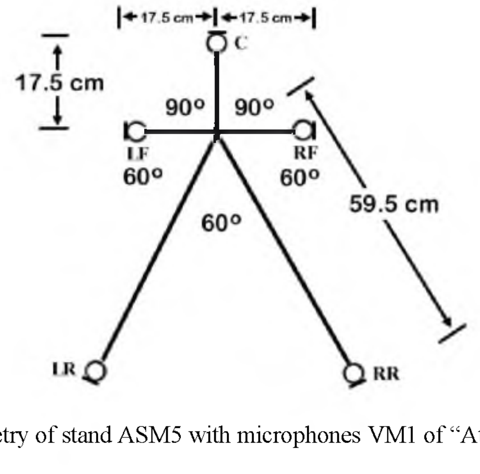


Fig. 2. Geometry of stand ASM5 with microphones VM1 of "Atmos" system

The producer recommends that during the recording the axes of the microphones` maximum sensitivity should overlay the axes of respective arms of the stand (standard set-up).

The "Atmos" array was used several times to do test recordings of concerts performed by the symphony orchestra at the Philharmonic in Szczecin. The signal was recorded by the Zaxcom "Deva" digital recorder with the resolution of 24 bits and the sampling frequency of 96 kHz. During the test the axes of the maximum sensitivity of each microphone overlaid the axes of respective arms of the stand (standard set-up). During the playback of the performance recorded in the way already described, it turned out that, despite loudspeaker systems' set-up fulfilling the standard ITU-R-BS.775-1, in the front sound stage an inaccurate apparent sound source localization could be noticed, with the clear domination of the background (rear channels) and center channel. It can be assumed that the reason for this negative effect lies solely in the wrong set-up of angles of particular microphones. The quality of the microphones themselves is beyond question – the VM1 microphones are among the best studio microphones in the world. In order to possibly validate this hypothesis, simulation tests were conducted using the method described in Section 2 with the assumption that the sound source is located far from the microphone. Hence the simulation tests were conducted for the so called far field.

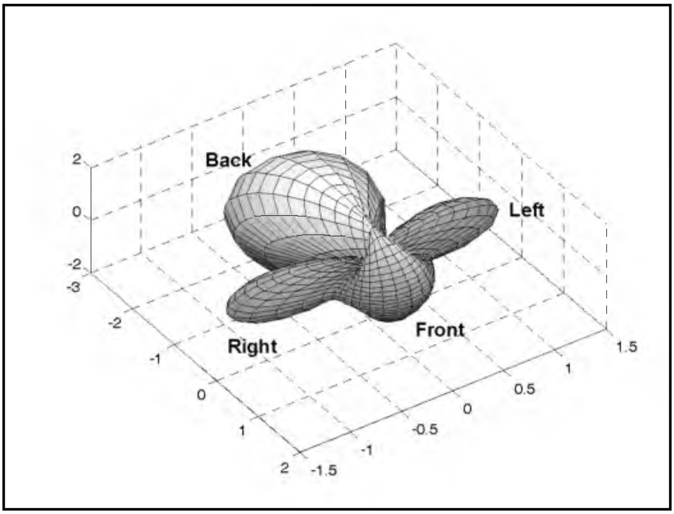
a)

b)

Fig. 3. "Atmos" system: a) system set-up in Studio S1 in Polish Radio in Szczecin, b) recording event at the Philharmonic in Szczecin

The directivity pattern of the “Atmos” system with standard set-up has a shape as shown in Fig. 4.

a)



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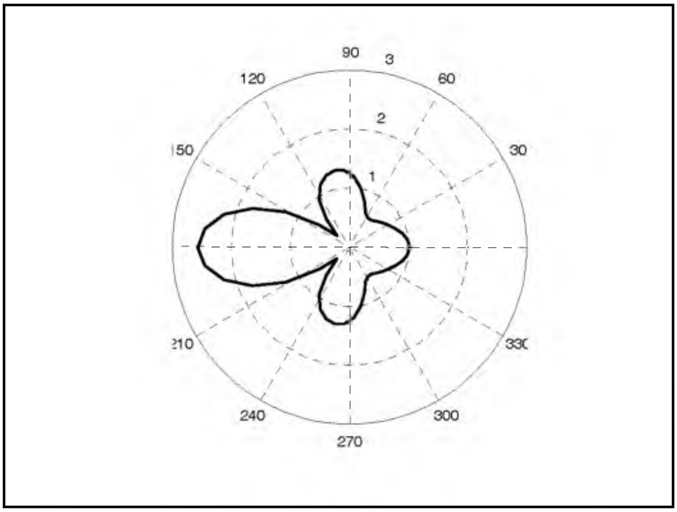
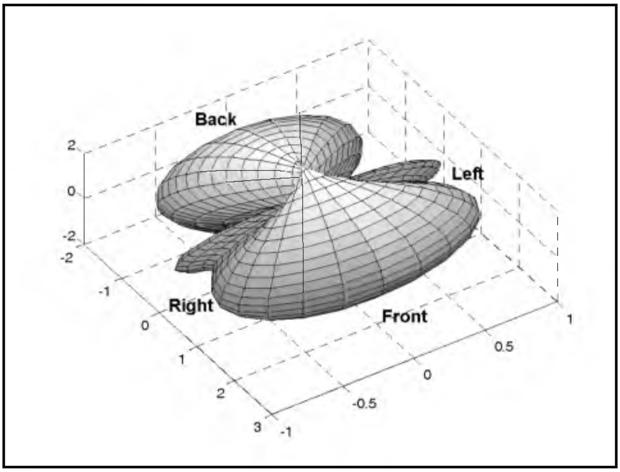


Fig. 4. Directivity pattern of “Atmos” system with microphones with standard set-up angles for the frequency f =500 Hz: a) 3-D graph, b) directivity pattern in the XY plane

In the 3-D graph the axes are dimensionless. Each point of the surface presented represents the effectiveness of array along the direction: surface point – the center of the coordinate system XYZ. The obtained patterns substantiated the observations related to degradation of the front sound stage. The directivity pattern is very uneven (far from omnidirectional) especially in the area of the front sound stage. The following test phase consisted in searching for optimal angles of rotation of particular microphones. While changing these angles, with the use of numerical modeling, such set-ups were sought which would provide maximally omnidirectional directivity pattern, especially in the front sound stage. The tests yielded the following conclusion: the LF microphone should be turned through an angle of $+70^0$ in relation to the standard set-up (clockwise direction of rotation), the RF microphone -70^0, the microphones C, LR and RR should be left as in the standard set-up. The pattern obtained is presented in Fig. 5.

a)



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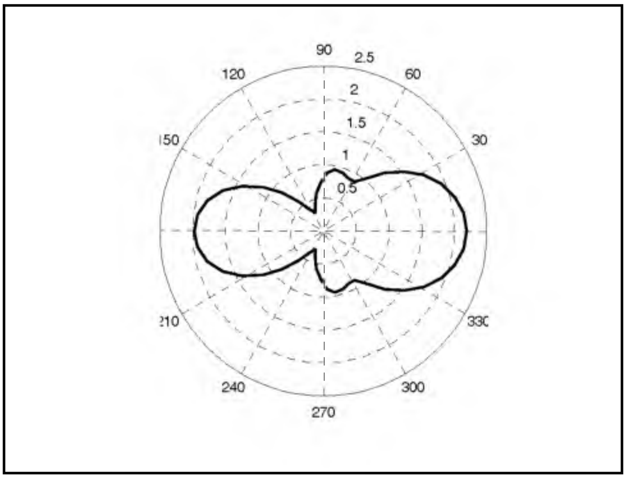

Fig. 5. Directivity pattern of "Atmos" system after the adjustment of angles for the frequency f =500 Hz: a) 3-D graph, b) directivity pattern in the XY plane

Having altered the set-up of the microphones, the recording session in the concert hall took place one more time. Subjective “readability” and the localization of apparent sound sources in the front sound stage was considerably enhanced in comparison with the standard set-up of the microphones.

The recordings were subsequently submitted for subjective verification to the Laboratory of Sound Engineering and Ambiophonics at the Faculty of Electrical Engineering, West Pomeranian University of Technology in Szczecin – Fig. 6.

Fig. 6. Laboratory of Sound Engineering and Ambiophonics at the Faculty of Electrical Engineering, West Pomeranian University of Technology in Szczecin

The tests of that kind are commonly applied. Due to their vital importance in quality assessment of multichannel signals and systems of recording and surround sound playback, the procedure and conditions under which the tests are conducted are standardized [5, 6]. The test - concerning a subjective assessment of the “Atmos” system after its optimization – was conducted with 28 participants, listeners-experts. The participants’ task was to compare the quality of recording samples obtained using the “Atmos” system with the microphones set-up both in a standard and optimized way. The assessment concerned the following parameters:

1. The sound quality of front channels understood as:
   - stability of front sound image,
   - width of front sound stage,
   - precision of apparent sound source localization,
   - sense of appropriate localization of sound sources depending on the type and character of a recorded event (e.g. instrument groups’ localization during the playback of symphony orchestra concert being in line with expectations).
2. The sound quality of rear channels understood as:
   - stability of rear sound image (analogous to Point 1.),
   - coherence of sound space (no feeling of void in certain spot, e.g. directly behind the listener, in the area of rear sound space),

– appropriate arrangement and localization of sound sources in the space.

3. The sense of spatiality, which is:
   – feeling of the size of the place,
   – appropriate length of reverberation time for a given event conditions (for a given space),
   – realism of sound space,
   – sense of "presence": feeling of being in a place where a recording was done as a measure of sound realism,
   – ratio of direct sounds to reflected sounds: if reflected sounds dominate, the sense of the so called "artificial acoustic perspective" arises.
4. Clarity (lucidity), which is:
   – speech clarity,
   – ability to identify and differentiate between voices and sounds to be heard simultaneously,
   – separation of individual short sounds occurring in short time intervals.
5. Balance:
   – dynamics of a played back track (appropriate to the nature of an acoustic event),
   – loudness ratio between front and rear channels.
6. Overall assessment: subjective assessment of a recording comprising formerly described parameters as well as the quality of the recording as a whole and general impression the recording made on the listener.

The listener-expert assessment consisted in completing a questionnaire form and marking individual parameters from 1 (bad) to 6 (excellent).

The results of the tests proved (Fig. 7) that the optimization of the microphones' set-up was especially beneficial to the quality of the front sound stage. Furthermore, the feeling of spatiality of played back recordings was greatly enhanced. And the balance of the sound surrounding the listener was highly marked.

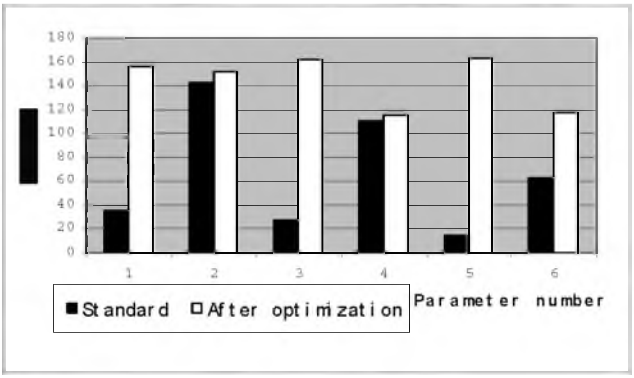

Fig. 7. Test results for the "Atmos" system before and after optimization of rotation angles of microphones

The tests in question concerned the case of recordings conducted in a large concert hall. It can be claimed with all probability that for recording in spaces of small cubic capacity and a small number of sound emitting objects, an optimal configuration of the "Atmos" system will be undoubtedly different from the one presented above. In such a case, simulation tests of patterns should be conducted for the so called near field [7].

References

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