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MEMS Microphone Arrays Structures in Sound Source Localization

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Abstract: *The main characteristics of microphone arrays, usually applied in sound source localization, greatly depend from their structures. Some possible structures of arrangement of the microphones in the arrays are: linear, triangular, circular, spherical, etc. The choice of using each of these structures depend from constructive parameters (size, dimension, etc.) and from concrete application (for example in mobile robots sound localization systems). The relative new MEMS (Micro Electro Mechanical Systems) technology sets some specific requirements related to their little size, digital sound output interface, phase diagram, etc. Therefore, the main goal of this article is to describe and use the already developed microphone array module with MEMS microphones, to propose different structures of MEMS microphone arrays and to test the precision of each of the proposed MEMS microphone array structures in dependence of their electrical and constructive characteristics, important for determination of direction of sound source. In the article are presented briefly the theoretical base of microphone arrays (Direction of Arrival – DOA and Beam Forming) and their related equations describing the main characteristic of microphone array – to localize direction of sound arrival. The presented theoretical basis and equations are then applied in the proposed in this article different structures of MEMS microphone arrays to define and test the quantitative estimation of precision of each proposed MEMS microphone array structures in the determination of direction of sound source.*

Keywords: *MEMS microphones; microphone arrays; sound source localization; style; modules of MEMS microphones*

I. Introduction

Microphone arrays can be regarded as devices or systems suitable to focus the audio attention of intelligent information systems like mobile robots, conference systems, hand free mobile phones devices, etc. in direction of sound source [1]. They are also an artificial model of the human hearing system with their ability to detect and properly determine the direction from which the sound arrive [2]. The applications of microphone arrays is wide spread in the following domains: sound localization for intelligent mobile robot systems [3] human machine interfaces [4] and in video conference systems [5], noise cancelation systems [6], sound quality enhancement in hand free mobile phone devices [7], etc.

There are a lot of methods and algorithms for output microphone arrays signal processing [8]. Most of them are concerning to calculations of the angle of sound source direction of arrival, i.e. to solve the problem of sound source localization. The precision of sound source localization, possible to achieve from each of the existing methods and algorithms using microphone arrays signal processing, depend not only from the effectiveness of the chosen method and algorithm, but also from the chosen microphone arrays structures (linear, triangular, circular, spherical, etc.), from the constructive parameters (size, distance between microphones, etc.) and from concrete application (for example in mobile robots sound localization systems). Despite of these traditional requirements the relative new MEMS (Micro Electro Mechanical Systems) technology is connected with some specific requirements (little size, digital sound output interface, phase diagram, etc.). Therefore, the main goal of this article is to apply the already developed microphone array module with MEMS microphones in proposed different structures of MEMS microphone arrays and to test the precision of each of the proposed MEMS microphone array structures in dependence of their electrical and constructive characteristics, important for determination of direction of sound source. In the article are presented briefly and used the theoretical base of microphone arrays (Direction of Arrival – DOA and Beamforming) and their related equations describing the main characteristic of microphone array – to localize direction of sound arrival.

II. Theoretical Bases of Sound Source Localizations using Microphone Arrays Structures

The most popular and wide spread methods for sound source localization are Direction of Arrival – DOA [9] and Beamforming [10]. In this article are presented briefly and used the theoretical base of microphone arrays

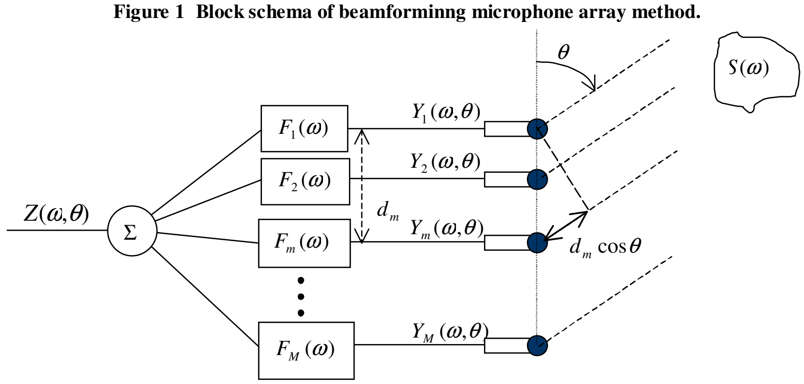
(Direction of Arrival – DOA and Beamforming) and their related equations describing the main characteristic of microphone array – to localize direction of sound source of arrival.
Methods for determining the DOA are based on sound intensity, the phase of cross-spectral functions and cross-correlation functions [11], [12], [13], [14]. The estimate of the DOA is obtained from time-delays and geometry. Direction of arrival (DOA) of the signal can be obtained from the measured time-delays. The time-delays $TDOA_{ij}$ are estimated for each pair of microphones M_i and M_j in the microphone array:

$$TDOA_{ij} = \frac{(\|m_i - s\| - \|m_j - s\|)}{c} \quad , \quad \text{for } i \text{ and } j \in [1, N] \tag{1}$$

where
m_i and m_j are the spatial coordinates of microphones M_i and M_j in the microphone array with $[1 \div N]$ microphones;
s - the spatial coordinates of sound source;
c – the speed of sound in the acoustical medium (usually air).

The values of $TDOA_{ij}$ calculated from equation (1) for each pair of microphones M_i and M_j in the microphone array are used to estimate sound source of arrival like angular direction representing with the value of angle θ. Methods of sound source localization with beamforminng microphone arrays, show in Figure 1 as block schema [15], used the means of multi-channel time varying filtering systems.

Figure 1 Block schema of beamforminng microphone array method.



The following equation describe the beamforming method using a bank of filters $F_m(\omega)$ ($m = 1,2,......M$) for processing in frequency domain the input signals $Y_m(\omega,\theta)$ from each microphone M_m ($m = 1,2,......M$) in beamforming microphone array to achieve the output sound signal $Z(\omega,\theta)$ and use calculated angle θ from equation (2) as useful information for sound source localization:

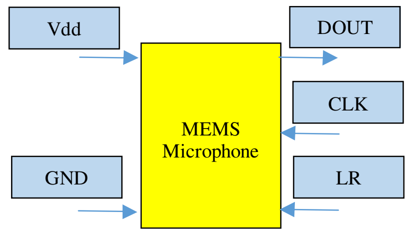
$$Z(\omega,\theta) = \sum F_m^*(\omega) Y_m(\omega,\theta) \tag{2}$$

The brief review above of two most popular groups of methods for sound source localization and they related descriptive equations (1) and (2) allow their comparison in relation of the goal of this article - to apply them in different structures of microphone array in the already developed microphone array module with MEMS microphones [16]. It is evident, that the methods based on determining the DOA with measurement and calculation of time-delays $TDOA_{ij}$ (equation 1) are easily realizable in real-time, unlike the beamforming methods which are computationally intensive (equation 2 and Figure 2). Therefore, the methods based on determining the DOA are very often applied in practical and real working source localization systems because of their simple hardware or software realization, if the precision is not so important and if can be compensate combining sound source localization, for example in robotic or surveillance systems, with visual detection and tracking of speaking person or other object sound source [17], [18], [19].
The results from comparative conclusions can be applied in the proposed here in this article MEMS microphone array structures, testing their precision in dependence of their electrical and constructive characteristics, important for sound source localization. The following characteristics of MEMS microphones are specific and different from other types of microphones: output signal in digital serial format, surface-mount technology (SMT) and SMD compliant, miniature dimension (3x4x1mm).

III. The Models of MEMS Microphone Arrays Structures for Sound Source Localization

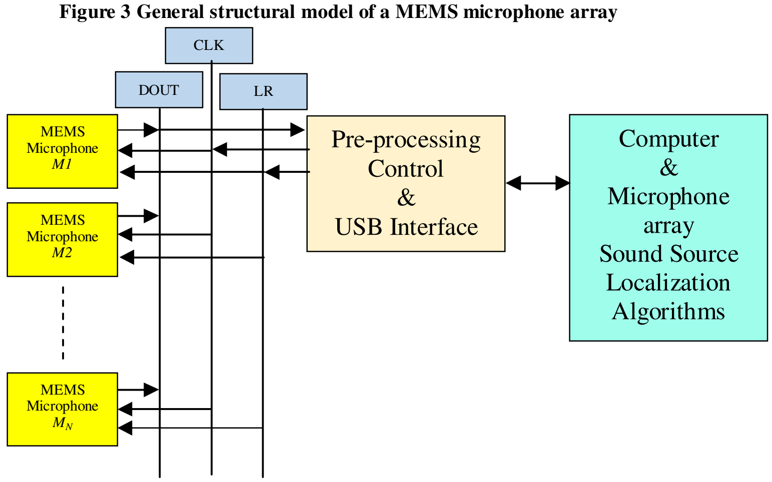
The MEMS technology [20], used in MEMS microphones, which leads to the mentioned above the specific features, can be described building a model of MEMS microphone (Figure 2) as elements of different structures proposed in this article.

Figure 2 Model of MEMS microphones



From Figure 2 is convenient to notice, that for design a general structural model (Figure 3) of a MEMS microphone array, it is important to apply the following signals: one information signal DOUT (Digital Output MEMS microphone signal) in format PDM (Pulse Density Modulation), two control signals CLK (Synchronization Clock Input) and LR (Left/Right control input signal for channel selection).

Figure 3 General structural model of a MEMS microphone array



The general structural model of a MEMS microphone array presented on Figure 3 is already realized as a practical developed microphone array module with MEMS microphones [16]. The main advantages of this microphone array module is the ability of flexible changes of the structure of the developed microphone array module with MEMS microphones. These changes can be the following: the type of MEMS microphones, the number of MEMS microphones, the distance with each pair of microphones in array, the geometric arrangement of microphones, etc.

IV. Testing the The Models of MEMS Microphone Arrays Structures Using the for Sound Source Localization

With the presented on Figure 3 general structural model of a MEMS microphone array and using the practical developed microphone array module with MEMS microphones are prepared the appropriate tests varying the above mentioned important MEMS microphone array characteristics. The tests are made in the following sequence:

- it is necessary to choose the structure of tested microphone array with the corresponding type and the number of the MEMS microphones in the chosen structure of the microphone array;
- to configure the developed microphone array module (using Control block shown in Figure 3) with chosen the structure, corresponding type and the number of the MEMS microphones;
- using the defined configuration of the developed microphone array module to input (in Computer shown in Figure 3) test sound signals from all of the MEMS microphones after pre-processing (in the corresponding block shown in Figure 3) and transferring via USB Interface shown in Figure 3;

- processing inputted with microphone array module test sound signals from all of the MEMS microphones applying developed for this purpose simulation or real time sound localization algorithms;
- analysing the results after processing the test sound signals, when applying different chosen types, number of microphones and their geometric structures in MEMS microphone array module.

The listed above steps of testing different structures of arrangements of MEMS microphones in the microphone array are present briefly in this article as the result achieved after numerous experiments conducted with the developed MEMS microphone array module. Tests are prepared as simulation programs in Matlab [21] using Phased Array System Toolbox. This Toolbox allow to represent the MEMS microphones and MEMS microphone array structures in the form of legitimate in Matlab system objects for object-oriented implementation of algorithms as follow:

Microphones: **phased.OmnidirectionalMicrophoneElement** - Omnidirectional microphone;
phased.CustomMicrophoneElement - Custom microphone;

Microphone array structures:**phased.ULA** – Uniform Linear Microphone Array;
phased.URA – Uniform Rectangular Microphone Array.

In the presented above Omnidirectional and Custom Microphone Elements can be chosen and defined the following main characteristics: **FrequencyRange** (from 20 Hz to 20 kHz) and **FrequencyResponse** (The frequency response in decibels corresponding to the specified frequencies).

In the presented above Uniform Linear and Rectangular Microphone Arrays can be chosen and defined the following main characteristics: **Elements** (The Microphone Elements of the array), **ElementSpacing** (The spacing between Microphone Elements in meters) and **NumElements** (The number of Microphone Elements in the ULA or URA). With these Matlab system objects, describing MEMS microphones and MEMS microphone array structures, are prepared the suitable simulation programs.

The main results from these simulations are summarized in the following Table 1(left part) for a chosen sample frequency f_s = 8 KHz, the structure of MEMS microphone array with number of microphones - 2 and distance between them – 0.022 m. The analysis of results show insufficient number of samples to ensure a suitable precision of sound source localization for real applications as mobile robots motion control. Therefore, from the electrical and constructive characteristics of developed microphone array module is known, that the microphones are placed in a constant distance 0.022 m, but from theoretical point of view (equation 1) is suitable to increase this distance to achieve bigger values of $TDOA$, respectively bigger values of number of samples in Table I and therefore to increase the precision of sound source localization. If the distance between microphones is not possible to change, it is proposed in this article to increase the sample frequency. This is possible for MEMS microphones, because their maximal sample frequency is 2.4 MHZ, which is very high in relation to sound signal maximal sample frequency – 44.1 KHz. In the right part of Table I are shown other results from for a chosen sample frequency f_s = 192 KHz. These results demonstrate the approximatively increasing of number of samples, respectively the precision of sound source localization, which is very important in applications like mobile robots motion control.

Table I The results from testing MEMS microphone array

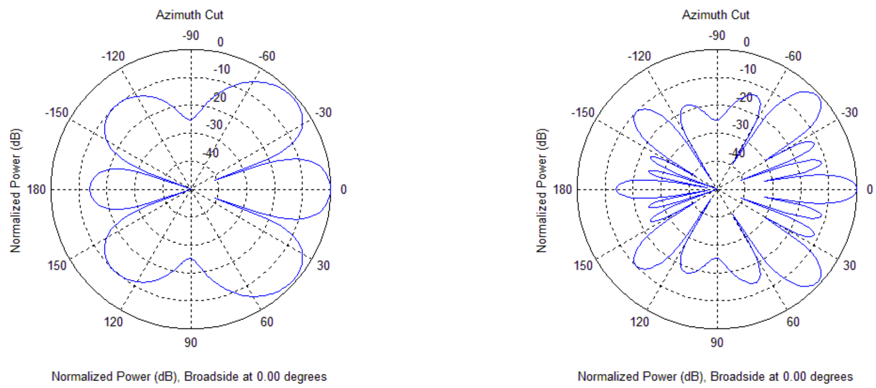
Sample Frequency 8 KHz Number of microphones - 2 Distance between microphones – 0.022 m		
Sound source direction in degree	Calculated *TDOA* in seconds	The number of samples corresponding to calculated *TDOA*
10	0.0715	19
20	0.0670	18
30	0.0618	17
40	0.0566	15
50	0.0454	12
60	0.0281	10
70	0.0225	6
80	0.0045	5
90	0.0000	0

Sample Frequency 192 KHz Number of microphones - 2 Distance between microphones – 0.022 m		
Sound source direction in degree	Calculated *TDOA* in seconds	The number of samples corresponding to calculated *TDOA*
10	0.0715	80
20	0.0670	76
30	0.0618	74
40	0.0566	60
50	0.0454	48
60	0.0281	36
70	0.0225	24
80	0.0045	20
90	0.0000	0

Other important characteristic – polar diagram of microphone array response as results from testing different configurations, structures and characteristics of the developed microphone array module is presented in Figure 4 for two cases of number of microphones (left – for two microphones and right – for four microphones). It is shown the tendency of increasing the precision of the direction of arrival, when the number of microphones in the microphone array is bigger.

The little part of the results from testing the different cases of characteristics of the developed microphone array module are presented when only one parameter is changed, but this particular result can be analyzed and used to make decision for an assembly optimal choice of all microphone array characteristics to achieve a concrete optimal precision in a real working microphone array with MEMS microphones.

Figure 4 Polar diagram of microphone array response to the sound source arrival



V. Conclusion

It is possible to conclude, that the goal set in the beginning of this article - to use the developed microphone array module with MEMS microphones, to propose different structures of MEMS microphone arrays and to test the precision of each of the proposed MEMS microphone array structures in dependence of their electrical and constructive characteristics, important for determination of direction of sound source is satisfy. The results achieved after numerous experiments conducted with the developed MEMS microphone array module show the possibility to use the prepared simulation programs in many cases when it is necessary to estimate a concrete design of MEMS microphone array structure in dependence of their electrical and constructive characteristics, important for precise determination of direction of sound source. It is foreseen to extend the prepared simulation programs with adding in Matlab new user defined system objects not only for Uniform Linear and Rectangular Microphone Arrays, but for arbitrary chosen geometrical structures, electrical and constructive characteristics.

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