Enhanced MVDR Beamforming for MEMS Microphone Array

Martin Papež, Karel Vlček

***Abstract*—** Microphone array technology has been widely used for the localization of sound sources. In particular, beamforming is a well-established signal processing method that maps the position of acoustic sources by steering the array transducers toward different directions. In this paper an implementation of DAS and MVDR beamforming were chosen for further simulations and testing, based on the terms of capabilities of MEMS microphone array. Both beamformers were extensively simulated and tested in Matlab. The results were used to compare the two beamformers based on their noise cancelling, frequency range and spatial filtration. The goal of this paper is comparing the accuracy of the MEMS embedded localization system when the different beamforming algorithms are used.

***Keywords*—** Beamfroming, embedded system, Matlab Simulation, MEMS, MVDR, signal processing.

I. Introduction

Processing of wideband signals using microphone array has been used in a number of science disciplines. First attempts of localization and noise filtering are within the 1970s, when the acoustic telescope based on an array of microphones was constructed and applied [1]. The former signal processing suffered from an insufficient performance in computing demands. This issue has already been resolved in the past and the interest has shifted to maximize accuracy and efficiency of broadband signal processing. Sensor arrays are currently utilized in many different applications, ranging from consumer electronics to military systems (Sound source separation, sound navigation, sound imaging, speech recognition or audio-video surveillance).

Beamforming with phased arrays of microphones is a well-established method for visualization of sound fields. However, because the sound field is mapped with a discrete number of microphones, beamforming techniques present intrinsic limitations, specifically the frequency dependence of the array resolution and the appearance of side lobes that contaminate the beamforming map with sometimes unexpected results. In the almost 40 years of development of acoustic array technology, numerous beamforming algorithms, as well as array geometries, have been suggested to improve the overall performance of array systems [2].

Nowadays beamforming is an essential tool widely used in the industry for all sorts of applications, such as vehicle assessment, computer games and surveillance, among others. Depending on the application, the most adequate processing techniques and array geometries vary. An ideal sound source localization system should present a delta function on the focusing direction and nulls elsewhere. However, beamforming presents two inherent limitations; firstly, an imperfect resolution on the focusing direction, due to a main beam instead of a delta function, and secondly, the appearance of sidelobes in directions other than the focusing direction. The frequency range of operation of an array is determined, at low frequencies, by the dimensions of the array, and by the microphone spacing, at high frequencies. However, the dimensions of the array and the number of microphones are usually limited by practical issues, such as the variability of the array and the overall cost of the equipment.

Currently, the problem of wideband signal processing by multiple sensors has established relatively low costs solutions especially due to designing and utilizing the Micro-Electro-Mechanical Systems (MEMS). MEMS achieved even with sufficient power and are used for challenges such as locating and filtering incident broadband signal. Electrical components such as electromagnetic transducer can be improved significantly compared to their integrated counterparts if they are made using. With the integration of such components, the performance of communication circuits will improve, while the total circuit area, power consumption and cost will be reduced. Utilization of embedded signal processing by MEMS microphone array has advantages such as

- price / performance
- compatibility / broad connectivity
- small device size

In practice, the implementation of MEMS systems utilizing microphone arrays is challenging, and its complexity will depend on the application for which the system is designed to be engaged. For source separation and localization there are several important characteristics that the system must have including array spatial resolution, low reverberation and real-time data acquisition and processing. The objective of this research is to present the of simulation MEMS sensor array.

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The results of adjusted embedded system are obtained on the basis of simulations which were created in MATLAB.

II. MICROPHONE ARRAY

Using multiple sensors in arrays has many advantages; however it is also more challenging. As the number of signals increases, the complexity of the electronics to acquire and process the data will grow as well. Such challenge can be quite formidable depending on the number of sensors, processing speed, and complexity of the target application. The design of the acoustic array board was based on three basic requirements;

- good spatial resolution,
- high signal-to-noise ratio (SNR)
- user selectable unidirectional/omnidirectional acoustic aperture (i.e., beam steering).

Spatial resolution is governed by the number of MEMS microphones, inter-microphone distance, and microphone sensitivity. The sensitivity of the array increases monotonically with the number of sensors, and the MEMS microphone type chosen for this array. The spatial sampling rate of this array can be approximated by dividing the sound speed c by the inter-microphone distance d and then further dividing the result by two in order to satisfy the Nyquist-Shannon theorem (1).

$$d \le \frac{\lambda}{2} = \frac{c}{2f}, \quad f_s \ll 2 f_{\max} \tag{1}$$

The MEMS array design takes into account mechanical factors such as array geometry, sensors disposition, microphone reverberation, and form factor. MEMS microphones are mostly packaged with wire bonding on a PCB substrate. Frequency response of MEMS microphones depending on the diameter of holes ranged from 50µm to 300µm, (Figure 1.) [4].

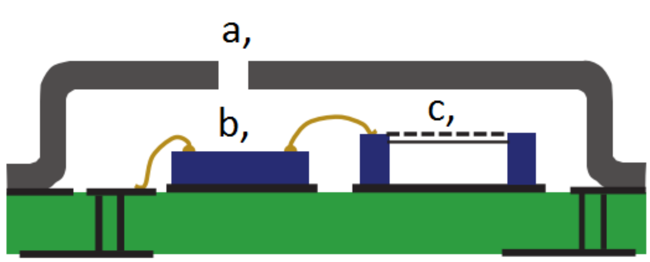

Fig. 1 Wire bounding MEMS microphone (MP34DB01) packed on a PCB substrate (*a, sound hole, b, MEMS c, ASIC*) [3]

The electronics of the array, on the other hand, have demanding issues to deal with, such as

- electronic noise
- power decoupling
- cross-talk
- connectivity

The MEMS microphone MP34DB01 is a fundamental piece of the array due to the small size, high sensitivity, and low reverberation. Its frequency response is essentially flat from 1 kHz - 10 kHz and it has a low limit of 100 Hz and a high limit of 30 kHz [3][4].

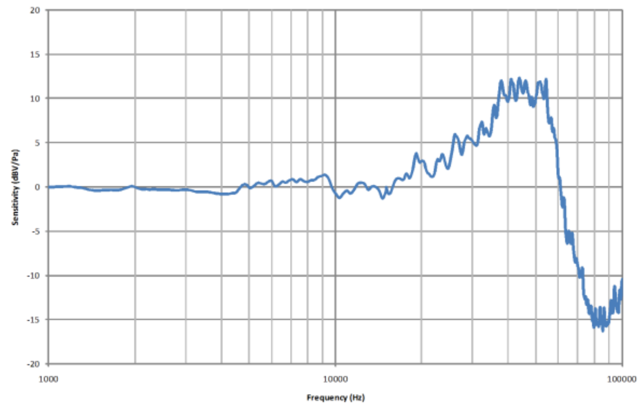

Fig. 2 Frequency response of MP34DB01 normalized to 1 kHz

III. BEAMFORMING

The primary goal of a beamformer array design is to achieve a beampattern with high directivity, a sharp mainlobe in conjunction with small sidelobe levels. Since beamformer is actually the M-point spatial filter, its beampattern is defined as the module of the directional response of this filter [6]. The beampattern is function of the array geometry and position of the source. Moreover, it also depends on the number of microphones used in the array and on the frequency of the signal arriving from the source. Each beampattern consists of specific number of beams having various levels. The number of lobes in the range from 0° to 180° degree is rising with scaling frequency and number of microphones. Uniform linear array of microphones (ULA) is shown in Figure 3. The challenge thereby is to reach this goal with a minimum amount of microphones, with a small array size, and with a robustness against sensor position errors and sensor noise.[7]

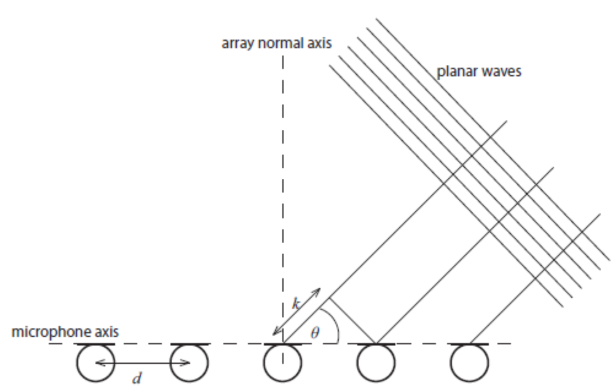

Fig. 3 Linear microphone array with equidistant microphone placement (ULA)

A. DAS Beamforming

Delay-and-sum (DAS) beamforming is one of the simplest array signal processing algorithm. Delay-and-sum array processing can also be applied to a MEMS array by compensating for the delays at the microphones due to a single plane wave, such that all processed microphone outputs have the same amplitude and phase. Figure 4 shows a conventional DAS beamformer for a N microphone array [8].

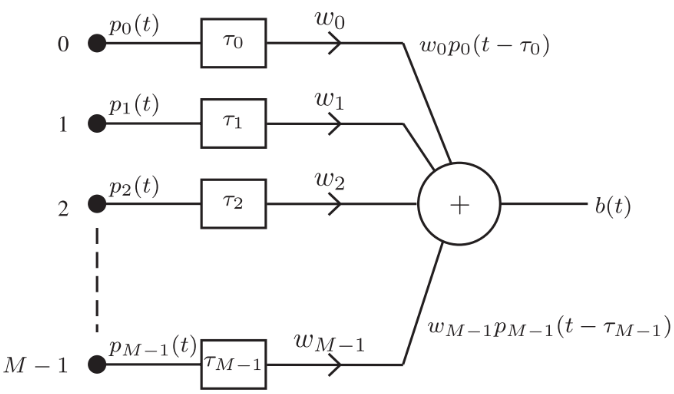

Fig. 4 Principle of DAS beamforming

The principle behind this technique is shown in Figure 4 in the presence of a propagating wave, the signals captured by the microphones are delayed by a proper amount before being added together, to strengthen the resulting signal with respect to noise or waves propagating in other directions (2),

$$y(t,\hat{\kappa}) = \sum_{m=0}^{M-1} x_m \omega_m \left(t - \tau_m(\hat{\kappa})\right), \tag{2}$$

where M is the number of microphones, x_m is the signal amplitude measured with the m microphone, ω_m is its associated amplitude weighting and $\tau_m(\hat{\kappa})$ is the delay applied.

B. MVDR Beamforming

MVDR beamforming is a well-known and extensively used beamforming technique that offers a good spectral characteristic of the output and is therefore well suited to acoustic beamforming and wideband signal enhancement. This method results from modifying DAS beamforming in the frequency domain which offers much higher dynamic range than other techniques. The main advantage of MVDR beamformers is the possibility to improve a given array by software and not by changing the array layout or increasing the number of input sensors. Array interpolation is carried out by preprocessing the N sensor signals to create $M \geq N$ sensor signals. As shown in Figure 5, virtual sensors are introduced by a multichannel mapping system $T\{\ \}$. Array processing like beamforming is then applied to the M virtual sensor signals [8][9].

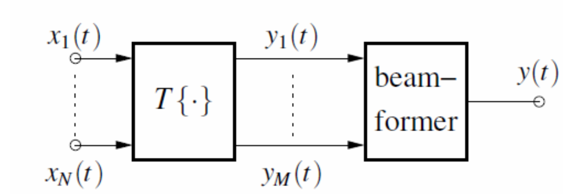

Fig. 5 Mapping of sensor signals $x_i(t)$ to virtual sensor $y_i(t) = T\{x_i(t)\}$

Sensor mapping in Figure 5 is a very general technique that can be fitted to a large variety of array configurations. For each selection of sensor positions, are computed the beamformer weights by a minimum variance distortion less response MVDR design. The aim of MVDR beamforming is to minimize the power of the output signal of the array while maintaining unity gain in the look direction and also maximizing the white noise gain. MVDR beamforming is based on filter-delay-sum beamforming and its frequency domain output signal $Y\left(e^{j\omega}\right)$ is defined as (3) [7] [11],

$$Y\left(e^{j\omega}\right) = \sum_{m=0}^{M-1} w_m\left(e^{j\omega}\right) X_m\left(e^{j\omega}\right) = \mathbf{w}^{\mathbf{H}}\mathbf{X}, \tag{3}$$

where $w_m\left(e^{j\omega}\right)$ denotes the filter coefficients of the beamformer for sensor m at frequency $\omega, X_m\left(e^{j\omega}\right)$, are the microphone input signals and $[\ ]^H$ denotes the matrix transpose conjugate. In addition to is suggested to minimize the total output power $Y_m\left(e^{j\omega}\right)$ under the assumption of a diffuse noise field in order to optimize spatial filtering with respect to reverberant environments. This leads to the super-directive beamformer with weight vector (4),

$$\mathbf{w}(\omega) = \frac{T^{-1}(\omega)\mathbf{v}(\omega)}{\mathbf{v}^H(\omega)T^{-1}(\omega)\mathbf{v}(\omega)}, \tag{4}$$

where $T_{i,j}(\omega)$ denotes the coherence of an isotropic noise field with absolute magnitude matrix of the microphone inter-distance.

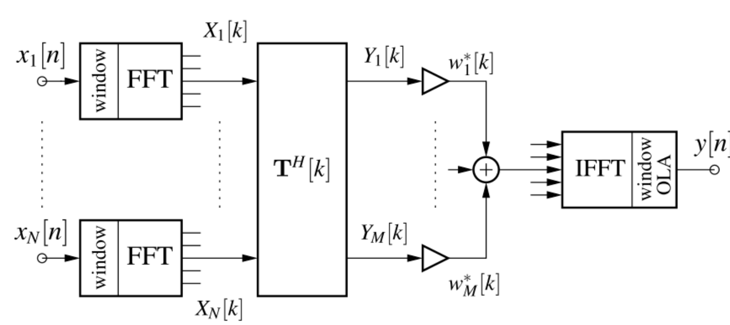

Fig. 6 MVDR Beamformer with FFT filterbank

The output of MVDR beamformer is recovered from evaluation (3) using inverse FFT followed by overlap. The advantages of an MVDR design are its computational efficiency and an easy inclusion of desired spatial nulls in the beampattern [8][11].

IV. SIMULATION OF AN ARRAY

In this section, design of wideband beamformer and simulation results are presented. MEMS microphone array was assembled in MATLAB simulations to verify receiver characteristics, which is used for spatial filtering of the processed signal are obtained from the simulations. Increasing of the number of MEMS microphones in array lead to more accurate signal processing and localization properties. However, it also increasing computing demands. The required relative bandwidth of processed wideband signal was the main criteria for design of MEMS array. Another demand was placed on the size of the device, which should not exceed the maximum size of $0{,}02\ \mathrm{m}^2$. Accordingly to these conditions the minimum distance between individual MEMS sensors was adjusted to $d \geq 0{,}03\ \mathrm{m}$. Due to this conditions, the simulations of array has been limited to a maximum number of MEMS sensors that were used. In the simulations of the uniform array were established the maximum of MEMS sensor to $N = 8$.

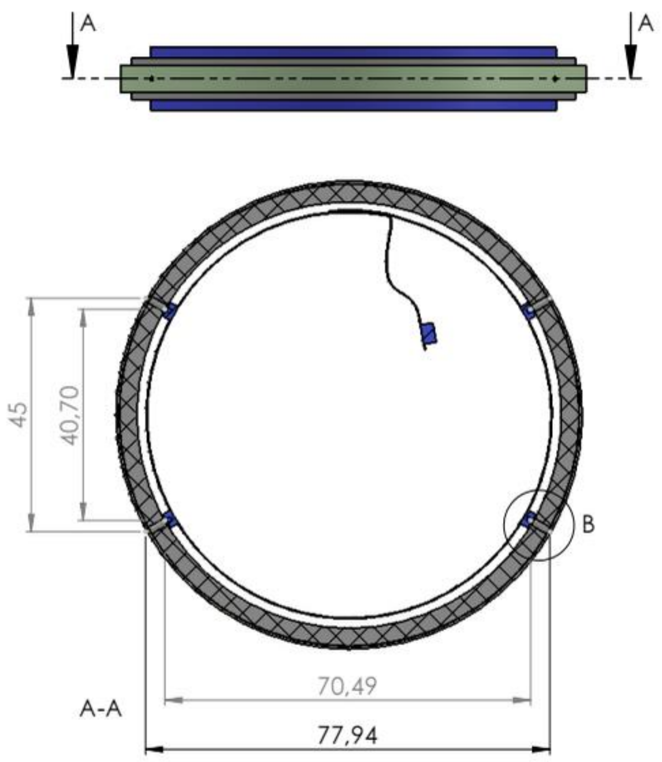

Fig. 7. Geometry of MEMS array, size $0{,}016\ \mathrm{m}^2$, $d \geq 0{,}03\ \mathrm{m}$, $N = 4$

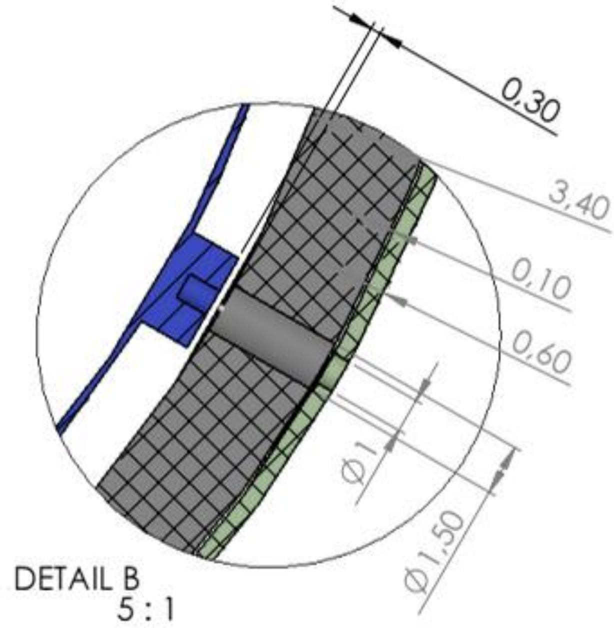

Fig. 8 Detail of placing MEMS microphone

The best MEMS microphone array distribution is illustrated in the Figure 7, 8, where the heterogeneous array structure is used. The array geometry was chosen in order to allow beamsteering on the horizontal and vertical planes. It should be noted that the all simulated array produced similar results frequency band within the range 10 kHz – 30 kHz.

Matlab simulation has been carried out for proposed heterogeneous MEMS array. Beamformers (DAS, MVDR) were considered for incoming signal sources from different angles. The estimation of the weight vector of a null steering beamformer calculated by using weight equation (2), (4). SNR is assumed to be 10 dB for all incoming sources. Based on these conditions and acquaintances wideband signal, array simulations were performed.

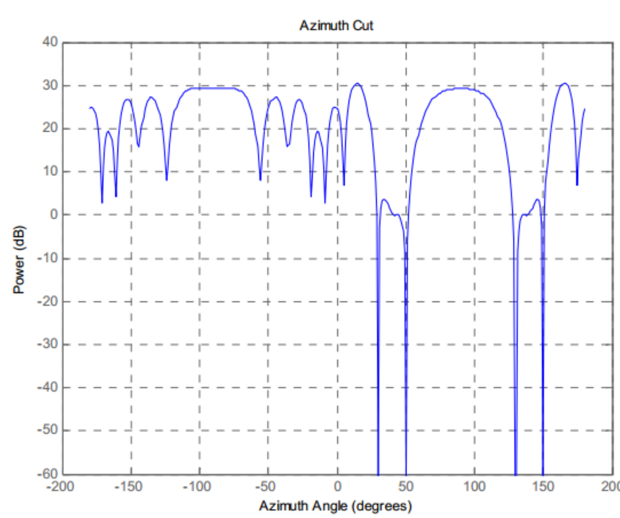

Fig. 9 Power spectrum and polar plot - DAS beamforming

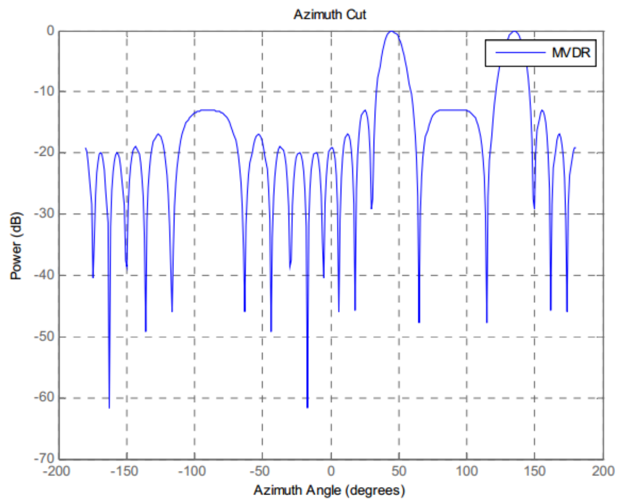

Fig. 10 Power spectrum and polar plot - MVDR beamforming

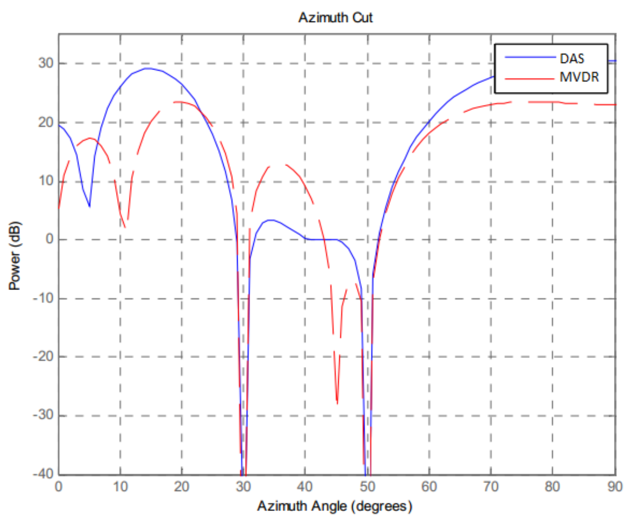

Fig. 11 Comparison between DAS and MVDR beamformer

The simulation results showed that proposed MEMS microphone array provides high accuracy while processing the wideband signal. The relative bandwidth of microphone arrays is much larger than that of wideband arrays operating at the range of frequencies 100 Hz – 30 kHz. As a consequence, microphone arrays are a special challenge for wideband beamformer designs.

Two different beamforming techniques have been examined in Matlab simulation. The normalized outputs obtained for DAS and MVDR beamformers are shown in Figure 9, 10. The effect of constraints can be better seen in Figure 11, when DAS and MVDR beamformers response pattern is compared. It can be noted that the DAS beamformer is able to maintain a flat response region around the 45° degree in azimuth, while the MVDR beamformer creates a null. The advantage of MVDR beamformer is that provides higher dynamic range and can be modified for various application without changing array layout or increasing number of sensors.

V. CONCLUSION

In this research the proposal of MEMS microphone array for wideband signals processing is analyzed and described. Design of MEMS microphone array is dependent on the signals pattern that this system will process. Defined requirements are advisable for use of various array layouts. The design of MEMS microphone array is based on MATLAB simulations. In addition, the proposed array positively affects receiving characteristics, and spatial filtering possibilities of broadband signal. Signal processing is based on limited computing capabilities of this embedded system and therefore it is necessary to take it into account during the design of algorithm. The simulation results show that the MEMS array is capable of performing admirable achievement for sound source separating applications when the algorithm based on DAS Beamforming is used. Furthermore, it has been presented wideband beamformer design method based on MVDR beamforming. This method exhibits smaller computing demands, a smaller array size, and offer a sharper mainlobe than DAS beamformer designs. The combination of chosen MEMS microphone array and MVDR beamformer enables broad scalability for more complex applications such as signal processing, spatial filtration or audio-video surveillance.

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