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Beamforming in near-field - metaheuristic approach and experimental tests in an anechoic chamber

Agnieszka Wielgus, Bogusław Szlachetko, and Michał Łuczyński

***Abstract*— A set of microphones spatially arranged in space in a specific pattern is called a microphone array. It can be used to extract and enhance the signal of interest from its observation corrupted by other interfering signals, such as noise or to estimate the direction of arrival of a source. In this paper we focus on a problem in which the desired signal (speech signal) is interfered by other signal with fully overlapping bandwidth but with different localization. Our goal is to attenuate the interfering signal. We experimentally study the method in which microphones do not have to be equally spaced and all information regarding signal phase is hidden in a transfer function of the microphone. We focus on determining the microphones positions and FIR filter coefficients so that the actual output the beamformer is as close as possible to the desired one (the level of speech signal remains unchanged, while the interfering signal is attenuated) in the sense of l_2 norm. To solve this problem, we use a metaheuristic algorithm. Next, we construct the array and make an experiment in anechoic chamber. The initial results of the experiment show that the proposed method can be applied for array designing.**

***Keywords*—microphone array; noise cancellation; spatial filtering**

I. INTRODUCTION

Spatial filtering is a well-known problem, that appears in many real-life areas, both civil and military, e.g. sound systems, radar or sonar systems [1]-[3]. If there is an array of sensors (i.e., antennas, microphones) we face a problem how to create a correct beam pattern, i.e., how to consolidate signals received by individual sensors. In acoustic, a microphone array can be used as a spatial filter, that sums up the acoustic signals received by individual microphones. It can also be used to separate signals that come from different points in space and their frequencies coincides, thus the classical, temporal filtering cannot be applied. Another important application of microphone arrays is the problem of source localization, i.e. they can be used to estimating the direction of arrival of a signal. Moreover, in real-world applications, speech quality can deteriorate due to, for example, background interference, noise, or reverberation and microphone arrays are applied to extract and enhance the signal of interest from its observation. In case of microphone array and speech signal (broadband signal), the system can be perceived as a set of FIR filters, where each filter has different set of filter coefficients and all outputs of these filters (all signals after filtration process) are summed up. As a result, a new signal is formed. The efficiency and the quality of the solution depend on system configuration, i.e. microphone placement and FIR filter coefficients (their values and filter order). For each placement a different set of coefficients provides the best solution, i.e., for each microphone placement, the set of optimal FIR filter coefficients must be recalculated.

Beamforming and spatial filtering have been studied for many years but are still important and up to date [4], [5]. In the classical approach to array designing, a standard microphone layout (e.g. horizontal, vertical, spherical, or equally spaced in a rectangular array) is used. Next, for a fixed microphone placement, a set of filter coefficients is calculated. In modern, sophisticated systems, arrays with a great number (e.g. a hundred) of unequally spaced microphones are used, however, they are designed based on signal bandwidth (it is considered directly).

Since increasing the array size does not have to improve the system quality, and systems with great number of sensors are characterized by a significant power consumption, it is important to design systems with small number of sensors. One of the approaches is applying the thinning technique on a big array, i.e., to reduce the number of microphones maintaining the desired system response/quality [6]-[10]. Another technique is to formulate the considered issue as the optimization problem and apply modern optimization techniques.

In [11] the authors considered broadband beamforming problem in which the system response should be as close as possible to the desired one in the sense of l_2 norm. The small change in array configuration leads to the change of optimal set of filter coefficients. It means that with each shift of microphone, we must calculate all filter coefficients for all microphones once again. The system quality is improved as the filter length is increased. However, there can be determined a limit of the system performance (performance limit) for a finite length filter, for which further increasing of filter length do not improve the system response [11]. Based on it, the authors have proposed the simultaneous optimization of microphone placement, along with the determination of filter parameters (in two-dimension space). Next, the problem of continuous optimization with constraints was formulated, in which the objective function is non-linear and has many local minima. To

All Authors are with Wrocław University of Science and Technology (e-mail: agnieszka.wielgus@pwr.edu.pl, boguslaw.szlachetko@pwr.edu.pl, michal.luczynski@pwr.edu.pl).

solve it, MATLAB method *fmincon* (the sequential quadratic programming method) was used. Then, for the analogically defined problem, in [12] the authors proposed a hybrid method, which is a combination of a genetic algorithm and a gradient method, and its efficiency has been verified during numerical experiments (simulations).

Since the approaches based on metaheuristics, presented in [12] and [13], look promising, are easy to implement and calculation time is short, we decided to check if the proposed method can be applied in real-life. For this purpose, we decided to hold a preliminary experiment in an anechoic chamber. To design the system, i.e. to determine microphone placement and FIR filters coefficients, we have implemented and run the algorithm based on simulated annealing technique. Based on simulation results, we built a small array in the chamber, and made recordings of speech signal disturbed by the noise. The bandwidth of both signals was the same, thus it was impossible to use classical frequency-domain filters. Next, we formed the beam – we applied the FIR filters for each signal and then summed up all the signals. The main achievement presented in the paper is as follows: a simple approach based on an optimization technique, which has been applied in near-real conditions, while in the literature such a solution was only considered theoretically. The results are promising and encourage for further tests in anechoic chambers and different rooms.

The paper is organized as follows: Section II describes the problem, the proposed numerical solution (algorithm for the microphone placement problem) is presented in Section III, while Section IV contains description of the numerical experiment and test that was held in an anechoic chamber. The work ends with a short summary (Section V).

II. Problem formulation

There is a set of $N = 1, \dots, n$ microphones. All microphones must be used to build the array. There are not any restrictions on array shape and microphones do not have to be equally spaced. Signals from microphones are sampled synchronously.

In case of broadband signals (like speech signal) broadband beamformer is equivalent to applying a finite-duration impulse response (FIR) filter of order L-1 (each array element is an L-tap finite impulse response filter) to each microphone output and then the signals are summed up.

For the fixed N-element microphone array, where microphone positions are denoted by vector $\boldsymbol{\lambda} = [r_1, r_2, \dots, r_I, \dots, r_N]$, where r_i denotes the position of microphone I (its coordinates), the transfer function of the i-th microphone in the near-field is defined as:

$$A_i(r,f) = \frac{1}{||r-r_i||} e^{-j2\pi f ||r-r_i||/c} \tag{1}$$

where c is the speed of sound in the air, r is the location of the sound source and f is a frequency.

An impulse response of the i-th filter is given by:

$$\mathbf{h_i^T} d_0(f) = H_i(L,f), \quad i = 1, \dots, N, \tag{2}$$

where

$$\mathbf{h_i} = [h_i(0), h_i(1), \dots, h_i(L-1)]^T, \quad h_i \in R^L$$

denotes the coefficients of the i-th FIR filter of length L and

$$\mathbf{d_0}(f) = \left[1, e^{\frac{-j2\pi f}{f_s}}, \dots, e^{\frac{-j2\pi f(L-1)}{f_s}}\right].$$

For the given microphone placement, system response can be found by solving the following equation:

$$G(r,L,f) = \textstyle\sum_{i=1}^{N} H_i(L,f) A_i(r,f) = \mathbf{A}(r,f)\mathbf{H}(L,f) \tag{3}$$

where $\mathbf{A}(r,f) = [A\_1(r,f), \dots, A\_N(r,f)]$ is a vector containing transfer functions of all microphones and $\mathbf{H}(L,f) = [H_1(L,f), \dots, H_N(L,f)]^H$ is the frequency filter response vector. The issue of finding the optimal set of coefficients (i.e. frequency response) for the fixed microphones' position can be determined by solving the quadratic problem, which can be solved very quickly using quadratic programming techniques as in [11].

We assume, that the desired system response $G_d(r,L,f)$ is described as follows: interfering signal should be attenuated, while the desired signal level should remain unchanged. The desired and interfering signal sources have different localization in space.

The problem addressed in this paper is to design the microphone array (i.e. the position of microphones in 2D space and the FIR filter coefficients) so that the actual output of the beamformer is as close as possible to the desired one G_d in the sense of l_2 norm:

For a given placement of the microphone array, it is necessary to calculate a vector of filter coefficients $\boldsymbol{H}$ that minimizes the objective function:

$$E(\boldsymbol{H}) = \frac{1}{||\Omega||} \int_{\Omega} \sigma(r,f) ||\mathbf{A}(r,f)\mathbf{H}(L,f) - G_d(r,L,f)||^2 \, dr df \tag{4}$$

where $\sigma(r,f)$ is a positive weighting function, and Ω denotes spatial-frequency domain. Ω consists of the region of passband Ω_P and stopband Ω_S, i.e. $\Omega = \Omega_P \cup \Omega_S$.

Formally, for a given microphones' placement in the array one can write the beamformer design problem as:

$$\min_{H \in R^{NxL}} E(\boldsymbol{H}) \tag{5}$$

Based on [11] and [12] we assume that there is a performance limit for finite filter length designs and further increasing of filter length do not improve criterion value significantly. Thus, for each microphone placement $\boldsymbol{\lambda} = \lambda_1, \lambda_2, \dots, \lambda_N$, and $\boldsymbol{\lambda} \in \boldsymbol{\Lambda}$, where $\boldsymbol{\Lambda}$ denotes the feasible region of microphone placement, we set the same filter length L. Each element of $\boldsymbol{\lambda}$ is a two-dimensional vector (2-D coordinates).

Considering the real and practical limitations, the minimum and maximum distance of microphones from the source were considered, and for the proper functioning of the system the correct distances between the individual microphones were considered (based on the microphone size) to:

$$\left||r_i - r_j|\right|^2 < \epsilon_d, \quad i,j = 1,2, \dots, N, j \neq j \tag{6}$$

Consequently, to find the optimal solution of the analysed problem, i.e. microphone placements vector $\boldsymbol{\lambda}$ and FIR filter coefficients $\widetilde{\boldsymbol{H}}$ it is necessary to determine:

$$\min_{\lambda \in \Lambda, \widetilde{H} \in \Gamma^N} E(\boldsymbol{\lambda}, \widetilde{\boldsymbol{H}}), \tag{7}$$

with the assumption (6), where:

$$E(\boldsymbol{\lambda}, \widetilde{\boldsymbol{H}}) = \frac{1}{||\Omega||}\int_{\Omega} \sigma(r,f)\left\|\mathbf{A}(\boldsymbol{\lambda},r,f)\widetilde{\mathbf{H}}(f) - G_d(\boldsymbol{\lambda},r,f)\right\|^2 drdf \quad (8)$$

Algorithm 1: Simulated annealing

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Define objective function min_{h∈R^{M,L}} E(h, λ_a, r)
Generate random initial solution λ_a, E_curr = E(λ_a)
Set T, max_it and γ, E_best = E(λ_a)
while (iter < max_it) or (stop criterion) do
    Choose λ_iter by a random change in a position of a random microphone
    Assign λ_a = λ_iter with probability P(T, λ_a, λ_iter) = min{1, exp (−(E(λ_iter)−E(λ_a)))/T}
    if E(λ_iter) < E_best then
        λ_best = λ_iter
    end
    T = γT
end
Show the best solution λ_best
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The problem defined in (8) is a nonlinear continuous optimization problem with constraints. The objective function (8) is non-convex and highly nonlinear with respect to the placement variables $\boldsymbol{\lambda}$. Thus, the placement problem cannot be solved efficiently with classical optimization techniques. However, for the given placement, the optimal frequency response (i.e. FIR filter coefficients), can be calculated by solving the quadratic problem (7).

III. Algorithm

Based on results presented in [13], we decided to determine microphone array placement and its FIR filters coefficients using simulated annealing approach. Since the number of microphones is small, this algorithm is easy to implement and can provide a good-quality solution. It's efficiency for similar instances of the problem is comparable with population-based algorithms (e.g. genetic algorithm, particle swarm optimization, etc.).

Simulated annealing (SA) [14] is a local search algorithm (we search the neighborhood of the current solution), which gives a possibility of escaping from local minima. The name of the algorithm refers to the thermodynamic cooling process in which the crystalline substance is heated and then slowly cooled down to the reaching of regular crystal structure. During each iteration of the algorithm, the new solution is constructed based on the current solution. If the criterion value of the new solution is better than the current one, then this solution is accepted and next solution is created based on this new solution. In the opposite case (if the new solution has worse criterion value), it can be accepted with the certain probability. Such an approach helps to escape from local minima (it provides diversification of searching process). To intensify the searching, the probability of worse solution acceptance decreases. To control this probability, a parameter called temperature is used. The temperature value is a function depending on the iteration of the algorithm, i.e. it decreases with each iteration of the algorithm. As the temperature goes down, the probability of accepting a worse solution decrease. For the considered problem of microphone placement, the general form of the simulated annealing algorithm can be described in the following way.

The simulated annealing algorithm starts with a feasible solution (microphone placement) $\lambda_a \in S$, where S is the set of all acceptable solutions (we discretized the solution space). In each iteration of the algorithm the solution $\lambda_{iter} \in N(\lambda)$ is generated randomly from the neighborhood of solution λ_a. A neighborhood of the solution is a set of solutions that do not differ dramatically, i.e. each solution from the neighborhood can be obtained from the current solution with a small move (change), the transition from the λ_a solution to the λ_{iter} solution cannot involve construction of a new solution from scratch. The neighborhood cannot cover the entire solution space, it covers only a certain area of the solution space, and it changes with every new solution. The new solution is accepted with probability $P(T, \lambda_a, \lambda_{iter})$ - function which value depends on temperature T and difference between criterion value of the current and new solution. In each iteration, the parameter called temperature T is decreased, thus probability that worse solution will be accepted as a new solution decreases. During the search process the best-known solution is stored. The cooling scheme is controlled by parameter γ. The SA algorithm is presented below (see Alg. 1)

IV. Experiment

At first, in this section, we present results of the numerical experiments. Next, we focus on measurements that were done in an anechoic chamber. All codes are implemented in MATLAB platform and run on PC with Intel(R) Core i7 CPU with 2.5GHz.

The desired response function of the spatial filter formed by microphone array is specified over a region that would fit into a room with a speaker (human) and two interfering sources. We assume that the speaker is exactly in the middle, between two interfering sources. It includes the frequency range of human voice. The frequency of the interfering noise fully coincides with the frequency of the desired signal. Since we should allow for the delay of the speech to reach the microphones, the desired response function in the passband region Ω_p is defined as:

$$G_d(\boldsymbol{\lambda}, \boldsymbol{r}, f) = e^{-j2\pi f\left((\|\boldsymbol{r}-\boldsymbol{r_c}\|)/c+(L-1)/2\mathrm{T}\right)}, \quad (9)$$

where $r_c = 1/N\sum_{i=1}^{N}\boldsymbol{r}_i$ denotes the center position of the placement variable λ and c = 340.9 m/s is the sound speed in the air, group delay $\tau = \frac{(L-1)}{2}$, Ts=125μs. The sampling rate is set to 8 kHz, and the maximum frequency is chosen as 3 kHz. In addition, it is assumed that the minimum distance parameter between two different microphone elements ϵ_d cannot be smaller than 0.015^2m^2. The weighting function is chosen as $\sigma(r,f) = 1$. The filter length L was chosen experimentally, based on system performance limit assumptions, and is equal to L=57.

The desired response function (its cross-section in space-frequency domain) is presented in Fig.1. In the stopband the desired response (signal attenuation) should be as big possible. Thus, it was assumed as -120dB. Such big attenuation guarantee that the filtered signal is negligibly small. The rest of the values, outside Ω_p and Ω_s are represented by the smallest possible value (floating-point relative accuracy).

The placement configuration problem is considered in two dimensions (we assume that the speaker, interfering signals, and microphones have the same z coordinate). The microphone array consists of N=6 elements.

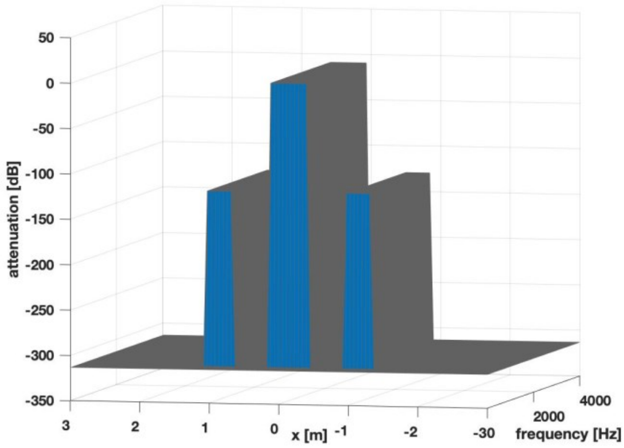

Fig. 1. Desired response of the microphone array

The passband region is defined as follows (see Fig. 2):

$$\Omega_p = \{(r, f): -0.3\text{m} \leq |x| \leq 0.3\text{m}, y = 0\text{m}, 0.4\text{kHz} \leq f \leq 3.0\text{kHz}\}$$

and the stopband:

$$\Omega_s = \{(r, f): 0.8\text{m} \leq |x| \leq 1.2\text{m}, y = 0\text{m}, 0.4\text{kHz} \leq f \leq 3.0\text{kHz}\} \cup \{(r, f): -1.2\text{m}! \leq |x| \leq -0.8\text{m}, y = 0\text{m}, 0.4\text{kHz} \leq f \leq 3.0\text{kHz}\}$$

The microphone placement feasible region:

$$\Lambda = \{(x, y): -1.2\text{m} \leq x \leq 1.2\text{m}, 1.0\text{m} \leq y \leq 4.0\text{m}\}$$

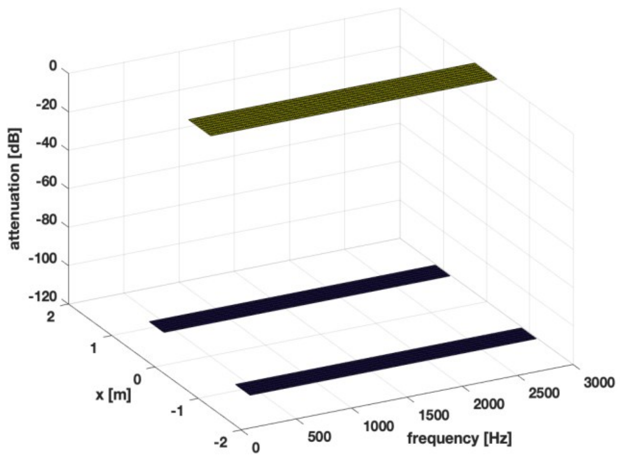

Fig. 2. Passband (yellow) and stopband (navy) regions

Both passband and stopband are discretized, the frequency points are taken every 0.1kHz and the spatial points are taken every 0.057m.

The number of iterations for SA is equal to 1000. We implemented geometric cooling scheme with $\gamma = 0.9$ and initial temperature T=10000. To avoid getting stuck in a local minimum, if there is no better solution found during 20 iterations, the current temperature is increased to its initial value. Such an approach allows us to jump to different regions of the solution space in search of promising solutions. We start with an initial microphone placement: microphones are in vertical line (all have coordinate $x = 0$) and distance between microphones is set to 4 cm (0.5 of wavelength for 4250Hz). The initial microphone placement is presented in Fig. 3, and its response is presented in Fig. 4. The criterion value, calculated according to (8), for this initial solution is -38.54dB. It is a mean value of attenuation calculated over the pass and stopband frequencies.

The SA algorithm provided the following microphone placement: λ = [(0.16; 2.64), (0.09; 2.73), (-0.15; 2.79), (0.06; 2.56), (-0.20; 2.39), (-0.05; 2.88)] (see Fig. 5) with the system response as in Fig. 6 and criterion value -54.03dB. The microphones are unequally spaced and are no longer in a single line. The general shape of system response (in space-frequency domain) is close to the desired one, i.e. interfering signal is attenuated.

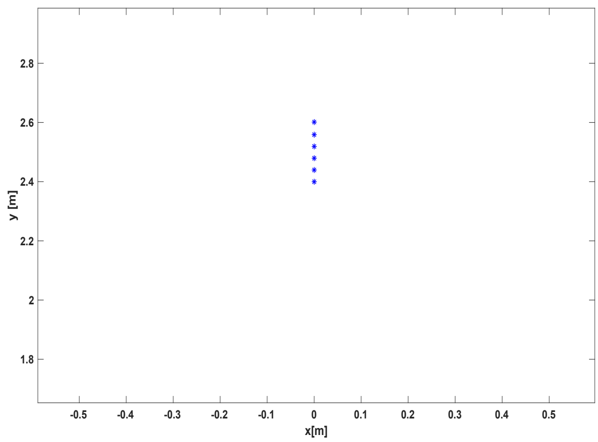

Fig. 3. Initial microphone placement, 6 microphones

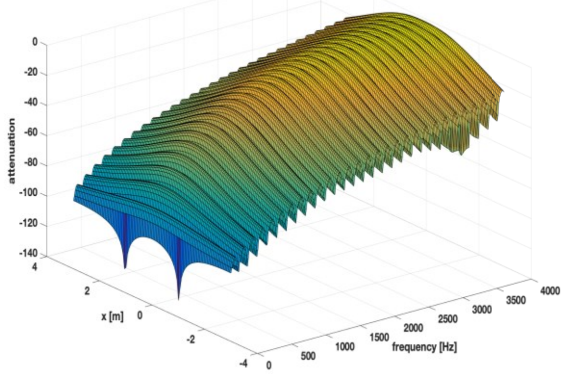

Fig. 4. Theoretical (calculated during simulation) system response for the initial array configuration

Then, an experiment with a real signal was performed. The measurement was conducted in an anechoic chamber of the Wrocław University of Science and Technology. The dimension of the chamber is 4.1m x 2.9m x 2.1m. As part of this experiment, sound sources, both the speech signal and the interference signal, were reproduced using APS Klasik loudspeakers (active monitors near field, two-way, crossover frequency 3.2kHz). To measure sound, 1/2-inch Superlux ECM999 measurement microphones were used, which were placed on stands. The microphone locations determined from the simulation results were precisely reproduced. All microphones and speakers were in one plane, determined using a laser level. The height of this plane was 1.1m above the chamber floor level. Before starting the measurement, the alignment was checked twice. The interfering signal was white noise, and the speech signal was a voiceover recording made in studio conditions.

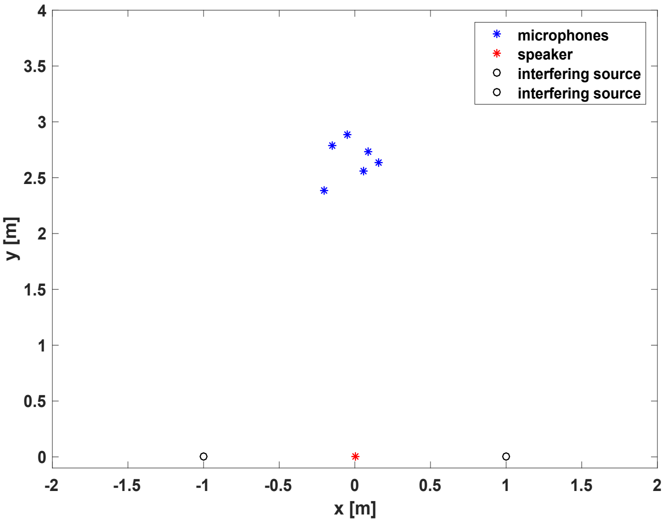

Fig. 5. Proposed microphone placement, 6 microphones, initial solution vertical

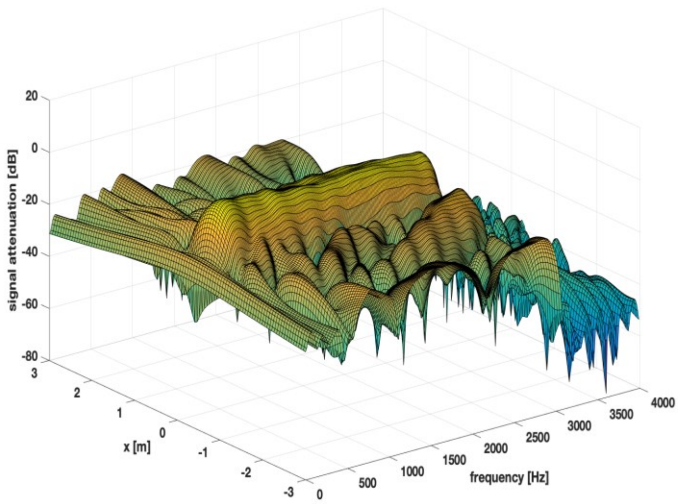

Fig. 6. Theoretical system response for the designed microphone array

During the preprocessing, the emitted signal from each microphone were bandpass filtered in the range 400Hz-3000kHz to meet previous assumptions. Filtering was performed in Matlab with 201-tap FIR filter.

Next, the spatial filtering was performed. The signals were filtered once again, we applied the FIR filter coefficients that were calculated by our algorithm. After this filtration process, we summed all signals. The waveform of the signals is presented in Fig. 7 (both signals are normalized). The blue one is the sum of the signals from each microphone (only after the preprocessing), the red one is the signal after beamforming. As it can be seen that noise level was reduced. The personal impressions of the authors, during listening of the filtered signal, indicate an audible reduction in noise. Regardless of personal impression, one can see that the noise level between voiced parts of the sound is smaller comparing to its level before spatial filtering.

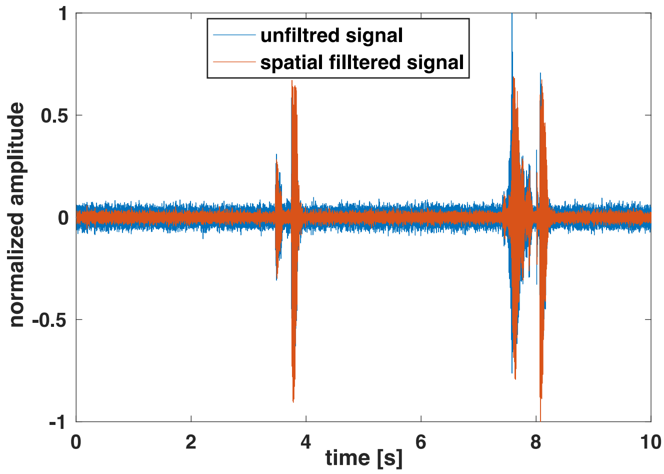

Figure 7. The original signal (blue) and signal after spatial filtering (red)

Since it can be hard to place microphones with high precision, we analyzed the impact of microphone placement on the criterion value and spatial filtering results. Small microphone's shift does not change the criterion value significantly; however, it requires different FIR filter coefficients.

For example, for the considered placement: λ=[(0.16; 2.64), (0.09; 2.73), (-0.15; 2.79), (0.06; 2.56), (-0.20; 2.39), (-0.05; 2.88)] the criterion value is -54.03 dB.

If one of the microphones is shifted 0.5cm: λ_1 = [(0.155; 2.64), (0.09; 2.73), (-0.15; 2.79), (0.06; 2.56), (-0.20; 2.39), (-0.05; 2.88)] then the best criterion value for this placement is -53.90dB.

The optimal FIR filter coefficients have changed for all microphones. The difference in coefficient value for the first microphone, for these two placements, is presented in Fig. 8. It is worth noticing that small changes in microphone positions does not have a significant impact on FIR filter coefficients. The microphone array is not very sensitive to small changes in microphone position.

There is an assumption, in the model, that the microphones are omnidirectional. However, in practice it can be unreachable, since microphones may be omnidirectional but in limited range of frequencies. Moreover, the array (its elements) gives some additional signal's distortions and reflection and has influence on systems efficiency. Despite these problems, the tested method looks promising, having only six microphones, we were able to attenuate the interfering noise.

V. CONCLUSIONS

This paper was devoted to an experimental analysis of microphone array designing method based on metaheuristic approach. Microphones positions and FIR filter coefficients were calculated simultaneously. The array with 6 microphones was tested in the anechoic chamber. The initial results are promising.

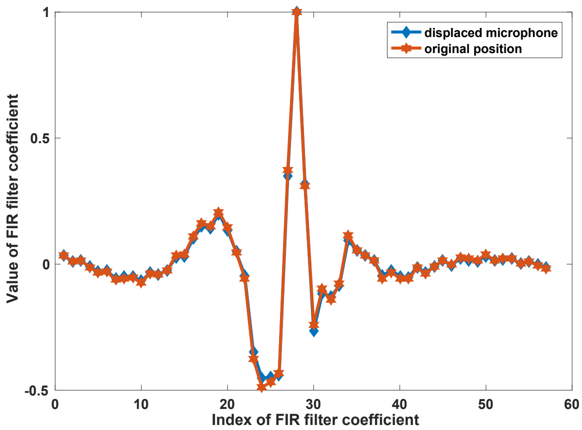

Fig. 8. Comparison of optimal FIR filter coefficients for two microphones. There is small difference in microphones placement (0.5cm on x axis)

We faced some problems that must be taken into consideration. During our experiment in anechoic chamber, it was hard to place microphones precisely. However, despite possible small shifts we were able to do the spatial filtering. Moreover, the array (its elements) gives some additional signal's distortions and reflection. Regardless of these additional problems that were omitted in the model and calculations, tests indicate possible applicability of the analyzed method during beamforming system design. In near-real environment it would be important to take reflections as additional source of the interfering sounds. Further works will focus on speech intelligibility tests.

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