MICROPHONE ARRAYS BEAMFORMING METHODS WITH LEAST MEAN SQUARE SPATIAL FILTRATION FOR NOISE SUPPRESSION

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Abstract

Beamforming is a group of methods with spatial filtering of sound signals captured from an array of sound sensors or microphones. The objective is to estimate the direction of arrival of sound source signal in the presence of noise and other interfering signals. A spatial filtering is prepared from beamformer to separates the signals with overlapping frequency content arriving from different directions. The goal of this article is to study the different beamforming methods using the algorithm of Least Mean Squares (LMS) filters for spatial filtering of sound signals from microphone array sensors. It is proposed to use a least mean square algorithm capable of iteratively adapting the weights of the sound sensor microphone array to minimize the noise in output microphone array signal. Two methods of microphone arrays are simulated and compared: the simple delay and sum beamformer and statistically optimum beamformer. The simulation results of are analyzed to estimate the SNR improvement of statistically optimum beamforming method in comparison to simple delay and sum beamforming method.

1. INTRODUCTION

The sound signals arriving to the microphone arrays are spatially propagating signals containing also interfering signals and noise. When the original sound signals and interfering signals are in the same temporal frequency band, it is not possible to use temporal filtering for separating the useful sound signal from the interfering signals [1]. Therefore it is necessary to apply the existence of different spatial locations to separate original sound signals from interfering signals and noise. This difference allow spatial separation the original sound signals from the interference signals using a microphone array beamforming methods. The microphone array contain a number of sound sensors and are made with different configurations. A simple way to use the microphone array beamforming is to apply signal filtering of each of sensor or microphones and to add up the filtered outputs of all the sensors. Here is proposed to study the different beamforming methods using the algorithm of Least Mean Squares (LMS) filters for spatial filtering of sound signals from microphone arrays.

2. THE BASIC PRINCIPLES AND DIFFERENCES BETWEEN BEAMFORMING METHODS

The different beamforming methods can be classified on two basic types: data independent and statistically optimum methods. In Fig. 1 is presented the simplest example of data independent beamforming method the delay and sum beamformer methods.

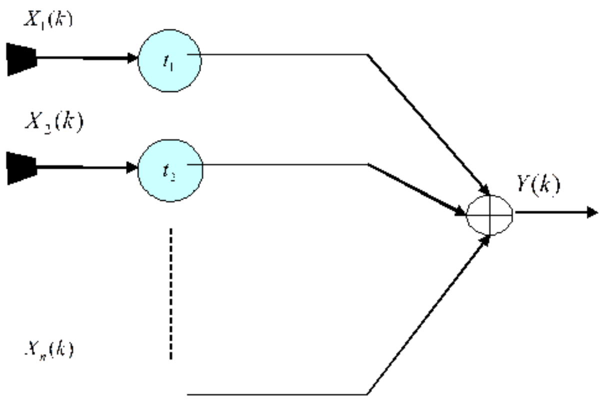

Fig.1. Delay and Sum Beamformer with *"n"* sound sensors or microphones

The major disadvantage of delay and sum beamforming method is the large number of sensors required to improve the SNR.

The main difference between shown in Fig. 1 simple delay and sum beamforming method and the statistically optimum beamforming method is the dependence on how the weights in the beanforming fitters are chosen: the weights in data independent beanforming methods do not depend on the array data; the weights in a statistically optimum beamformer are chosen based on the statistics of the array data.

3. THE STATISTICALLY OPTIMUM BEAMFORMING METHODS

The statistically optimum beamforming method can be modeled as array processor shown in Fig. 2 with *"N"* sound sensors or microphones and *"J"* taps per microphone [2]. The delays after each sensor are not shown here.

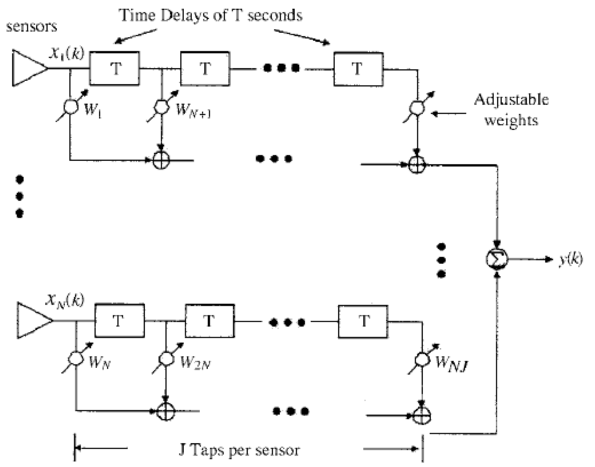

Fig. 2. Array processor with *"N"* sound sensors or microphones for statistically optimum beamforming method modelling

The presented in the Fig. 2 array processor work as a single tapped delay in which each weight is equal to the sum of the weights in the vertical column of the processor. These summation weights are set so as to determine the desired frequency response characteristic in the desired direction. The array processor is assumed to be steered to the required desired direction by appropriate delays after the sensors. Out of *J* weights of each *N* tap determine the desired direction frequency response. The remaining *NJ – J* weights are used to minimize the total power in the array output, which is equivalent to minimizing the non- desired direction noise power because the signal and the noise is usually uncorrelated. The vector of tap values at the *k-th* sample is written as $X(k)$:

$$X^T(k) = \left[x_1(k),\, x_2(k), \ldots\ldots\ldots\ldots, x_{KJ}(k)\right] \quad (1)$$

The tap values are the sums of the values due to desired-direction waveforms and the non- desired-direction noises:

$$X(k) = L(k) + N(k), \quad (2)$$

where

$L(k)$ and $N(k)$ are the *KJ* dimensional vectors of desired-direction and non-desired-direction at the *kth* sample, respectively:

$$L^T(k) = \left[l_1(k) \quad l_2(k) \quad . \quad . \quad . \quad l_{KJ}(k)\right] \quad (3)$$

$$N^T(k) = \left[n_1(k) \quad n_2(k) \quad . \quad . \quad . \quad n_{KJ}(k)\right] \quad (4)$$

The vector of weights at each tap is *W*:

$$W^T = \left[w_1 \quad w_2 \quad . \quad . \quad . \quad w_{KJ}\right], \quad (5)$$

With the equation:

$$E\left[N(k)L^T(k)\right] = 0 \quad (6)$$

is assumed that the condition of uncorrelation between desired direction vector and the vector of non-desired direction noise is satisfied. Therefore, the output of the array at the time of the *k-th* sample is:

$$y(k) = W^T X(k) = X^T(k) W \quad (7)$$

It is necessary to define and expressed the requirements of the weights on the *j-th* vertical column of the taps sum to a chosen number f_i with a *KJ* dimensional vector c_j, a constraint matrix *C* and a *J* dimensional vector *F* of weights of the desired-direction-equivalent tapped delay line:

$$c_j^T W = f_i \qquad j = 1, 2, \ldots\ldots, J \quad (8)$$

$$C = \left[c_1 \quad . \quad . \quad c_j \quad . \quad . \quad . \quad c_J\right] \quad (9)$$

$$F^T = \left[f_1 \quad . \quad . \quad f_j \quad . \quad . \quad f_J\right] \quad (10)$$

Now, the constraint can be written as:

$$C^T W = F \quad (11)$$

To achieve the optimal values of vector of weights W_{opt} the necessary minimization of equation (11) is carried out:

$$\begin{array}{c} \min\limits_{W} \; W^T R_{XX} W \\ subject\ to\ C^T W \end{array}, \quad (12)$$

is proposed to use the algorithm of Least Mean Squares (LMS) filters [3] for spatial filtering of sound signals from microphone arrays. In the equation (12) is applying R_{XX} one of the existing in theory of statistically optimum beamforming method correlation matrixes defined as:

$$\begin{array}{l} E\left[X(k)X^T(k)\right] = R_{XX} \\ E\left[N(k)N^T(k)\right] = R_{NN}\,, \\ E\left[L(k)L^T(k)\right] = R_{LL} \end{array} \quad (13)$$

assuming that the equation (6) is satisfied. Therefore, the optimal values of vector of weights W_{opt} can be found as:

$$W_{opt} = -R_{XX}{}^{-1} C\lambda , \qquad (14)$$

by the method of Lagrange multipliers λ:

$$H(W) = \frac{1}{2} W^T R_{XX} W + \lambda^T (C^T W - F) \quad (15)$$

From equation (15) is found the gradient with respect to W and is setting to zero:

$$\nabla_W H(W) = R_{XX} W + C\lambda = 0 \qquad (16)$$

to calculate the optimal values of vector of weights W_{opt} as is shown in equation (14). In this equation R_{XX} can be considered as positive semi definite. Therefore, this give the reason to substitute equation (14) in equation (11):

$$C^T W_{opt} = F = -C^T R_{XX}{}^{-1} C\lambda \qquad (17)$$

and find the Lagrange multiplier λ from the following equation:

$$\lambda = -\left[C^T R_{XX}{}^{-1} C\right]^{-1} F \qquad (18)$$

Finally the optimum weight vector can be written as:

$$W_{opt} = R_{XX}{}^{-1} C\left[C^T R_{XX}{}^{-1} C\right]^{-1} F \qquad (19)$$

In the calculated above optimal values of vector of weights W_{opt} from equation (19) the correlation matrix R_{XX} is not known. The direct substitution, when the correlation matrix R_{XX} is calculated, leads to a number of multiplications at each iteration proportional to the cube of the number of weights. Then, the adaptive algorithm [4], described briefly below, can be used to avoid this difficulties:

— initializing the vector of weights *W*

$$W(0) = C(C^T C)^{-1} F \; ; \qquad (20)$$

— at each iteration move the weight vector in the negative direction of the constrained gradient, scale by a constant *μ* and after *k-th* iteration calculate the next weight vector as:

$$\begin{aligned} W(k+1) &= W(k) - \mu \nabla_W H[W(k)] \\ &= W(k) - \mu[R_{XX} W(k) + C\lambda(k)] \end{aligned} ;(21)$$

— the Lagrange multipliers $\lambda(k)$ are chosen by requiring W(k+1) to satisfy the constraint:

$$\begin{aligned} &F = C^T W(k+1) = \\ &C^T W(k) - \mu C^T R_{XX} W(k) - \mu\, C^T C\lambda(k) \end{aligned} ;(22)$$

— calculating the Lagrange multipliers λ(k) from equation (22);

— substituting multipliers λ(k) into (21):

$$\begin{aligned} &W(k+1) = W(k) - \mu\left[I - C(C^T C)^{-1} C^T\right].. \\ &R_{XX} W(k) + C(C^T C)^{-1}\left[F - C^T W(k)\right] \end{aligned} ;(23)$$

— defining vector $\tilde{F}$ and matrix P:

$$\tilde{F} = C(C^T C)^{-1} F \; ; \quad P = I - C(C^T C)^{-1} C^T \; ;(24)$$

— finally the iteration steps of adaptive algorithm is the following:

$$W(0) = \tilde{F} \qquad (25)$$

$$W(k+1) = P\left[W(k) - \mu R_{XX} W(k)\right] + \tilde{F} \quad (26)$$

The statistical optimized LMS algorithm described above can be summarized in two main steps using the following equations:

– Initialization step (25)

– Iterative steps (26).

4. SIMULATION OF SIMPLE DELAY AND SUM BEAMFORMER AND STATISTICALLY OPTIMUM BEAMFORMER

Linear microphone sensor (Fig. 3) for are two types of beamformers are simulated: the simple delay and sum beamformer and statistically optimum beamformer. One of the speech signals, which are used in the numerous simulations is shown in Fig. 4 without and with added noise to test the ability of two simulated methods, the simple delay and sum beamformer and statistically optimum beamformer, to suppress the interfering signals and noise. The resultant simple delay and sum beamformer and statistically optimum beamformer diagrams are shown in Fig. 5.

Table.1 show that the values of SNR measured for statistical optimized with LMS algorithm for spatial filtering is higher, that these of simple delay and sum beamformer method.

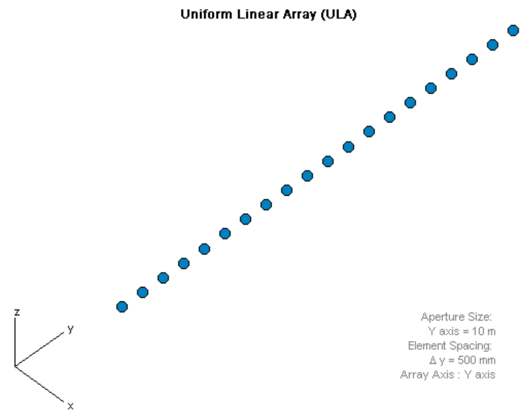

Fig. 3. Simulation model of linear microphone sensors

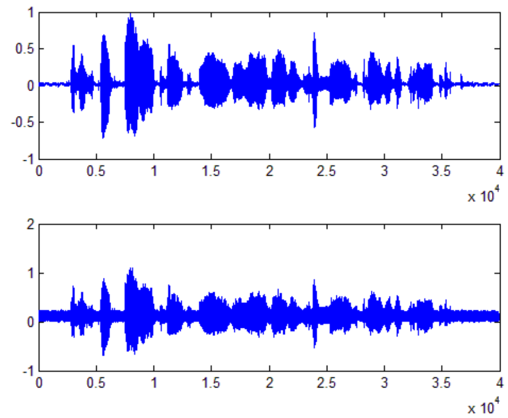

Fig. 4. One of the speech signals, used in the numerous simulations without and with noise

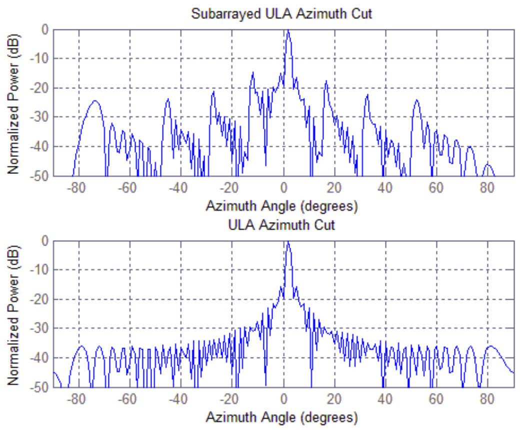

Fig. 5. The resultant simple delay and sum beamformer and statistically optimum beamformer diagram

Table. 1.

Simulation number	Delay and sum beamformer SNR(dB)	Statistical beamformer SNR(dB)
1	01.97	04.26
2	-06.04	01.23
3	-05.11	00.04
4	01.87	05.34
5	-06.02	02.34
6	-05.40	00.02

5. CONCLUSION

The results of simulations presented in graphics and tables shown the ability of the proposed least mean square algorithm to iteratively adapting the weights of the sound sensor to minimize the noise in output microphone array signal.

6. ACKNOWLEDGMENT

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