Equalization Frequency Characteristics of the Microphone for Acoustic Camera

Ognjen Jovanović, Miloš Bjelić, Jelena Ćertić

Abstract— **Acoustic camera represents a modern tool in acoustic measurements. Many measuring microphones in this measurement system would significantly increase the price of the system. Measuring microphones nearly have a flat amplitude characteristic and a linear phase characteristic. The motivation for this research was to cut down the cost of the acoustic camera system realization, which was expensive do to the large number of microphones used. The idea was to use microphones of lesser quality compared to the measuring microphones and with specific digital processing get a resulting signal that is comparable to the signal we got with the measuring microphone. In this paper, an equalization technic was used with the help of an adaptive algorithm to improve the frequency characteristics of the microphone of lesser quality. Results of the experiment show that this method can lead to budget cuts and still preserve the accurate location of the sound sources.**

Keywords— **microphone, microphone array, beamforming, equalization, LMS**

I. Introduction

In the last two decades, the development of algorithms for space-time signal processing for microphone arrays leads to an important new tool in acoustic measurements [1]. A microphone array in a combination with a video camera forms a system known as an acoustic camera and it allows us visualization of the sound field in space. By using an acoustic camera, a microphone array to be precise, locating of a sound source can be performed. Therefore, this tool found its way not only in acoustics but in other engineering fields, such as the car industry, airplane industry, traffic control, etc. [2]. The algorithms for space-time signal processing that are used to determine the direction of the incoming sound on the microphone array are called beamforming algorithms [3].

Microphone array consists of omnidirectional microphones placed in a certain order and this order is called the geometrics of the array. The order of the microphones in space is defined by the microphone array beampattern, also known as the directional diagram. Microphones that are used should have a nearly flat amplitude characteristics and a nearly linear phase characteristics. The reason for this request is the fact that the algorithms for signal processing are based on the exact information of the phase difference between signals on different microphones. Also, the flat amplitude characteristic is important because the specific level of the sound the sources make in space needs to be preserved. Microphones that fulfil these requests belong to the measuring microphone group. Measuring microphones are present on the market and can be found with many manufacturers, but their price is significantly higher than the price of other microphones. Since more than 20 microphones are used in a typical microphone array [4], the price of the whole system can be very high. The high price of the measuring microphones was the motivation for this research.

The basic idea for this paper is to improve the performance of omnidirectional microphones, that are of lesser quality than the measuring microphones, by using certain signal processing to get an equivalent microphone that can be used for the construction of an acoustic camera. In this paper, the process of amplitude and phase characteristics equalization in the digital domain will be shown. Electrical microphones with a ¼ inch membrane diameter, that are usually used for computer communication, were used in this research. To improve the amplitude and phase characteristics of the microphone, measurements were made in which the lesser quality microphone was compared to the reference measurement microphone. To perform the equalization of the microphone, an experiment was needed in which the microphone and measuring microphone are in the same sound field on very small distance from each other. Signals from the lesser quality microphone are the input signal for the equalization algorithm. For the equalization, the Least Mean Square (LMS) algorithm was used [5]. An output of the measuring microphone, which has a flat amplitude and a linear phase characteristics in almost the whole audio band, was used as a reference signal for equalization. Therefore, the quality of the speaker is not important because we compare the characteristics of the two microphones. Lesser quality microphones that were used in this experiment have an amplitude characteristic that has a deviation of more than 5 dB in some frequency bands compared to the measuring microphone. After the equalization of the microphones characteristics with the adaptive algorithm, the tap weights of the adaptive filter are obtained and they represent the calibration coefficients. Recorded signal needs to be filtered with the obtained tap weights before the beamforming processing. The advantage of this suggested method is that it only requires one measurement to determine the tap weights of the adaptive filter. The adaptive filter and the used microphone make the equivalent microphone that can be used for the design of the microphone array that is used in the acoustic camera. In this

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paper, the characteristics of the microphone were analyzed in a frequency band from 200 Hz to 8000 Hz. The listed band was determined because the microphone will be used for a microphone array which purpose is to locate sound sources in this band. The remaining of the paper is organized as follows. In the second section the adaptive algorithm that was used is going to be described. In the third section the experiment results will be shown and discussed. In the final chapter the conclusions that we made are presented.

II. Least-mean-square algorithm (LMS)

The least-mean-square (LMS) algorithm is a linear adaptive filtering algorithm that consists of two basic processes - a filtering process and a filter coefficient adaptation process. Together they form a feedback loop around the LMS algorithm. Fig. 1. The transversal filter tap weights $w(n)$ are being adapted using the LMS algorithm and those weights are expected to approach the value of the Wiener solution as the number of iterations n approaches infinity.

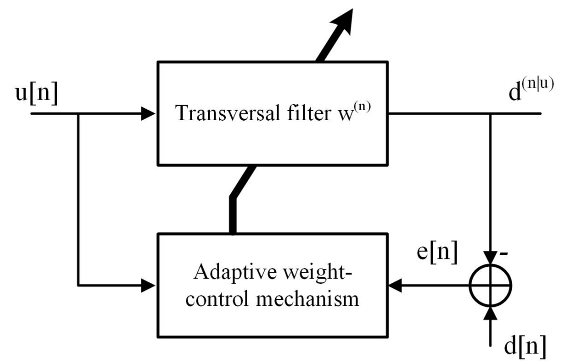

Fig. 1. Block diagram of the LMS algorithm

For this algorithm, the desired response $d(n)$ needs to be known, as well as the tap-input vector $u(n)$. An estimation error $e(n)$ is also defined as the difference between the desired response $d(n)$ and the actual filter output $y(n)$. The estimation error $e(n)$ and the filter output $u(n)$ are applied in the estimation of the tap weights. Since the LMS algorithm involves feedback it raises the issue of stability, therefore it must fulfill criteria that:

$$J\left(n\right) \rightarrow J\left(\infty\right) \text{ as } n \rightarrow \infty \tag{1}$$

where $J(n)$ is the cost function, mean-squared error of the filter output at time instant n, and its final value $J(\infty)$ is a constant.
The coefficients of the optimal Wiener filter $\mathbf{w}_o=[w_{o0}\ w_{o1}\ \dots\ w_{oN-1}]^T$ (^{T} denotes matrix/vector transpose) are calculated as:

$$\mathbf{w}_o = \mathbf{R}^{-1}\mathbf{p} \tag{1}$$

where **R** is the correlation matrix of the tap inputs and **p** is the cross-correlation vector between the tap inputs and the desired response. The LMS algorithm is based on the simplest choice of estimators for **R** and **p:**

$$\hat{\mathbf{R}}(n) = \mathbf{u}\left(n\right)\mathbf{u}^T\left(n\right) \tag{3}$$

$$\hat{\mathbf{p}}(n) = \mathbf{u}(n)d(n) \tag{4}$$

where $\mathbf{u}(n)=[u(n)\ u(n-1)\ \dots\ u(n-N+1)]^T$ is the vector of the samples of the filter input signal, and H denotes Hermitian operator. The LMS algorithm is performed by a few easy steps:

1) *Initial tap-weights values, typically* $\mathbf{w}(0)=0$;
2) *Calculating the filter output*
   $y(n)= \mathbf{w}(n)\cdot\mathbf{u}(n)$,
3) *Determining the estimation error*
   $e(n)=d(n)-y(n)$
4) *Tap-weight adaptation*

$$\hat{\mathbf{w}}(n+1) = \hat{\mathbf{w}}(n) + \mu\mathbf{u}\left(n\right)e\left(n\right) \tag{7}$$

5) *Repeating the algorithm form step 2, until the end of the signal.*

In our experiment, we used two microphones, one microphone was of better quality and it was used as a reference, while the second was of lesser quality and was in a cascade with a LMS adaptive filter, so that the filter could improve the microphone's performance. The output of the reference microphone is the desired response $d(n)$ and the output of the second microphone is the tap-input vector $u(n)$ as it is shown on Fig. 2.

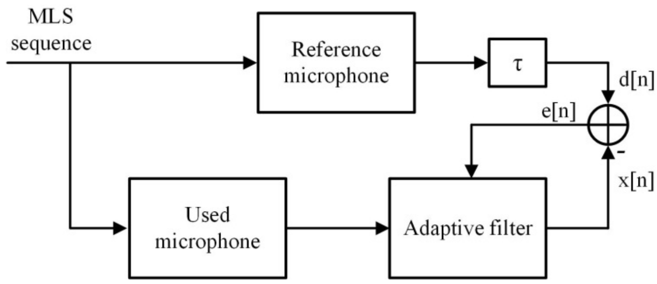

Fig. 2. Block diagram of the experiment

III. Experiment results and discussion

The experimental setup is based on the scheme given in Fig. 2. The delay block denote by τ compensates delay introduced by the adaptive filter. The proposed method is validated by comparison of the signals at the output of the reference microphone and the signals before and after additional filter equalizer. The coefficients of the additional filter are obtained by adaptive LMS procedure. The convergence of the LMS algorithm is verified by tracking the values of the filter coefficients in each step of the adaptation.

The experiment was done in anechoic chamber. As the reference microphone the Neutrik miniSPL E044 [6] was used. The microphone that needs improvement was placed near the reference microphone, as shown on Fig. 3. Both microphones were exposed to the same sound field made by a speaker. As a test sequence a maximum length sequence-MSL [7] was used with the length of 30 seconds.

Fig. 3. Experimental setup

The difference between the output of the reference microphone and the output of the second microphone, i.e. input of the adaptive filter, can clearly be seen on their power spectral density PSD on Fig. 4. The PSD of the adaptive filter input signal is lower than the PSD of the desired signal in the whole frequency range [200 - 8000] Hz.

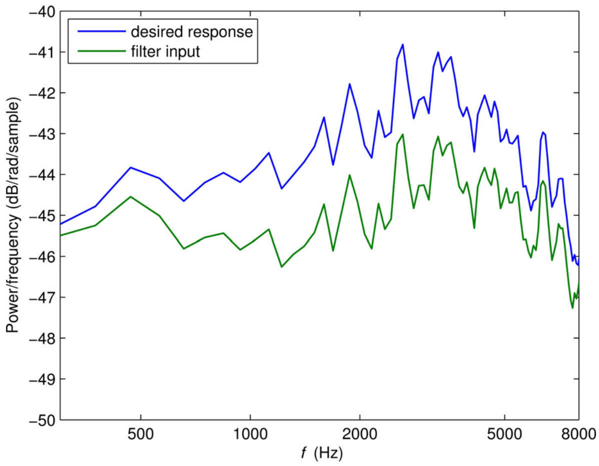

Fig. 4. Spectrum of the desired response and filter input signal

As we can see, there are large differences between the two PSDs. Therefore, we rely on the LMS adaptive filter to shape the output of the second microphone so it would resemble more to the output of the reference microphone. The filter order is N=33 and it was chosen empirically, as it gave the best results for our case and increasing it further did not make any improvement. As it was mentioned earlier, we need to be beware of the stability of the LMS algorithm. We choose appropriate value of the parameter μ in our experiment, which can be shown by looking at the tap weights as they converge smoothly to their final values (Fig. 5).

Fig. 6. shows the magnitude response of the adaptive filter for the final values of the coefficients. We can see that magnitude response compensates the difference between two PSDs presented in Fig. 4.

In Fig. 7. the overall response of the cascaded connection of the second microphone and adapted filter is presented. We can notice that the LMS adaptive filter improved the response of the second microphone. Resolving the problems, we had in the part of the spectrum that is important to our experiment. The area from 200 to 8000 Hz is almost identical.

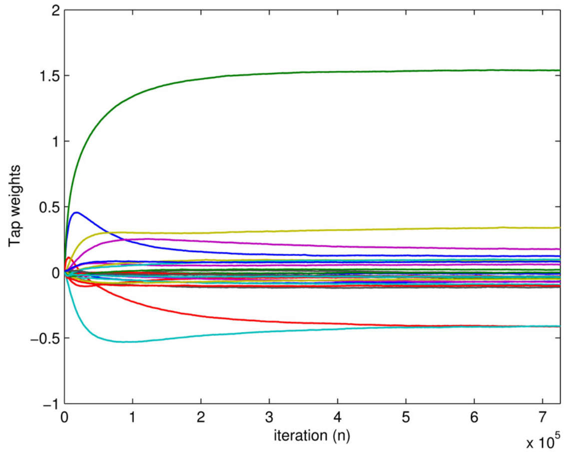

Fig. 5. Filter tap adaptation

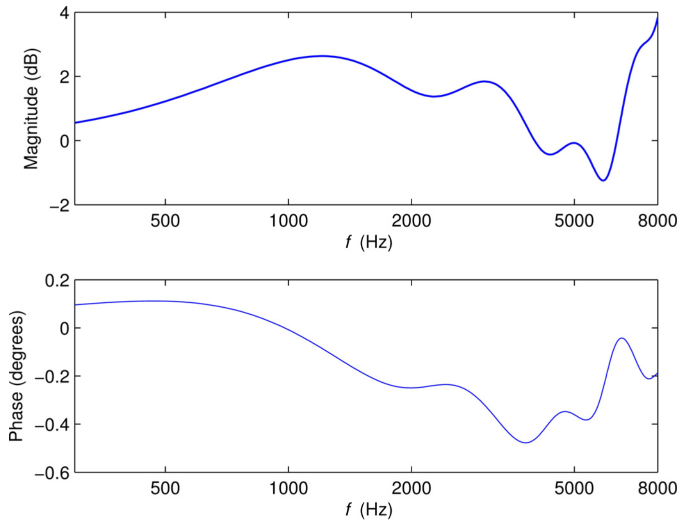

Fig. 6. Filter characteristics

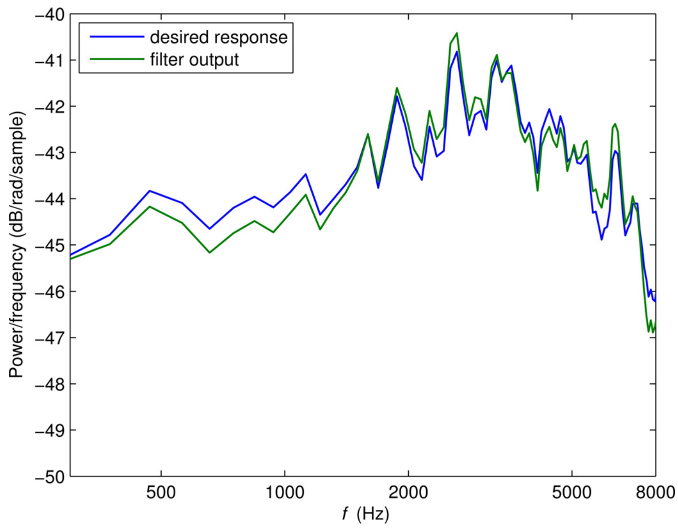

Fig. 7. Spectrum of the desired response and filter output signal

In Fig 8 we displayed the difference between the PSD of the filter input signal and the desired signal, and the difference between the PSD of the filter output and the desired signal As we can see, the filter output signal has smaller differences and is fluctuating around 0 dB which would be the ideal case with a maximum of about 2 dB. The difference of the filter input signal reaches a maximum of about 5 dB. This clearly states that the LMS filter improved the microphone response.

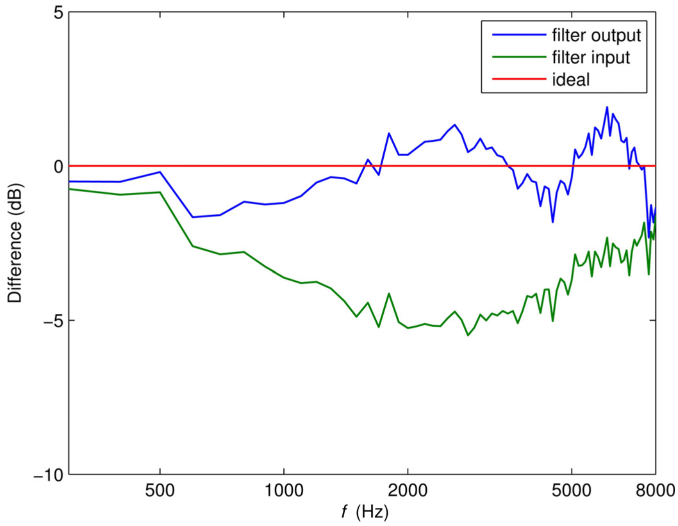

Fig. 8. Difference between spectrums

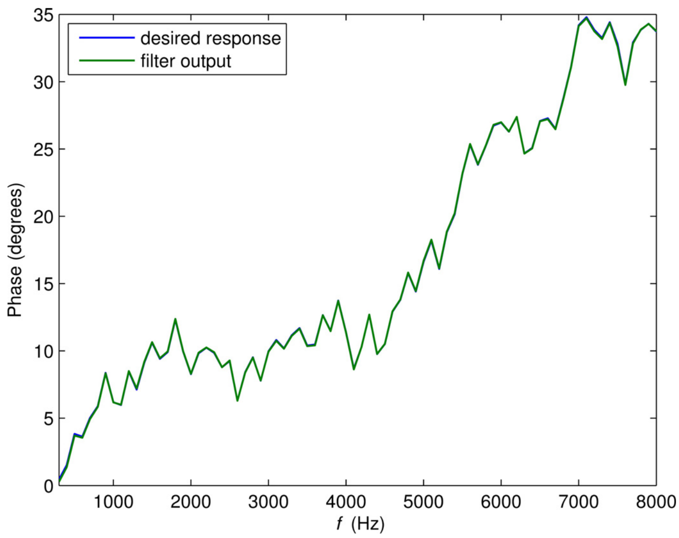

Fig. 9. Phase comparison

The phase spectrums of the two signals shows that the LMS filter did not corrupt the phase (Fig. 9).

In our experiment, we used two microphones placed very close to each other, since one of them was of lesser quality the received signals were quite different. We then showed that the performance of the microphone can be improved without any additional cost using digital processing, to be specific LMS adaptive filtering. This method provided us with a 3 dB improvement.

To see the effect of the microphone calibration on the accuracy of the localization in space two experiments were done with a planar microphone array with 24 microphones [8]. In the first experiment, none of the microphones were calibrated, while in the second experiment all of them were calibrated with the procedure described in this paper. In experiment, as the source of the sound we used a car horn that is located 10 meters from the microphone array. The horn of the used car is located behind the front left wheel, so the results of the localization using the acoustic camera was expected at this position. In Fig. 10. the results of both experiments are shown. The sound levels of sound sources were coded with different colors, relative to the value of the maximum sound level. The localization was done with the CLEAN-SC algorithm [9]. In the experiment that was done without the calibration, the resulting position of the localization was near the expected position, but it was not precise because the real sound source was located a meter away. In Fig. 10b). the position we got using the acoustic camera with all its microphones calibrated is shown. It can be noticed that the place marked as the source is significantly closer to the exact position then it was in the experiment without the calibration of the microphones. The location of the front left wheel was marked as the dominant sound source, which suits the position of the horn. Through these experiments it has been shown that the calibration of the acoustic camera microphones provides us with a more accurate localization of sound sources in space.

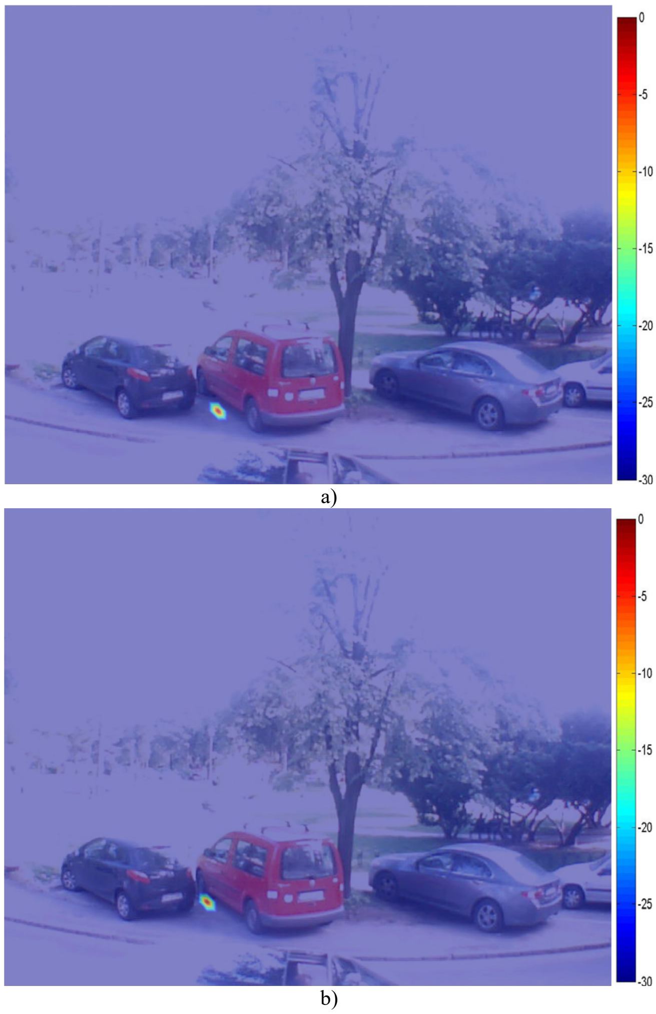

Fig. 10. Acoustic camera localization a) without calibration of the microphone, b) with calibration of the microphone

IV. CONCLUSION

Two microphones placed very close to each other received the same signal (sound) and because one of them is of lesser quality, the received signals were different with deviations in PSD up to 5 dB. Therefore, we wanted to enhance the ability of the microphone using the LMS adaptive filter. The signal obtained at the output of the better microphone was considered as the desired signal. After many experiments, we concluded that the best results are gotten when the filter order is N=33 and that the stability request of the adaptive filter is fulfilled. After the signal was filtered, improvement was noticeable since the deviations between the signals decreased from 5 dB to 2 dB. Also, it was important for us that the filter does not affect the phase considerably. This solution proves to be useful because we achieve a 3 dB gain without any additional spending, with only an easy programable software.

V. Acknowledgments

This paper is realized as a part of activities supported by Ministry of Education, Science and Technological Development, project TR-36026 and project TR-32023.

VI. References

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