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Comparison of LMS Adaptive Beamforming Techniques in Microphone Arrays

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Abstract: This paper discusses principles of microphone array beamforming, specifically the use of LMS algorithm with training sequence. The problem of wideband nature of acoustical signals and its impact on the techniques of beamforming are discussed. Detailed explanation of classic narrowband and wideband LMS beamformers is presented, as well as the modification of narrowband algorithm with pre-steering. Experimental testing and comparison of algorithm performances was conducted and measurement results are presented. The used microphone array is part of Brüel & Kjær acoustical camera, and is comprised of 18 omnidirectional non-uniformly spaced microphones.

Keywords: Microphone array, Adaptive beamforming, LMS, Pre-steering.

1 Introduction

Microphone arrays are microphone systems that are able to achieve far greater directivity as opposed to standard directional microphone constructions [1]. Their operation is based on the superposition of signals from multiple microphones, which are spatially distributed following a certain geometry. Prior to summation, signals undergo certain modifications in order to adapt system characteristics to the desired application. For example, one practical application of these systems is in conference audio systems [2], where an array of microphones is used to locate the active speaker, maximize the beam in his direction and then acquire the signal. Another popular application is the acoustical camera which can separate dominant sound sources or sound sources of certain frequency content on a given visual scene [3].

The key characteristics that describe the microphone array operation are the direction of maximum sensitivity (main lobe) and the shape of directivity function. Both characteristics depend on the spatial distribution of microphones as well as on the type of processing performed on the signals [4]. Predominant methods of processing the microphone signals are delay and scaling. By delaying the signals, one performs the directional steering of the main lobe [5]. It should be noted though, that this action inevitably leads to the distortion of

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the shape of directivity pattern. Shape itself can be manipulated with by scaling the microphone signals with certain weighting coefficients. Operations of delay and scaling can be realized by multiplying the analytical waveform of the signals with complex weighting coefficients. These coefficients are also called the spatial filter coefficients, and the process itself: spatial filtering or beamforming [4]. The laws which apply to filtering signals in time domain are also valid in the case of spatial filtering. Optimization of spatial filter characteristics is the result of algorithms used in sensor arrays. A great number of these algorithms originate from the field of antenna arrays in radio communications (array processing). Their application in the domain of acoustical signals is often impossible due to wide frequency bandwidth of the signals, hence certain modifications must be made.

This paper covers the experimental performance comparison of two adaptive LMS beamforming techniques in real, closed space conditions. Adaptive LMS algorithm [6, 7] is one of the basic algorithms used in array processing and this paper presents two variations of it: narrowband algorithm and wideband algorithm. These algorithms are widely spread in literature. This paper also suggests a modification of the narrowband algorithm that is referred to as the wideband algorithm with pre-steering. Pre-steering is a technique that attempts to overcome the issue of high computational complexity which is inherent in wideband LMS algorithms. With this technique, the issue of large frequency bandwidth of acoustical signals is dealt with by assessing the relative time differences of the signals arriving at different microphones, and then aligning the signals according to those delays. Time delays are determined by cross-correlating the signals from the microphones using a training sequence, with a resolution equal to the sampling period. After the pre-processing of signals, narrowband LMS algorithm is used. The performance of the modified algorithm is compared with the performance of the classic wideband LMS beamforming.

Signal acquisition is performed using a planar non-uniform microphone array consisting of 18 microphones, which is a part of Brüel & Kjær acoustical camera. The utilized training sequences were MLS sequence [8] as well as real acoustical signals.

The paper states the theoretical basis of adaptive narrowband and wideband LMS algorithms applied to microphone arrays. Afterwards, the pre-steering modification of the narrowband algorithm is described, followed by the description of the experiment along with the utilized equipment. At the end, the experimental results are presented followed by the most significant conclusions derived therefrom.

2 Theoretical Basis

2.1 Signal bandwidth interpretation

In terms of modelling signals in sensor array theory, the signal can be considered narrowband if mutual delays of complex envelopes of the signals detected by the microphones can be neglected. Then, for an array of L sensors, the following holds:

$$s(t-\tau_n) \approx s(t); \quad n = 1, 2, \ldots, L, \tag{1}$$

where $s(t)$ stands for the complex envelope of the signal, and τ_n is the delay between the signal on the sensor n in respect to some reference point. Equation (1) will be satisfied if the following condition is met:

$$B\Delta T_{\max} \ll 1, \tag{2}$$

where B is the frequency bandwidth of the complex envelope, and $\Delta T_{\max}$ is the maximal delay time. From (2) it is clear that both the frequency bandwidth of the signal and array geometry dictate whether the signal can be considered narrowband.

2.2 Narrowband beamforming

As was mentioned earlier, one of the means of beamforming is multiplication of signals with complex coefficients. The selection of coefficients dictates the direction of maximal sensitivity as well as the shape of the directivity function (beampattern) [4]. If the narrowband condition is met than the structure from Fig. 1 can be used for beamforming.

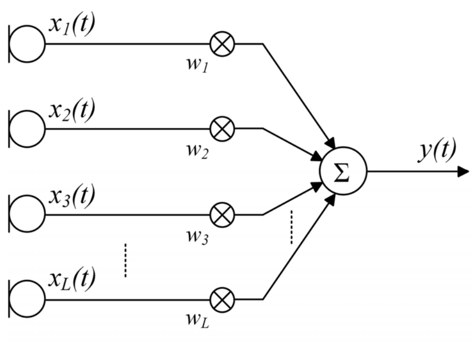


Fig. 1 – *Narrowband beamforming.*

Signals from microphone outputs are represented in their analytical form and then multiplied with the set of complex coefficients [9]. Beampatterns of a linear microphone array consisting of eight omnidirectional microphones are presented in Fig. 2. The topmost figure represents the beampattern of the array when no processing is applied to the signals, only simple summation of all signals. Basic parameters that describe the direction of the maximal sensitivity are the azimuth and elevation angle and main lobe width. The shape of directivity is also a function of frequency and Fig. 2 presents the beampattern for only one frequency. The middle picture in Fig. 2 shows a 30 degree shift of the direction of the main lobe (beamsteering). Main lobe shift is realized by using a set of complex coefficients of the following form:

$$w_n = 1\mathrm{e}^{\mathrm{j}\omega\tau_n}, \tag{3}$$

where τ_n is a time delay for the nth microphone. It is clear that the direction of main lobe is shifted towards the azimuth 30º. An unavoidable side effect that follows beamsteering is the distortion of the side lobes to a certain extent. The bottommost picture of Fig. 2 presents the manipulation over the shape of directivity function using coefficients of the following form:

$$w_n = A_n\,\mathrm{e}^{\mathrm{j}\omega\tau_n}, \tag{4}$$

where A_n is the modulus of the coefficient in the n^{th} branch of the array. The directivity function shaping presented in the bottommost picture of Fig. 2 is called null steering, which means setting a direction in which the directivity function will have the value of zero [5]. Null steering can only be achieved by coefficients of the type presented in (4).

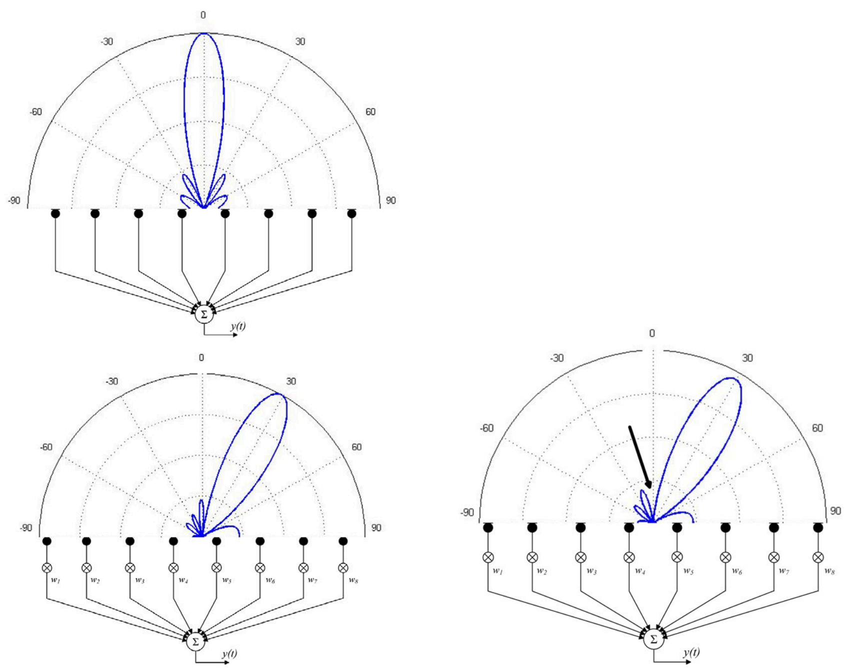


Fig. 2 – *Linear array beampattern without processing (top left), phase shift (bottom left), amplitude scaling and phase shift – null steering (bottom right).*

2.3 Wideband beamforming

Narrowband beamformers are based on the assumption that selected coefficients will introduce the same delay for all spectral components of the signal from the sensor. However, when signals are wideband, it is necessary to process the signals with a structure that takes into account different phase shifts of different spectral components of a wideband signal. This paper presents the so called TDL (tap delay line) structure [9] that achieves this. Instead of one coefficient in every branch of the microphone array, TDL consists of a number of delay lines, outputs of which are multiplied with complex coefficients and then summed up. This structure is presented in Fig. 3.

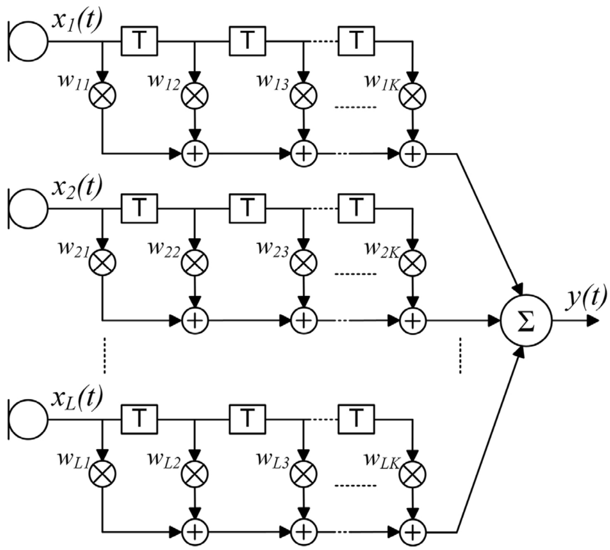


Fig. 3 – *Wideband beamforming.*

In Fig. 3 L is the number of microphones and K is the number of delay elements in each branch of the array. When K equals 1 the structure represents a narrowband beamformer. This structure allows the control of phases and amplitudes of multiple spectral components within a frequency band of interest, hence it can be used for wideband signal processing. Obviously, the computational complexity of the operations performed in this structure is vastly greater than that of a narrowband beamformer.

2.4 Adaptive beamforming

Application of adaptive algorithms in microphone array field is based on using a predefined signal to train the array. From a desired spatial direction a training signal is emitted and it is used as a reference for adapting the weighting coefficients. When the signal is acquired and processed it is compared to a reference training signal with the goal to minimize the difference between the two. If the signal acquisition is performed in the presence of interfering signals, the algorithm will create a beampattern that favours the direction from which the training signal comes, and suppresses the signals arriving from the direction from which interference occurs. After coefficient optimization, the array can be used to extract the signal from the direction for which it is adapted. Optimization algorithm can be any of the usual algorithms such as: LMS (Least Mean Square), NLMS (Normalized Least Mean Square), RLS (Recursive Least Squares) etc. [9]. Fig. 4 shows the structure of adaptive beamforming with LMS algorithm.

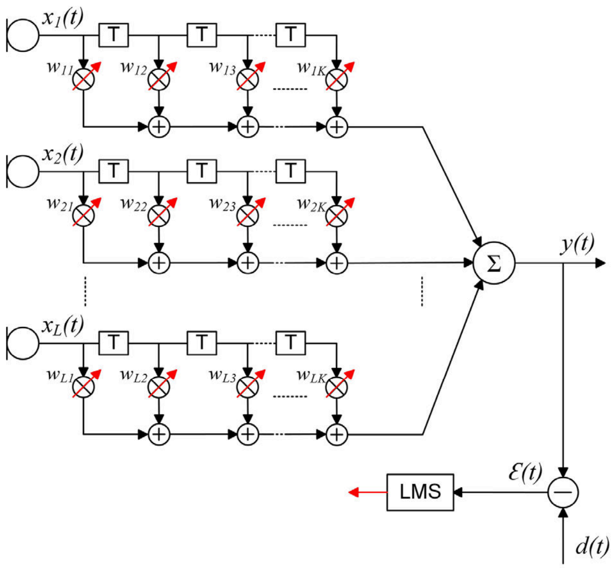


Fig. 4 – *Adaptive beamforming.*

In Fig. 4 $d(t)$ denotes the predefined training sequence, and $\varepsilon(t)$ the error signal. This paper looks into application of LMS algorithm for minimization of mean square error in adaptive beamforming. Calculation of coefficient values for the optimal spatial filter is based on the gradient descent optimization method [5]. In the n^{th} iteration coefficients are calculated for $n+1^{st}$ iteration, which will filter the training sequence. In the $n+1^{st}$ iteration the mean square value of the error signal is calculated, which is the difference between the signal on the output of the array processing structure and the reference signal. Based on this value, optimization of coefficient values for $n+2^{nd}$ iteration is performed. This algorithm is performed iteratively until the optimal value of coefficients is obtained for which a certain condition related to the size of the error is met. Expression for the mean square error represents a positively defined quadratic form that in theory has one global minimum. Spatial filter coefficients values that give a minimum of this function represent the optimal values.

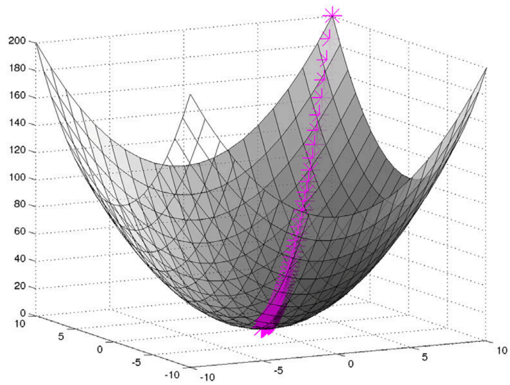


Fig. 5 – *Gradient descent method.*

Graphical illustration of the gradient descent method is shown in Fig. 5. The goal is to find the quickest route from the start point on the surface of this function to the minimum which corresponds to optimal coefficient values. Values in the n+1st iteration are calculated as in (5):

$$w(n+1) = w(n) + \mu\nabla\xi(w(n)), \quad (5)$$

where μ is gradient step size, which is a positive scalar value that controls the speed of convergence, and $\nabla\xi(w(n))$ represents the gradient of mean square error. From Fig. 4 it is obvious that the error signal can be calculated using the following formula:

$$\varepsilon^*(w(n)) = w^* x(n+1) - d(n+1), \quad (6)$$

where (.)* denotes complex conjugate transposition. From (5) and (6) the final expression for the coefficient values can be written as:

$$w(n+1) = w(n) - \mu 2\, x(n+1)\, \varepsilon^*(w(n)). \quad (7)$$

The value of parameter μ determines the speed of convergence and the size of the residual error. For bigger steps the speed of convergence is greater, but so is the residual error, and vice versa.

2.5 Pre-steering method

Pre-steering represents a modification of narrowband beamformer which gives a structure that is suitable for wideband signal processing, but is much less computationally complex with respect to classic wideband beamformers. As with the basic narrowband beamformer every branch consists of only one complex weighting coefficient. However, prior to multiplication, signals from each microphone are delayed by a time interval called a pre-steering delay time (τ_n in Fig. 6). The purpose of delay blocks is to achieve mutual time alignment of all signals, so that mutual delays are no longer than one sampling period. The values of delays for certain signals are determined based on cross-correlation of received signals. In order to precisely determine relative delays, based on cross-correlation, the training sequence must possess good cross-correlation characteristics. Therefore, in experiments presented in this paper, an MLS pseudo-random sequence is used (Maximum Length Sequence). After the alignment with pre-steering, coefficient optimization is carried out so as to create a null in the interference direction. Adaptive beamformer with pre-steering is illustrated in Fig. 6.

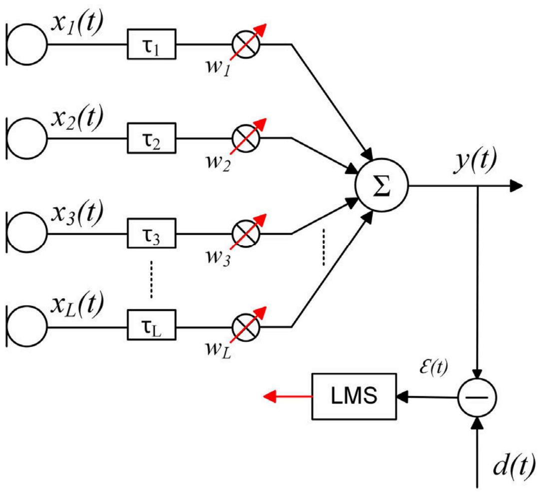


Fig. 6 – *Pre-steering method.*

3 Experimental Setup

This chapter gives a brief description of the experimental setup and equipment that has been used. Measurements were performed in closed space conditions in a reverberant space, where multiple reflections are present. As the source of acoustical signals studio monitors JBL LSR6325P-1 were used [10]. The microphone array consists of 18 non-linearly spaced microphones which are a part of B&K acoustical camera [11]. Prior to measurement, the calibration of each microphone has been carried out. Signal acquisition was performed utilizing a hardware-software solution PULSE from B&K. Experimental setup is shown in Fig. 7.

Fig. 7 – *Experimental setup.*

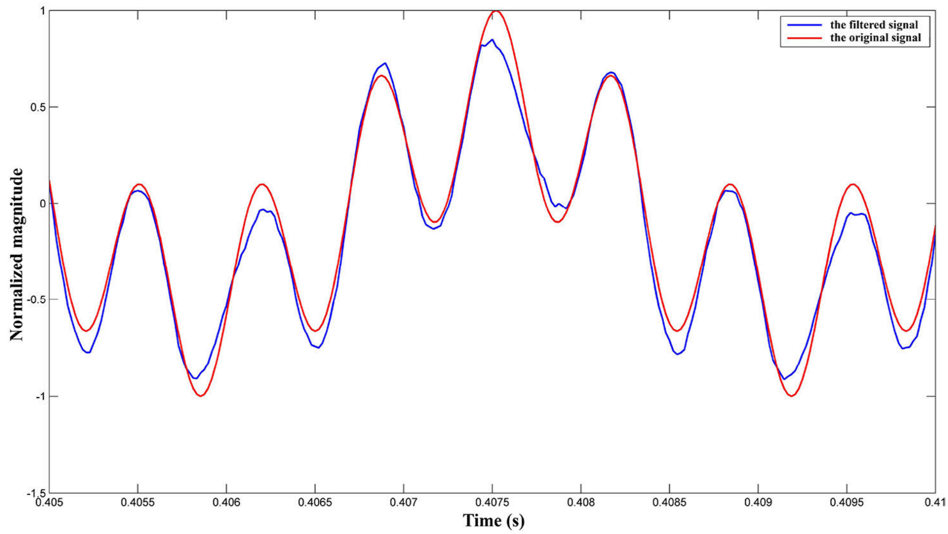
4 Experimental Results

This chapter presents the results obtained using the three different adaptive algorithms. Comparison of results obtained with adaptive narrowband beamformer, adaptive wideband beamformer and adaptive beamformer with pre-steering is performed. Comparison criteria are speed of convergence, error convergence and residual error.

In the first scenario the performance of narrowband algorithm is tested with a narrowband excitation, which is the sum of two sinusoidal signals with frequencies 300 Hz and 1500 Hz, and with a wideband excitation using the MLS sequence. The results are displayed in Fig. 8.

In Fig. 8 red colour represents the original signal, whereas blue represents the signal from the output of the array processing algorithm. In the ideal case these two signals would overlap. It is noticeable in the upper figure that the output signal follows the original one to a certain degree. This means that the algorithm is able to perform the adaptation. However, in the case of a wideband excitation, the narrowband algorithm is unable to adapt. This is obvious from the bottom figure, since it is clear that the output signal does not follow the changes of the input. This test was designed as an illustration of the algorithm behaviour and its qualitative assessment, and no quantitative evaluation was performed.

The second scenario compares the performances of two variants of wideband algorithms. The excitation signal is an MLS sequence reproduced from the “right” speaker located at azimuth 30º and elevation of 90˚. The comparison of signal waveform and error convergence is performed. Both algorithms utilize step size of 10^{-3} for the LMS algorithms, and the adaptation is performed over 30000 samples. TDL structure consists of 300 taps in each branch of the processing block.



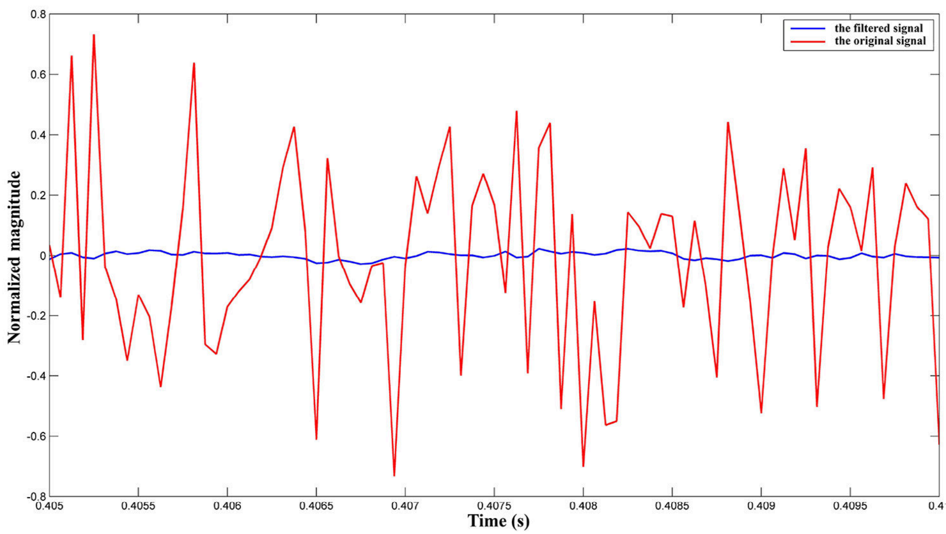


Fig. 8 – *Narrowband algorithm. Sinusoidal excitation (top), MLS excitation (bottom).*

Fig. 9 shows the error convergence for the two algorithms. Convergence time is approximately the same for both algorithms. It should be noted however, that the error value at the beginning of adaptation process is about 30 times smaller in the case of the TDL algorithm. After the adaptation process, signals are filtered with the optimized set of coefficients to give the output signal. Fig. 10 shows segments of output signals from both algorithms together with the corresponding training sequence. Results displayed in the figure demonstrate a slightly better match for the TDL algorithm, which comes at the expense of greater computational complexity.

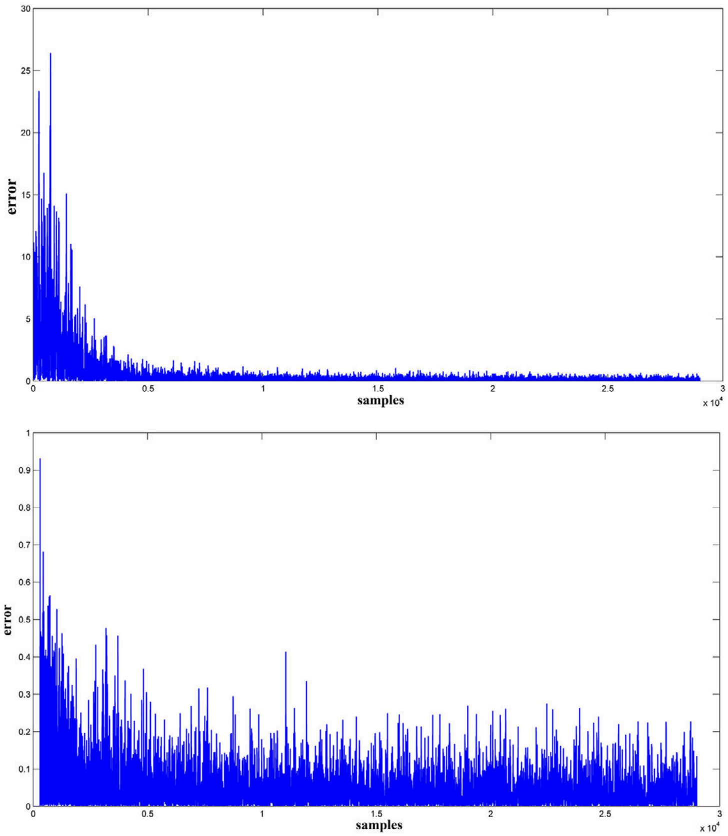
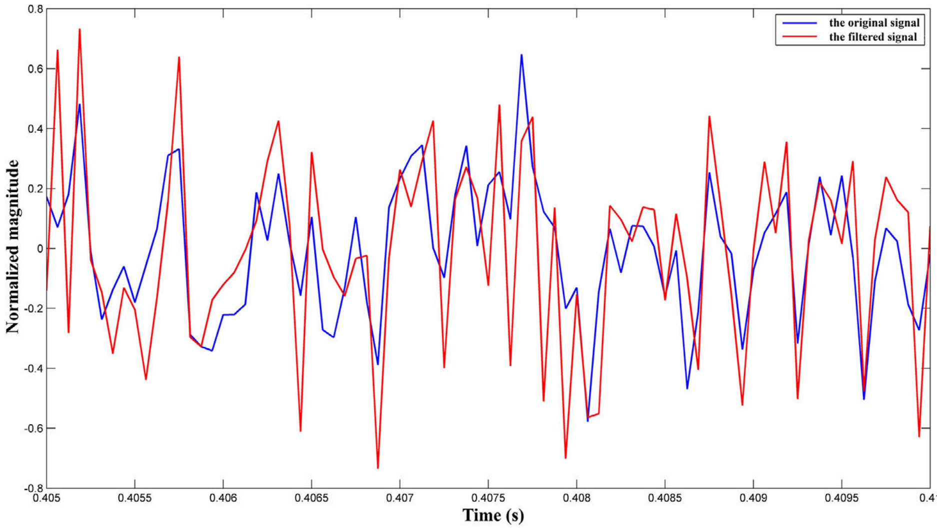


Fig. 9 – *Error convergence: pre-steering algorithm (top), TDL algorithm (bottom).*

The comparison of residual error of the two algorithms is performed. The error signal is computed as the difference between the original signal and the output signal after array processing. Residual error power is approximately 0.052 in the case of pre-steering algorithm, whereas in the case of TDL algorithm it is 0.026 . This 3 B difference in favour of TDL algorithm corresponds to the difference seen in Fig. 10. The acceptable difference between the output signal and the original one is a matter of subjective impression and concrete conclusions can be made only after conducting extensive psychoacoustic research.



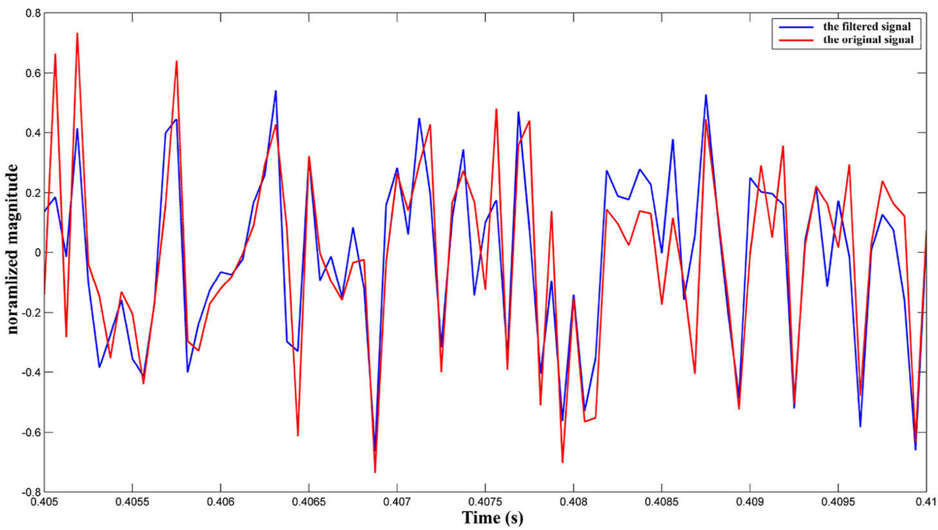


Fig. 10 – *Output signal and training sequence: pre-steering algorithm (top), TDL algorithm (bottom).*

In the third experiment signals are reproduced simultaneously from both speakers. Right speaker reproduces the signal of interest, whereas the left speaker reproduces the interfering signal. The purpose of this experiment is to separate the signal of interest by suppressing the interference. Reproduced signals are guitar tones. The right speaker reproduces the note G (signal of interest), left speaker reproduces note H (interfering signal). Each microphone will detect the signal that represents the sum of the two mentioned signals, along with all the reflected components that occur in closed space. Ratio of the signal of interest and the interference will depend on the position of the microphone relative to the speakers. Adaptive algorithms define coefficients of spatial filter that has maximum in the direction of the speaker that reproduces the signal of interest, and forms a minimum in the direction of interference. Ideal case would be complete suppression of the interfering signal. Since this is not practically possible it is necessary to quantify the gain in interference suppression achieved by utilizing a microphone array compared to using only one microphone.

Signals of guitar tones consist of fundamental frequency component and higher harmonics. Fundamental frequency component of note G is at 196.00 z, while of note H is at 246.94 z. Most of the signal energy is concentrated within the first couple of harmonics. Therefore, this experiment considers fundamental frequency component and the first two harmonics. Fig. 1 shows the spectrum of the signal detected by one microphone. This signal consists of spectral components of the signal of interest and of the interfering signal. Arrows indicate the spectral components of the signal of interest. It can be seen in this figure that the level of the fundamental component of the signal of interest is relatively low compared to the interfering signal. To separately measure the energy of every component in the spectrum, filter banks are used. A filter bank has been designed using previously developed software [12]. The resulting filter bank has all-pass complementarity property, which makes it suitable for application in audio signal processing. In Fig. 1 the amplitude characteristics of the filter bank that extracts the signal of interest are indicated with green colour, whereas the red colour represents the bank that extracts the interfering signal.

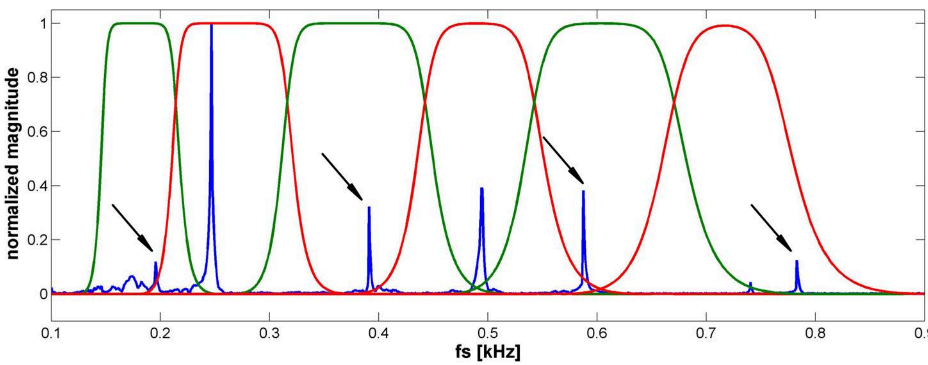


Fig. 11 – *Signal of one microphone with complementary filter banks.*

Since the frequencies of components are known, it is possible to make a complementary bank so that each filter in this bank will separate only one spectral component. By calculating the signal levels at the outputs of filters it is possible to quantify the ratio of signal to interference before the use of microphone array structure. **Table 1** shows the levels of individual spectral components as well as the total signal level detected with one microphone. The levels indicated in Table 1 are obtained using the following formula:

$$L_{spl} = 20\log\left(\frac{p_{rms}}{p_0}\right), \tag{8}$$

where p_{rms} is the root mean square of sound pressure signal after passing through the filter corresponding to the given harmonic, and p_0 is the reference value of 20 Pa. It should be noted though, that signals from the output of the filters will be comprised not only of note harmonics, but also of any spectral content present in the filtered band. This spectral content is the consequence of ambient noise.

Table 1
Signal levels detected with one microphone.
Note G – *signal of interest. Note* H – *interfering signal*
Fundamental component level, 2. *1st harmonic level,* 3. *2nd harmonic level*
All levels indicated are expressed in dB.

	1.	2.	3.	**Sum**
Note G	69.90	71.92	73.93	**77.07**
Note H	82.53	77.36	57.98	**83.7**

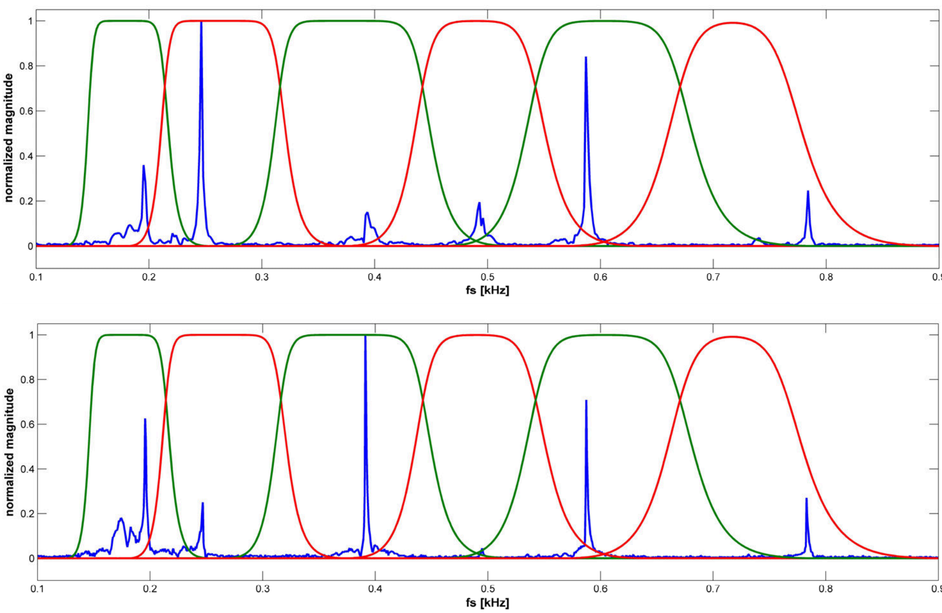


Fig. 12 – *Filtered signals. Pre-steering algorithm (top), TDL algorithm (bottom).*

The signal of interest (note G) represents the training sequence. The training process is performed while reproducing simultaneously both the signal of interest and interference. After obtaining the optimal set of coefficients and filtering the signal, we use the previously described methodology to determine the levels of signal and interference from the output of the array structure. Fig. 2 shows the spectrum of the signals for pre-steering and TDL algorithms. It can be concluded that TDL algorithm separates the signal of interest more efficiently. However, it can be seen that with pre-steering algorithm certain gain is also achieved compared to using one microphone, which is manifested as the greater relative level of fundamental component of the signal of interest compared to one presented in Fig. 1. The results are quantified in **Table 2**. The levels indicated are obtained using (8).

Table 2

Signal levels after array processing.
Note G – signal of interest. Note H – interfering signal
1. *Fundamental component level,* 2. *1st harmonic level,* 3. *2nd harmonic level*
All levels indicated are expressed in dB.

	1.	2.	3.	**Sum**
Note G pre-steering	64.77	57.93	69.99	**71.31**
Note H pre-steering	70.67	59.90	40.87	**71.02**
Note G TDL	72.46	73.33	70.45	**77.07**
Note H TDL	64.22	40.83	44.87	**64.29**

4 Conclusion

The comparison of three different beamforming algorithms utilizing LMS optimization was performed. Results of the experiments with narrowband algorithm illustrate the simplicity and good performances as long as the condition for narrow frequency bandwidth of the signal is met. The second experiment compares the performances of two wideband algorithms with MLS training sequence. Results show that convergence time is approximately the same for both algorithms, but that the residual error is about two times smaller when using the TDL algorithm. The third scenario uses a real acoustical signal as the training sequence as well as interfering signal. TDL algorithm suppresses the interference by approximately 19.5 dB, whereas pre-steering achieves 12.68 dB, but also weakens the signal of interest by 6 dB. Based on the results it can be concluded that TDL algorithm will always perform better at the expense of increasing the computational complexity. Practically, this means that the execution time of operations for TDL algorithm is much greater, and pre-steering might prove to be an optimal solution for certain applications due to its simplicity and ease of implementation. The main restriction for the pre-steering algorithm is the necessity for having a training signal with good cross-correlation characteristics. If that is not the case, this algorithm is much more sensitive to reflections of the original signal which exist in closed space, compared to TDL algorithm. The optimized set of coefficient produced by any of the algorithms can be used to extract the signal coming from the direction of the training sequence as long as it has spectral characteristic similar to those of the training sequence.

5 References

[1] M. Mijić: Audio sistems, Akademska misao, Belgrade, Serbia, 2011. (In Serbian).

[2] Y. Tamai, S. Kagami, H. Mizoguchi, K. Sakaya, K. Nagashima, T. Takano: Circular Microphone Array for Meeting System, Sensors, Vol. 2, pp. 1100 □ 1105, 2003.

[3] M. Eric: Some Research Challenges of Acoustic Camera, 19th Telecommunications Forum, TELFOR, 22□24 Nov. 2011, Belgrade, Serbia, pp. 1036 – 1039.

[4] H. L. Van Trees: Optimum Array Processing, John Wiley and Sons, New York, USA, 2002.

[5] L. C. Godara: Smart Antennas, CRC Press, New York, USA, 2004.

[6] S. Haykin: Adaptive Filter Theory, Prentice Hall, New Jersey, USA, 2002.

[7] B. Widrow, J.M. McCool, M. Larimore, C.R. Johnson: Stationary and nonstationary learning characteristics of the LMS adaptive filter, Proceedings of the IEEE, Vol. 64, No. 8, 1976, pp. 1151 – 1162.

[8] A. Mitra: On the Properties of Pseudo Noise Sequences with a Simple Proposal of Randomness Test, World Academy of Science, Engineering and Technology, International Scholarly and Scientific Research and Innovation, Vol. 2, No .9, 2008, pp. 631 – 636.

[9] W. Liu: Wideband Beamforming, John Wiley and Sons, New York, USA, 2002.

[10] Datasheet and technical documents. http://www2.jblpro.com/catalog/General/Product.aspx?PId=24&MId=7.

[11] Datasheet and technical documents. http://www.bksv.com/products/transducers/acoustic/acoustical-arrays.

[12] J. Ćertić, D. Šumarac, I. Pavlović, Salom: Nonuniform Complementary Filter Bank for Analysis of Audio Signals, Forum Acusticum, Alborg, Denmark, 2011.