ANALYSIS OF MULTI-SENSOR BEAMFORMERS

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Abstract

Two basic multi-channel noise suppression methods are compared — Frost's adaptive beamformer and Zelinski's beamformer with adaptive postfiltering. Experimental results are shown and compared with theoretical assumption.

Introduction

Many different noise suppression systems have been proposed during last years. A group of multi-channel methods should be pointed out. The principal advantage of these methods is a spatial selectivity realized by summing signals from input channels of the system during sampling. This idea leads to relatively easy and robust algorithms designed without making strict hypotheses about the acoustical environment or processed signals.

The main approaches of multi-channel noise suppression methods can be represented by three systems: Frost's adaptive beamformer [1], Zelinski's beamformer with adaptive postfiltering [2] and Simmer's Linearly constrained beamforming with adaptive constraint values [3].

Adaptive beamformer and Beamformer with adaptive postfiltering are studied and compared in this paper. Theoretical background and limits of both algorithm are presented in the theoretical part. The second part introducing used comparative methods. Results of simulations are shown and discussed in the experiments and the conclusion part respectively.

Algorithms description

Diagram of *adaptive beamformer* described by Frost in his work [1] is on Fig. 1. The system consists of K microphones. There is an adaptive filter with J adjustable weights behind each microphone. The output of the system is summation of all channel outputs. The filters weights are counted in order to solve expression:

$$\min W^T R_{xx} W, \quad \text{subjected to } C^T W = H, \tag{1}$$

where W is a matrix of the filter weights, R_{xx} is a covariance matrix of the input signals, H is a vector of the coefficients of a filter with required frequency response and C is a constant vector. The way how to count the optimal filter weights and adaptive LMS algorithm solving (1) is described in [1]. The system keeps the frequency response in the look-direction defined as direction orthogonal to sensors plane and minimizes the output power of nonlook-direction signals.

Main conditions of proper work of the system are correlation between noise samples and no correlation between signal and noise samples. The noise source with correlated samples is called source of coherent noise. Frost's system works well if the number of the sources is lower than number of channels. Performance of the system in case of incoherent noise is limited.

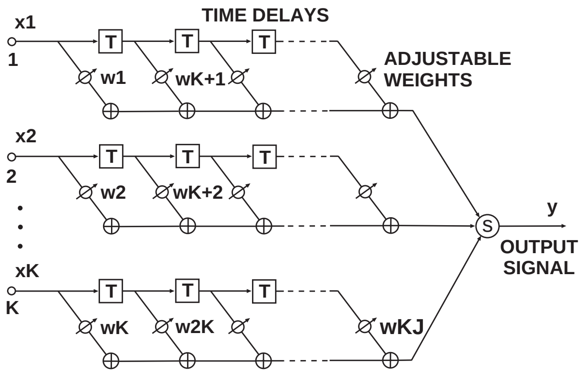

Fig.1: Adaptive beamformer.

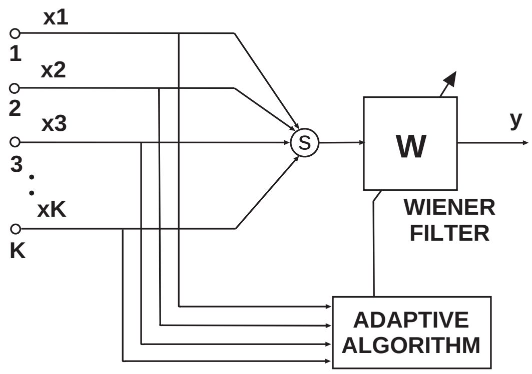

Fig.2: B'former with adaptive postfiltering.

Diagram of basic *beamformer with adaptive postfiltering* published by Zelinski [2] is on Fig. 2. The system can by divided into two parts. The first part, called conventional beamformer (DAS), is a microphones array followed by a summation block. The second part is a Wiener filter. Weights of the filter are derived from input signals. If the signal and the noise are uncorrelated, the weights can be evaluated using this expression:

$$W(f) = \frac{\Phi_{ss}(f)}{\Phi_{ss}(f) + \Phi_{nn}(f)}, \tag{2}$$

where $W(f)$ if the transfer function of the Wiener filter, $\Phi_{ss}(f)$ is the power spectral density (PSD) of desired signal and $\Phi_{nn}(f)$ is the PSD of the noise at the output of the beamformer. A way how to obtain optimal weight was showed by Zelinski in [2].

This system achieves high incoherent noise reduction with relatively small number of microphones. If a coherent noise appears, distortions will be noticeable in the output signal.

Comparative criteria

The most important criteria used in multi-channel processing are (see [4]): geometry of microphone array, type of suppressed noise, directivity pattern, directivity index, frequency response, desired signal distortion, steering error behaviour, speed of convergence etc. Criteria used in the next section are described below.

Since beamforming algorithms rely on numeration of the correlation between input signals samples, it is necessary to distinguish types of noise taking into consideration this fact. Useful criterion is the coherence defined as:

$$\Gamma(f) = \frac{\Phi_{ij}(f)}{\sqrt{\Phi_{ii}(f)\Phi_{jj}(f)}}, \tag{3}$$

where $\Phi_{ii}(f)$ is the PSD of i-th channel signal and $\Phi_{ij}(f)$ is the cross power spectrum density of i-th and j-th channel signals. Three basic noise type are distinguished according to coherence function: incoherent, coherent and diffuse. Relevant coherence functions are on Fig. 3. The coherent noise arises from a small number of noise sources in area without reverberation. The diffuse noise arises from high number of noise source or in an reverberation area.

Spatial selectivity of microphone array for coherent and diffuse noises is given by the directivity pattern and the directivity index respectively. Since coherent noise comes to a system from one direction, the directivity pattern is defined as the signal reduction coefficient in relevant angle. The incoherent and the diffuse noises come to a system from all directions, therefore the directivity index is the ratio of the signal power received in look-direction to the average power received from others directions.

Last characteristic used in this paper is the frequency response. This criterion represent dependency of the transmission on a frequency and a angle.

Experiments and results

Frost's and Zelinski's algorithms were modelled and tested in MATLAB. Synthetic as well as real signals were used for the tests. The parameters of the tested systems and tests results are given in this section.

Frost's algorithm — Configuration of the system: number of sensors $K = 4$, linear microphone spacing, $d = 5cm$ for coherent noise, $d = 21cm$ for diffuse noise, number of filters taps: $J = 21$, sampling frequency of used noise: $f_{sm} = 22kHz$, sampling frequency of simulated noise: behind microphones $f_s = 8kHz$, sampling frequency in the front of microphones: $f_{se} = 80kHz$.

The directivity characteristic of the array averaged over all frequencies and the transfer function for coherent noise from direction of 30 degree are shown on Fig. 5. The same characteristics but for diffuse noise are shown on Fig. 6. It can be seen that while the algorithm works well for coherent noise — suppression of signal in the noise direction is the highest (the ratio of the output power to the input power is about 13dB) and the transfer function have minimum on the relevant angle, it behaves as a DAS in the case of diffuse noise (compare with Fig. 7 below) where the periodic peaks of the transfer functions are caused by periodicity of microphone arrays. It can be concluded from performed tests that the suppression rate and shape of the directivity characteristic depend on the angle of arriving noise. The same holds about the relation between the microphone spacing and the directivity characteristic.

Zelinski's algorithm — Configuration of the beamformer with adaptive postfiltering was similar to previous system: $K = 4$, lin. spac. of microphones, $d_{coh} = 5cm$, $d_{diff} = 21cm$, $J = 21$, sampling frequencies: $f_{sm} = 22kHz$, $f_s = 8kHz$, $f_{se} = 80kHz$.

Performed tests shown that the system behaves similarly to a DAS in case of coherent noise (compare directivity characteristics and transfer functions on Figs 7 and 8 above). On the Figs can be seen that the main difference is in the transfer function where Zelinski's algorithm makes zeros due to the Wiener filter. Another situation becomes in the case of diffuse noise. The Zelinski's algorithm sets the filter weights to minimize the power of the output signal. It results to the directivity characteristic showed on figure 8 below. Since diffuse noise affects the beamformer from all directions the algorithm tries to minimize the output power form all directions. The noticeable difference can be find in comparison with the DAS on Fig. 7 below where the transfer function reach higher level of suppression on all frequencies.

At last the directivity index was measured. Comparison of the characteristics for both algorithms as a function of frequency is showed on figure 4. The picture shows that Zelinski's algorithm gives better performance, especially on lower frequencies. The directivity index of Zelinski's algorithm of the particular tested case were 7.30 dB while of the DAS 5.83dB.

Conclusions

The basic features of the algorithms were verified. Conclusions as follows can be made.

Frost's algorithm works well with coherent noise. Greater number of microphones increases a quality of the directional response and the noise suppression ratio. Short distance between microphones is necessary to reach good noise suppression ratio in wide angle. The problem of the algorithm is the requirement of great number of microphone to reach good features.

Zelinski's algorithm works well with diffuse noise. This fact leads to smaller number of microphones but also to greater microphone spacing causing worse directional response. Zelinski's system is based on a DAS therefore these two systems behave similarly. When the DAS works well Zelinski's algorithm increases noise suppression with the same number of microphones. The main problem of Zelinski's algorithm is the distortion of a useful signal if there is a coherent noise in the input signal.

The fact that both algorithms are restricted to the given types of noise limits applications of these systems. Since the natural environment contains all types of noises it is necessary to use a system independent of a noise type. One of solutions of the problem is shown in [3]. Next work will be aimed to this way.

Acknowledgement

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References

[1] Frost III, O.L.: *An Algorithm for Linearly Constrained Adaptive Array Processing*, Proceedings of the IEEE, vol. 60, no. 8, 1972

[2] Zelinski, R.: *A Microphone Array With Adaptive Post-Filtering for Noise Reduction in Reverberant Rooms*, ICASSP'88, New York, April 1988

[3] Fisher, S., Simmer K.U.: *Beamforming Microphone Arrays for Speech Acquisition in Noisy Environments*, Speech Communication, vol. 20, Elsevier Science B.V. 1996

[4] Ingerle, J.: *Beamformers Comparison Criteria*, Contribution on Poster 2001, Prague, May 2001

[5] Ingerle, J., Štrupl, M.: *Study of Multisensor Beamformer*, Contribution to Czech-German Speech Processing Workshop 2000, Prague, September 2000

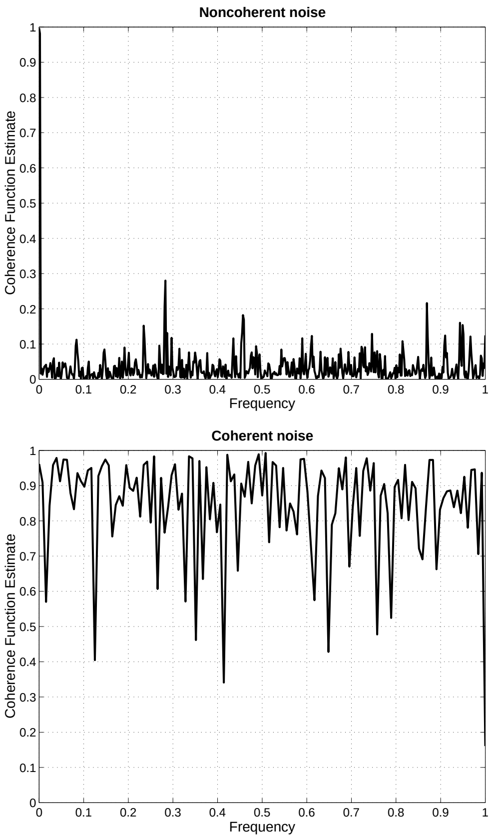

Fig.3: Coherence function of noise types

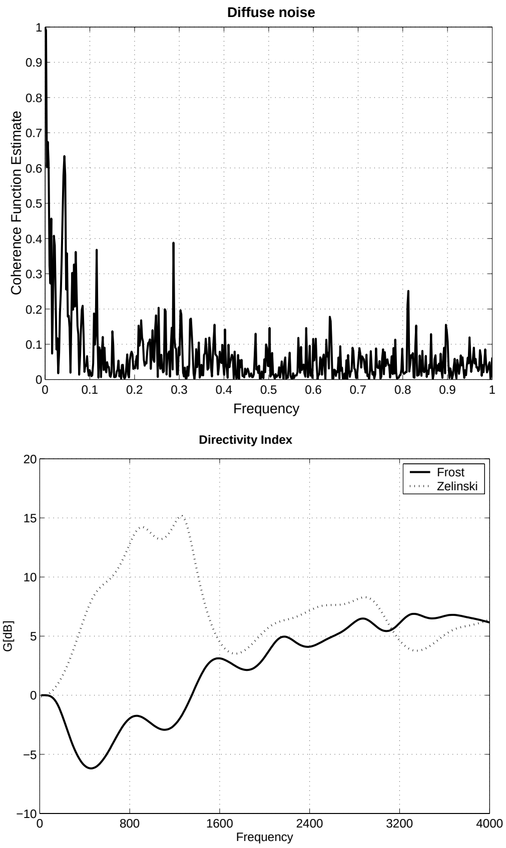

Fig.4: Directivity index of the systems

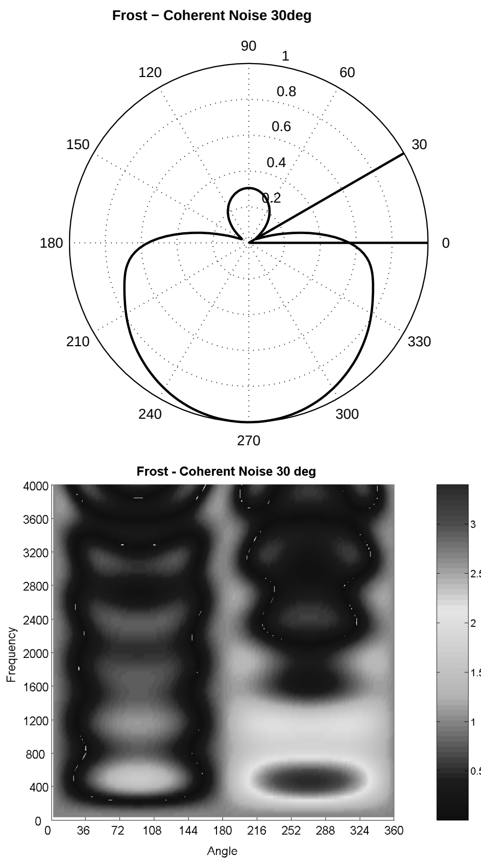

Fig.5: Frost's: Coherent Noise

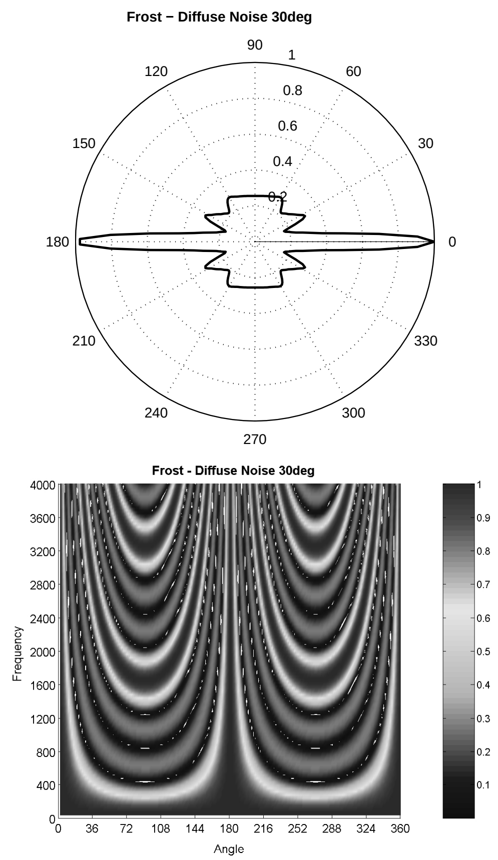

Fig.6: Frost's: Diffuse Noise

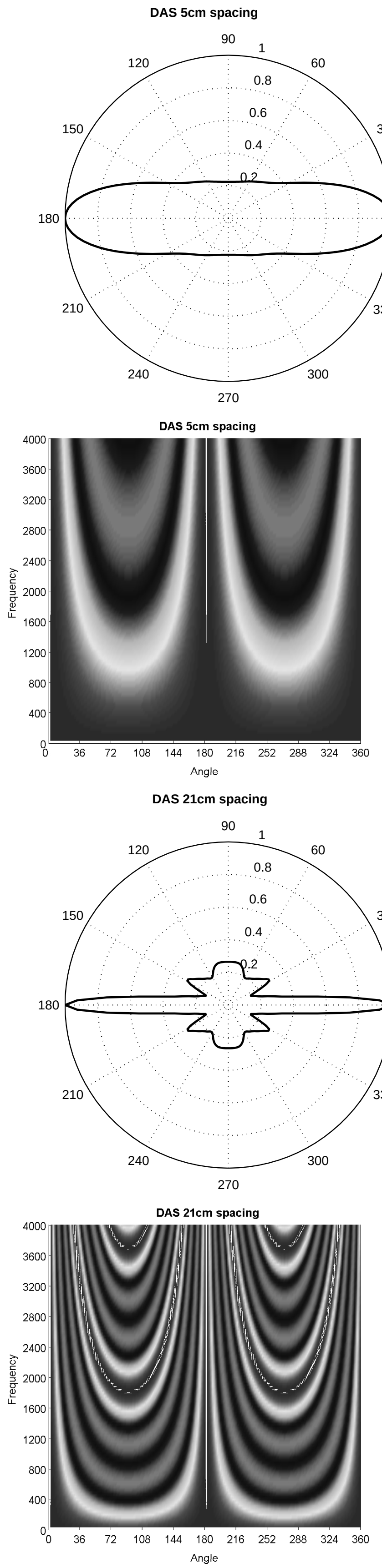

Fig.7: Directivity and tr. function of D&S 5cm and 21cm microphone spacing

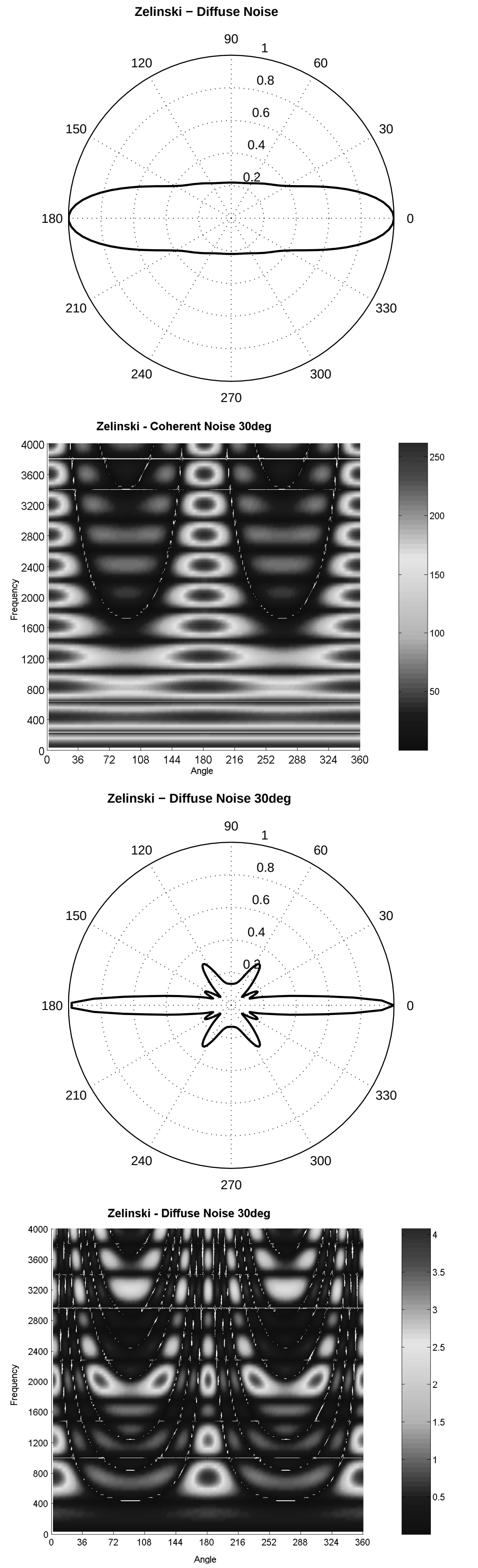

Fig.8: Directivity and tr. function of Zelinski's b'mformer for coherent and diffuse noise