Basic investigations of microphone arrays

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Introduction

In order to localize and separate sound sources at structures in airflow a microphone array was developed. In advance, basic investigations about the generation of directional characteristics of microphone arrays were carried out [1]. The knowledge obtained during these investigations illustrated on the example of a line array is content of this article. For the improvement of the localization results a frequency band averaging technique was applied. The obtained computer simulation results were verified by measurements on real model sound sources.

Directional Characteristic

Moving a single-frequency sound source radially around a microphone array and adding the sound pressure values at the microphones in phase causes the development of a directional characteristic (also called array pattern). For a two-microphone arrangement, the smallest form of a microphone array, a main lobe is generated at a minimum distance d between the two microphones greater than a half wavelength λ (see Figure 1 left). The measured sound pressure values correspond to spatial sample values of the sources sound wave at a fixed time. Increasing d causes spatial under-sampling. This effect is reflected in the occurrence of spatial aliasing in terms of side lobes that do not allow a straightforward localization (see Figure 1 right).

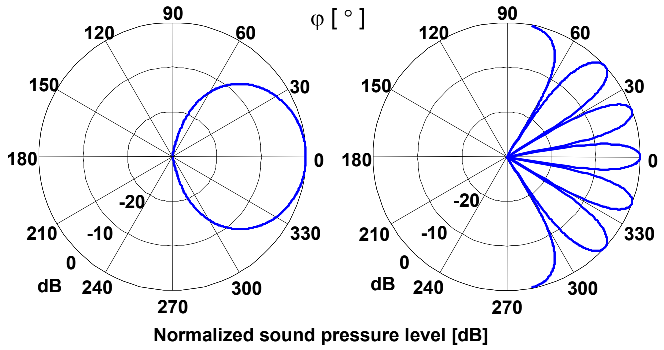

Figure 1: Directional characteristics of sound pressure for two microphones of different distance, left: $d = \lambda/2$, right: $d = 3\lambda$.

The spatial under-sampling can be counteracted by increasing the number of microphones. The spatial aliasing effects are reduced by the attenuation of the array patterns side lobes. This increases the range of straightforward source localization (usable analysis width). For the investigations described here the number of microphones was fixed to 8.

The array pattern also possesses a frequency dependency. Raising the frequency and thus lowering the wavelength λ of the source signal corresponds to an extension of the array width at a constant frequency. Figure 2 shows the array pattern of an equidistant line array dependent on frequency over a range of 12800 Hz. It can be recognized that the main lobe remains at the center position and decreases in width with increasing frequency. By contrast the side lobes differ in their positions for different frequencies. This fact is the basis for a frequency band averaging technique described later.

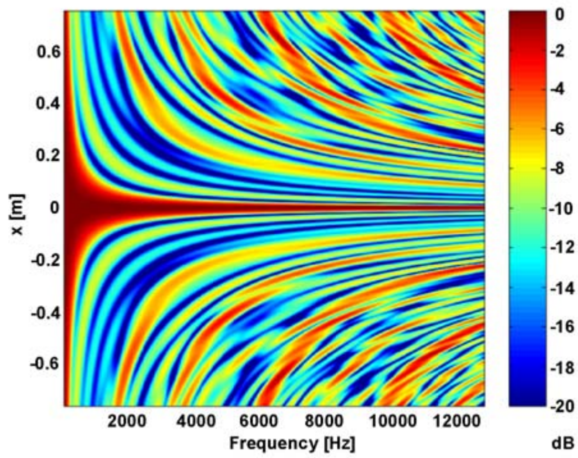

Figure 2: Array pattern of equidistant line array with 8 microphones over a frequency range of 12800 Hz.

Computer Simulations

For further investigations the far field beamforming algorithm in the frequency domain [2] was implemented in Matlab (GUI). This allowed a free choice of experimental setup and array parameters. A wide range of simulations referring the dependency of array patterns of different parameters like source position, microphone weighting and so on were realized. Here, the influence of different microphone distributions (equidistant, random, geometric, logarithmic, sparse) consisting of 8 microphones shall be shown.

For the comparison of the different microphone distributions two main criteria of the array pattern were decisive, main lobe width and side lobe suppression. The main lobe width as a measure for resolution was defined as the width of area of the main lobe where the normalized sound pressure level is greater than -3 dB. The side lobe suppression defined as the difference of sound pressure level between main lobe and highest side lobe contains the information of strength of aliasing. Figure 3 and Figure 4 show these criteria versus frequency for the different microphone distributions.

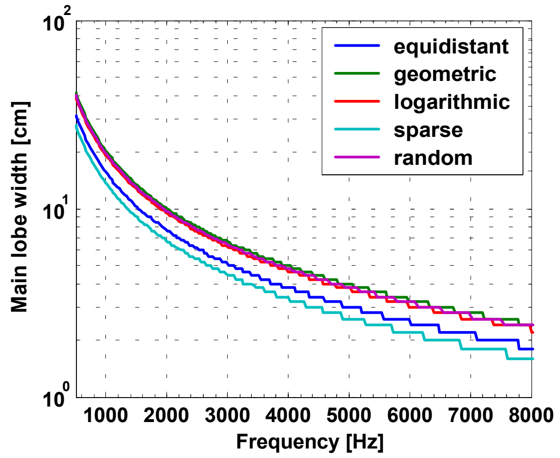

Figure 3: Frequency dependence of main lobe width for different microphone distributions.

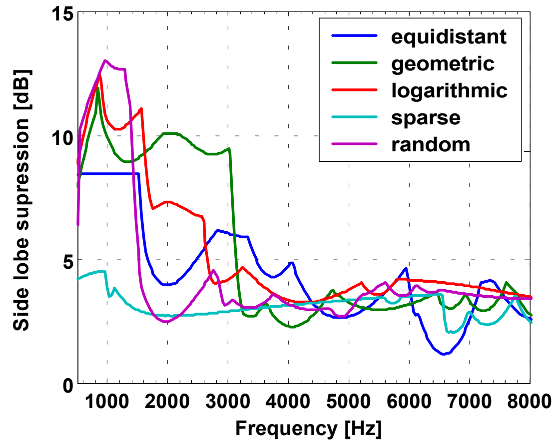

Figure 4: Frequency dependence of side lobe suppression for different microphone distributions.

It can be recognized that the sparse and the equidistant array yield the smallest main lobe width and thus the best resolution. By contrast, the logarithmic array causes the highest average side lobe suppression.

Frequency Band Averaging

Based on the fact of the changing positions of the side lobes for different frequencies and with prospect to the localization of broadband sound sources a frequency band averaging algorithm was applied. It corresponds to an energetic addition of the array patterns over a chosen frequency range. Figure 5 shows the array pattern at a single frequency of 8000 Hz in comparison to the result of an octave band averaging around a center frequency of 8000 Hz.

The results of the frequency band averaging technique showed a negligible change in main lobe width. The side lobe suppression instead increases with bandwidth, caused by the attenuation of spatial aliasing effects. By applying a third octave / octave band averaging especially for the equidistant distribution an improvement of side lobe suppression of about 3 dB / 5 dB can be achieved.

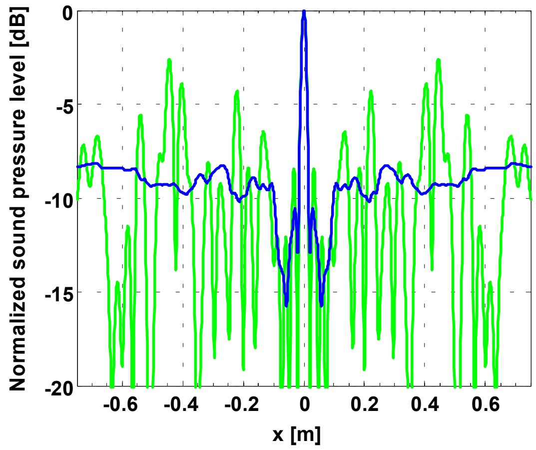

Figure 5: Array patterns of equidistant line array, green: single frequency of 8000 Hz, blue: result of octave filtering at a center frequency of 8000 Hz.

Model Sound Sources

Using this knowledge different point sound sources should be localized. The result for the localization of a dipole sound source is shown in Figure 6. The dipole consists of two microphones parallel to the microphone array and having opposite phase. The line array positioned on the left side allows a localization of the dipole parallel but also perpendicular to the array. As can be seen the resolution into the depth is only a quarter to a third of the resolution parallel to the array. Therefore the two monopole sound sources appear in the shape of ellipses instead of circles. The typical directional characteristic of a dipole in the shape of the figure 8 can be clearly recognized. Also the gap between the two monopoles where the two monopole signals are cancelled out is noticeable.

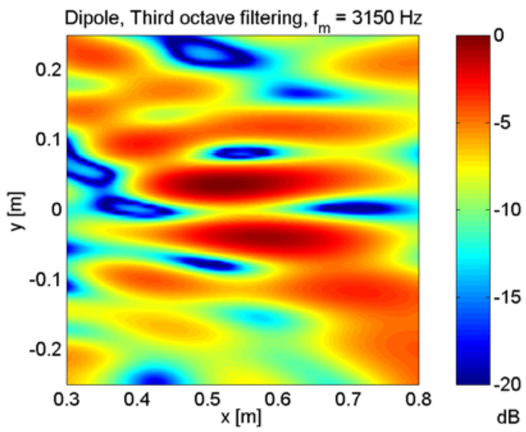

Figure 6: Dipole localization, spatial distribution of normalized SPL, third octave averaging, f_m = 3150 Hz

References

[1] Schulze C., Untersuchungen zu Linienarrays, Studienarbeit, TU Dresden, Inst. f. Akustik u. Sprachkomm., 2003

[2] Johnson D., Dudgeon D., Array Signal Processing: Concepts and Techniques, Prentice Hall, 1993