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PRACTICAL APPLICATION OF ORTHOGONAL BEAMFORMING

Ennes Sarradj[1] and Christian Schulze[2]

[1]Brandenburgische Technische Universität Cottbus, Institut für Verkehrstechnik,
Siemens-Halske-Ring 14, 03046 Cottbus, GERMANY, ennes.sarradj@tu-cottbus.de
[2]Gesellschaft für Akustikforschung Dresden mbH, Stauffenbergallee 15, 01099 Dresden, GERMANY,
christian.schulze@akustikforschung.de

ABSTRACT

A very important aspect in the process of machinery noise control is the proper identification of noise sources. One of the more recently developed options to do this is the beamforming technique using a microphone array. With this technique, it is possible to localize and rank sound sources. However, there are some limitations with beamforming that may become problems for the practical application of the technique. First of all, due to the limited dynamic range it is not possible to detect minor sources with a level well below the main source. A second problem is the limited spatial resolution that will not allow to separate sources situated closely together. These problems are addressed by an advanced signal processing method, the orthogonal beamforming technique. After a brief overview of the theory of orthogonal beamforming, the focus is on the practical application of the technique. Examples of measurements on machinery and on vehicles are presented and the results are compared to that of the classical beamforming technique.

1 Introduction

The identification of a noise sources in a practical situation requires to choose a technique that that meets some basic requirements. It should allow for a quick measurement and it should provide the necessary information at the same time. That information includes the number of sources or source mechanisms contributing and the location and strength of the sources. The location must either be given with a spatial resolution good enough to distinguish multiple sources or the number and strength of sources per location must be known. The strength of different sources has to be estimated in order to rank the sources correctly. In many cases it is necessary to measure not only the very strong, dominating sources, but also the minor ones as they may contribute sufficiently to the noise in a changed design. An additional requirement for the chosen technique is that it should give good results not only in the case when the probable locations of noise sources are known a priori.

There are many possible options for a technique to identify noise sources. A more recently developed technique is the beamforming technique using a microphone array. The beamforming technique allows for fast measurements and produces a map of sources with their contribution to the sound pressure level. Thus, the location and an indicator of source strength are available. However, there exist practical limitations in the spatial resolution and dynamic range. These limitations may disallow the use of beamforming as a practical tool in may cases.

After a brief introduction into the theory of beamforming, an advanced signal processing method, orthogonal beamforming, is presented here that addresses the limitations of spatial resolution and dynamic range and yields additional insight in the source mechanisms. Then, the use of orthogonal beamforming and the results are demonstrated on practical examples.

2 Theory of Beamforming

Assume to have number of microphones (a microphone array). A directional receiver may be constructed from these microphones following the working principle of a concave mirror. The wave front stemming from the source, arrives at the single microphones at different times and with different amplitude. The amplitude and temporal differences between the microphone signals are compensated by signal processing. This is done by applying a certain delay Δt_i to each signal y_i and by amplification of each signal by a factor of A_i. After the proper compensation, all signals have equal phase. The summation of the signals results in amplitude amplification. If sound from another source at a different location arrives at the microphones, the temporal and amplitude differences are not the same as in the first case. The summation will not result in an amplitude amplification if the same Δt_i and A_i are applied. So, the array acts as a directional sound receiver.

If different t_i and A_i are applied, the effect is the same as if a mirror is rotated. Thus the directional characteristic is altered; the focal point is 'steered' to another location.

The procedure of applying certain delays and amplitude factors is also called beamforming; the device to do so is a beamformer. In short, the output of the beamformer with M microphone signals y_i is given by:

$$z(t) = \sum_{i=1}^{M} w_i A_i y_i (t - \Delta t_i). \tag{1}$$

w_i is an optional weighting factor that may be applied to a signal. A_i and Δt_i depend on the point

$\overrightarrow{x_0}$ to which the array is focused. The output spectrum of the beamformer reads then as:

$$Z(\omega) = \sum_{i=1}^{M} w_i e_i(\overrightarrow{x_0}) Y_i(\omega), \tag{2}$$

where e_i incorporates the combined effect of A_i and Δt_i. If the beamformer output spectrum is multiplied by its conjugate $Z^*(\omega)$ and normalized by , the result is a power spectrum:

$$|Z(\omega)|^2 = Z(\omega)Z^*(\omega) = \mathbf{e}(\overrightarrow{x_0})^H \mathbf{W}\mathbf{Y}(\omega)\mathbf{Y}(\omega)^H \mathbf{W}^H \mathbf{e}(\overrightarrow{x_0}). \tag{3}$$

$\mathbf{e}(\overrightarrow{x_0})$ is the so called steering vector (that steers the directional characteristic to $\overrightarrow{x_0}$) and has all e_i as its column elements. $\mathbf{W}$ is a diagonal matrix where w_i weighting factors populate the main diagonal and $\mathbf{Y}$ is a column vector containing the spectra of all microphone signals. The most inner term in equation (3) yields the cross spectral matrix S_{xy} of the microphone signals. If the power spectrum from (3) is normalised by the squared trace of W, the result is the normalised (or true) beamformer output power spectrum:

$$S(\omega) = \frac{\mathbf{e}(\overrightarrow{x_0})^H \mathbf{W}\mathbf{S_{xy}}\mathbf{W}^H \mathbf{e}(\overrightarrow{x_0})}{(\mathrm{trace}\,\mathbf{W})^2}. \tag{4}$$

This spectrum corresponds to the sound level contribution

$$L_S(\omega, \overrightarrow{x_0}) = 10\lg\left(\frac{S_(\omega)}{p_0^2}\right) \tag{5}$$

in the array centre location. The beamformer may be applied successively for different locations $\overrightarrow{x_0}$ on a plane parallel to the array. Thus, a map of sound level contributions may be calculated and plotted for this plane. Some examples are shown in section 3. Of course may the sound level contribution mapped on other surfaces, too. This includes non-planar surfaces. Also, the possible distances for the surface ranges from very near (around 10% of the array diameter) to very far. This makes the beamformer a very versatile instrument.

In most cases there is not only one sound source present. A beamformer will be applied with the aim to separate different sound sources. Therefore, that the directional characteristic has a very narrow peak in the steering direction (resp. the location on which the array is focused) and is as insensitive as possible to sound from other directions or locations. These two properties depend mainly on the geometric layout of the array and on the w_i weighting factors. For different situations and measurement tasks, optimised geometries may be developed and used [1].

Further, some properties of the sound field in the presence of multiple sources may be applied to yield extra information that makes it possible to separate the sources.

In a practical situation, different source mechanisms of aerodynamic, thermal or mechanical nature coexist. Each of these mechanisms give rise to one sound source or, more likely, a distribution of sound sources that are not necessarily restricted to one distinct location only. The sound fields from these sources or source distributions are temporally uncorrelated if (as assumed) different source mechanisms act. Moreover, they are also spatially uncorrelated and can be assumed to be orthogonal. An array of microphones effectively samples the sum of the sound fields from different source mechanism spatially. Assuming that N different source mechanisms exist and may be described by source strengths q, the signal from i-th microphone is:

$$y_i = \sum_{j=1}^{N} f_{ij} q_j. \tag{6}$$

The f_{ij} are factors that depend on the location of the microphone and on the source location(s) of the source mechanism.

Using equation (2), the beamformer output maybe expressed as:

$$Z(\omega) = \sum_{i=1}^{M} w_i e_i(\overrightarrow{x_0}) \sum_{j=1}^{N} f_{ij} Q_j(\omega). \tag{7}$$

If the sound fields of the source mechanisms are spatially orthogonal, the vectors $\mathbf{f}_j q_j$ can be assumed to have this property too, at least for large numbers of microphones. If we define the source mechanisms to be different, if the resulting vectors $\mathbf{f}_j Q_j$ are orthogonal, then it is possible to apply an algorithm that may be called orthogonal beamforming. Based on the orthogonality, $\mathbf{f}_j Q_j Q_k^* \mathbf{f}_j^H = 0$ is true for $j \neq k$. Thus, when transforming (7) it is possible to write the cross spectral matrix as a sum of cross spectral matrices:

$$\mathbf{S_{xy}} = \sum_{j=1}^{N} \mathbf{S_{xy}}_j. \tag{8}$$

Each of these matrices correspond to a single source mechanism. The use of the matrices in a beamforming algorithm allows to map the sound level for each source mechanism j separately:

$$S(\omega)_j = \frac{\mathbf{e}(\overrightarrow{x_0})^H \mathbf{W} \left(\mathbf{S_{xy}}_j - \operatorname{diag} \mathbf{S_{xy}}_j \right) \mathbf{W}^H \mathbf{e}(\overrightarrow{x_0})}{(\operatorname{trace} \mathbf{W})^2 - \operatorname{trace} \mathbf{W}}. \tag{9}$$

This allows a very efficient separation of sources and source mechanisms. The usable dynamic range grows substantially.

3 Practical examples

3.1 Equipment

To apply orthogonal beamforming in practice, the practical limits of the measurement equipment must be taken into account. First, the number of microphones is a major cost driver, so it is limited. For the examples below, 32 microphones were arranged as a planar array in two circles with different diameters around the same center. Each circle incorporated 16 microphones. The outer circle had a diameter of 1.30 meters. A further problem is the noise from the microphone channels, that is not included in the above theory. The theory becomes much more complicated with the channel noise considered, but it can be shown that the beamformer output is still a reasonable estimation of the true situation.

3.2 Application to aerodynamic noise

It is well known that a flow around a cylinder generates noise with a tonal character. The tonal frequency coincides with the vortex shedding frequency of the von Kármán vortex street downstream the cylinder. In an experiment a cylinder of 4 mm diameter was placed in the open jet of an aeroacoustic wind tunnel. In the core of the jet the flow is laminar while in the mixing region of the jet boundaries the flow is turbulent. It can be expected that the degree of upstream turbulence affects the noise generation. In particular, the noise generated at the cylinder in the jet core will be more tonal and of more power. The noise generated in the mixing region will be

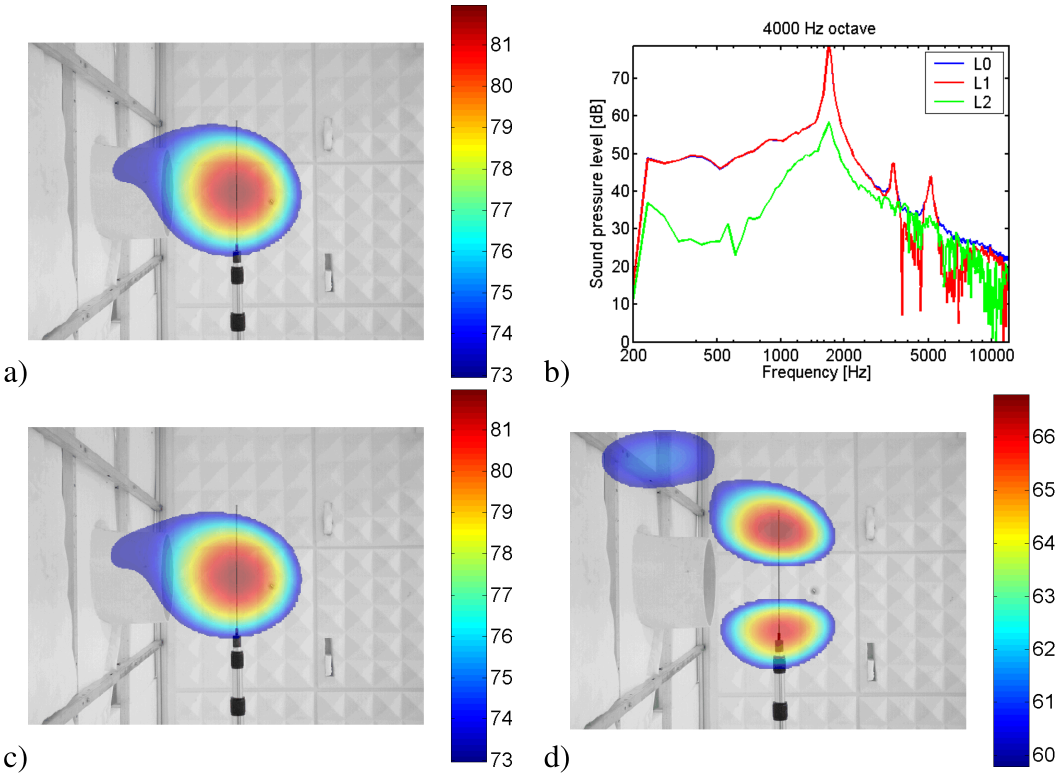

Figure 1: a) cylinder in Mach 0.11 open jet (nozzle at the left side of the picture), beamforming result map for the 2 kHz octave band, b) spectra of the overall result (L0) and the two first components (L1,L2) of the orthogonal beamforming result, c) first and d) second component of orthogonal beamforming result map for the 2 kHz octave band

more broadband. The noise from both regions can be thought of as two independent and thus uncorrelated sources.

A measurement using the classical beamforming technique shows that the cylinder is a noise source (Fig. 1a), but no further information on the sources can be given. The application of the orthogonal beamforming algorithm shows the source in the core region as the first component and the source in the mixing region as the second component, see Fig. 1b-d. Note that the second component makes only a minor contribution to the overall result. Thus, the level is so low that the source from the mixing region can not be noticed in the classical beamforming result in Fig. 1a.

3.3 Application to a cleaning vehicle

The tested cleaning vehicle has several potential sound sources: the engine and powertrain components beneath the vehicle, the blower and air exhaust in the upper part, the sucker (air intake beneath the front of the vehicle) and the rotating brushes in front of the vehicle. If the vehicle is operated with all equipment switched on, it is not obvious which components contribute the most. Thus, in order to apply efficient measures against noise the sources must be located and ranked. Orthogonal beamforming was used and led to practical results while pure classical beamforming did not.

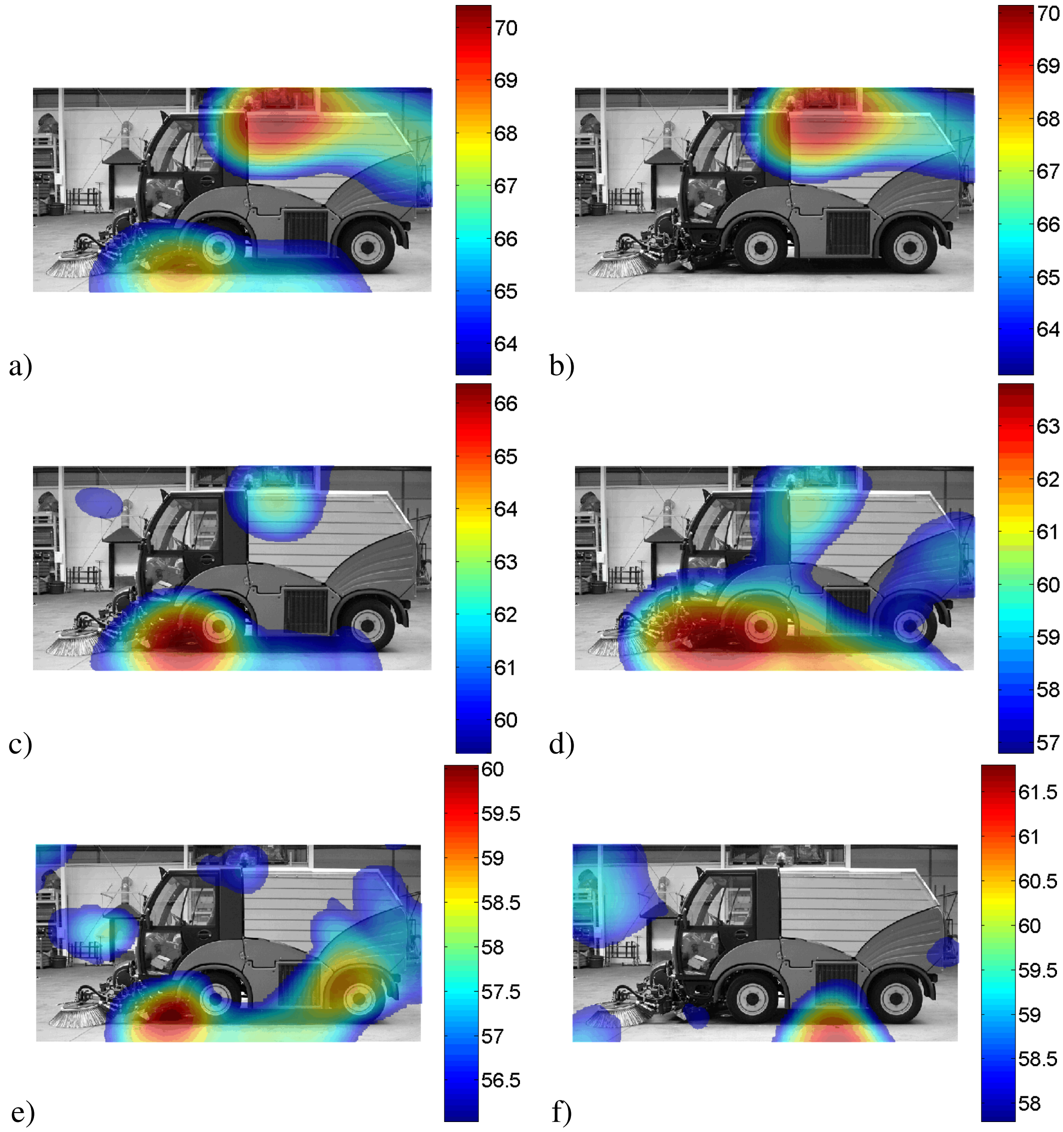

Figure 2: result maps from measurements on the cleaning vehicle (1.25 kHz third octave band): a) total and b)-f) first to fifth component corresponding to the blower, sucker(2x), brush and engine cooling

4 Conclusion

Orthogonal beamforming is able to effectively separate noise source mechanisms even when multiple sources with very different source level are present. Practical application was shown on two examples and the underlying theory of the technique were explained in short. It was demonstrated that orthogonal beamforming is not only a tool for measurements at laboratory setups but is also useful for practical application.

References

[1] C. Schulze, E. Sarradj and A.Zeibig: Characteristics of Microphone Arrays. Proc. Inter-Noise 2004, Prague