Identification of Noise Source Mechanisms using Orthogonal Beamforming

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Abstract

Conventional beamforming is able to perform source localisation with good spatial resolution. The output of a microphone array beamformer is typically a map of relative sound pressure contributions projected on a plane surface. No information is given on how these sound pressure contributions are related to several source mechanisms. Moreover, noise and so called ghost images limit the dynamic range included in the map of sound pressure contributions. As a result, only the dominating sound sources are included in the map, while minor sources are masked by noise.
In a practical situation, the sound field is composed of contributions from different sound source mechanisms. Each of these mechanisms may result in a number of distinct "sources". If the mechanisms are uncorrelated, the respective sound pressures signals are orthogonal. Based on this idea, the beamforming algorithm may be modified to result in partial maps each of which correspond to a single source mechanism. Thus, sound source mechanism may be separated and also minor sources may be detected.
The technique is explained and practical examples from wind tunnel measurements and from a vacuum cleaner are shown.

1 Introduction

The experimental identification and localisation of noise sources is a task which is important in many design stages of quieter vehicles and equipment. In the course of the development of acoustical measurement technology, an number of methods were developed to solve this task. One possible approach utilises a directional sound receiver to detect the location and strength of noise sources. In this approach, the directional characteristic of the receiver is steered successively into different directions. This procedure results in a mapping of sound levels to directions. Thus, it is in principle possible to detect sources and their location. Such directional sound receiver may be a single directional microphone, e.g. a microphone with a concave mirror. A more comfortable approach, that is also called beamforming, is to use a phased microphone array with an electronically steerable directional characteristic.
Most often, the acoustic situation is quite complex in that a larger number of noise sources are present. Different source mechanisms may act to produce the noise. In this case, not only should the sources be identified but also should the sources assigned to those source mechanisms. This is possible by applying appropriate signal processing techniques to the output from a microphone array.

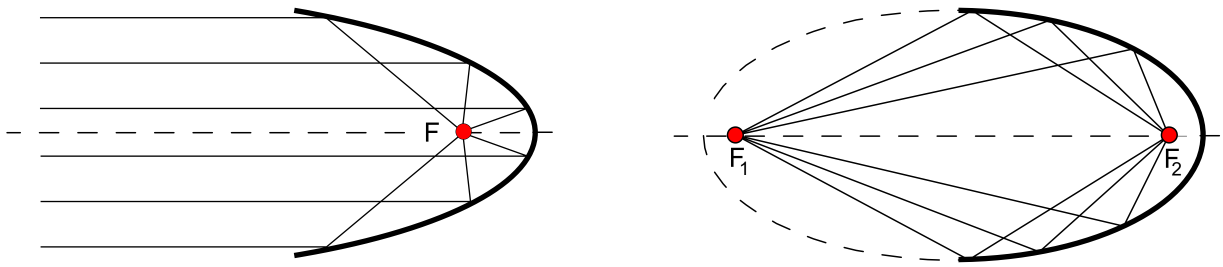

Figure 1: Concave mirror with microphone in the focal point: parabolic (left) and elliptic (right) mirror

2 Beamforming

2.1 Concave mirrors

One possible approach to construct a directional sound receiver is the use of a concave mirror together with a single microphone. The microphone is then put into the focal point of the mirror. Thus, for a parabolic mirror shape, the sound waves incident parallel to the mirror axis will be focused at the microphone (see Figure 1 left). Sound waves from other directions will not be focused. Therefore, the microphone is more sensitive in the direction of the mirror axis. One underlying assumption is that the sound source to detect with this directional receiver is far away from the mirror, so that the wave front coming from that source is approximately a plane wave front.
If that assumption does not hold, an elliptic mirror (see Figure 1 right) may be used. In that case the spherical sound waves from a source in one focal point are focused in the other point. For a practical application the mirror must be rotated and its shape must be altered if the focus is to set on another point. This turns out to be impracticable, so the electronic simulation of the mirror using a microphone array is a more favourable approach.

2.2 Microphone array

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, see Figure 2. 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 the signal by a factor of A_i. After the compensation, all signals have equal phase. The summation of the signals results in amplitude amplification. If sound from another sound source at another 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, see Figure 3. Thus the directional characteristic changes.
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

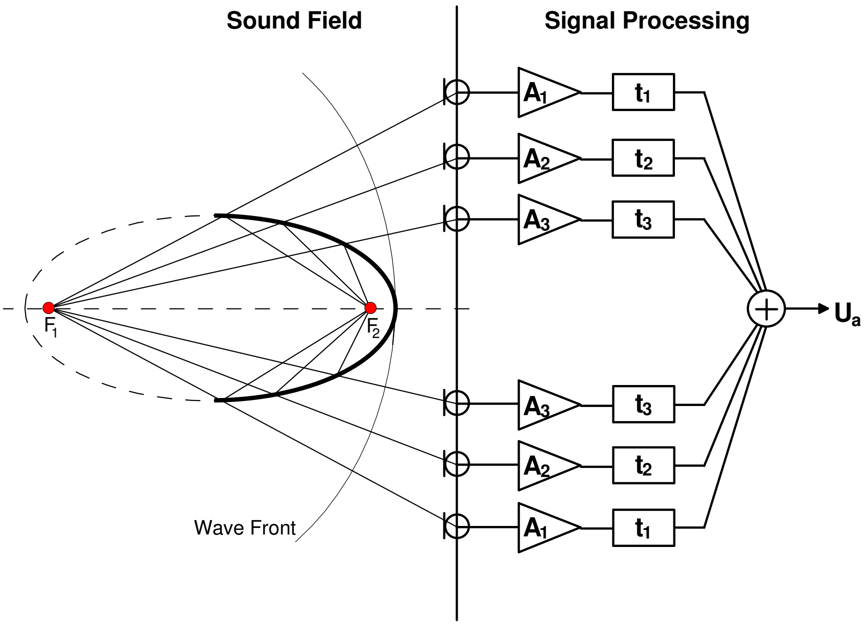

Figure 2: Analogy between elliptic concave mirror and microphone array with delay and amplitude compensation

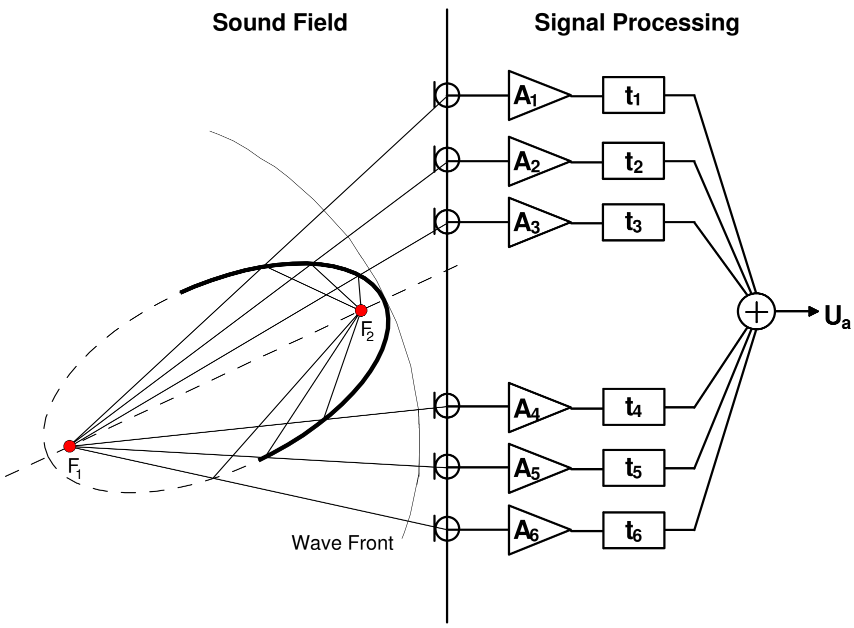

Figure 3: Rotated elliptic concave mirror and equivalent microphone array with a delay and amplitude compensation different to that in Figure 2

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 4. 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.

3 Multiple sources

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.

3.1 Noise source mechanisms and orthogonal sound fields

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.

3.2 Orthogonal beamforming

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.

4 Example results

To apply the technique explained in practice, the practical limits of 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 used simultaneously. The array microphones were arranged in

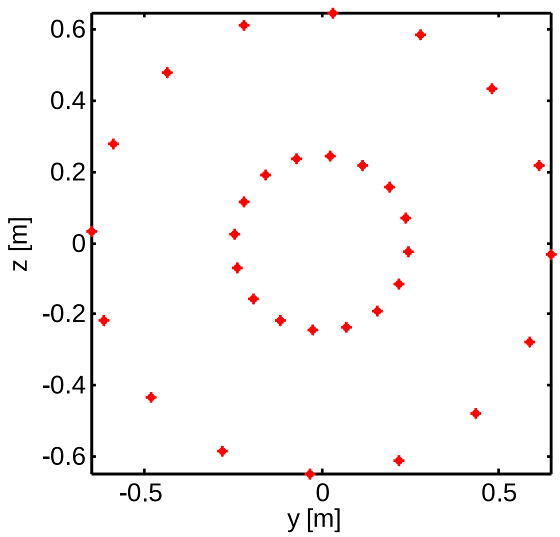

Figure 4: Layout of the array microphones

two circles with different diameters around the same centre. Each circle incorporated 16 microphones. The outer circle had a diameter of 1.30 metres (see Fig. 4).
A further problem is the noise from the microphone channels, that is not incorporated 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.

4.1 Example 1: application to trailing edge noise

The first example deals with a airplane wing profile in model scale that was measured in the aeroacoustic wind tunnel of TU Dresden. Fig. 5 shows the setup of the wing profile in the open jet of the wind tunnel. The wind speed for the reported results was Ma 0.11. At this speed the noise from the trailing edge was not much louder than the noise from the wind tunnel nozzle (Fig. 5 left). Therefore, it was difficult to separate the trailing edge noise.
The microphones were flush-mounted in a fully reflecting plate and the array was mounted 0.8 m above the wing profile. So the results in Fig. 6 are top views showing the xy-plane while the photograph in Fig. 5 is a side view.
The result from the conventional beamforming shows sound sources at the trailing edge as well as around the leading edge of the profile. If orthogonal beamforming is applied, the first component shows clearly the sound sources ar the trailing edge and the second component shows the noise from the nozzle that is only about 3 dB less. Both sources could be separated sucessfully.

4.2 Example 2: vacuum cleaner

The second example is a more practical ”real world” example: a vacuum cleaner (Figs. 7 and 8. The main sound sources at this vacuum cleaner are the three outlet vents (left and right at the cleaner body and also in the upper part it) and inlet (bottom of photographs). Different source mechanism of mechanical and aerodynamic nature act inside the vacuum cleaner and at in- and outlets. The main sound energy is centred at 1 kHz octave band. In the results from this band (Fig. 7) the overall result from conventional beamforming shows one single source at the lower body of the cleaner that can be interpreted as the turbine noise. The first component from orthogonal beamforming shows nearly the same result, but the second and third components show clearly different mechanisms: the inlet and

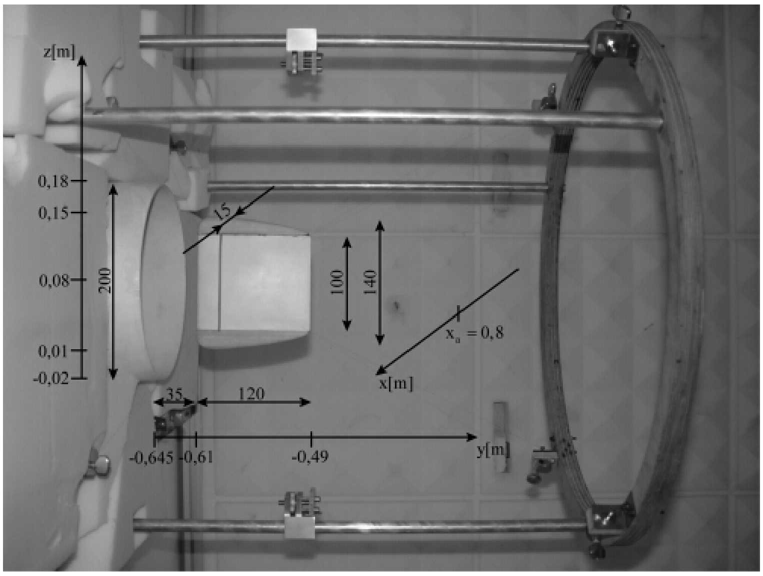

Figure 5: Setup of the model scale wing profile in the wind tunnel (side view, yz-plane)

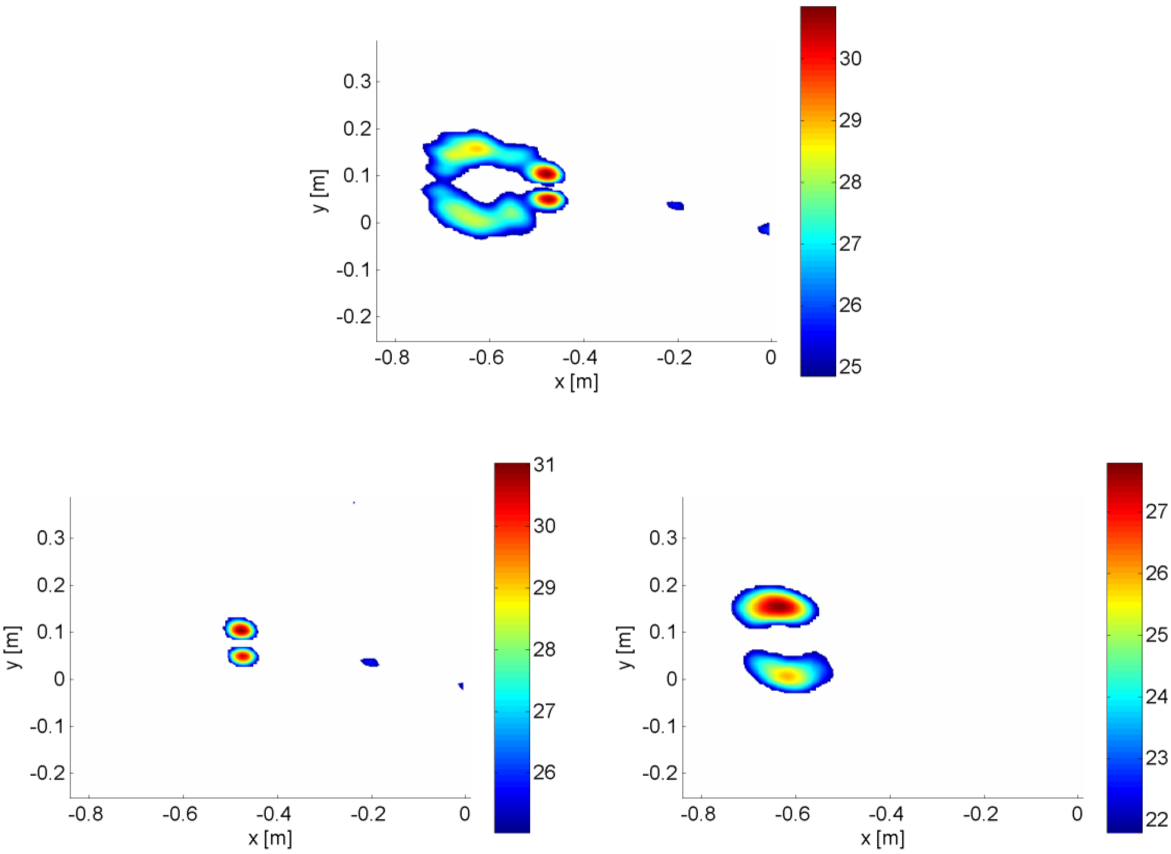

Figure 6: Results for the wing profile (top view, xy-plane)

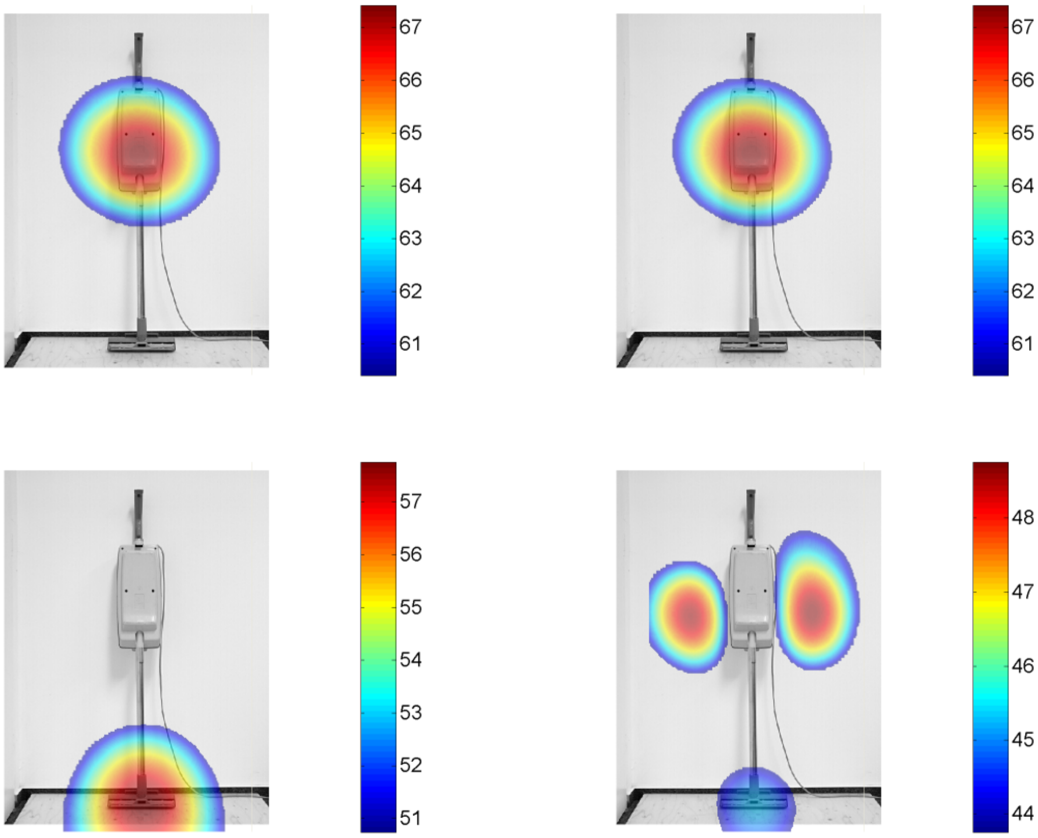

Figure 7: Results

outlet noise, 10 dB respective 20 dB lower in level than the first component. At 8 kHz, the spatial resolution of the array is much better. The sound insulation of the case of the cleaner is much higher at this frequency. Thus, sound sources are detected only at the inlet and the outlets. At the outlets, the noise is a combined product of the source mechanisms acting in the turbine and in the outlet flow. These mechanisms could be separated in the first four components (components 1 and 2 shown in Fig. 8). The noise from the inlet is shown in the 5th component although its peak level is about 18 dB below that of the first and dominating component.

5 Conclusion

The approach of the orthogonal beamforming is able to effectively separate noise source mechanisms even when multiple sources with very different source level are present. The theory of orthogonal beamforming was explained in short and the practical application of the technique was shown in two examples.

References

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

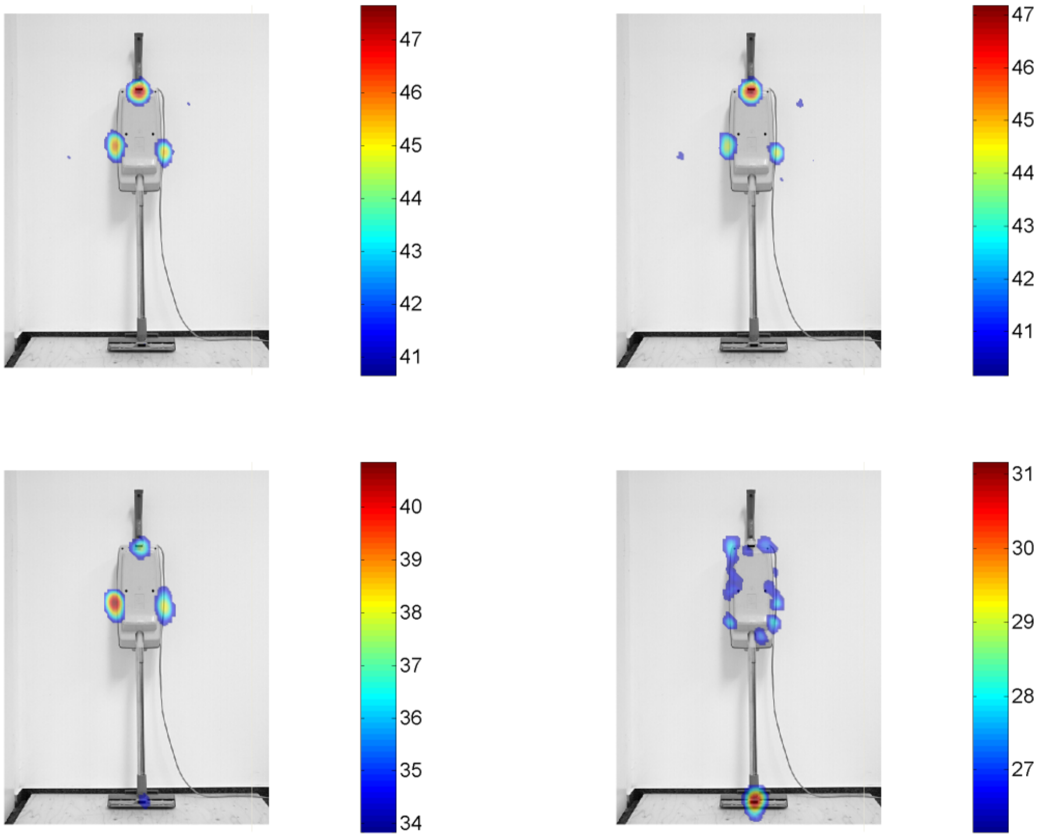

Figure 8: Results

[2] M. Brandstein and D. Ward: Microphone Arrays: Signal Processing Techniques and Applications. Springer, 2001.