Localization of Rotating Sound Sources Using Time Domain Beamforming Code

Christian Maier, Wolfram Pannert and Winfried Waidmann

Abstract The motion of an acoustic source produces a Doppler shift of the source frequency which is dependent on the source's motion relative to the receiver. Some applications in acoustics involve rotating sound sources around a fixed axis in space. For example, the noise emitted by fans is of interest and because of the fast rotation, the sound sources are not easy to locate with the standard delay-and-sum beamforming code. In the time domain approach for stationary sound sources, the delay-and-sum beamforming works with shifting the microphone signals due to their different delays caused by the different distances between the source and the microphones and summing them up. This approach is adapted to a moving source, resulting in time dependent delays. The delays are calculated via an advanced time approach where the time at the receiver is calculated from the emission time τ plus a time dependent delay due to the time dependent distance $r(\tau)$. In contrast to the standard beamforming code, this time domain beamforming code allows to treat rotating sound sources as well as stationary sound sources. In this chapter the differences between the standard delay-and-sum beamforming to the rotating time domain beamforming is shown and examples are presented.

Keywords Fan · Rotating sound sources · Retarded time · Beamforming

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1 Introduction

The sound emitted by moving sound sources such as a flowed airfoil or rotating fan blades are problems of technical interest. Visualising and analysing moving sound sources is much harder in comparison to stationary sound sources. The Doppler-shift and the retarded time due to the movement of the sound source have to be taken into account. In this work a time domain algorithm is presented which can be applied to rotating sound sources which are produced for example by fan blades. The theory is shown and the algorithm is proved with measurements using an acoustic camera at a rotating fan.

The standard Delay-and-Sum beamforming method can be applied in time and frequency domain [1]. However, the method is not suitable for moving sound sources. To compensate the movement of the sound source, special corrections are necessary. For this case the rotation of fans have to be compensated. The pressure field of a moving monopole is derived for a uniform flow in this approach. The approach presented below is original based on [2] and is based on the Delay-and-Sum, beamforming method in the time domain. A method to compensate rotating sound sources in the frequency domain, especially for high resolution beamforming techniques [3], is presented by Pannert [4].

2 Theory

The movement of a point source can be treated with the Greens function approach for solving the inhomogeneous wave equation.

Taking the inhomogeneous wave equation for a stationary source located at $\vec{x}$

$$\frac{1}{c^2}\frac{\partial^2 p'}{\partial t^2} - \Delta p' = q(\vec{x}, t) \tag{1}$$

where $q(\vec{x}, t)$ is a the source distribution, the solution for free space conditions without boundaries for p' can be calculated from an integral formulation

$$p'(\vec{x}, t) = \frac{1}{4\pi} \int_{\mathbb{R}^3} \frac{q(\vec{y}, t - |\vec{x} - \vec{y}|/c)}{|\vec{x} - \vec{y}|} d^3\vec{y} \tag{2}$$

with

$$r = |\vec{x} - \vec{y}| \tag{3}$$

and the retarded time τ

$$\tau = t - |\vec{x} - \vec{y}|/c \qquad (4)$$

The signal which was emitted at time τ at a position $\vec{y}$ and is observed at time t at the point $\vec{x}$. For the general source distribution a concrete source distribution can be inserted. The simplest model for a moving monopole is the distribution

$$q(\vec{x}, t) = Q(t)\delta(\vec{x} - \vec{x}_s(t)) \qquad (5)$$

With $\vec{x}_s(t)$ as the actual time dependent position and $Q(t)$ as the amplitude of the monopole sound source.

Figure 1 shows the situation for a moving sound source and a fixed observer position. The observer point is the microphone position (at the microphone array).

It is necessary to calculate the distance between sound source and the microphone position for every time step τ_n to calculate the time delay to the observer position [5, 6].

In the retarded time approach, the retarded emission time τ is calculated back from the receiving time t via

$$\tau = t - r(\tau)/c \qquad (6)$$

and cannot be calculated analytically in general cases due to the complicated dependence of $r(\tau)$ from τ. It can numerically found as a root of Eq. (7) [6]. Algorithms that treat that problem can be found in [7] or [8].

In the advanced time approach which is applied in this work, the receiver time t can be calculated via

$$t = \tau + r(\tau)/c \qquad (7)$$

Fig. 1 Movement of the source term to a fix observation point
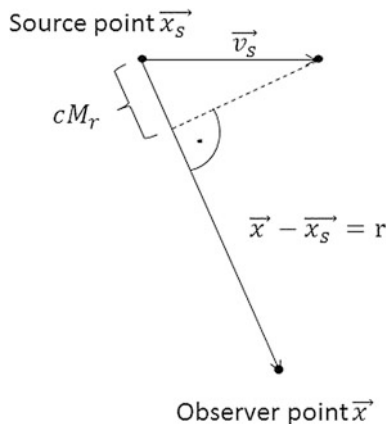
This is much easier, but results in unequally spaced time samples t_n when using equally spaced time samples τ_n.

In Fig. 2 the situation is shown for a moving source. An emitted signal at the time τ arrives at the observer position $\vec{x}$ at the time t. The speed of sound is c. In the case of a stationary source, the retarded time only depends on the position of the source $\vec{x}_s$. In the case of a moving source it depends on $\vec{x}_s(\tau)$. This time delay is calculated for every time step n

$$\Delta t_n(\tau_n) = t_n - \tau_n = \frac{|\vec{x} - \vec{x}_S(\tau_n)|}{c}. \tag{8}$$

The condition $\tau_n < t_n$ is always fulfilled for the case that the source term moves with subsonic speed and $\Delta t_n(\tau_n)$ is always positive. Working with these time delays the motion of the source can be compensated in the received microphone signals and the moving source is imaged at a fixed position which corresponds to the position at time $\tau = 0$.

In Fig. 3, the typical set up for investigating a fan with an acoustic camera is shown. It is necessary to compensate the movement of the sound source. To compensate this movement, in this case the rotation of the fan with its blades, it is necessary to shift the time signals of every microphone for every time step at an amount, which is due to the change in distance between the moving source and the selected microphone. These shifted signals are used then to calculate the beam pattern.
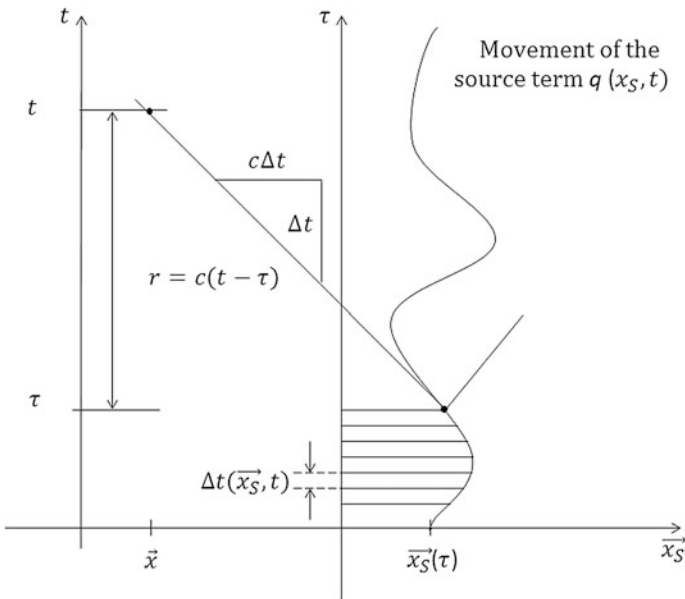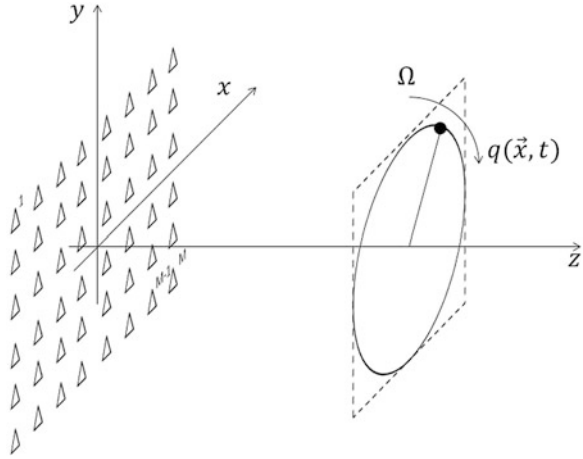

Fig. 2 Retarded time emitted from a moving sound source in the space–time

Fig. 3 Microphone array
with moving sound source in
front of it

In Fig. 4 simulated signals for a rotating source are shown. The pressure signal
shows clearly the varying frequency due to the Doppler shift (Fig. 4a). The radial
motion of the source is subsonic Fig. 4b shows the spectrum of the microphone
signal in (a). The radius at which the sound source rotates is 0.65 m and the
frequency of rotation is 100 Hz.

The frequency spectrum (Fig. 4b) shows the peak no longer at the position of
1500 Hz. This is the effect due to the Doppler shift; the peak is now shifted in
positive and negative frequency away from the emitted 1500 Hz. Moreover, this
frequency spectrum is not symmetric, because the motion between the source and
the receiver also has an influence to the amplitude of the signal.

3 Measurements

Software tests with a rotating sound source were carried out to proof the pro-
grammed algorithm. This software code treats the rotating beamforming theory
explained in the last section. For this tests an artificial rotating sound source was
simulated in software, that rotates with 600 rpm on a circle with a radius of 0.3 m
around the position $x = 0.2$ m and $y = 0.0$ m out of the middle. The distance between
the sound source and the virtual microphone array is $D = 1$ m. In Fig. 5a the rotating
monopole can be seen at the middle frequency 2500 Hz at the position x = 0.4 m
and y = 0.0 m for a simulation time of 0 s. Figure 5b shows the same monopole
sound source after 0.025 s at the position x = 0.2 m and y = 0.2 m. The monopole
sound source rotates over the time and can be captured to every desired point of
time.

Fig. 4 Received signal from
a rotating source. Distance
microphone—plane of
rotation in distance $D = 1$ m;
rotation speed $n = 6000$/min;
frequency of the source
$f = 1500$ Hz. **a** shows the
microphone signal and **b** the
spectrum of the pressure
signal
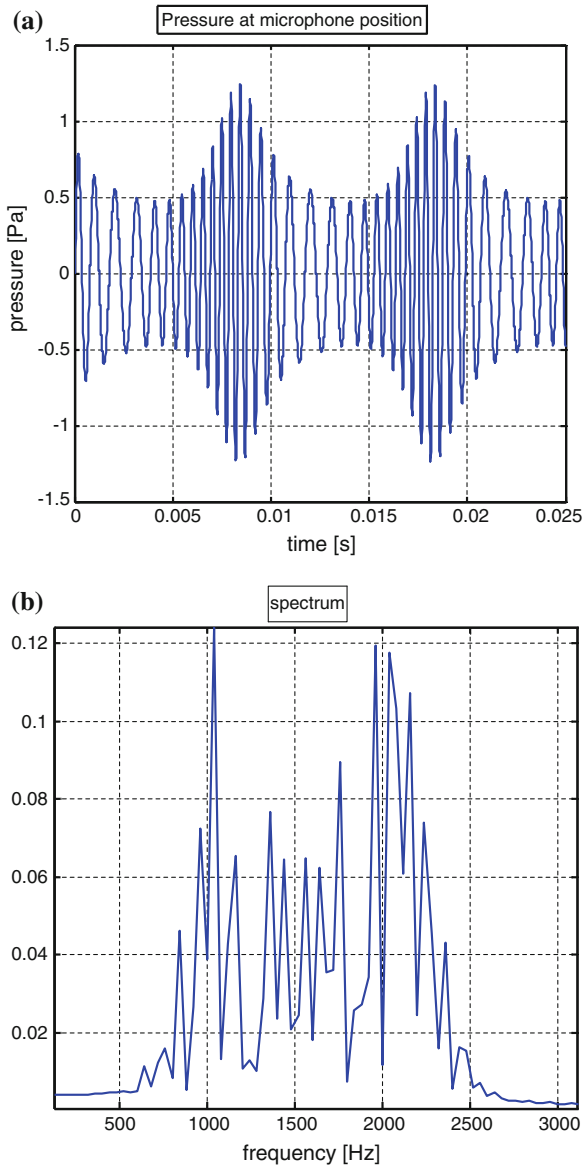
4 Results

The algorithm used for the validation is implemented in the acoustic camera. So the
analysis can be done and it is possible to compare it to the standard delay-and-sum
beamforming. The Delay-and Sum beamforming only locates stationary sound
sources and rotating beamforming locates moving sound sources. It also shows a

Fig. 5 Virtual software sound source rotating anticlockwise around $x = 0.2$ m and $y = 0.0$ m, **a** at time 0 s and **b** at time 0.025 s
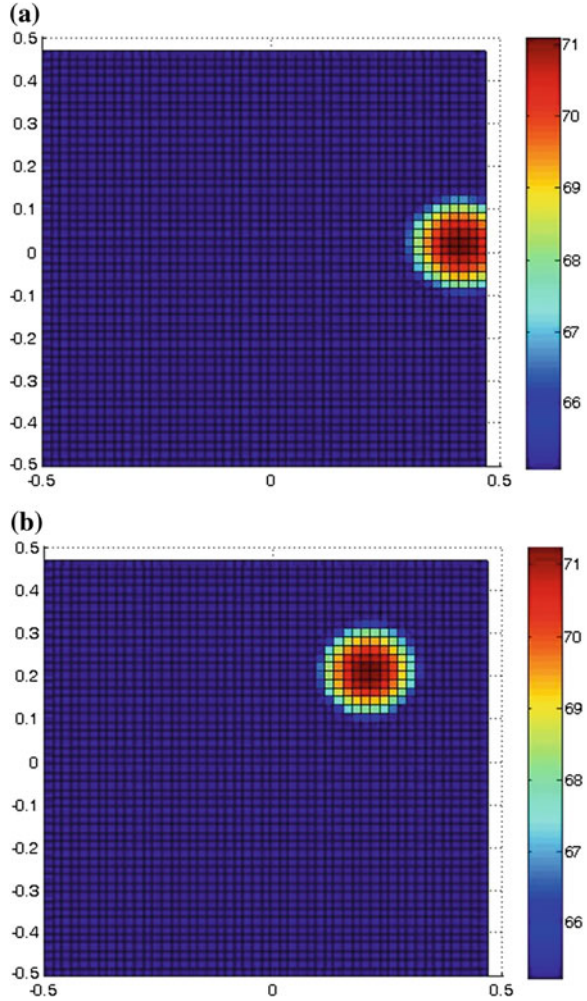
ring shaped distribution of sound sources in the gap between the blades and the wall whereas the rotating beamforming shows the spot shaped sound sources on the blades (Fig. 6).

Figure 7 shows similar beamforming results like Fig. 6, but with the rotation compensation. Opposite to the beamforming version without rotation compensation stationary sound sources should be averaged out whereas the rotating sources are visible and, in this case, they belong to the sound emitted by the blades itself. Beside this the acoustical photo is superimposed with a frozen-image to match the sound sources to the blades of the fan.
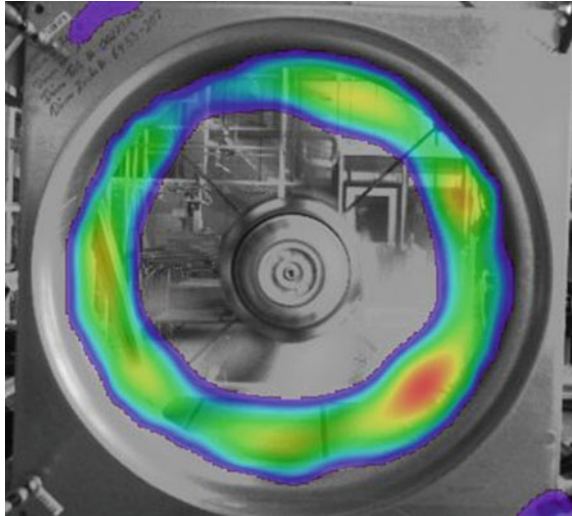
Fig. 6 Analysis of a fan with no rotation compensation. Only the stationary sound sources from the gap between blades and wall are visible
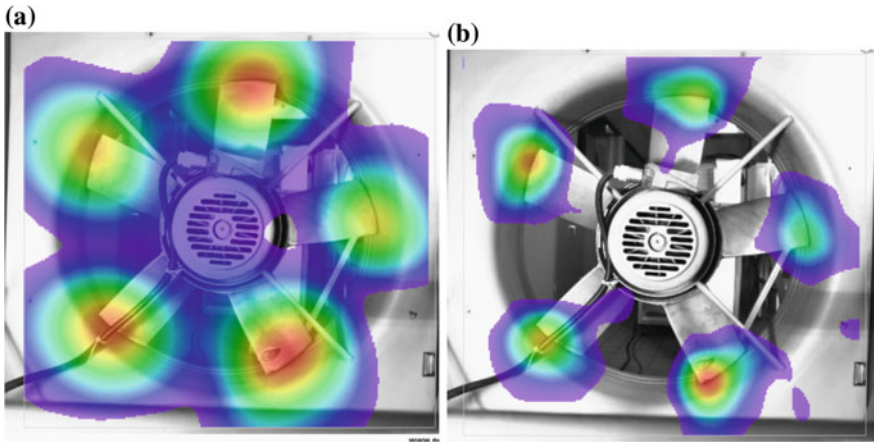

Fig. 7 Analysis with motion compensation. After an alignment of the static optical picture the moving sources at the blade tips are visible. **a** shows the Beamforming plot for 2 kHz and **b** the beamforming plot for 4 kHz

5 Conclusion

The rotating beamforming algorithm is possible to image the stationary sound sources as well as rotating sound sources. In combination with an acoustic camera it is a helpful tool for optimising fan geometries to reduce sound emission.

Moreover, with this algorithm it is possible to locate sound sources at their position on the blades. So it is possible to distinguish between the leading edge and the trailing edge of the blade and study the frequency dependence of the generated noise.

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