# COMPARISON OF MICROPHONE ARRAY METHODS FOR THE CHARACTERIZATION OF ROTATING BROADBAND NOISE SOURCES

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## SUMMARY

For the motion compensation of rotating sound sources, different microphone array methods are presented and compared to each other. At first, a simulated benchmark case with discrete rotating broadband sound sources with non-constant rotational speed is considered. Based on the estimation of the source position and source amplitude, recommendations of the number of microphones to be used for actual measurements on rotating systems are given. In a second case, a benchmark fan is analyzed to compare the source distribution and source amplitudes of the considered methods. Advantages, disadvantages and limits of the respective methods are shown.

## INTRODUCTION

The localization of rotating sound sources with a microphone array is a major task to reduce broadband axial fan noise. Depending on the adjusted operating point and considered frequency band, the sound sources are located at different components of the fan blades [1]. Considerable effort has been put into compensating the relative motion of the sound sources to the microphone array and to understand the rotating sound source mechanism. The motion compensation can be done in time domain or frequency domain. Sijtsma et al. [2] and Minck et al. [3] presented different methods in time domain, taking into account the time delays and the Doppler effect due to the rotation. In their approach, the emitted signals from the moving sources are reconstructed by continuously moving the focus of the microphone array along the rotational motion of the fan. Other methods presented by Dougherty et al. [4], Herold and Sarradj [5], Lowis and Joseph [6] and Pannert and Maier [7] work in the frequency domain. For this, the pressure signals are Fourier transformed prior to the application of beamforming algorithms. Motion compensation in the frequency domain can be done in different ways, e.g. by employing virtual rotating microphones or by a modal decomposition of the rotating sound field. The basic idea of the considered frequency domain methods is the transformation of the pressure signals at the microphones into a rotating reference frame and the correct calculation of the rotating sound field in the medium. To calculate the sound radiation of a rotating sound source, it should be kept in mind that in the rotating reference frame, the medium between the microphone array and the sound sources is rotating itself. Ignoring the rotating medium will lead to inaccurate sound travel times between the assumed source positions and the microphones. For the considered methods, the microphones are equidistantly arranged on a ring, coaxial to the rotational axis of the sound sources. After motion compensation, beamforming is used to localize the sound sources. High-resolution frequency domain beamforming methods are based on the evaluation of the cross spectral matrix (CSM), which is approximated by averaging cross spectra of the time signals. The averaging process cancels out non-continuous sources. By removing the main diagonal of the CSM, uncorrelated background noise can be reduced. Then, the CSM is multiplied with the steering vectors to shift the phase of the microphone signals according to the focus points.

The first method for motion compensation, that is used by Dougherty et al. [4], Herold and Sarradj [5], and Zenger et al. [1], defines a virtual rotating array. The pressure data at the virtual microphones are approximated through linear interpolation by the measured signals at the real microphones according to the momentary angular position of the measured object. The cross spectra are calculated from the measured pressure signals directly and can be averaged with Welch’s method. In contrast to this method, the method used by Lowis and Joseph [6], Pannert and Maier [7], and Ocker and Pannert [8], utilizes a modified free-space Green’s function in the rotating reference frame to describe the rotating sound field analytically. The modified Green’s function is derived by a modal decomposition of the rotating sound field and presented as harmonic series expansion. According to the rotational speed of the sources, the acoustic modes are shifted in the frequency domain. In a second step, a virtual rotating microphone array is used to calculate the pressure signals at the microphones in the rotating reference frame through mode shifting. With this method, no interpolation of the pressure signals is needed. Afterwards, the CSM in a rotating reference frame is inversely determined from the calculated pressure signals in the rotating reference frame. The averaging process of the reconstructed CSM is done in the frequency domain with Daniell’s method [9]. Objects with non-constant rotational speed can be investigated by analyzing short time sequences and averaging over these sequences.

The aim of this work is the comparison of the developed frequency-domain microphone array techniques for rotating sound sources, especially for the application on axial fans. With these advanced techniques for the acoustic characterization and sound source localization, optimization strategies can be developed to reduce the aeroacoustic sound emission of axial fans.

The outline of this paper is organized as follows. The theory section features a short summary of the methods of motion compensation including the calculation of the CSM and Clean-SC deconvolution. In the setup section, a description of the two considered benchmark cases is given. Then, the obtained results for the different methods of motion compensation are compared to each other and discussed. A conclusion is drawn in the last section.

## THEORY

The data processing for the localization of rotating sound sources with a microphone array follows the descriptions by Herold and Sarradj [5] for the virtual rotating array method, Pannert and Maier [7] for the modal decomposition method, Ocker and Pannert [9] for the calculation of the CSM with Welch’s and Daniell’s method, and Sijtsma [10] for Clean-SC deconvolution.

For motion compensation of rotating sound sources, the microphones have to be arranged equidistantly on a circular array and aligned to the axis of rotation. The methods can be extended for the use of several rings of microphones. Using the virtual rotating array method, the measured time data from the stationary system are transformed into a rotating reference frame. To compensate a varying rotational speed, the source position is tracked revolution by revolution and recorded synchronously to the microphone data. The pressure signals at the virtual rotating microphones are then calculated by linear interpolation from the measured sound pressure based on the angular position of the virtual microphones. After the interpolation, the time data are transformed into frequency domain using Welch's method [11]. For each microphone the time signal is divided into overlapping time blocks, onto which a fast Fourier transformation (FFT) is applied. The CSM $\mathbb{C}$ is the result of the cross correlated pressure signals $\mathbf{p}$ of all possible pairs of microphones

$$\mathbb{C} = \overline{\mathbf{p}\mathbf{p}^{\mathrm{H}}}, \tag{1}$$

where the superscript H indicates the Hermitian transpose.

As an alternative to the virtual rotating array method, the modal decomposition method can be used for motion compensation. Using this method, the first step is to transform the pressure signals at the stationary microphones from the time domain to the frequency domain via an FFT. The sound field of a rotating point source in the rotating reference frame is then defined as

$$\mathbf{P}_\Omega = \mathbb{G}_\Omega \mathbf{Q}. \tag{2}$$

$\mathbf{P}_\Omega$ are the pressure signals in the frequency domain in the rotating reference frame for all microphone positions. $\mathbb{G}_\Omega$ is the modal decomposed Green's function, which describes the sound radiation of a rotating sound source in the rotating reference frame. Note that the acoustic modes in the series expansion of the modal decomposed Green's function are shifted according to the rotational movement. $\mathbf{Q}$ are the spectra of the sound sources. To compensate the source motion, it is necessary to shift the coefficients of the azimuthal modes $\mathrm{P_m}$ of the pressure signals as

$$\mathrm{P}_\Omega(\mathbf{x}, \omega) = \sum_{\mathrm{m}=-\frac{\mathrm{L}}{2}+1}^{+\frac{\mathrm{L}}{2}} \mathrm{P_m}(\mathrm{r_x}, \Theta_\mathrm{x}, \omega + \mathrm{m}\Omega) \mathrm{e}^{\mathrm{i m}\varphi_{\mathrm{xl}}}, \tag{3}$$

where m is the mode number, $\mathbf{x}$ the microphone positions, $\Omega$ the rotational frequency and $\varphi_{\mathrm{xl}}$ the polar coordinates of the L microphones. The number of microphones of the array limits the resolution of propagating azimuthal modes m. According to Poletti [12], the negative limit of azimuthal modes to be considered can be approximated as

$$\mathrm{m}^- = \frac{-\mathrm{k_0 r_s}}{1 + \mathrm{k_R r_s}} \tag{4}$$

and the positive limit can be approximated as

$$\mathrm{m}^+ = \frac{\mathrm{k_0 r_s}}{1 - \mathrm{k_R r_s}}. \tag{5}$$

Here, $\mathrm{r_s}$ is the maximum sound source radius, $\mathrm{k_0} = \frac{\omega_\mathrm{k}}{\mathrm{c}}$ the wave number of the emitted frequency of the source and $\mathrm{k_R} = \frac{\Omega}{\mathrm{c}}$ the wave number due to the rotation. For the definition of the number of microphones to be used for actual measurements, the number of propagating azimuthal modes has to be determined and resolved. Positive azimuthal modes correspond to co-rotating modes and negative azimuthal modes to counter-rotating modes.

After motion compensation, the CSM can be calculated from the pressure signals in the rotating reference frame $\mathbf{P}_\Omega$. In contrast to the virtual rotating array method, averaging the CSM with Welch's method is more difficult, because of the mode shifting. Hence the CSM is calculated in the frequency domain with Daniell's method over neighboring frequency lines symmetrical about the focused frequency [13].

With the CSM, beamforming algorithms and the Clean-SC deconvolution can be applied. Most commonly, the amplitude of the considered focus point $\mathrm{B_s}$ is calculated in the frequency domain by the delay-and-sum beamformer as follows

$$\mathrm{B_s} = \mathbf{h}_\mathrm{s}^\mathrm{H} \mathbb{C} \mathbf{h}_\mathrm{s}, \tag{6}$$

where $\mathbf{h}_\mathrm{s}$ is the steering vector for a specific focus point s. Based on the delay-and-sum beamformer output $\mathrm{B_s}$, the Clean-SC deconvolution searches for the highest peak and copies it into an empty (clean) map. In a second step, all coherent portions according to the highest peak are identified and subtracted from the original (dirty) map. Iteratively the highest peaks from the dirty map are identified and copied to the clean map until a convergence criterion is reached [10].

## SETUP AND RESULTS

For the comparison of different data processing approaches for localizing rotating sound sources with a microphone array, two benchmark cases are investigated. First, the simulated benchmark case from Herold [14] is considered. Three point sources with different source levels are rotating clockwise with the same, but a slightly varying rotational speed emitting uncorrelated white noise. The challenge is to determine the correct position and the sound pressure level of the three sources at the trigger instant. To focus the sound sources correctly, the non-constant rotational speed has to be considered. In the second benchmark case from Zenger et al. [15], a generic fan with unskewed fan blades is investigated. Here, the pressure signals from the fan N1UG were measured with a microphone array for a specific operating point. The challenge is to localize the unknown source distribution and to determine the sound pressure level at the fan blades for different frequency bands.

### Three rotating point sources

Time data are generated at 64 microphone positions. The microphones are equidistantly distributed on a ring with a diameter of 1 m, which is placed axisymmetric to the axis of rotation of the sound sources. The sampling frequency is $\mathrm{f_{sample}} = 48\ \mathrm{kHz}$, the measurement time $\mathrm{t_{meas}} = 10\ \mathrm{s}$ and the distance from the sources to the microphone array $\mathrm{z} = 0.5\ \mathrm{m}$. On the left-hand side of Figure 1, the triggered position of the sources and the microphone array are presented. The right-hand side shows the non-constant rotational velocity. The source levels are $\mathrm{L_{p,1}} = \mathrm{L_{p,2}} + 3\ \mathrm{dB} = \mathrm{L_{p,3}} + 6\ \mathrm{dB}$.

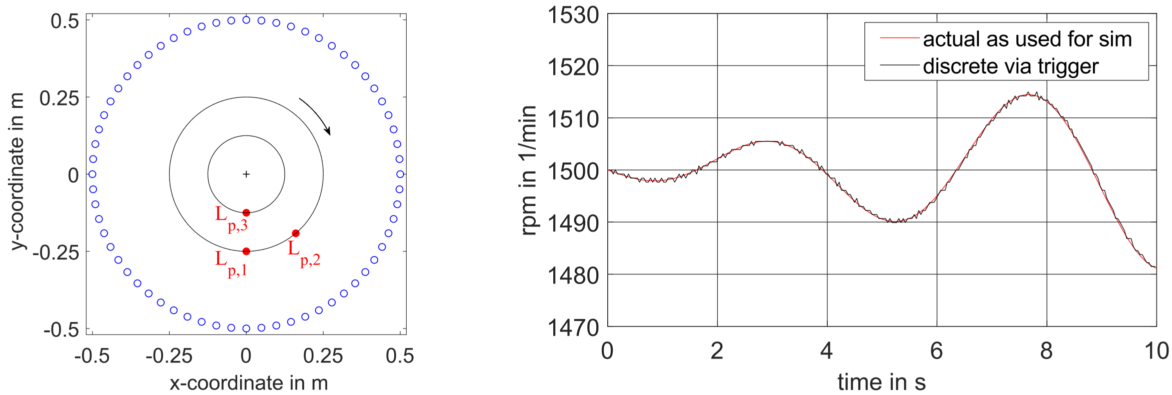



*Figure 1: Left: Triggered position of the three sources (red markers), positions of the 64 microphones (blue markers). Right: Non-constant rotational velocity.*

The focus plane is discretized with an equally-spaced cartesian grid consisting of 3600 points with 0.01 m spacing. The analysis is performed with Clean-SC deconvolution with a loop factor of 0.9.

The sound maps are evaluated for the third octave band with a center frequency 2000 Hz. The maps are plotted with a dynamic range of 20 dB.

Assuming an averaged constant rotational velocity $n = 1500$ rpm over the whole measurement time leads to a smeared beamforming source map. The amount of sound sources, the positions and sound pressure levels cannot be determined as it is shown in Figure 2.

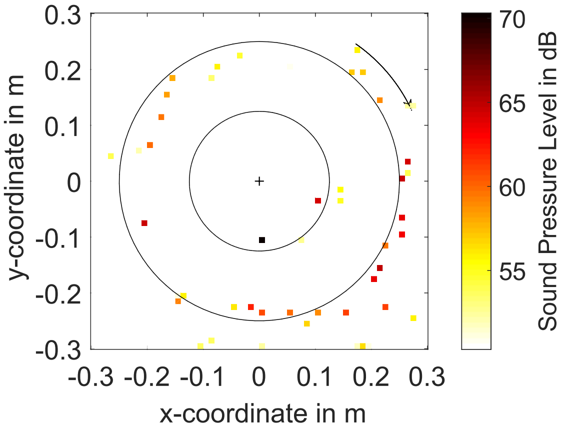



*Figure 2: Smeared sound source distribution for n = constant for the third octave band with center frequency 2000 Hz*

For better-focused sound sources, the varying rotational speed has to be taken into account. To produce the beamforming map on the left-hand side of Figure 3, the virtual rotating array method with averaging the CSM via Welch's method is used. The beamforming map on the right-hand side is generated with the modal decomposition method for motion compensation and Daniell's method for averaging the CSM.

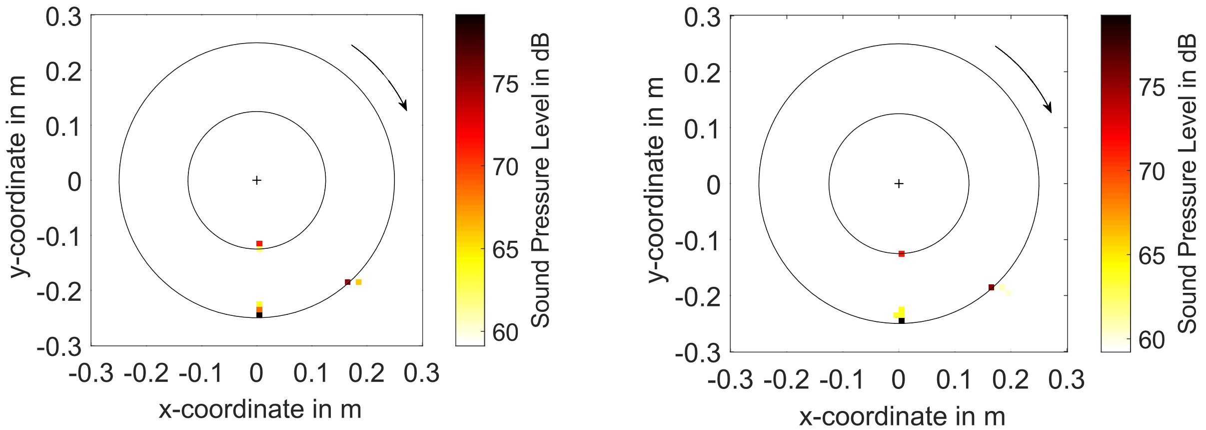



*Figure 3: Correct focusing of the sound sources evaluated with the virtual rotating array method (left) and the modal decomposition method (right) for the third octave band with center frequency 2000 Hz*

For this benchmark case, the results look quite similar, although the method of motion compensation and averaging the CSM are different.

In a second step, the number of microphones, necessary for a successful source reconstruction, is analyzed. Especially for measurements with a limited number of microphones, it is necessary to know the resolution limit for both methods of motion compensation. Using the virtual rotating array method with too few microphones leads to inaccurate interpolated values and limits the applicability of this method for higher frequencies. For the modal decomposition method, the number of microphones determines the resolution of the propagating azimuthal modes. According to Eq. 4 and Eq. 5, the limits of propagating azimuthal modes are $m^{-} = -9$ and $m^{+} = 12$. Not resolving these spinning modes can result to an incorrect source distribution. Figure 4 shows the beamforming map for this benchmark case for the third octave band with center frequency 2000 Hz. The number of microphones are 64, 32, 16 and 8. All microphones are evenly distributed on a ring with a diameter of 1 m. The position of the microphones is shown on the left, the beamforming map as calculated with the virtual rotating array method in the middle, and the beamforming map calculated with the modal decomposition method on the right.

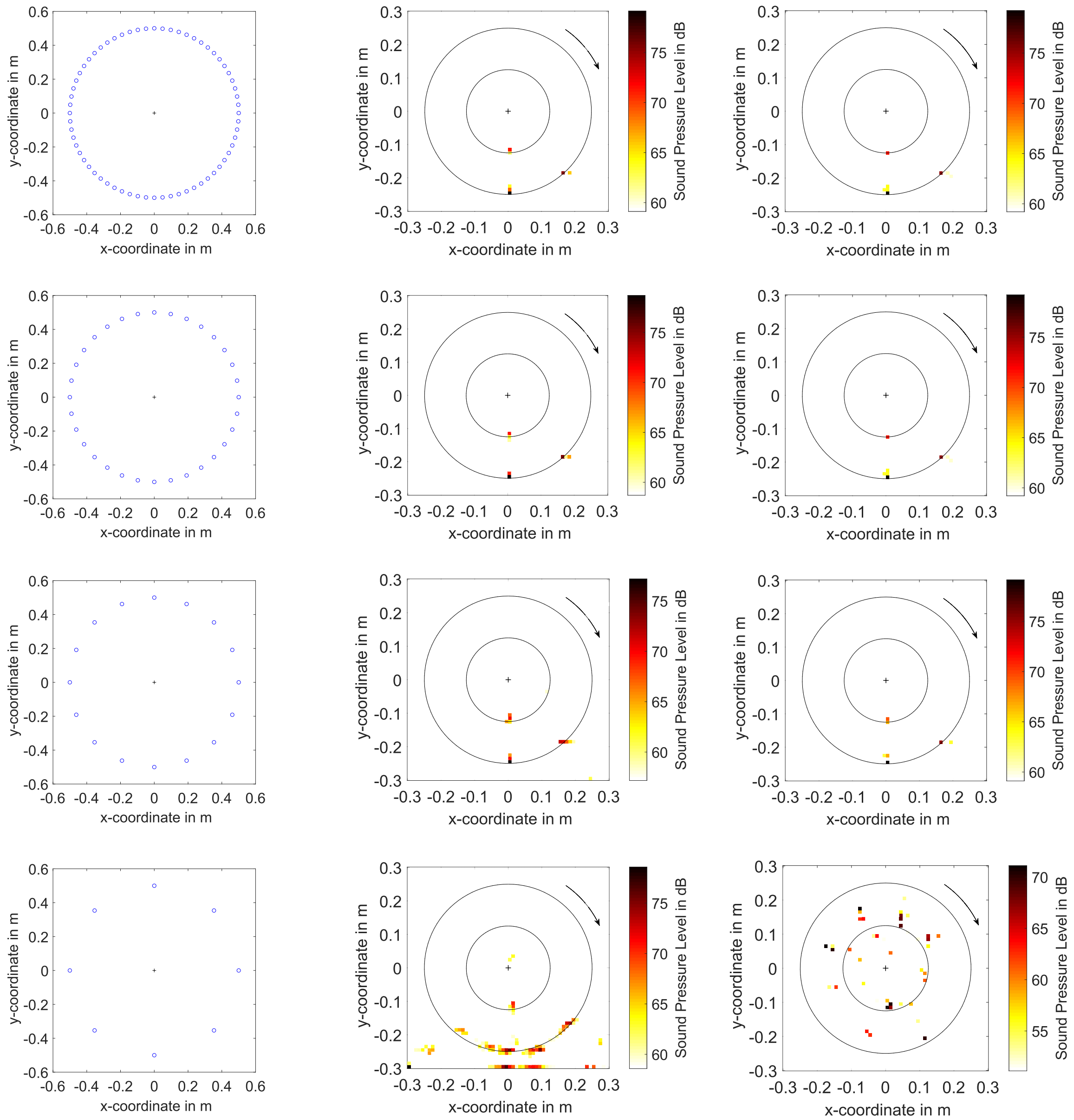



*Figure 4: Position of microphones (left), beamforming map with virtual rotating array method (middle) and with modal decomposition method (right) for the third octave band with center frequency 2000 Hz*

Both methods correctly identify the position and ranking of the sound sources as long as a certain number of microphones is not undercut. With a non-constant rotational rate, the data processing with the modal decomposition method could lead to a slight defocused point source if too large time sequences are used. Both methods work fine with 16 microphones for the analyzed third octave band with center frequency 2000 Hz. With 8 microphones, artifacts show up in the beamforming source maps. These artifacts result from energy leakage into points in vicinity of actual sources for the virtual rotating array method and energy leakage into other modes for the modal decomposition method.

## Low pressure axial fan N1UG

The benchmark is performed on a generic fan with unskewed fan blades. The experimental setup of the benchmark as well as the aerodynamic performance of the fan, fluid mechanical quantities from laser Doppler anemometer (LDA) measurements, wall pressure fluctuations in the gap region and sound characteristics are described in detail by Zenger et al. [15]. The test fan is mounted in a short duct and integrated into a standardized inlet test chamber according to ISO 5801 [16], as shown in Figure 5.

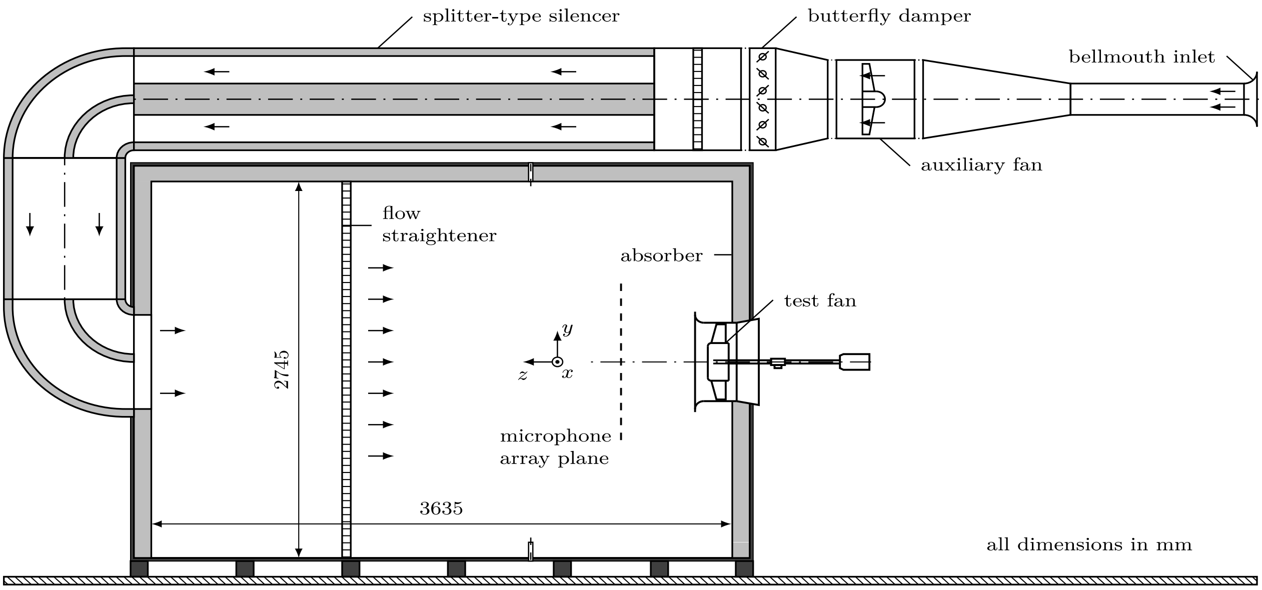



*Figure 5: Standardized inlet test chamber according to ISO 5801*

Fan design parameters and the operating point are listed in Table 1. The flow rate coefficient $\phi$ and the total-to-static pressure coefficient $\psi_{ts}$ are defined as

$$\phi = \frac{4\dot{V}}{\pi^2 D^3 n} \tag{8}$$

$$\psi_{ts} = \frac{2\Delta p_{ts}}{\rho(\pi D n)^2} \tag{9}$$

where $\dot{V}$ is the volume flow rate, D the fan diameter, n the rotational speed in clockwise direction as seen in Figure 6, $\Delta p_{ts}$ the total-to-static pressure difference and $\rho$ the air density.

*Table 1: Fan design parameters and operating point*

| Parameter | Value |
|---|---|
| Rotor diameter D | 0.495 m |
| Hub diameter $D_{hub}$ | 0.248 m |
| Number of blades z | 9 |
| Rotational speed n | 24.8 Hz |
| Flow rate coefficient $\phi$ | 0.18 |
| Total-to-static pressure coefficient $\psi_{ts}$ | 0.18 |

To determine the angular position of the fan, an optical fork sensor (one 3 V pulse per revolution) is used. The measurement is carried out with 64 microphones arranged on a ring-shaped array axisymmetric to the rotation axis of the fan. Figure 6 shows the arrangement of the microphones of the type 40PH-Sx (G.R.A.S. Sound and Vibration A/S) on a ring with a diameter of $D_{array} = 1$ m. The distance from the microphone array to the fan is 0.49 m.

*Figure 6: Generic test fan (rotational direction: clockwise) and microphone array*

The sampling frequency of the measured pressure signals is $f_{sample} = 48$ kHz, recorded with a PXIe-1075 front end with PXIe-4496 data acquisition modules (National Instruments Corporation). The evaluation time is $t_{meas} = 20$s, which corresponds to 500 revolutions.

The focus plane is discretized with an equally-spaced cartesian grid consisting of 3600 points with 0.01 m spacing. The analysis is performed with Clean-SC deconvolution with a loop factor of 0.9. For the plot, a dynamic range of 20 dB is used. Figure 7 shows on the left-hand side the beamforming map evaluated with the virtual rotating array method and on the right-hand side with the modal decomposition method for the third octave band with center frequency 2000 Hz.

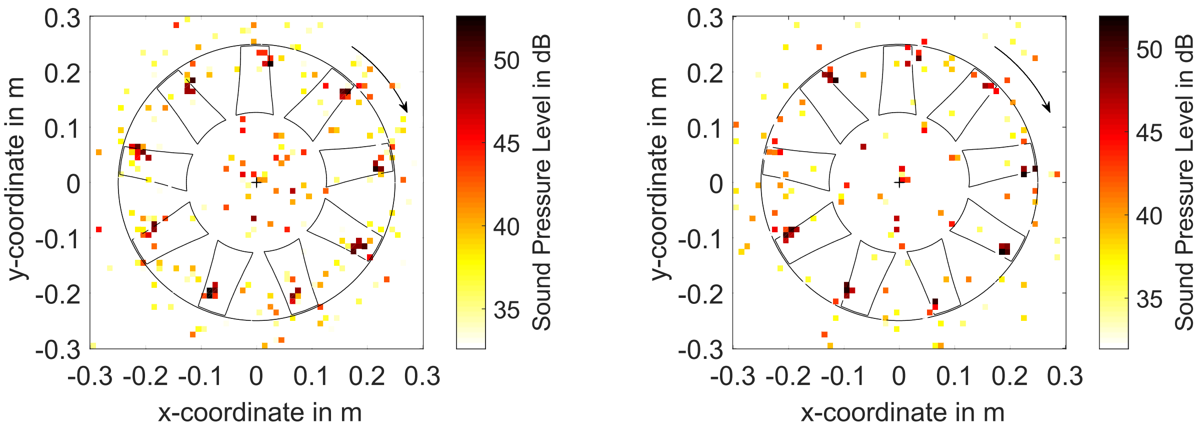
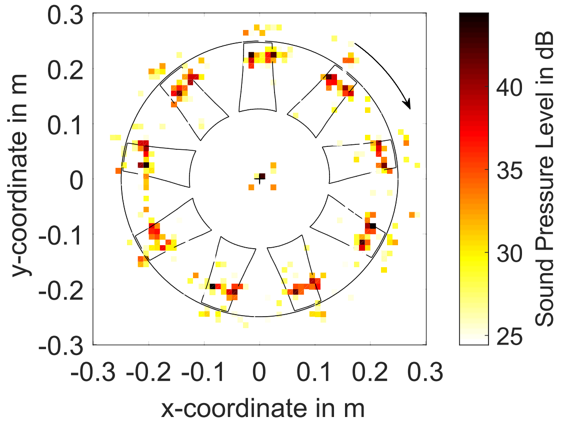



*Figure 7: Beamforming map of the fan N1UG evaluated with the virtual rotating array method (left) and the modal decomposition method (right) for the third octave band with center frequency 2000 Hz*

At this frequency band, the main sound sources are located near the leading edge. According to [1], for $\phi = 0.18$, the sound sources result from the interaction of the fan blade leading edge with the inflow. The higher angular velocity at greater radii leads to a stronger interaction with the inflow and therefore to a higher source strength near the blade tip. In Figure 8 the same evaluation is presented for the third octave band with center frequency 5000 Hz.



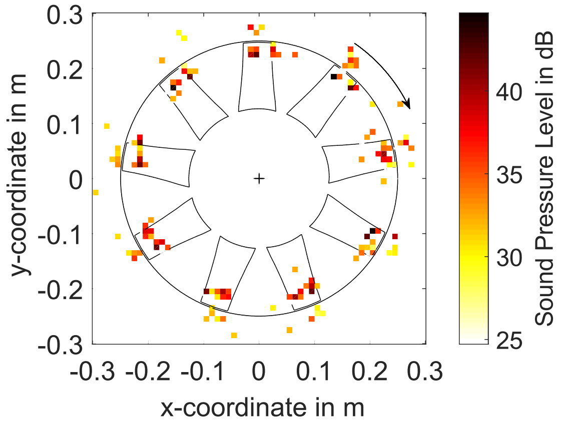




*Figure 8: Beamforming map of the fan N1UG evaluated with the virtual rotating array method (left) and the modal decomposition method (right) for the third octave band with center frequency 5000 Hz*

The sound sources are distributed from the fan blade leading to the fan blade trailing edge. At this operating point, the source distribution could result from the turbulent boundary layer and its interaction with the fan blade trailing edges.

Both methods yield comparable results in terms of localization and quantification of the sources at different frequency bands.

## CONCLUSION

The virtual rotating array method and the modal decomposition method, which aim to compensate the motion compensation of rotating sound sources, are presented and compared to each other. Two benchmark cases are considered for the assessment of rotating sound sources. At first, a simulated case with discrete rotating broadband sound sources with a non-constant rotational speed is analyzed. Secondly, a measured fan with distributed sound sources is considered. The methods' applicability is limited by the number of microphones employed. For the considered third octave band with center frequency 2000 Hz and a rotational speed around n = 1500 rpm, using less than 16 microphones leads to reconstruction errors, which manifest in method-characteristic artifacts. The minimum number of microphones to be used for measurements depends on the rotational frequency and the analyzed frequency band of the sources. Both methods correctly identify the position and the sound pressure levels of the sound sources at different frequency bands. Comparable results could be obtained for simulated and measured sound sources.

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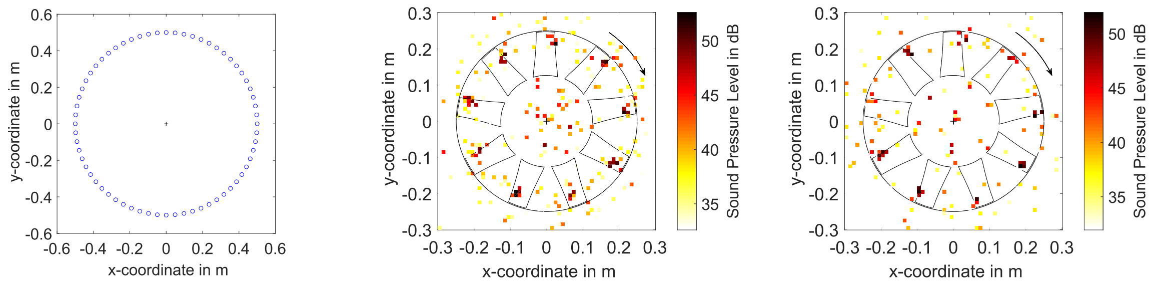
# ANNEXES

Figure A shows the beamforming map for the benchmark fan for the third octave band with center frequency 2000 Hz. The number of microphones are 64, 32, 16 and 8. The position of the microphones is shown on the left, the beamforming map as calculated with the virtual rotating array method in the middle, and the beamforming map calculated with the modal decomposition method on the right.



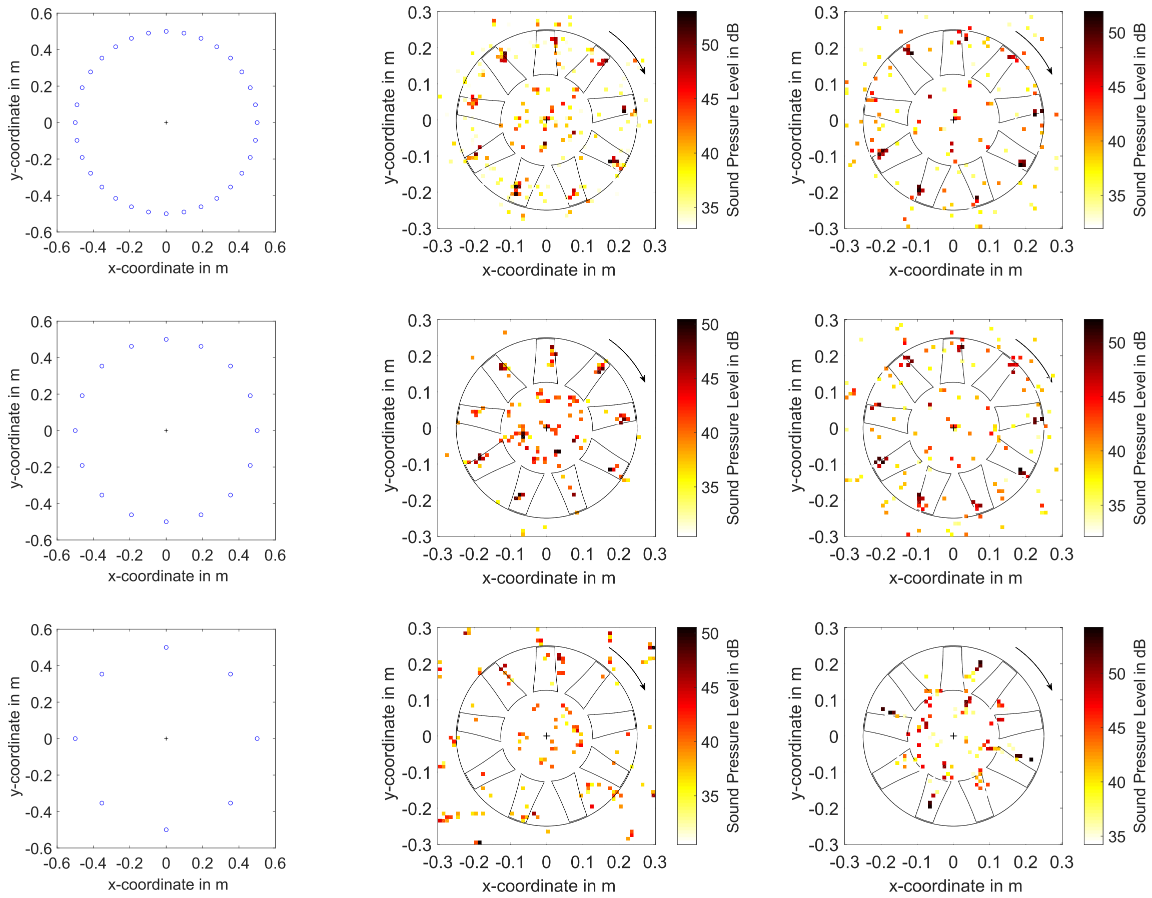



*Figure A: Microphone positions (left), beamforming map with the virtual rotating array method (middle) and with the modal decomposition method (right) for the third octave band with center frequency 2000 Hz*

The results of the benchmark fan are similar to the results of the first benchmark with three rotating point sources considering the source reconstruction with a reduced number of microphones. Both methods work fine with 16 microphones for the analyzed third octave band with center frequency 2000 Hz and a rotational speed around n = 1500 rpm. A source reconstruction with 8 microphones leads to artifacts in the beamforming source maps for both methods. To avoid artifacts at higher frequencies and higher rotational speeds, the number of microphones used for the measurement has to be increased.