BeBeC-2020-D31

Microphone array measurement for the determination of the absorption coefficient

Artur Paszkiewicz, Gert Herold and Ennes Sarradj
Technische Universität Berlin, FG Technische Akustik
Einsteinufer 25, 10587 Berlin, Germany

Abstract

Microphone array methods can be used to obtain the angle-dependent absorption coefficient in an in-situ measurement. The method presented in this contribution calculates the absorption coefficient on the base of determining the incidence and reflected sound waves. It consists of a two-step algorithm. First, the position of the source is localized using delay-and-sum beamforming. Then, an adaptive nulling algorithm, which is also based on the concept of beamforming, is employed. The method is examined in a simulation of a sound field above an absorber generated by the image source model. Furthermore, it is verified experimentally by an in-situ measurement in an anechoic room with a typical acoustical material. Parameters like sample type and number or arrangement of microphones are varied to detect the limits of application of the method. The results of the investigations are compared with standard procedures for determining absorption coefficients.

1 INTRODUCTION

These days it is not a challenge to preciseley calculate a sound field in a room due to well-advanced simulation algorithms. The refined algortihms consider more and more the wave propagation effects such as diffraction and reflection [11]. A simulation is dependent on the boundary conditions that are determined by the acoustical properties of a material. For an accurate description of the wall properties, the material parameters must be known.
To obtain such parameters, there exist standard measurement procedures, like the determining of the absorption coefficient in the reverberation room [2] or impedance tube [1]. With these methods the absorption coefficient will identify only for diffuse field conditions or in vertical angle of incident. Both methods are for laboratory conditions. However the properties of a material conduct in an other manner in an installed condition or often there is no chance to bring the test material in such a laboratory. For that reason, it is essential to measure the absorption coefficient or other acoustic parameters of material in an in-situ procedure.

One know method to measure the angle-dependent reflection coefficient in an in-situ method is from [6]. The basis of the measurement is a time-dependent separation of the direct and reflection part of a stimulation impulse.
One more proposed method is from Tamura [10]. In this approach the reflection coefficient will be calculated on the basis of a spatial Fourier transform. The pressure will be measured in two planes above the sample and the incident and reflective wave components are figure out due to the propagation theory and previous decomposing into plane-wave components. For this is a large area of a sample necessary.
As every method has its own limits, a procedure on the foundation of beamforming will be presented in this contribution. The method will be performed in a source image simulation and a measurement in an anechoic room with two different test materials.
To determine the absorption or reflection coefficient, the complex sound pressure will be measured at several microphones arranged in a linear array above the sample for different angle of incident. The incident and reflective wave will be estimated by using an adaptive nulling algorithm [5]. As the measurement is angle-dependent, the incident and reflective angle will figured out with conventional beamforming algorithms.

2 MICROPHONE ARRAY PROCEDURE

The presented microhone array procedure based on a beamforming algorithm to calculate a reflection coefficient. After the estimation of the incident and reflective signals with the use of a microphone array, the result is generated from the ratio. Here the adaptive nulling algorithm according to Sun et al. [9] will be used. Previously, the acoular software package [8] is used to calculate the incident and reflected angle.

2.1 Approach to calculate the reflection coefficient

The adaptiv nulling algorithm is based on the idea of maximize the signal from the regarding direction.
Assuming plane waves conditions, the transfer function $H_m = \frac{p_m}{p_0}$ between the pressure at the reference microphone and microphone m for a known angle of incidence θ in the frequency domain submits to Eq. (1) due to the phase difference between each microphone. In Eq. (1), s_i and s_r are the incident and reflected signal components, whereas x_o defines the distance from the material under test to the nearest sensor. As illustrated in Fig. 1, the incident and reflected signal can be evaluated by using a microphone array with M linearly spaced microphones with distance x_m.

$$\begin{bmatrix} H_0 \\ H_1 \\ \vdots \\ H_{M-1} \end{bmatrix} = \begin{bmatrix} 1 & 1 \\ e^{-jk(x_1-x_0)\cos\theta} & e^{jk(x_1-x_0)\cos\theta} \\ \vdots & \vdots \\ e^{-jk(x_M-x_0)\cos\theta} & e^{jk(x_M-x_0)\cos\theta} \end{bmatrix} \begin{bmatrix} s_\mathrm{i} \\ s_\mathrm{r} \end{bmatrix} + \begin{bmatrix} \frac{n_0}{p_0} \\ \frac{n_1}{p_0} \\ \vdots \\ \frac{n_{M-1}}{p_0} \end{bmatrix} \tag{1}$$

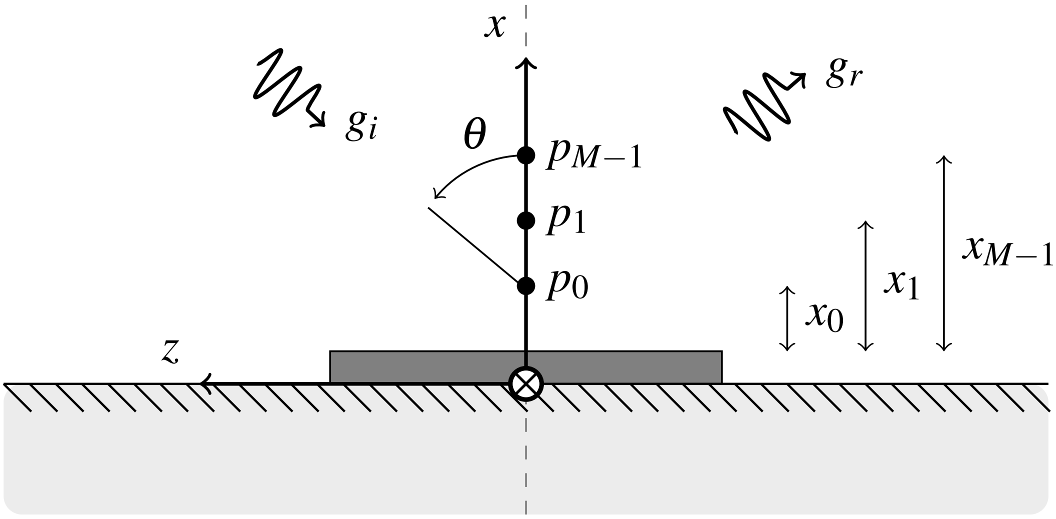


Figure 1: Illustration of the microphone setup.

Equation (1) can be written as an outer product:

$$\boldsymbol{H} = [\boldsymbol{g_i} \;\; \boldsymbol{g_r}]\,\boldsymbol{s} + \frac{1}{p_0}\boldsymbol{n}\,. \tag{2}$$

The unknown incident s_i or reflected signal s_r will be calculated by estimating the beamforming steering vector $\boldsymbol{q}_\mathrm{i}$ or $\boldsymbol{q}_\mathrm{r}$. According to Kang and Hwang [5] the beamforming steering vector $\boldsymbol{q_i}$ for the incident signal s_i fulfills following conditions:

$$\max|\boldsymbol{q_i}\cdot\boldsymbol{g_i}|, \text{ subject to } \|\boldsymbol{g_i}\| = 1, \tag{3}$$
$$\boldsymbol{q_i}\cdot\boldsymbol{g_r} = 0\,. \tag{4}$$

To solve these conditions, the orthogonal properties of the vectors will be utilized and following [9] the beam steering vector $\boldsymbol{q_i}$ are given by

$$\boldsymbol{q_i} = \frac{\sum\limits_{k=2}^{M}(\boldsymbol{e_k}\cdot\boldsymbol{g_i})\,\boldsymbol{e_k}}{\left\|\sum\limits_{k=2}^{M}(\boldsymbol{e_k}\cdot\boldsymbol{g_i})\,\boldsymbol{e_k}\right\|}. \tag{5}$$

In Eq. (5) the vector $\boldsymbol{e_k}$ represents the textitk-the eigenvector of the matrix $\boldsymbol{g_r}\boldsymbol{g_r}^\mathrm{H}$. Thereby, only those eigenvectors are used, which correspond to an eigenvalue equal to zero. The incident signal s_i submits to

$$s_\mathrm{i} = \frac{\boldsymbol{q_i}\cdot\boldsymbol{H}}{\boldsymbol{q_i}\cdot\boldsymbol{g_i}} \tag{6}$$

because of the use of inner product between the beam steering vector $\boldsymbol{q}_\mathrm{i}$ and the the transfer function vector $\boldsymbol{H}$ [9]. Following the procedure described by Eq (5) and Eq. (6), the reflected signal s_r can be calculated. Therefore, the two conditions in Eq. (3) and Eq. (4) are modified by changing the direction vector $\boldsymbol{g_i}$ of the incident wave with the reflective wave $\boldsymbol{g_r}$ to establish the beamforming steering vector $\boldsymbol{q}_\mathrm{r}$. In analogy to Eq. (6), the incident signal s_r is given by

$$s_\mathrm{r} = \frac{\boldsymbol{q_r} \cdot \boldsymbol{H}}{\boldsymbol{q_r} \cdot \boldsymbol{g_r}}. \tag{7}$$

The reflection coefficient is defined as the ratio of the reflective s_r and incident s_i signal:

$$R(\theta) = \frac{s_\mathrm{r}}{s_\mathrm{i}} e^{-jk2x_0 \cos\theta}. \tag{8}$$

The complex function in Eq. (8) accounts for the phase difference caused by the distance of the array to the object to be measured.

$$\alpha(\theta) = 1 - |R(\theta)|^2. \tag{9}$$

2.2 Beamforming to determine the incident and reflective angle

Beamforming is used for the estimation of incident and reflective angle. The squared sound pressure of a source with an sound incidence angle θ is given by

$$B(\theta) = \boldsymbol{h}^H(\theta)\boldsymbol{C}\boldsymbol{h}(\theta), \tag{10}$$

where $\boldsymbol{C}$ is the cross spectral matrix and $\boldsymbol{h}(\theta)$ is the steering vector. The steering vector depends on the source properties and according to [7] there exist different formulations for it. In this contribution, a plane wave assumption is used and the transfer function for the incident wave $\boldsymbol{g_i}$ is defined in Eq. (1).Here, the location θ of the source is specified as an angle of incident.
Conventional and Functional beamforming is used to obtain the squared sound pressure of an incident wave at the reference position. In the delay-and-sum beamforming the sensor outputs are weighted dependent on the time of arrival and sum up together [4]. According to [3], functional beamforming determines the source strength from the eigenvalue decomposition of the cross spectral matrix. The functional beamforming map is given by

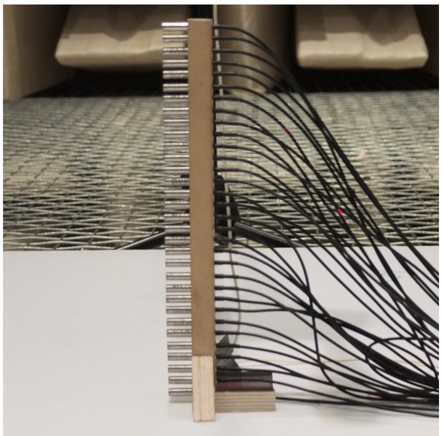
$$B_\nu(\boldsymbol{h}) = \left[\boldsymbol{h}^H(\theta)\boldsymbol{C}\boldsymbol{h}(\theta)\right]^\nu \tag{11}$$

with $\nu \geq 1$. In this contribution, the parameter ν will be set to 8.

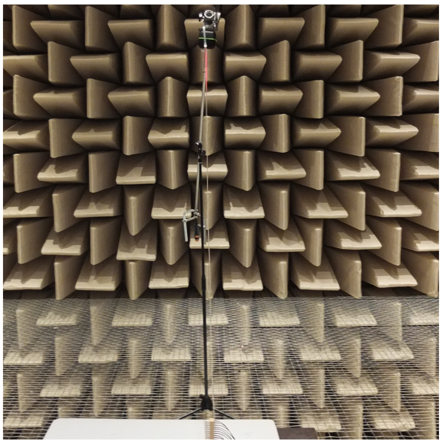
3 MEASUREMENT SETUP

For a verification of the microphone array procedure a simulation and a measurement in an anechoic room are performed. For the simulation the image source model is used. Therefore a sound field with two sources with different levels, but same white noise, are generated at mirrored positions using *Acoular* [8].
For the experimental investigation two different samples are used. One sample is a wooden plate with the size of 1.5 m x 1.4 m and a thickness of 0.02 m. The other one is a porouse material named Basotect in the dimensions of 2.0 m x 1.25 m and a thickness of 0.05 m. Furthermore, the array for the measurement is shown in Fig. 2a. The measurement is conducted with 32

microphones, which are positioned at a distance of 0.01 m to each other in a linear arrangement. But for the evaluation later, different arrangement will be taken into account. The space between the sample and the first microphone amount 0.01 m.
The used source is a small portable loudspeaker with a membrane diameter of 0.07 m. The stimulate signal is a white noise with a measurement time of 3 minutes. The loudspeaker is placed at various angles between 0 ° and 90 °, but every time at least in 2.00 m distance to keep the plane wave condition. As an example the loudspeaker position for an incidence angle of 0 ° is imaged in Fig. 2b.
The transfer functions H_m between each microphone are determined with a FFT size of 16384 and a Hanning window.



(a) The microphone array arrangement.



(b) The loudspeaker postion at an angle of 0 °.

Figure 2: Experimental setup of the measurement in an anechoic room.

4 RESULTS

4.1 Determining of the incidence and reflective angle

Before the introduced procedure is used to determined the absorption coefficient α of the two samples, the incidence angle is found out due to conventional beamforming. The result of the measurement with a wooden plate for four different angle of incidence is shown in Fig. 3. All 32 microphones are used. For a comparison, Fig. 4 presents the result of the simulation for a source image with reflection coefficient one. All displayed spectra are indicated relative to its maximum per narrow frequency to delete the influence of the speaker.
Both results are shown, that for the delay-and-sum algorithm the right angle of incidence are founded. The maximum can be located at the angle of incidence for a large frequency area. There is also a second maximum with the same frequency behaviour like the other one. This maximum is the mirror source of the original source. For the deep frequency area, the maximum line is inaccurate due to the short microphone distance in comparison with the wave length.In the higher frequency there are some artefacts. Here are the limits of the beamforming algorithm.
In according to [9], Fig. 5 and Fig. 6 present the results for a beamforming with only three

microphones. The simulation results in Fig. 6 shows, that the beamforming maps have larger areas with equal sound pressure levels. The determination of the incidence angle will be more inexact, due to a lack of information.
For the evaluation of the measurement with the wooden plate with only three microphones, the results are calculated with the functional beamforming algorithm. The incidence angle is not good hited as that as in the previous results, but a trend to the target angle is apparent.
Because the use of 32 microphones is not practical for an in-situ-mesurement, one further microphone arrangement is tested. In Fig. 7 is the result of the measurment with the use of six microphones in an non-redundant arrangement. This arrangement has different distances between source and every seperate microphone [4]. The results show clearly a line above the frequency with the maximum at the target angle. Only for the almost at grazing-incidence angle of 85 ° the main source and the mirror source merge together. Some beamforming artefacts can be viewed.
Figure 9 to Fig. 11 show the beamforming results for the Basotect sample with the same microphone arrangement like above. Also here it is clear, that the target angle fits with the maximum of the map. Now the mirror source is not visible, due to the higher absorption of the material. Also the results in Fig. 10 with the use of only three microphone describe more precise ares of the incidence angle.

4.2 Estimation of the absorption coefficient

After estimation of the incidence angle, the absorption coefficient is calculated with the procedure described in Sec. 2.1. The results are only determined for the microphone arrangement with three and six microphones.
For the wooden plate Fig. 12 and Fig. 13 are shown the absorption coefficient α in according with Eq. (9) for four different angles. The angels are yielded from the beamforming results above with the same loudspeaker positions. Here, the maximum in the beamforming map is taken. Is the maximum equivalent or in the near to the angle of the mirror source, it will be converted to the incidence angle of the original source.
A wooden plate should have a absorption coefficient above zero. In Fig. 12 archives the absorption coefficient values till 0.7 for an angle of 14 °. In case of the measurement with three microphones the absorption coefficient increases with the frequency till 800 Hz and then falls again. This fact shows, that here is the limit of the presented algorithm. Also for low frequencies the error increases because the plane wave condition is not longer fulfilled. In the results of the measurement with six microphones, the absorption coefficient are clearly smaller.
Also the absorption coefficient for the Basotect material are calculated. The results for the use of three microphones are presented in Fig. 14. The absorption coefficient increased with the frequency and achieved the value of one. This behaviour indicates every angle. But for the grazing-incidence angle of 85 the development of the absorption coefficient is untypical for such a porous material. The result for the measurement with six microphones shows defects in the low and high frequency range.
The comparison between the absorption coefficients determined with the approach of this contribution and the absorption coefficient from the manufactum specification out of a measurement in the impedance tube in Fig. 14 and Fig. 15 shows a good accordance at orthogonal sound incidence.

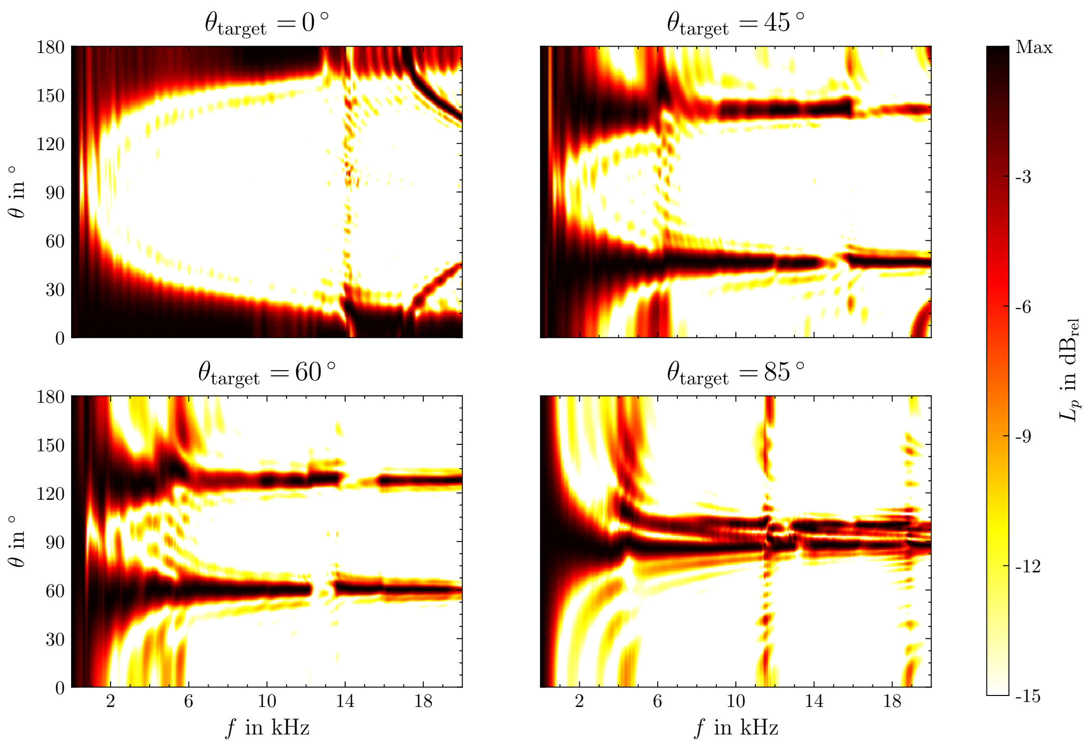


Figure 3: Beamforming results of the measurement with 32 microphones and a wooden plate to find out the incidence and reflective angle.

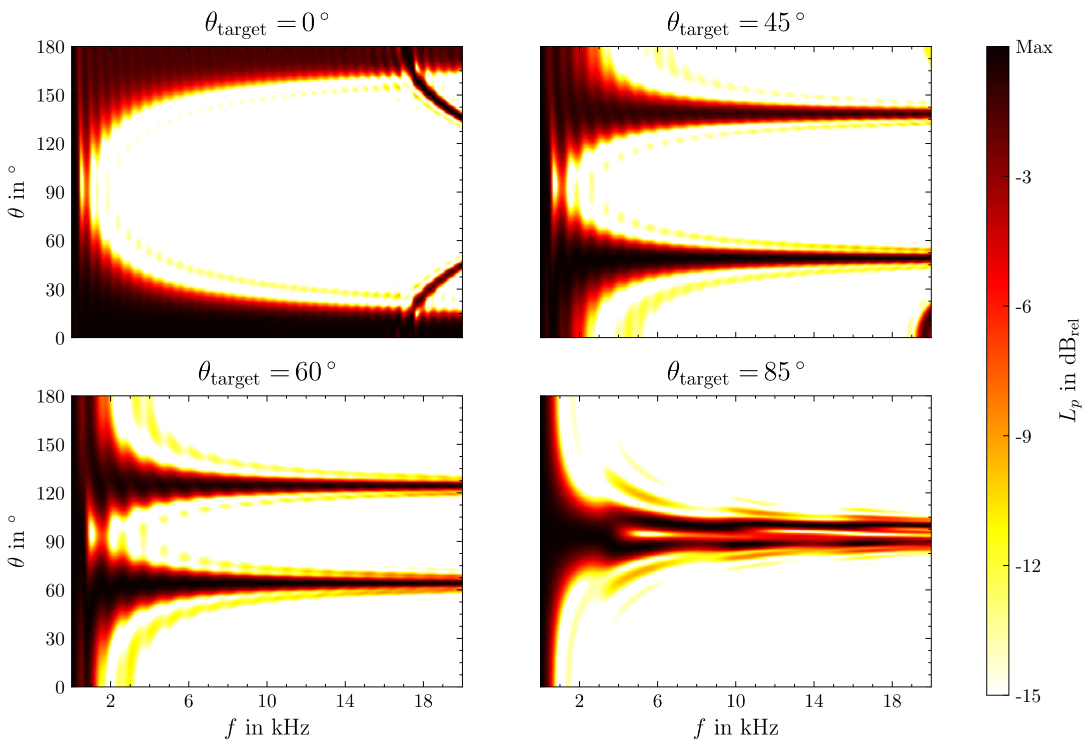


Figure 4: Beamforming results of the simulation with 32 microphones to find out the incidence and reflective angle.

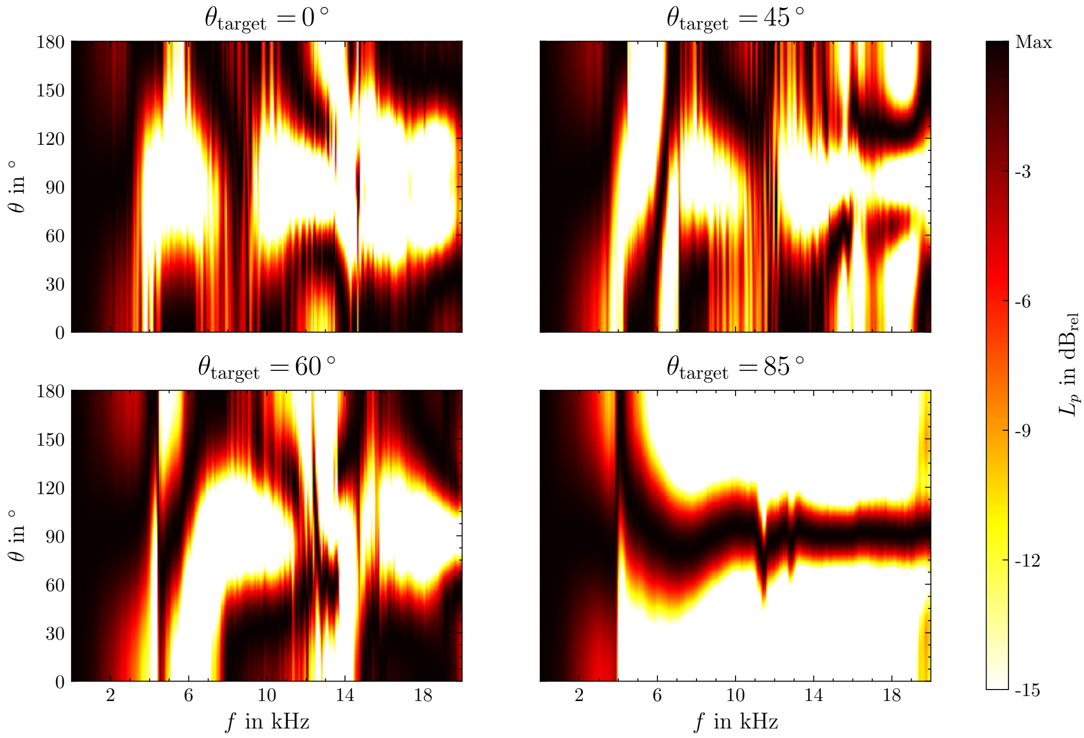


Figure 5: Beamforming results of the measurement with three microphones and a wooden plate to find out the incidence and reflective angle.

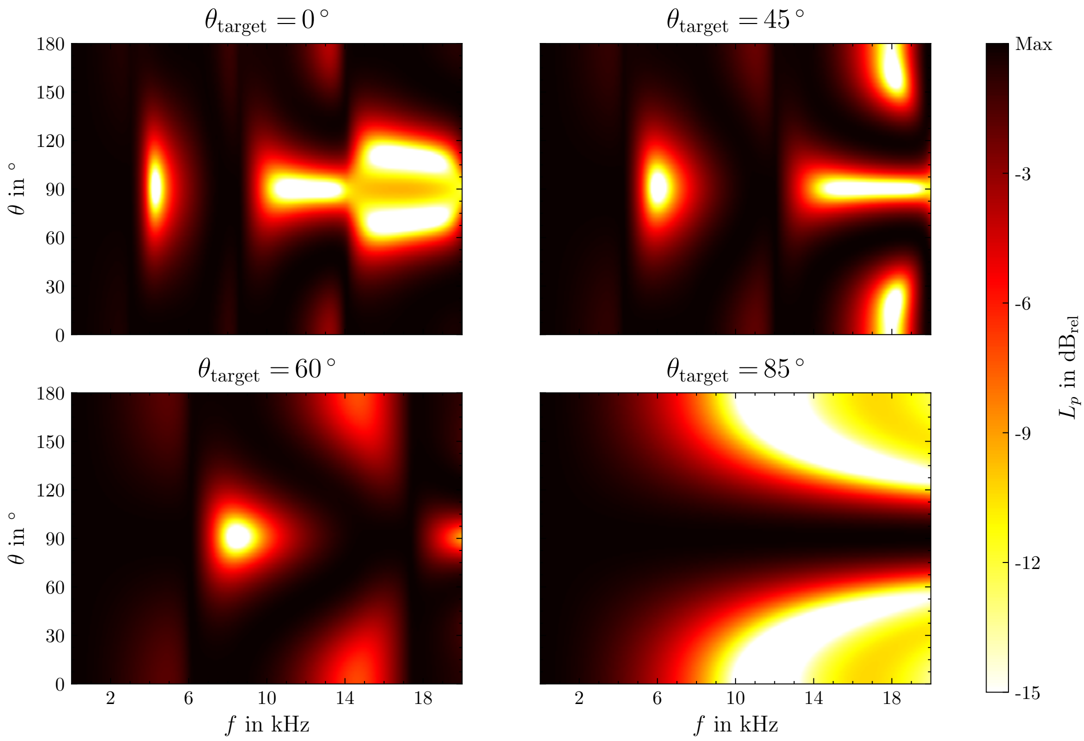


Figure 6: Beamforming results of the simulation with three microphones to find out the incidence and reflective angle.

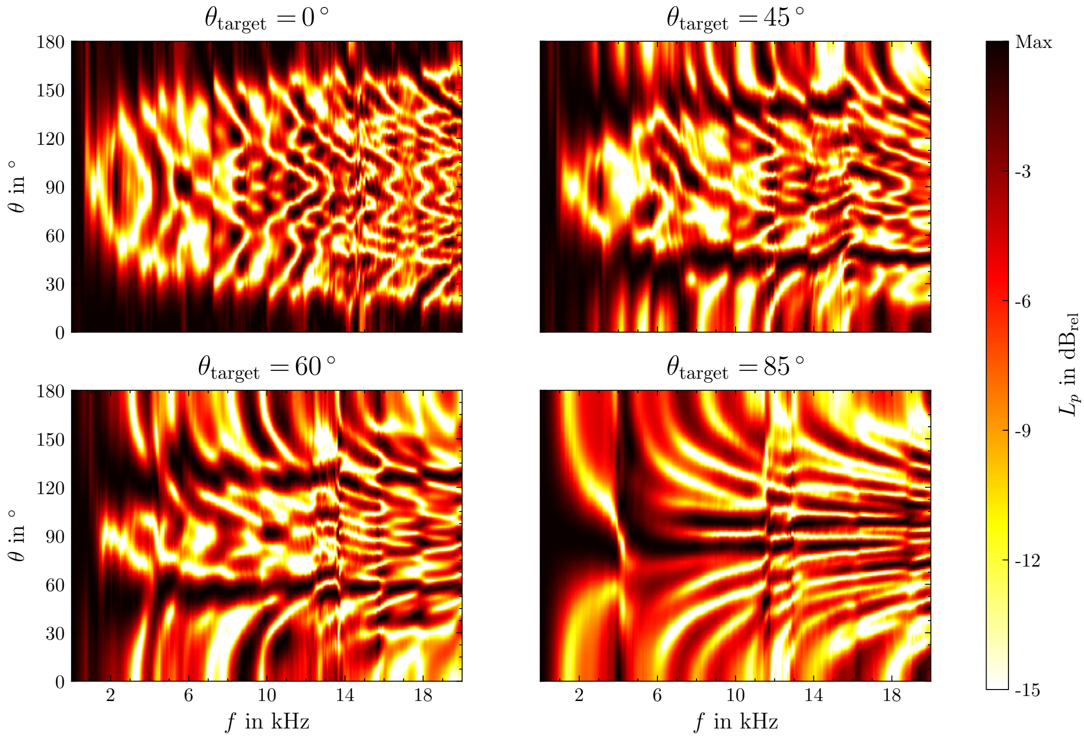


Figure 7: Beamforming results of the measurement with six microphones in an non-redundant arrangement and a wooden plate to find out the incidence and reflective angle.

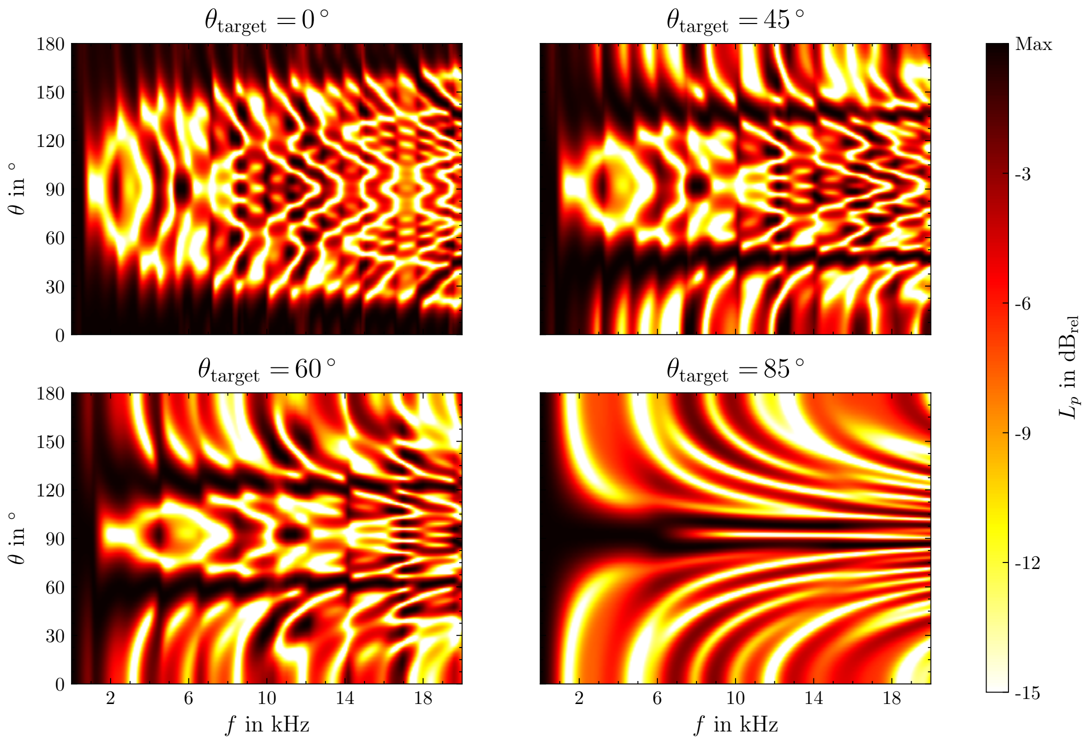


Figure 8: Beamforming results of the simulation with six microphones in an non-redundant arrangement and a wooden plate to find out the incidence and reflective angle.

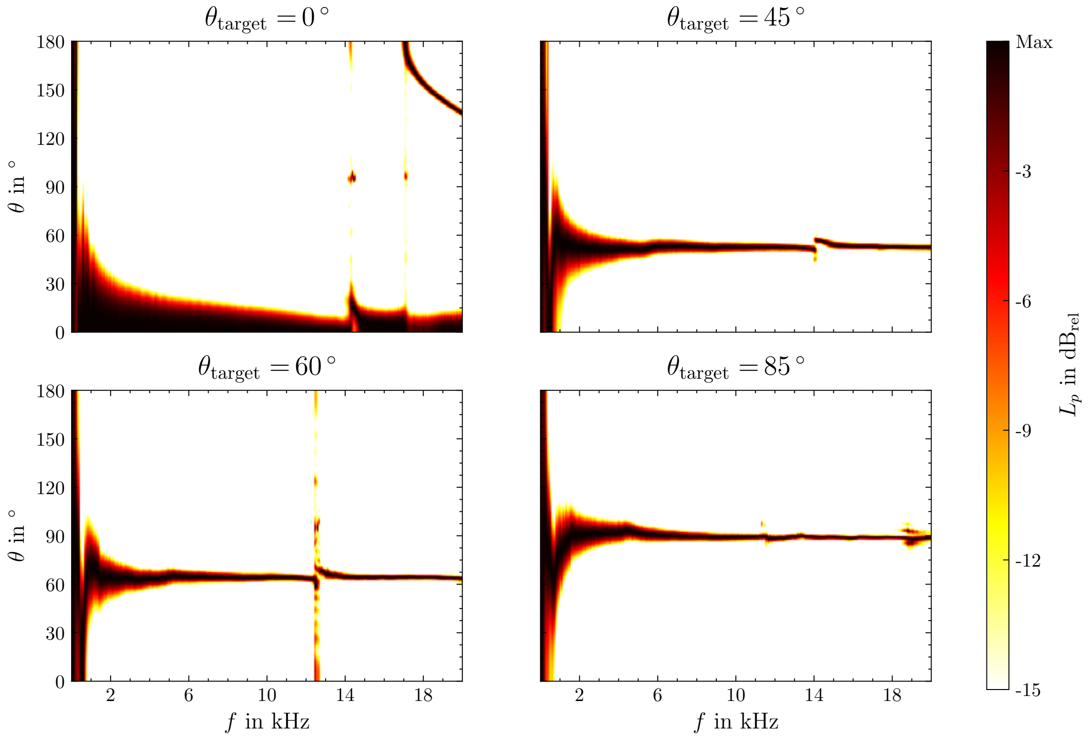


Figure 9: Beamforming results of the measurement with 32 microphones and a Basotect sample to find out the incidence and reflective angle.

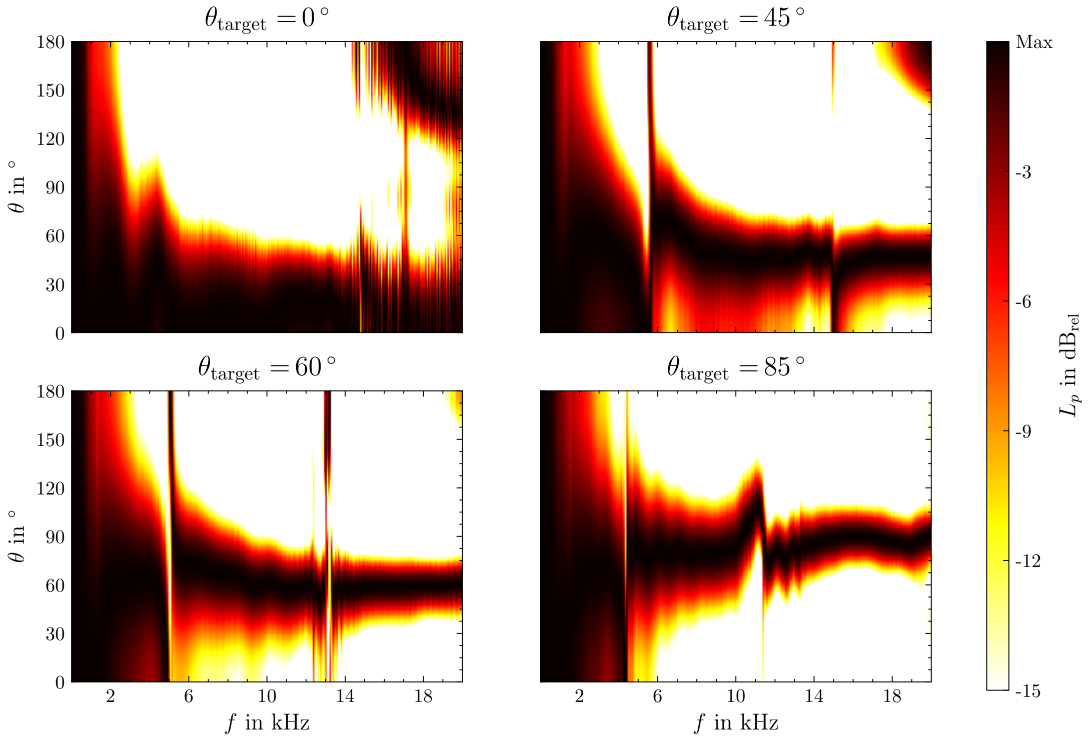


Figure 10: Beamforming results of the measurement with three microphones and a Basotect sample to find out the incidence and reflective angle.

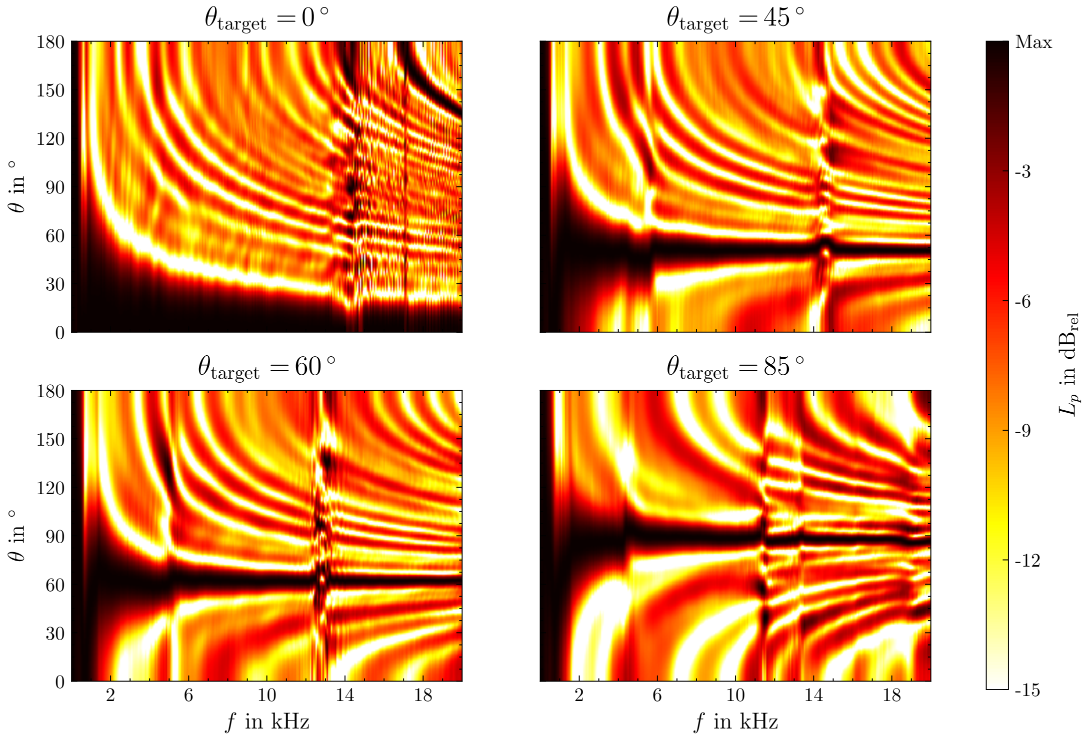


Figure 11: Beamforming results of the measurement with six microphones in an non-redundant arrangement and a Basotect sample plate to find out the incidence and reflective angle.

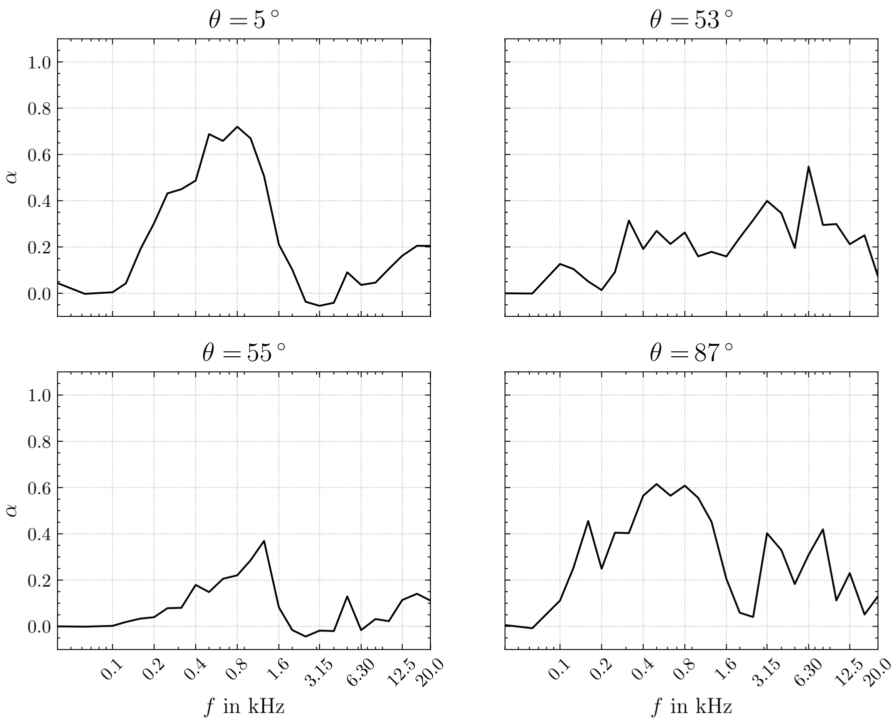


Figure 12: Absorption coefficient of the measurement with the wooden plate and three microphones per third octave band.

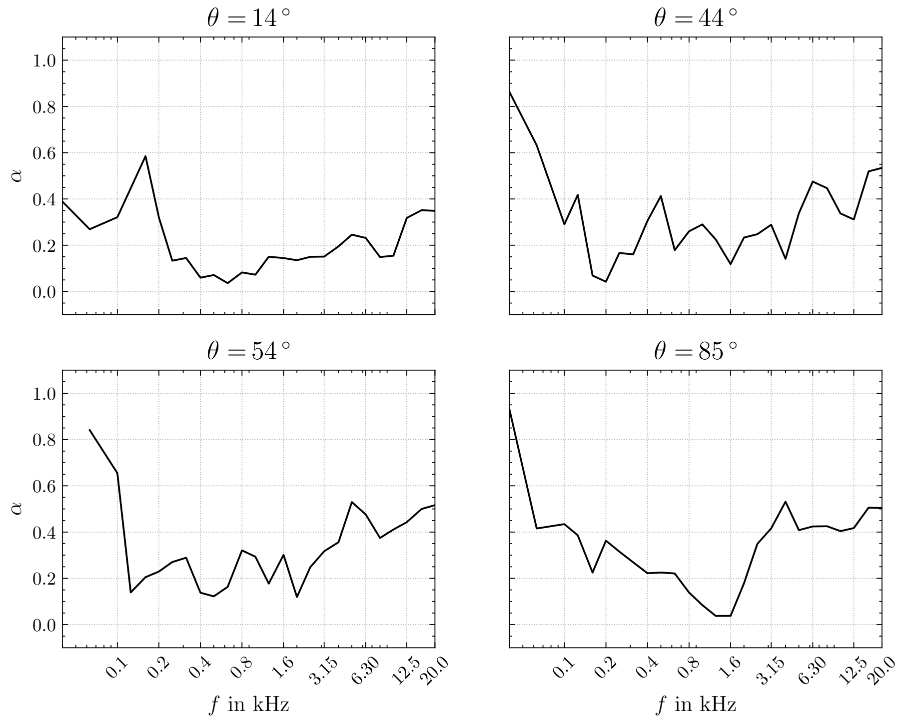


Figure 13: Absorption coefficient of the measurement with the wooden plate and six microhpones in an non-redundant arrangement per third octave band.

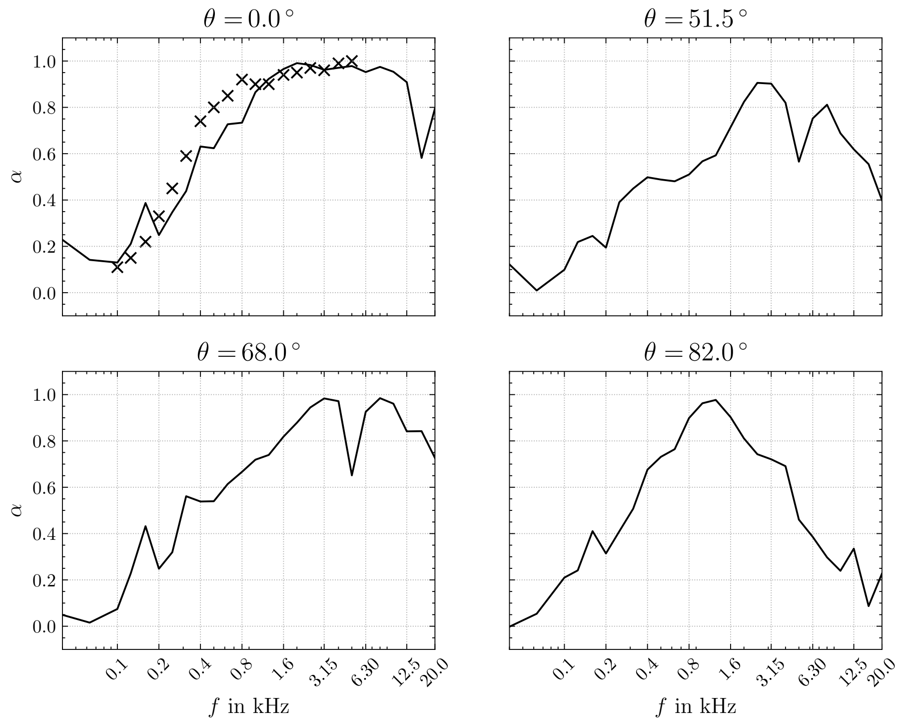


Figure 14: Absorption coefficient of the measurement with the Basotect sample and three microphones per third octave band. x: Manufacturer specifications for the absorption coefficient from an impedance tube measurement.

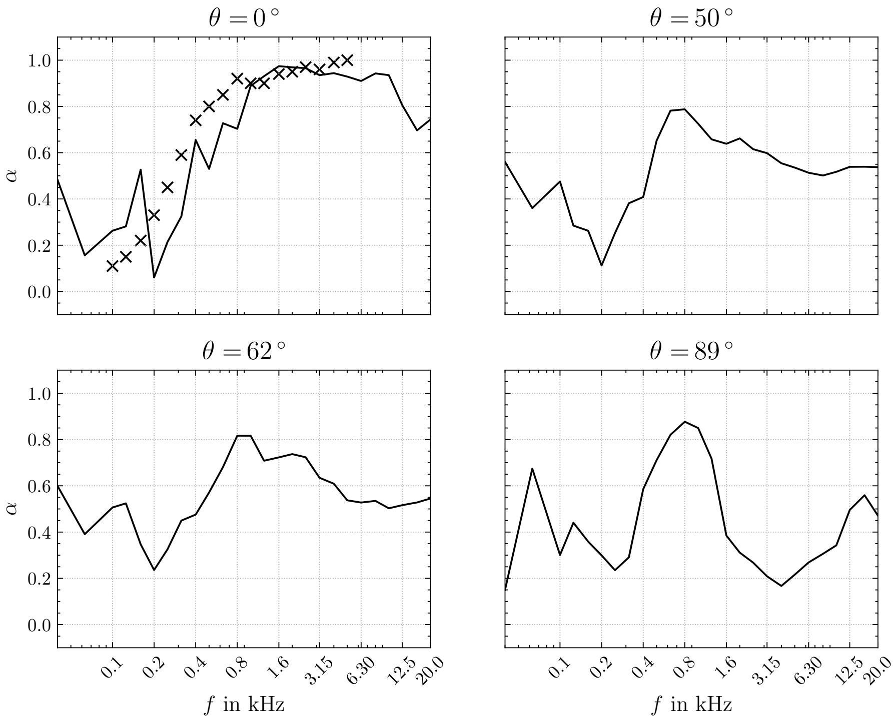


Figure 15: Absorption coefficient of the measurement with the Basotect sample and six microphones in an non-redundant arrangement per three third octave band. x: Manufacturer specifications for the absorption coefficient from an impedance tube measurement.

5 CONCLUSIONS

The angle-dependent absorption coefficient for two different materials were determined from a measurement in an anechoic room and a source image simulation. The right angle of sound incidence could be specified with conventional beamforming algorithms in spite of two coherent sources. At this the results discover the limits caused of the lack of information because of too few microphones.
The absorption coefficient of a porous material could be determined correctly in comparison to the manufacturer specifications. Limits are shown for a high frequency range and nearly grazing angle of incidences. The use of the algorithm presented in this contribution with heavy reflecting materials should be further developed.

References

[1] "ISO 10534:1998 - Acoustics - Determination of sound absorption coefficient and impedance in impedance tubes."

[2] "ISO 354:2003 - Acoustics - Measurement of sound absorption in a reverberation room."

[3] R. P. Dougherty. "Functional Beamforming." *5th Berlin Beamforming Conference, February 19-20 2014, Berlin, Germany.*, 2014. doi:10.2514/6.2014-3066.

[4] D. H. Johnson and D. E. Dudgeon. *Array Sginal Processing:Concepts and Techniques*. P T R Prentice Hall, Englewood Cliffs, 1933. ISBN 9780130485137.

[5] Y. J. Kang and E. S. Hwang. "Beamforming-based partial field decomposition in NAH." *Journal of Sound and Vibration*, 314(3-5), 867–884, 2008. ISSN 0022460X. doi:10.1016/j.jsv.2008.01.010.

[6] E. Mommertz. "Angle-dependent in-situ measurements of reflection coefficients using a subtraction technique." *Applied Acoustics*, 46(3), 251–263, 1995. ISSN 0003682X. doi:10.1016/0003-682X(95)00027-7.

[7] E. Sarradj. "Three-dimensional acoustic source mapping with different beamforming steering vector formulations." *Advances in Acoustics and Vibration*, 2012, 2012. ISSN 16876261. doi:10.1155/2012/292695.

[8] E. Sarradj and G. Herold. "A Python framework for microphone array data processing." *Applied Acoustics*, 116, 50–58, 2017. ISSN 0003682X. doi:10.1016/j.apacoust.2016.09.015.

[9] J. C. Sun, C. W. Shin, H. J. Ju, S. K. Paik, and Y. J. Kang. "Measurement of the normal acoustic impedance using beamforming method." *Journal of Mechanical Science and Technology*, 23(8), 2169–2178, 2009. ISSN 1738-494X. doi:10.1007/s12206-009-0435-z.

[10] M. Tamura. "Spatial Fourier transform method of measuring reflection coefficients at oblique incidence. I: Theory and numerical examples." *The Journal of the Acoustical Society of America*, 88(5), 2259–2264, 1990. ISSN 0001-4966. doi:10.1121/1.400068.

[11] M. Vorländer. "Computer simulations in room acoustics: Concepts and uncertainties." *The Journal of the Acoustical Society of America*, 133(3), 1203–1213, 2013. ISSN 0001-4966. doi:10.1121/1.4788978.