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A beamforming system based on the acousto-optic effect

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Summary

Beamforming techniques are usually based on microphone arrays. The present work uses a beam of light as a sensor element, and describes a beamforming system that locates sound sources based on the acousto-optic effect, this is, the interaction between sound and light. The use of light as a sensing element makes this method immune to spatial aliasing. This feature is illustrated by means of simulation and experimental results. For ease of comparison, the study is supplemented with results obtained with a line array of microphones.

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1. Introduction

Beamforming techniques localize sound sources using a finite number of input signals that are captured with an ensemble of transducers. The lower frequency range of analysis is typically related to the dimensions of the array, that is, below a certain frequency the information captured with the transducers of the array is almost identical, and thus, the beamforming output becomes omnidirectional. The upper frequency is usually limited by the dimensions of the transducers (to avoid scattering effects) and/or the spatial sampling performed by the layout of the array (to prevent aliasing effects). The scattering can partly be compensated for applying a frequency response correction that will only counteract the possible effects for a certain direction of incident sound, typically axial incidence. The spatial aliasing problem is in practice more difficult to circumvent because the straightforward solution of increasing the density of transducers in the array layout also compromises the transparency of the array versus the incident sound field. In any case, spatial aliasing above a certain frequency is unavoidable when using a finite number of transducers.

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The present work proposes a beamforming system that uses the acousto-optic effect to localize sound sources in the far field. The use of a beam of light as a sensing element makes this beamforming system immune to spatial aliasing. The article is organized in the following manner: First, for ease of comparison, a brief section describing the effects of spatial aliasing in conventional Delay-and-Sum beamforming (DSB) is given. Then follows a section about the physical principles governing the acousto-optic beamformer. The performance of the acousto-optic beamformer is assessed by means of experimental results and compared with results obtained with DSB. Before the conclusions, a few comments on the possibility of using this method in nearfield beamforming are discussed.

2. Conventional beamforming and spatial aliasing

In the far field, the localization of acoustic sources can usually be reduced to a simpler task: the identification of direction of propagation of plane waves. Thus, when using microphone arrays, no relevant information is contained in the sound pressure levels of the sensors, these are approximately the same at all positions, but the key feature that indeed unravels the location of the source is the phase difference between the signals captured by the transducers. Classical DSB clearly benefits from this, that is, the beamforming output is maximum when the phase shifts applied to the sensor signals compensate for the propagation delays caused by the layout of the microphone array. For the particular case of a line array of microphones, the beamforming output can be written analytically as follows[1],

$$\begin{aligned} b_{DS}(\theta) &= \left| \frac{1}{M} \sum_{m=0}^{M-1} \tilde{p}_m e^{-jkmd\sin\theta} \right|^2 \\ &= \left| \frac{\tilde{P}_0}{M} \frac{\sin\Big(k(\sin\tilde{\theta} - \sin\theta)Md/2\Big)}{\sin\Big(k(\sin\tilde{\theta} - \sin\theta)d/2\Big)} \right|^2, \end{aligned} \quad (1)$$

where all microphones are assumed to be equally important, $\tilde{p}_m$ represents the Fourier transform of the sound pressure of the m'th microphone, d and M are the microphone spacing and the total number of microphones, k is the wavenumber, $e^{-jkmd\sin\theta}$ is the phase shift introduced to the m'th microphone in order to compensate for the delay experienced by incident waves coming from the direction θ, and $\tilde{P}_0$ and $\tilde{\theta}$ correspond to the amplitude and the incident direction of the actual sound measured at the microphone positions. Figure 1 shows the beamforming output of a line array of 19 microphones when the acoustic source is located at three different positions. The reference angle $\theta = 0°$ corresponds to the perpendicular direction of the line array. As can be seen in panels (a) and (b), the output consists of two mainlobes because the symmetry of the line array makes it impossible to distinguish between the real source and its corresponding image source. Note that these two mainlobes are overlapped in panel (c) and that they become wider as the incident sound approaches grazing incidence.

The prominent C-shaped curves observed at high frequencies are the result of spatial aliasing. The number of transducers per unit length is not sufficient to map the location of the source correctly at high frequencies, yet misleading to the presence of ghost sources. The Nyquist theorem states that the spacing between the transducers must always be smaller than half the wavelength (or equivalently $f < c/(2d)$), otherwise spatial aliasing can corrupt the measurement. The Nyquist frequency of the line array simulated in Figure 1 is approximately 4.5 kHz. This corresponds to the worst case scenario, which occurs when the source is aligned with the line array, see panel (c).

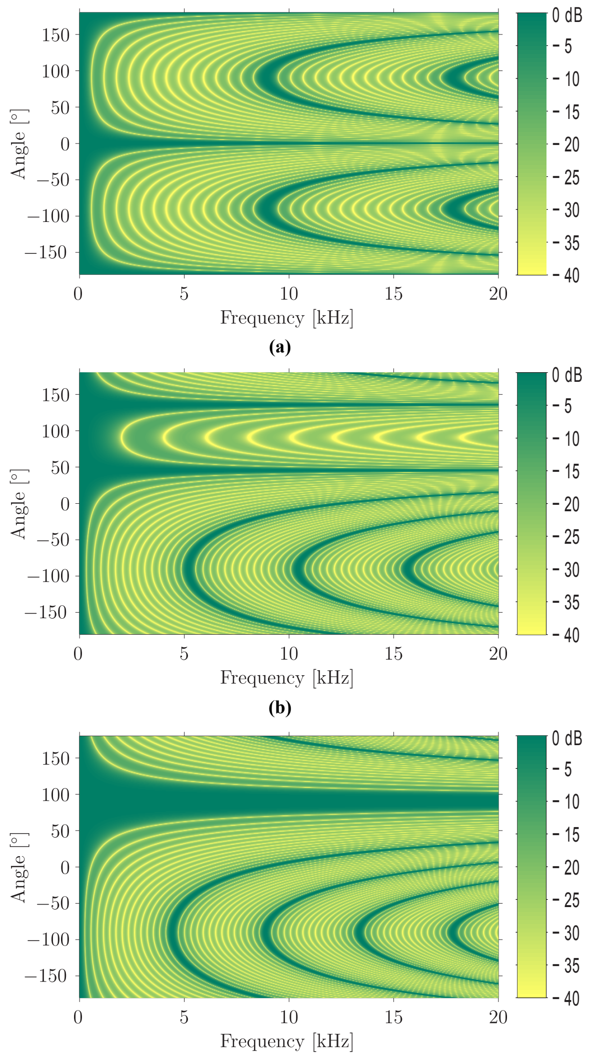


Figure 1: Beamforming ouput of a line array of 19 microphones as a function of frequency. Three source positions are simulated: (a) 0°, (b) 45° and (c) 90°. The spacing between the microphones is 3.75 cm.

3. Acousto-optic beamformer

Waves travel differently depending on the medium of propagation. For instance, electromagnetic waves generally travel slower in dense media and faster otherwise (vacuum corresponds to the extreme case). The phenomenon of sound in air inherently involves pressure changes that at the same time cause density fluctuations. Hence, sound waves can in principle influence the propagation of electromagnetic waves such as light. This is the so-called acousto-optic effect, a physical phenomenon that has been widely investigated with ultrasonic waves [2, 3, 4]. Recent studies show that the acousto-optic effect can also be measured within the audible frequency range using a laser Doppler vibrometer (LDV)[5, 6]. In this frequency range and with sound pressure levels below the threshold of pain, sound waves cannot diffract light, but only make it travel a bit faster (pressure decrease) or slower (pressure increase). All the information about the acousto-optic effect is thus contained in the phase of light, not its amplitude. In particular, it can be shown that the phase of a beam of light traveling through an acoustical space is [6]

$$\phi = k_0 n_0 L_0 + k_0 \frac{n_0 - 1}{\gamma p_0} \int_{\mathbf{L}} p(x,y,t) \mathrm{d}l, \quad (2)$$

where k_0 is the wavenumber of light in vacuum, n_0 and p_0 are the refractive index and the atmospheric pressure under static conditions, γ is the ratio of specific heats, $p(x, y, t)$ corresponds to the sound pressure along the line integral, $\mathbf{L}$ represents the propagation path, and L_0 is the total length.

With a proper measurement setup, carefully designed to minimize the mechanical vibrations of the structure where the LDV and the reflecting point are mounted, the apparent velocity captured by the LDV due to the interaction between sound and light is [6]

$$v(t) = \frac{n_0 - 1}{\gamma p_0 n_0} \frac{\mathrm{d}}{\mathrm{d}t} \left(\int_{\mathbf{L}} p(x, y, t) \mathrm{d}l \right). \tag{3}$$

Figure 2 sketches the measurement setup that makes it possible to measure this phenomenon. The most interesting feature of Equation (2) is that it contains the value of the continuous integral of the sound pressure along the path traveled by the light. Beamforming techniques base their calculations on a finite number of input signals, and this is indeed what causes aliasing. In particular, when no delays are applied to the sensor signals, for $\theta = 0°$ in Equation (1), the beamforming output is simply the summation of the measured signals. Theoretically, if an infinite number of sensors were placed along a finite line of length L_0, the following result could be achieved,

$$\lim_{M \to \infty} \sum_{m=0}^{M-1} \tilde{p}_m d = \int_0^{L_0} P(x, y, \omega) \mathrm{d}l, \tag{4}$$

where $P(x, y, \omega)$ is the temporal Fourier transform of $p(x, y, t)$. Such an ideal situation would yield an infinite spatial resolution ($d \to 0$) that would make any aliasing effects disappear. Even though the use of an infinite number of transducers is in practice not feasible, the continuous integral of the sound pressure can alternatively be calculated using the acousto-optic effect. Fourier transforming Equation (3) yields,

$$V(\omega) = j\omega \frac{n_0 - 1}{\gamma p_0 n_0} \left(\int_{\mathbf{L}} P(x, y, \omega) \mathrm{d}l \right), \tag{5}$$

and thus,

$$\int_{\mathbf{L}} P(x, y, \omega) \mathrm{d}l = \frac{\gamma p_0 n_0}{n_0 - 1} \frac{V(\omega)}{j\omega}. \tag{6}$$

This expression can be used to define a beamforming output that is immune to spatial aliasing:

$$\begin{aligned} b_{AO} &= \left| \frac{1}{L_0} \int_0^L P(x, y, \omega) \mathrm{d}l \right|^2 \\ &= \left| \frac{1}{L_0} \frac{\gamma p_0 n_0}{n_0 - 1} \frac{V(\omega)}{j\omega} \right|^2. \end{aligned} \tag{7}$$

Analytically, the continuous integral of plane waves traveling through a beam of light of length L_0 yields the following beamforming output:

$$\begin{aligned} b_{AO} &= \left| \frac{1}{L_0} \int_0^{L_0} \tilde{P}_0 e^{j(\omega t - \vec{k} \cdot \vec{r})} \mathrm{d}x \right|^2 \\ &= \left| \frac{\tilde{P}_0 e^{j\omega t}}{L_0} \int_0^{L_0} e^{jk \sin \tilde{\theta} x} \mathrm{d}x \right|^2 \\ &= \left| \tilde{P}_0 \mathrm{sinc} \left(k \sin \tilde{\theta} L_0 / 2 \right) \right|^2. \end{aligned} \tag{8}$$

Taking into account that $L_0 = (M - 1)d$, the similarity between Equations (1) and (8) when $\theta = 0$ is worth noting. The immunity to spatial aliasing of the defined acousto-optic beamformer is illustrated in the beamforming pattern depicted in Figure 3. Note that the acousto-optic beamformer b_{AO} is independent of θ, that is, it cannot steer the looking direction of the beamformer electronically as in conventional beamforming systems. Instead, one has to steer the laser beam manually towards those angular directions of interest.

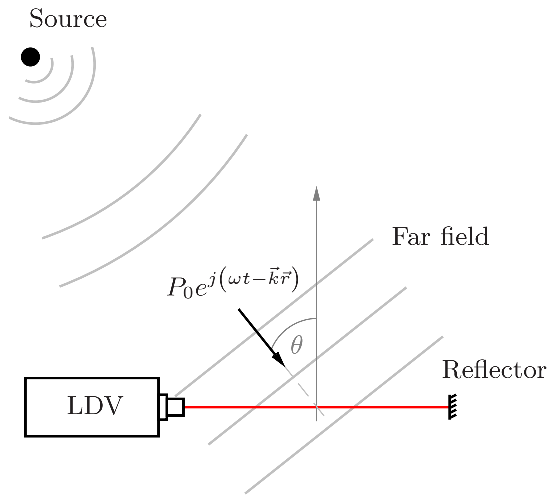


Figure 2: Sketch of the acousto-optic beamforming system.

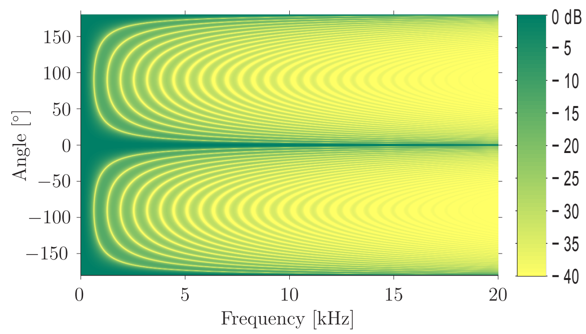


Figure 3: Beamforming output of the acousto-optic beamformer as a function of frequency ($L_0 = 54$ cm).

4. Results and discussion

The measurements were carried out in an anechoic room of about 1000 m^3. A loudspeaker radiating white noise was placed at a distance of 5 m from the beamforming system in order to fulfill the far field conditions described in the theory. A picture of the measurement setup is shown in Figure 4. Quite some efforts were spent on minimizing the mechanical vibrations of the structure holding the beamforming system. Moreover, an accelerometer (B&K Type 4344) was mounted on the reflecting point to monitor the residual mechanical vibrations that could bias the velocity measured with the LDV (a Polytec OFV-505). The output of the accelerometer showed that the measurement setup was fairly immune to sound generated by mechanical vibrations above 2 kHz. The setup was covered with absorbing material in order to reduce the possible effects of scattering from the equipment. The beamforming system was installed on a turntable allowing to steer the laser beam of the LDV towards different directions. Given the symmetry of the beamforming pattern, the acousto-optic beamformer was only rotated 180° instead of 360°. The results are presented in Figure 5. Despite the mechanical vibrations of the structure below 2 kHz, the measured acousto-optic beamforming pattern resembles very much the one presented in Figure 3. Although the background noise, some resonances of the structure and the scattering from the equipment do not allow to see the sidelobes as clearly as in the ideal simulation, the experimental results confirm that the beamforming output is definitely free of aliasing artifacts.

For comparison's sake, supplementary measurements were carried out with a line array of 19 microphones (B&K Type 4957) using conventional DSB.

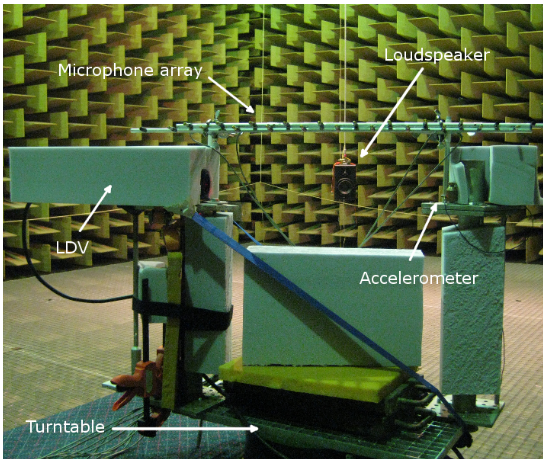


Figure 4: Measurement setup. The loudspeaker was placed 5 m far away from the beamforming systems ($L_0 = 54$ cm, $d = 3.75$ cm).

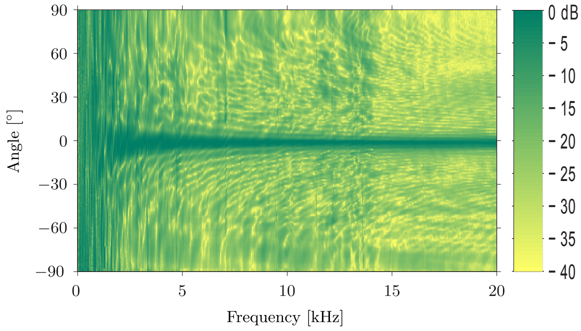


Figure 5: Beamforming pattern measured with the acousto-optic beamformer as a function of frequency ($L_0 = 54$ cm).

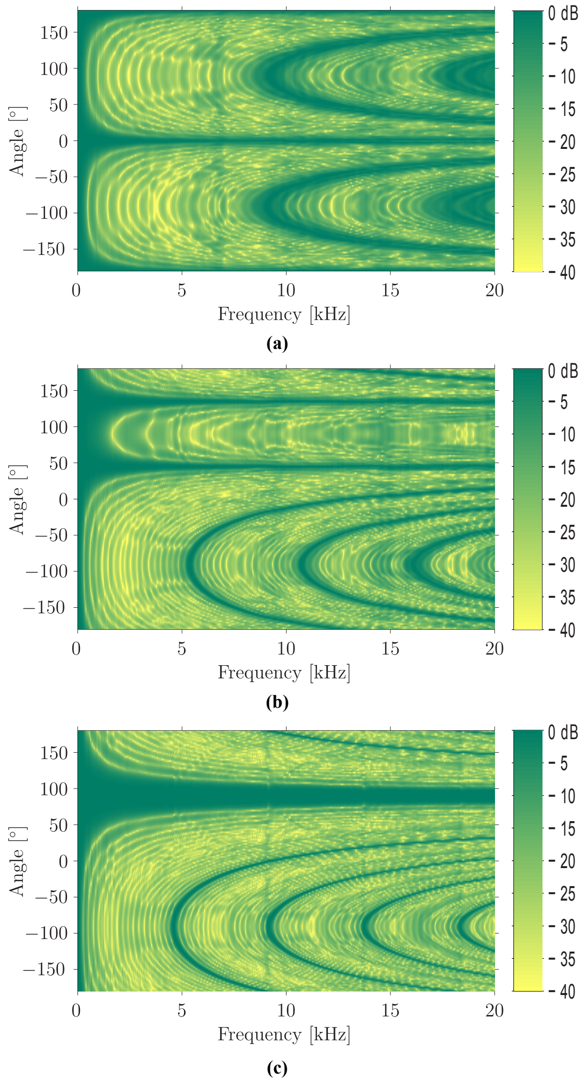
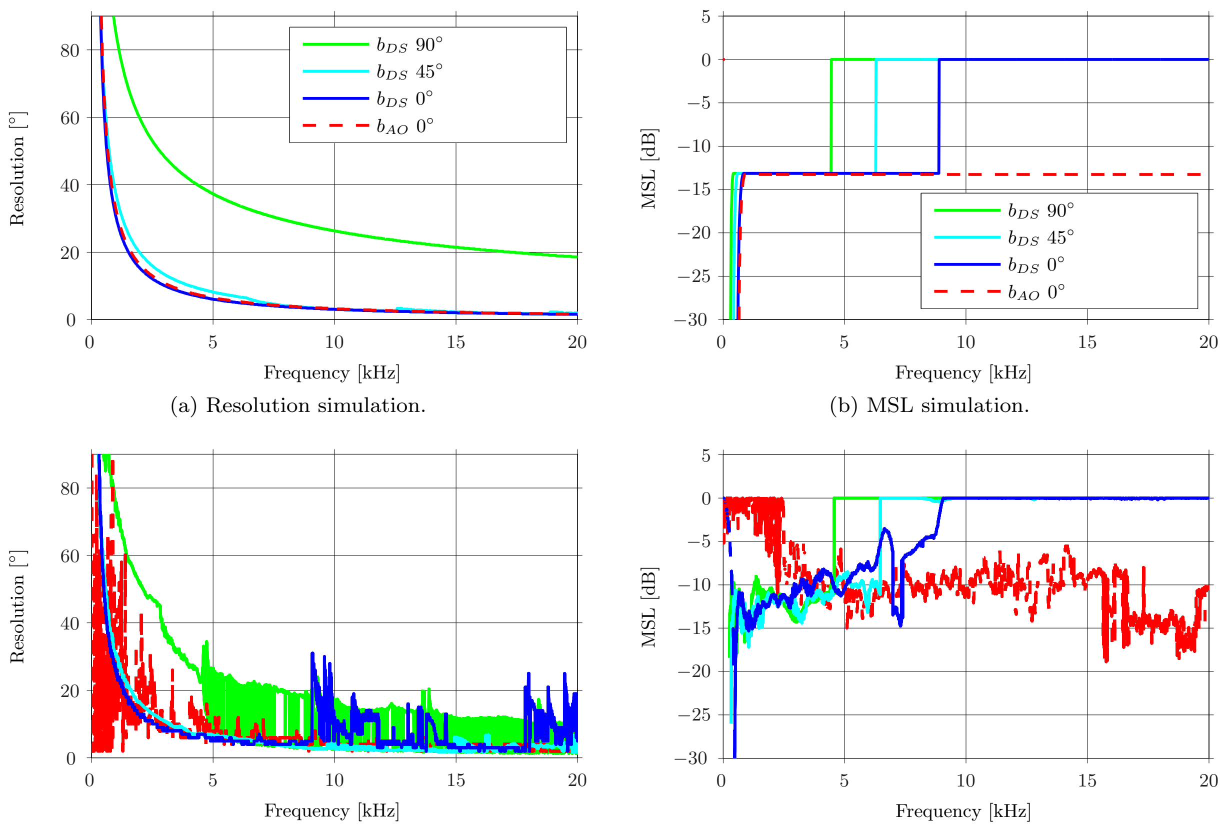


Figure 6: Beamforming patterns measured with a line array of 19 microphones as a function of frequency. Three source positions were measured: (a) 0°, (b) 45° and (c) 90°. The spacing between the microphones was 3.75 cm.



(a) Resolution simulation.

(b) MSL simulation.

(c) Resolution measurement.

(d) MSL measurement.

Figure 7: Resolution and MSL of the microphone array and the acousto-optic beamformer. Three different source positions were tested with the microphone array (0°, 45°, 90°) and only one with the acousto-optic beamformer (0°).

The results can be seen in Figure 6. Again, the beamforming outputs are subject to noise, vibrations of the structure and scattering effects, but in addition to these, there could be some misalignment between the microphones caused by the homemade array layout. Nevertheless, aliasing artifacts arise at approximately the same frequencies predicted in the simulations.

A good way to assess the performance of a beamforming system is to use the so-called resolution (normally defined at -3 dB of the mainlobe) and the maximum sidelobe level (MSL), that is, the level difference between the mainlobe and the most prominent secondary lobe. The former characterizes the minimum angle necessary to resolve two sources, whereas the latter indicates the possible influence of ghost sources. Figure 7 shows the resolution and the MSL of the line array and the acousto-optic beamformer analyzed in the simulations and the measurements. As can be seen, both resolution and MSL are quite noisy in the measurements. None of the possible sources of error (noise, vibrations, etc.) are included in the simulations. However, the main features observed in the simulations can also be identified in the measurements, that is, the acousto-optic beamformer has almost the same resolution as the line array when the source is located at $\tilde{\theta} = 0°$ and its MSL does not equal 0 dB at high frequencies due to its immunity to spatial aliasing. Instead, the MSL obtained with the line array clearly illustrates the limitations of using a finite number of microphones above the Nyquist frequency.

5. A note on nearfield beamforming

Sound sources can be detected with the proposed acousto-optic beamformer successfully provided that the waves impinging on the beamforming system have a planar wavefront. Roughly speaking, wavefronts parallel to the laser beam produce larger amplitudes of the LDV's signal than those whose symmetry do not fit the straight light of the laser beam, e.g. spherical waves. Hence, the localization of sound sources in the nearfield using the acousto-optic effect seems so far complicated. An example of the beamforming pattern that would be obtained in the nearfield is simulated in figure 8. As can be seen, the energy of the mainlobe seems to smear out around the angular position of the actual source ($\tilde{\theta} = 0°$) and the sidelobes also become more prominent (cf. Figure 3).

Alternatively, sound sources could in principle be localized in the nearfield using acousto-optic tomog-

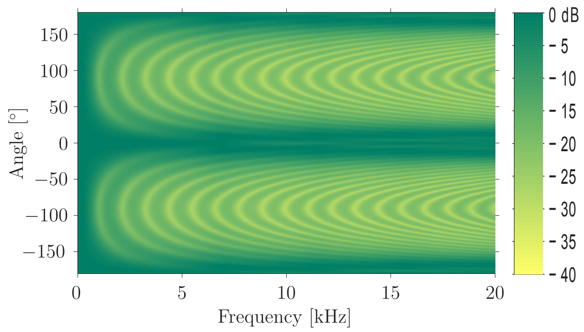


Figure 8: Acousto-optic beamforming output simulated in the nearfield. The source is located 0.4 m from the center of the laser beam ($L_0 = 30$ cm).

raphy [6]. In this measurement technique, the acousto-optic effect is used to reconstruct the acoustic pressure over a surface. The reconstructed sound field could be processed with a conventional nearfield beamforming technique. However, the output of the tomographic reconstruction provides the sound pressure distribution over a finite number of points, and thus, spatial aliasing plays again an important role.

6. Conclusions

The acousto-optic beamformer is a beamforming system immune to spatial aliasing. This makes it possible to extend the frequency range of analysis to all audible frequencies. The price to pay for such an ideal property is that the acousto-optic beamformer cannot steer the looking direction of the system electronically. One has to rotate the laser beam manually into the angular positions of interest. This could be implemented in practice in a similar way as the radar antennas used in air traffic control, that is, the acousto-optic beamformer could rotate at a constant speed, sweeping the acoustical space under investigation. In some cases where only one looking direction is of interest, e.g. in some surveillance applications, this drawback would not even be a problem.

The results also show that when comparing an acousto-optic beamformer to a conventional line array of microphones, both present approximately the same resolution and MSL figures when the source is located in front of the beamforming system (and below the Nyquist frequency). However, although the line array can steer electronically the looking direction towards the sides, both resolution and MSL worsens considerably as the source approaches a grazing angular position.

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