BEAMFORMING METHOD: SUPPRESSION OF SPATIAL ALIASING USING MOVING ARRAYS

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ABSTRACT

In the scientific and industrial community a growing interest is nowadays devoted to sound source localization techniques. Among all the existing methods, the use of the beamforming algorithm in sound source localization problems is increasing, thanks to its many advantages if compared to the other available methods (Nearfield Acoustic Holography and Helmholtz Equation Least Square Method). The beamforming limits are well known and deeply investigated in literature and the study of possible solutions to overcome these limitations is a key point. The present paper deals with the spatial aliasing problem and proposes an innovative approach to improve the method performances. The spatial aliasing phenomenon causes the appearance of “ghost sources” in the reconstructed images. An innovative solution based on the use of moving microphone arrays is here proposed. The array motion, multiplying the measurement points, allows to reduce the aliasing errors. This paper presents the theoretical background of the innovative method and the results obtained with numerical simulations, in comparison with the traditional “Delay & Sum” method. The influence of some parameters, such as the array speed, and the geometry are investigated and a first experimental validation is given.

1 INTRODUCTION

Different methods exist in literature and are practically employed to localize and separate sound sources. Among them the beamforming ([1], [2]), the Nearfield Acoustic Holography (NAH, [3], [4]) and the Helmholtz Equation Least Square Method (HELS, [5], [6]) can be considered, as all these techniques use microphone arrays in order to measure the acoustic pressure in a defined number of points. Each method then processes the measured data in order to reconstruct the acoustic field on the source surface, allowing the localization of the emitting areas. Beamforming has some specific aspects that make the method a powerful analysis tool. The main advantages, if compared to NAH and HELS, are: a less number of measurement points, the applicability both to static and moving sources, and the possibility of placing the array in the far field. On the other hand, beamforming presents two intrinsic drawbacks that limit its applications. The first one is the worse spatial resolution, i.e. the capability of separating two adjacent emitting sources, if compared to NAH and HELS, especially in the low frequency range. The second one is the so called spatial aliasing, that causes the appearance of “ghost sources” beside the actual ones in the reconstructed acoustic map (beampattern). This problem is strictly linked with the considered frequency range, and it is possible to define an upper limit for each array configuration. The present work is focused on this latter issue and proposes an innovative solution, based on the use of moving arrays, to reduce the identification errors.

2 ALIASING SUPPRESSION USING MOVING ARRAYS

As reported in literature ([7]) it is possible to define the critical frequency (f_{max}) of an equally spaced array as:

$$f_{\max} = \frac{c}{d} \tag{1}$$

where c is the sound speed and d is the distance between two following microphones. If the sound source frequency exceeds this critical value, ghost sources appear in the beampattern.

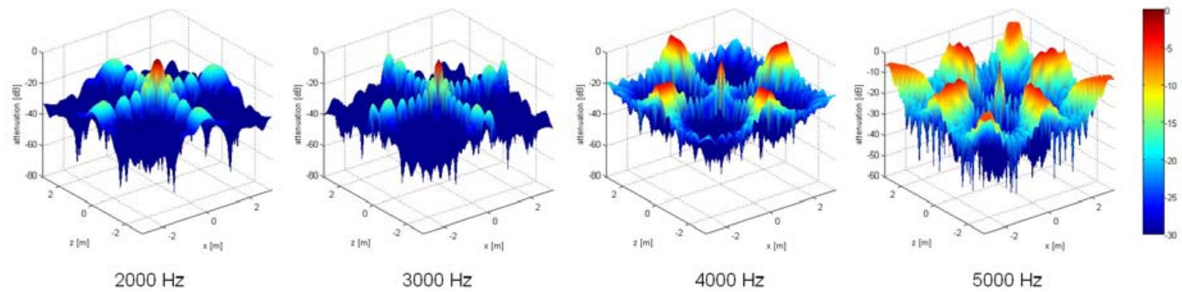


Fig. 1. Numerical beampattern for a 2000-5000 Hz sound source placed in x=0, y=0 m; spatial aliasing effects and “ghost sources” appearance.

The ghost sources, numerically generated by the beamforming algorithm, do not correspond to real emitting sources and cause therefore identification errors. The aliasing effects can be observed in Fig. 1: an array with critical frequency of about 3400 Hz is used to identify a single source placed in front of it, at growing frequencies, from 2000 Hz to 5000 Hz. When the sound source frequency is less than the critical value, a main lobe identify the real emitting source, while the typical side lobes decrease ([1], [2], [7]). Exceeding the critical frequency, instead, many ghost sources appear.

The standard solution to avoid aliasing errors is the reduction of the microphone spacing *d*. As the overall array dimensions are linked to the resolution capability, maintaining the array aperture and reducing the microphone spacing lead to a very high number of transducers, being practically unachievable due to the high involved costs. This paper suggests a new approach, in which the number of measurement points is increased in a fictitious way, moving the array during the measurement session. Each pressure value is therefore sampled at a different position. The type of array movement, e.g. translation or rotation, represents a fundamental point, changing the achievable performances. This paper deals with the analysis and the development of rotating planar arrays; all the microphones follow circular trajectories around a fixed revolution centre (Fig. 2).

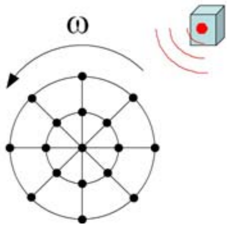
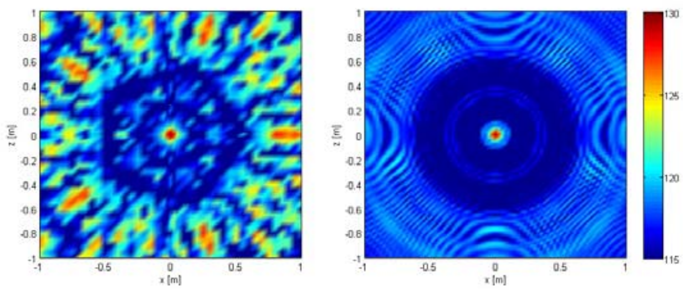


Fig. 2. Scheme of a planar rotating microphone array

The moving process allows to cancel the ghost sources, as shown in Fig. 3. A 8000 Hz sound source placed in front of an array with f_{max}≈2000 Hz is simulated. The beampattern obtained with the static measurement array is affected by the presence of many ghost sources (red areas), that disappear applying the moving approach.



(a) *(b)*

Fig. 3. Numerical beampattern for a 8000 Hz sound source placed in x=0, y=0 m; comparison between static (a) and moving array (b).

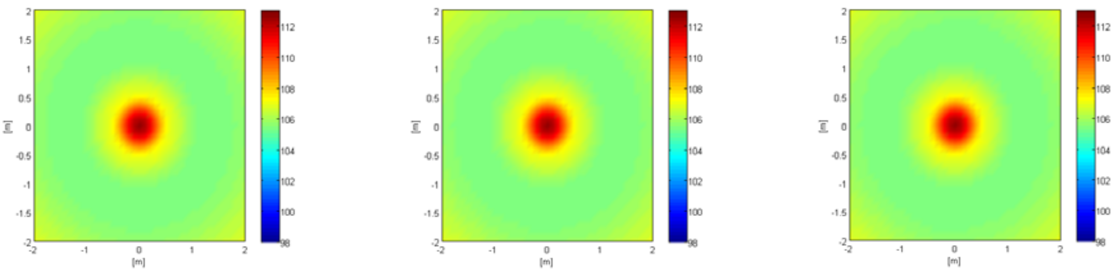
3 NUMERICAL SIMULATIONS

Since a formula that gives the method output for the moving approach is not available, the method performances have been investigated performing numerical simulations. Due to the kind of considered motion (rotating array) the influence of the following parameters has been considered: angular velocity, number of performed turns and array geometry.

The simulations have been performed using numerical models reproducing the sound propagation from the source toward the moving microphones. The sources have been modeled as monopoles, involving therefore a spherical propagation model, while the microphones have been considered as ideal transducers, i.e. not introducing amplitude or phase distortions. The array rotation has been simulated moving the measurement points at a constant angular velocity on circular trajectories. The general time domain *Delay&Sum* algorithm ([7]) has been applied to process data and to obtain the output beampattern.

3.1 Angular velocity

The angular velocity range is practically limited by the microphones itself. High speed in fact can produce non negligible air flow around the microphones, leading to an important background noise. Specific windscreens applied to the microphones allow to support air speed up to about 6 m/s ([8]). This means that, considering a common array with an outer diameter of about 2 meters, acceptable values for the angular velocity are lower than 6 rad/s. The effects of different possible speed values, 1.57 rad/s, 6.28 rad/s and 12.56 rad/s, have been investigated and the results are shown in Fig. 4. It is clear that speed variations in the considered range do not affect the method performances.



(a) *(b)* *(c)*

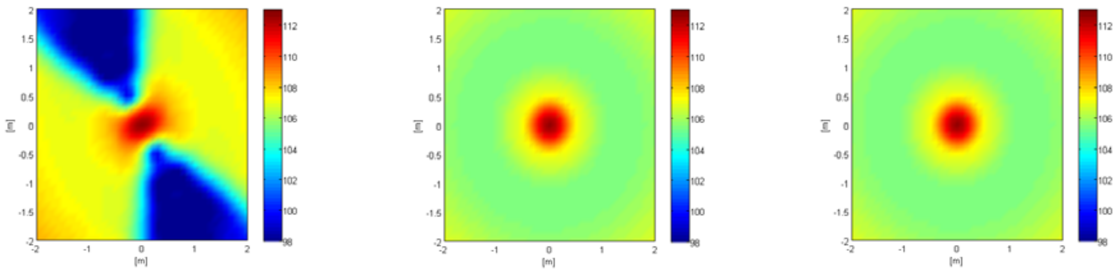
Fig. 4. Numerical beampattern for a 1000 Hz sound source placed in x=0, y=0 m; comparison between 1.57 rad/s (a), 6.28 rad/s (b) and 12.56 rad/s (c) angular velocity.

3.2 Number of turns

Two different situations can be identified for the rotation: partial and complete revolution. The effects on the beampattern have been investigated performing simulations with a 20 microphone linear array, rotating around its centre, in three different conditions: 1/3 of turn, 1 full turn and 5 full turns (Fig. 5).

Fig. 5-a shows the partial revolution case; only a third of turn is completed, thus achieving a measurement surface not uniformly distributed. The resulting beampattern shows an asymmetric main lobe in correspondence of the sound source position, and side lobes at two different suppression levels: a -5 dB zone corresponding to the space covered during the microphones motion, and a -15 dB zone elsewhere. This asymmetry in the side lobe levels disappears performing a complete rotation (Fig. 5-b). In this case the beampattern is uniformly shaped and the side lobe suppression level is about -7 dB. Even if this value is not much higher than the -5 dB obtained in the previous case, it is a great asset in case of multiple emitting sound sources, when the final beampattern comes from the sum of multiple contributions. Fig. 5-c shows that there are no differences performing 5 complete turns

instead of only 1 turn; no improvement are obtained covering the same measurement points more than once.



(a) *(b)* *(c)*

Fig. 5. Numerical beampattern for a 1000 Hz sound source placed in x=0, y=0 m; comparison between 1/3 of turn (a), 1 full turn (b) and 5 full turns (c) revolution angle.

3.3 Array geometry

Five different array geometries, named in the following type A-B-C-D-E (Fig. 6), have been proposed and tested. In order to perform a direct comparison of the performances in term of side lobes suppression, all the configurations are composed by the same number of transducers, i.e. 20, and used in the same test conditions, i.e. with two emitting sound source (a 1000 Hz source placed in 0,0 m and a 2000 Hz source placed in 1,0 m). The array configurations are:

A. linear array, 8 cm spacing, rotating around its centre;
B. linear array, 8 cm spacing, rotating around its extreme point;
C. pentagonal array ([9]), rotating around its centre;
D. geometric spiral array, rotating around its centre;
E. arithmetic spiral array, rotating around its centre.

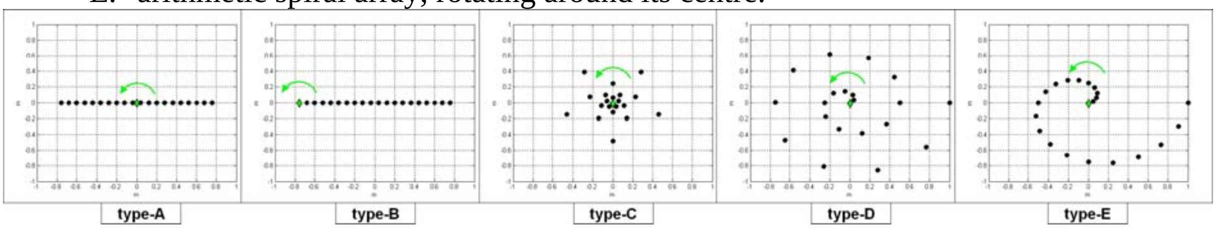
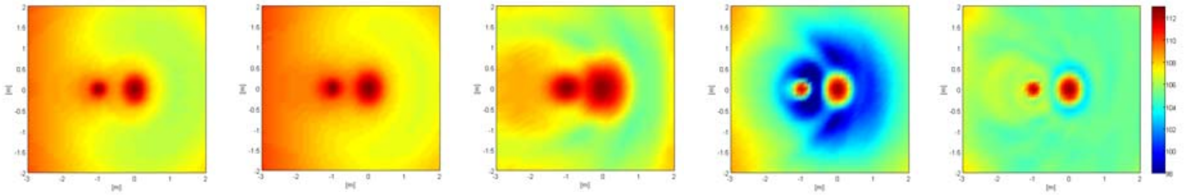


Fig. 6. Microphone array configurations.

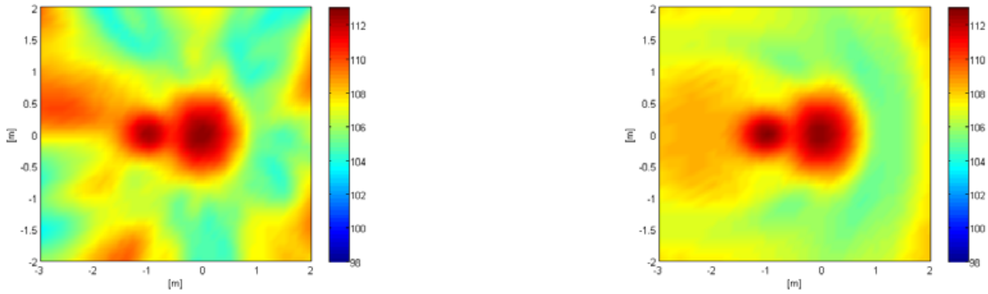
The obtained results are shown in Fig. 7. Type A array (Fig. 7-A) shows a generally good behavior, evidencing a good side lobe suppression beside a good resolution capability. Type B array (Fig. 7-B), based on the same static geometry but with a shifted rotation centre, shows worse results. Even if a better spatial resolution could be expected because of the greater array aperture ([1]), actually the effect is the opposite: both side lobe attenuation and spatial resolution are worse. This is due to the source position that is not centered with respect to the array during the motion, causing a strong decrease in the resolution capability (as stated by the Rayleigh formula, [2]). Type C array (Fig. 7-C), has better performances in term of side lobe suppression, due to its well balanced planar configuration. On the other hand it has a worse spatial resolution due to the narrow aperture. Better performances can be achieved placing microphones along spiral distributions, thus optimizing the static configuration. Both spiral geometries, geometric step spiral (Fig. 7-D) and arithmetic step spiral (Fig. 7-E), achieve better results in terms of resolution and side lobe suppression.



(A) (B) (C) (D) (E)

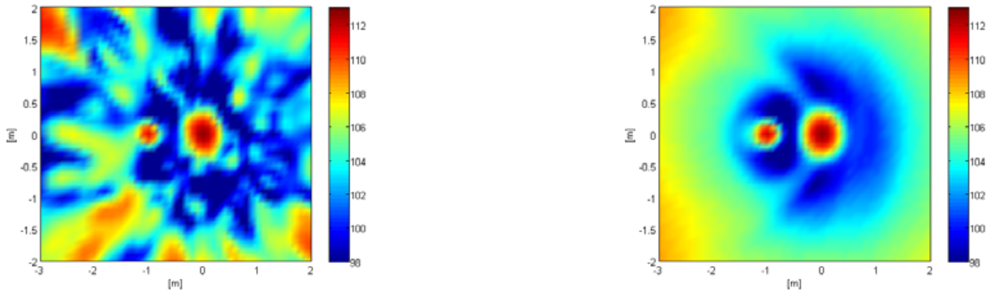
Fig. 7. Numerical beampattern for two close sound sources; comparison between array configurations (Fig. 6).

The improvements related to use of the proposed rotating approach are well visible comparing the static and the moving case. The comparison is shown in Fig. 8 for the pentagonal geometry (type C), and in Fig. 9 for the spiral geometry (type D).



(a) *(b)*

Fig. 8: Numerical beampattern for two close sound sources; comparison between static (a) and rotating (b) application for the type C array.



(a) *(b)*

Fig. 9. Numerical beampattern for two close sound sources; comparison between static (a) and rotating (b) application for the type D array.

It is to be highlighted that both these configurations, if used in traditional static way, are affected by aliasing errors at 2000 Hz. The array rotation allows to reduce and even avoid the problems.

4 EXPERIMENTAL VALIDATION

A first experimental validation of the proposed method has been achieved creating a rotating array and comparing the experimental to the numerical results. To this aim a 20 microphones pentagonal array has been designed and assembled (Fig. 10).

Fig. 10: View of the rotating pentagonal array (type C) and the measurement set-up

Some experimental tests have been carried out in an external environment (Fig. 10), using a loudspeaker placed in front of the array to reproduce the emitting sound source. Several tests have been performed in order to investigate the system performances in different conditions, varying the frequency in the range between 1000 Hz and 8000 Hz. Fig. 11 reports the comparison between the simulation (a) and the experimental (b) test in the case of rotating approach, showing a very good agreement, for a 4000 Hz emitting sound source. Fig. 11-c instead shows the experimental beampattern in the traditional static case, affected by some ghost sources, proving the actual improvements of the proposed approach.

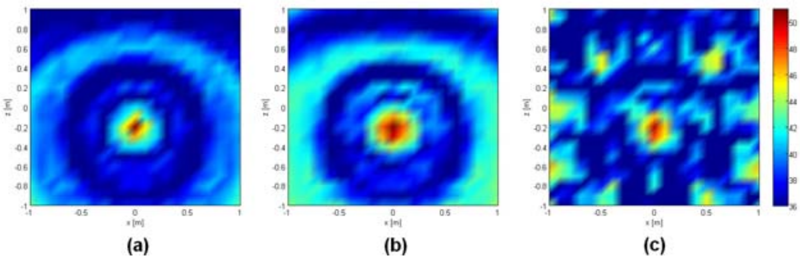


Fig. 11. Beampattern for a 4000 Hz sound source placed in x=0, y=0 m; comparison between numerical rotating (a), experimental rotating (b) and experimental static (c) case.

5 CONCLUSIONS

The present work proposed an innovative approach based on the use of moving arrays in order to reduce and even avoid the aliasing errors that affect the beamforming technique. The problem has been numerically investigated in order to analyze the parameters influence and the method output performances. A real measurement system, based on a 20 microphone rotating array, has been designed and built in order to validate the obtained numerical results. The work has proved the suggested approach capability in the aliasing errors reduction, allowing to increase the beamforming applications.

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