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An Introduction to Virtual Phased Arrays for Beamforming Applications

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Sound localization problems are usually tackled by the acquisition of data from phased microphone arrays and the application of acoustic holography or beamforming algorithms. However, the number of sensors required to achieve reliable results is often prohibitive, particularly if the frequency range of interest is wide. It is shown that the number of sensors required can be reduced dramatically providing the sound field is time stationary. The use of scanning techniques such as "Scan & Paint" allows for the gathering of data across a sound field in a fast and efficient way, using a single sensor and webcam only. It is also possible to characterize the relative phase field by including an additional static microphone during the acquisition process. This paper presents the theoretical and experimental basis of the proposed method to localise sound sources using only one fixed microphone and one moving acoustic sensor. The accuracy and resolution of the method have been proven to be comparable to large microphone arrays, thus constituting the so called "virtual phased arrays".

Keywords: beamforming, source localization, virtual phased arrays, measurement techniques.

1. Introduction

There are many applications which require the utilisation of microphone arrays in order to localise sound sources across a space. Traditionally, this implies investing a large amount of money into an acquisition system. Furthermore, the resolution of the measurements would depend upon the number of transducers used and their respective positions (the geometry of the array). If the array consists of too many sensors, it becomes acoustically significant, biasing the characterization of the sound field.

A "Virtual Phased Array" approach can be taken to avoid many practical constraints of conventional beamforming devices, assuming the sound field is time stationary. The proposed technique enables the characterization of a measurement area as a set of "virtual transducers" with a rather simple measurement system. A single moving sensor is utilised to continuously acquire data across the space whilst a static reference microphone also records the event. The acoustic signal is later split into blocks which have associated different spatial positions. Each block, or segment of the recorded signal, represents an element of the virtual phased array. The phase estimation is computed relative to the fixed reference sensor. This measurement method can potentially address many common problems due to its low cost and straightforward acquisition process. The use of a single moving sensor avoids array calibration issues and any other limitations derived from using fixed array geometry.

The idea of creating synthetic or virtual arrays using a limited number of sensors had also been explored in other disciplines, most being focused upon enhancing the possibilities of conventional radars (Synthetic Aperture Radar or SAR) (Wiley, 1985; Curlander, McDonough, 1991) and sonar systems (Synthetic Aperture Sonar or SAS) (Cutrona 1975; 1977). The majority of methods developed for SAS and SAR share common ground which differs from the proposed noise localization technique; they are active systems based on the coherent addition over many pings across the space (Hayes, Gough, 2009), whereas the presented approach is based upon passive synchronization via a fixed reference sensor.

In previous works, virtual phased arrays have been shown to work remarkably well in laboratory conditions for mid-high frequencies with multiple beamforming algorithms (Fernandez Comesaña *et al.*, 2011). Furthermore, the performance of virtual phased arrays has been tested successfully at lower frequencies in the surroundings of a gas plant (Fernandez Comesaña *et al.*, 2012). In addition, several deconvolution methods have been adapted and tested for virtual phased arrays achieving spatial resolution, dynamic range and accuracy improvements (Fernandez Comesaña *et al.*, 2013a). However, the foundations of the measurement technique have yet to be studied in detail. This article presents the theoretical basis of virtual phased array technology along with some simulations and an experimental validation of the measurement method applied to a sound source localization problem in outdoor conditions.

2. Theory

This section is divided into three main parts: first of all the equations which describe the sound field received by a moving transducer are presented; next, the method used to preserve the relative phase information of the sound field is introduced; lastly, a series of beamforming algorithms are adapted for virtual phased arrays.

2.1. Acoustic signal received by a moving sensor

The equations which describe the behaviour of a linear sound field are well known and utilised in multiple fields of acoustics. Nonetheless, non-linearities appear in an acoustic signal recorded with a moving transducer due to Doppler effects. This section not only evaluates the impact of the Doppler shift but also presents a definition of the sound pressure and particle velocity data acquired accounting for the arbitrary movement of the measuring probe. The derivation presented below is based upon the solution of an analogue problem in electromagnetism (Hoop, 2009) which has recently been adapted for acoustic moving sources (Camier *et al.*, 2012). The current work evaluates a reciprocal problem to the Camier study, since in our case the sound source remains static while the receiving microphone is moving.

Let us begin by defining the excitation of the system with a punctual sound source $Q(\mathbf{x}_0, t)$ located in an arbitrary position $\mathbf{x}_0 = \{x_0, y_0, z_0\}$ at which the temporal behaviour varies according to a function $q(t)$. Hence,

$$Q(\mathbf{x}_0, t) = q(t)\delta(\mathbf{x}_0). \tag{1}$$

In order to study the sound field produced by $Q(\mathbf{x}_0, t)$ is crucial to define the velocity potential Ψ associated with it. This allows for the derivation of the sound pressure and particle velocity at any point in the evaluated environment. The wave equation for free field conditions in the presence of a punctual source is defined as

$$\Delta\Psi - \frac{1}{c^2}\frac{\partial^2\Psi}{\partial t^2} = Q(\mathbf{x}_0, t), \tag{2}$$

where $\Delta\Psi$ indicates the laplacian of the velocity potential and c is the speed of sound in air. In order to introduce position changes of source or receiver in the analytical model, it is necessary to define the associated Green functions and convolve them with the excitation produced by $Q(\mathbf{x}_0, t)$. Following the derivation proposed by Camier (2012), it is possible to define the velocity potential as a function of the excitation source q, the distance between source and receiver r, and also the speed of the moving sensor $\mathbf{V}$, evaluated in a time instant which depends upon the propagation time T from the sound source to the receiver, following trajectory $\mathbf{x}$

$$\Psi(\mathbf{x}, t) = \frac{q\left(t - \dfrac{r(t-T)}{c}\right)}{a}, \tag{3}$$

where

$$a = 4\pi\left(r(t-T) - \mathbf{V}\left(t - \frac{r(t-T)}{c}\right)\cdot(\mathbf{x}(t-T) - \mathbf{x_0})\right).$$

Finally, the sound pressure p and the particle velocity vector $\mathbf{u}$ are obtained by temporally and spatially deriving the velocity potential, hence

$$p = -\rho\frac{\partial\Psi}{\partial t}, \tag{4}$$

$$\mathbf{u} = \nabla\Psi. \tag{5}$$

In conclusion, the sound pressure and particle velocity acquired by a moving sensor have been defined. In later sections, the implementation of the above analytical expressions will allow for the evaluation of how the measurement conditions impact upon the spectral estimation.

2.2. Phase acquisition

Absolute phase synchronization of data acquired at different time intervals it is only possible when dealing with strictly deterministic signals and therefore it is unsuitable for most practical cases (Percival, Walden, 1993). Consequently, it is common practice to acquire phase information simultaneously at multiple positions using sensor arrays to maintain time synchronism. Nevertheless, if the sound field can be assumed time stationary, relative phase variations can be characterized at different time instances, allowing the use of scanning techniques to also assess the phase spatial distribution. The relative phase differences between any pair of points of the sound field can be obtained by calculating their cross-spectrum, because of the time independent nature of the resulting expression. So, setting the fixed reference sensor to a position m, the cross-spectra with any other measurement point n can be defined as (Shin, Hammond, 2008)

$$S_{p_m p_n}(\omega) = \lim_{T\to\infty} \frac{E\{P_m^* P_n\}}{T}, \tag{6}$$

where P_m^* denotes the conjugate spectrum of the measurement point m and P_n the spectrum of the moving sensor when passing at n. Evaluating Eq. (6) for a simple case (a pulsating sphere of amplitude A in free field conditions), it can be seen that the cross-spectrum does not have a time dependency, it only changes proportionally to the evaluated frequency ω and the difference in distance between source to moving and source to fixed sensors $(r_m - r_n)$

$$S_{p_m p_n}(\omega) = \frac{A^2}{r_m r_n}\, e^{jk(r_m - r_n)}. \tag{7}$$

The computation of cross-spectra between two sensors allows for the study of phase spatial variations across a sound field if the two signals are linearly related. The degree of linear relationship can be measured by assessing the coherence between them. Therefore, it is possible to quantify the quality of each signal block and thereby select the representative parts of the data via coherence estimates. An extended discussion about this matter can be found in a previous work presented by the author (Fernandez Comesaña *et al.*, 2013b).

2.3. Beamforming

One common application for sensor arrays is to determine the direction of arrival (DOA) of propagating wavefronts. An array receives spatially propagating signals and processes them to estimate their direction of arrival, acting as a spatially discriminating filter (Manolakis *et al.*, 2005). This spatial filtering operation is known as beamforming. Conventional *sum-and delay-beamforming* steers a beam to a particular direction by computing a properly weighted sum of the individual sensor signals. As such, this procedure results in the addition of signals coming from the direction of focus which maximizes the energy in the beamformer output whilst signals from other directions will be attenuated.

The asynchronous time acquisition performed with virtual phased arrays implicitly constraints the range of applicable localization techniques to frequency domain beamforming methods. Setting the origin of coordinates to the reference sensor position, a compatible sum-and-delay algorithm can be defined as

$$B(\mathbf{l},\omega) = \frac{1}{N}\sum_{n=1}^{N} w_n S_{p_m p_n}(\omega) e^{j\varphi(\mathbf{l},\omega)}, \tag{8}$$

where N is the total number of virtual transducers covered by the moving sensor, w_n is a weighting factor applied to each cross-spectrum and $\varphi(\mathbf{l},\omega)$ is a phase term which allows for the beamformer to be focused towards a certain direction $\mathbf{l}$. If the distance to the source r is known beforehand, it is possible to express $\varphi(\mathbf{l},\omega)$ as a function of the wavelength k and the separation differences between source and sensors $(r_n - r_m)$. In addition, an attenuation term can be added to quantify the sound pressure emitted

$$B_{NF}(\omega) = \frac{1}{N}\sum_{n=1}^{N} w_n S_{p_m p_n}(\omega) r_m r_n e^{-jk(r_m - r_n)}. \tag{9}$$

On the other hand, if the distance to the source is unknown, then it is not possible to provide information about the sound pressure at the source. However, it is feasible to estimate the location of the noise source for far field conditions by defining $\varphi(\mathbf{l},\omega)$ as the scalar product between the moving sensor position $\mathbf{x}$ and a unitary vector $\zeta^{\mathbf{l}}$, oriented in the direction $\mathbf{l}$ (Johnson, Dudgeon, 1993)

$$B_{FF}(\omega) = \frac{1}{N}\sum_{n=1}^{N} w_n S_{p_m p_n}(\omega) e^{-jk(\zeta^{\mathbf{l}}\cdot\mathbf{x})}. \tag{10}$$

3. Simulations

The implementation of the fundamental principles introduced so far allow us to assess viability and accuracy of virtual phased arrays from a theoretical point of view. This section is focused upon the comparison of conventional multichannel measurements with virtual phased arrays via simulations.

Three pulsating monopole sources have been modelled in free field conditions with a sinusoidal excitation $q(t) = \omega^{-1}\sin(\omega t)$, for frequencies of 200, 400 and 800 Hz. Letting the fixed reference sensor be the origin of coordinates, the sources were positioned at 0, 20 and 40 degrees of azimuth. A measurement area of 2 meters wide was evaluated using a sensor moving at 0.1 m/s and a phased array of 40 equally spaced elements. The distance between the source and measurement areas was set to 10 meters. Figure 1 illustrates the geometry of the simulation undertaken.

The acoustic signal received by each microphone was computed by applying the superposition principle, adding individual source contributions. On the one hand, the sum-and-delay beamforming algorithm has been directly calculated using the data from the static microphone array; on the other, the signal acquired with the moving sensor has been divided into 40 blocks of 0.5 seconds length, associating each of those blocks to a position in the static array. Subsequently, it was

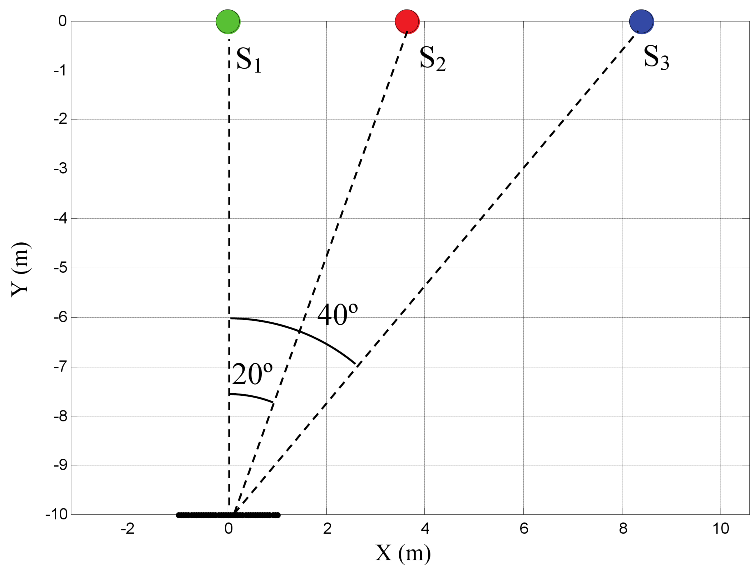


Fig. 1. Schematic view of the simulation environment.

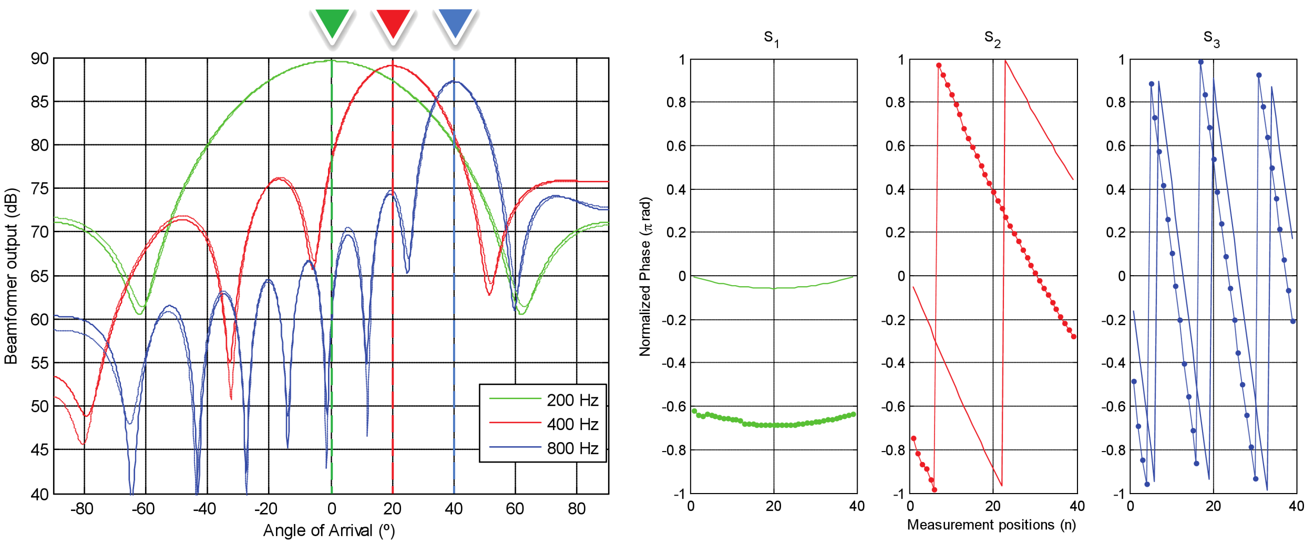


Fig. 2. Comparison of the beamformer output (left) and phase estimation (right) of data acquired with a static microphone array (solid line) and a virtual phased array (dotted line) for several frequencies. The triangles above the left figure indicate the real position of the sound sources.

possible to apply the beamforming algorithm to each of the blocks representing the elements of the virtual phased array. Figure 2 presents the obtained results in terms of beamforming output and phase estimation.

It can be seen in Fig. 2 that the simulation shows almost identical results either using data from either a static array or a virtual phased array. Moreover, the maxima of the beamforming output perfectly match in both cases with the theoretical location of the individual sources (0°, 20° and 40°). Furthermore, regarding the phase estimations (right hand side of Fig. 2), the relative phase (dotted line) follows the same pattern as the absolute phase, independent of frequency or source position. There is however a constant offset between the relative and absolute phase caused by the propagation delay from the source to the reference sensor. Nevertheless, it does not affect the beamforming results since this difference remains constant during the scanning process.

4. Measurement methodology: Scan & Paint

The measurement procedure to acquire the data is based upon the scanning technique "Scan & Paint" (Fernández Comesaña *et al.*, 2013c). The acoustic signals of the sound field are acquired by manually moving a probe or sensor across a measurement plane whilst filming the event with a camera. In the post-processing stage, the sensor position is extracted by applying automatic colour detection to each frame of

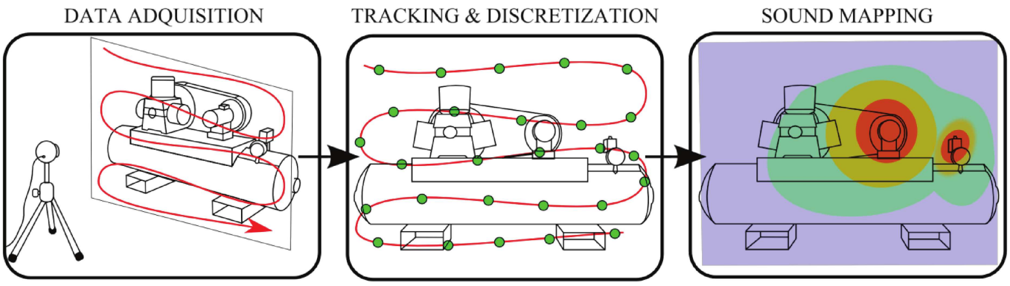


Fig. 3. Illustration of the basic steps undertaken with the Scan & Paint measurement method.

the video. The recorded signals are then split into multiple segments using a spatial discretization algorithm, assigning a spatial position depending on the tracking information. Therefore, each fragment of the signal will be linked to a discrete location of the measurement plane. Next, spectral variations across the space are computed by analysing the signal segments. The results are finally combined with a background picture of the measured environment to obtain a visual representation which allows us to "see" the sound. Figure 3 presents a sketch of the measurement methodology.

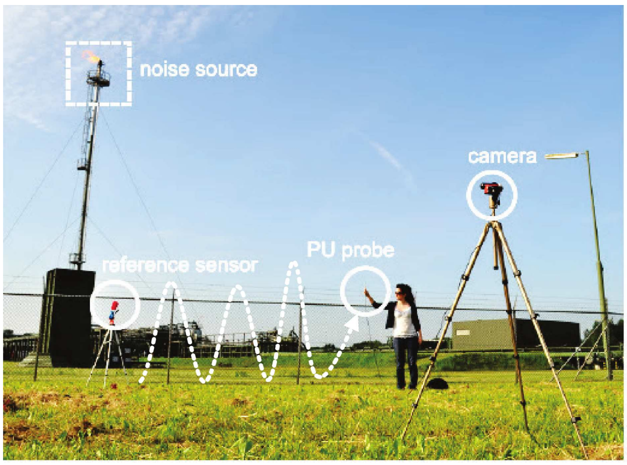
Only the 2D location relative to the background image is computed at this point so it is later necessary to define the relationship between 2D coordinates and 3D absolute coordinates, establishing a link between pixels and meters at the measured plane. Additionally, a fixed reference sensor shall be used to preserve the relative phase information across the sound field via cross-spectral estimates, as it has been introduced in the Subsec. 2.2.

5. Practical implementation

Since the simulation results support the potential of the proposed method, a practical measurement procedure for undertaking tests with virtual phased arrays has been developed. For that purpose the scanning measurement technique, Scan & Paint, has been integrated with the source localization algorithm presented above. This section explains in detail the measurement methodology, the validation procedure and the achieved results.

5.1. Instrumentation and measurement scenario

All measurements were carried out using a Microflown P-U probe which contains a pressure microphone together with a particle velocity sensor. Furthermore, a GRAS free-field microphone was used to measure the reference pressure at a fixed position. Windscreens, for reference and moving sensors, were used to avoid wind noise during the outdoor tests. In addition, a Logitech Webcam Pro 9000 camera was utilised to film the measurements. Sweep measurements were performed along a total surface of 6 meters horizontally and 2 meters vertically. The measurement time taken for the presented sweep was about 4 minutes. A grid of 0.25 meters was chosen to create an array of 85 virtual transducers. A picture of the experimental set-up can be seen on the left hand side of Fig. 1. Furthermore, a satellite picture of the measurement location is presented on the right hand side of the same figure.



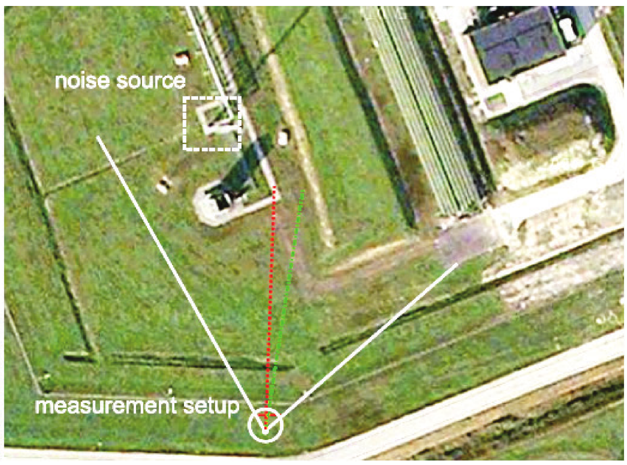


Fig. 4. Experimental set-up: performing a test measurement (left) and satellite picture of the measurement location (right). Green and red dotted lines indicate the camera's central axis and a normal axis to the measurement plane, respectively.

As shown on the right hand side of Fig. 3 there was a misalignment between the centre of the background picture (green dotted line) and the normal axis of the measurement virtual plane (red dotted line). Nonetheless, human errors were corrected during the post processing stage. The two axes were estimated by evaluating pictures of the set-up and satellite images. The measurement plane was situated parallel to the surrounding fence, whereas the central camera axis was estimated directly from the pictures taken during the measurements. The total time required for undertaking the experiment, including the set-up and acquisition stages, was about 15 minutes, therefore providing a fast source localization solution.

5.2. *Experimental results*

The localisation maps measured in the surrounding facilities of a gas plant are shown in this section. In the case studied, a flare stack was identified as the a priori dominant noise source in the area. Figure 4 presents several source localization maps for different frequency regions. A minor correction was applied (5 degree offset) in order solve the misalignment between the camera axis and the normal axis measurement plane shown in Fig. 1. It can be seen that localization in the azimuth axis gives good estimates even at 100 Hz. In contrast, the elevation of the noise source was not as accurate at low frequencies, mainly due to the limitations of the height of the virtual array (2 meters). As it has been mentioned above, the number of transducers and the total effective length of the array are asymmetrical, leading to better results for azimuth than for elevation estimations. Thus, a larger measurement area along the vertical axis should be covered in order to improve the accuracy at lower frequencies.

6. Multichannel solutions *versus* virtual phased arrays

One of the main problems of most conventional beamforming arrays is the cost of the measurement equipment. Not only the large number of transducers but also the multichannel acquisition systems required raise the costs remarkably.

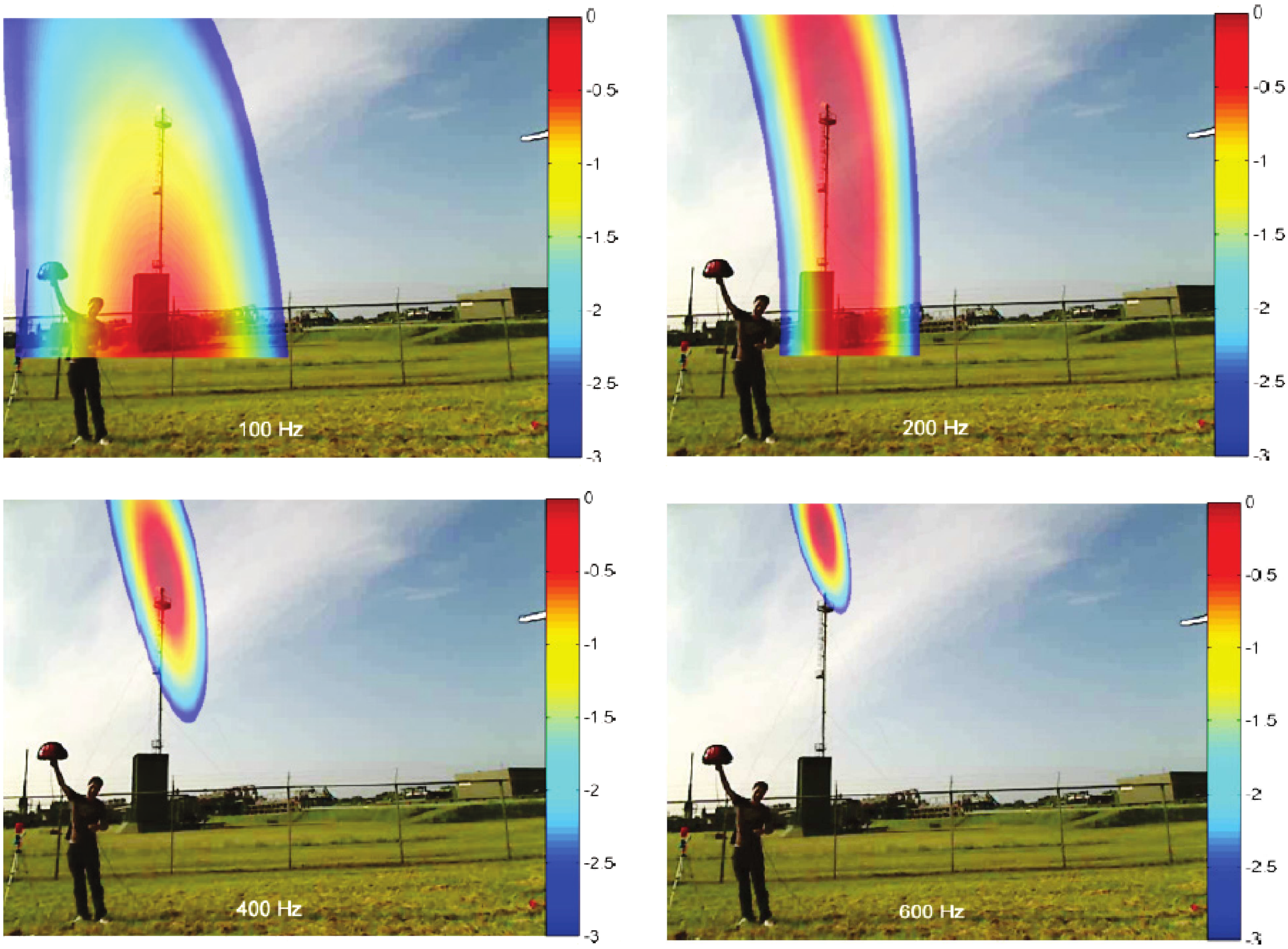


Fig. 5. Beamforming localization maps at 100 Hz (top left), 200 Hz (top right), 400 Hz (bottom left) and 600 Hz (bottom right) [dB].

The amount of time required to set up the instrumentation and perform the measurement is always an important issue. Manual sweeps of a single probe are a fast procedure to obtain information across a sound field in a direct way. The measurement presented in this paper was undertaken in less than 15 minutes, which can be seen as a reasonable amount of time for obtaining a solution to a noise localization problem at 100 Hz. There are several commercial solutions which are also portable and easy to set-up; however, their frequency range is very limited, especially in the low frequency region due to their reduced dimensions.

The flexibility of virtual phased arrays is one of its prime advantages *versus* multichannel solutions. The proposed method enables scanning of very small areas, for high frequency assessments, or large spaces, for localizing low frequency sources. In contrast, multichannel arrays have the transducers distributed along a fixed structure which is usually difficult to modify, making it unfeasible to optimize the array geometry for a frequency range of interest.

The absence of physical fixed positions leads to the minimization of discretization errors and spatial aliasing. The transducer spacing and the array size are fixed parameters implicitly associated with the resolution of the beamforming algorithm. Therefore, the resizable measurement grid of virtual phased arrays also provides an important advantage in the data post-processing stage to adapt the effective frequency range of the localization maps.

The assessment of sound fields created by partially-correlated sound sources might decrease the quality of the phase estimates due to coherence drops between the signals of the fixed and moving transducer. The investigation of this concept is out of the scope of this paper, but it should be further investigated to clarify the limitations that it can potentially impose.

7. Conclusions

A "Virtual Phased Array" approach has been successfully validated as a novel broadband source localization technique for assessing environmental noise problems under stationary conditions from both a theoretical and practical point of view. The insignificant impact of the Doppler effect in the data yields to identical results for simulation undertaken either for virtual phased array or conventional multichannel phased array. This fact clearly supports the robustness of the theoretical basis for the measurement technique. An experimental validation test had also been undertaken successfully. It is important to highlight that good results are obtained even at lower frequencies, which commercial multichannel solutions are not able to assess due to size limitations of the arrays.

It has been shown that low scanning speeds do not have a significant influence on the accuracy of the localisation maps. However, further investigation is needed to determine the limitations of the method imposed by the characteristics of the sound field, i.e. the presence of multiple partially-correlated sound sources.

Assessing a time stationary sound field, the measurement technique introduced reduces the number of transducers, measurement time and cost of conventional microphone arrays. Moreover, the remarkable flexibility of "virtual arrays" makes them a powerful tool for assessing broadband noise localization problems.

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