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Robust microphone array beamforming for long-term monitoring of industrial areas

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Abstract

Noise generated by industrial sources is often a cause of annoyance for people living in the surroundings of industrial areas, even when all individual plants adhere to regulations and possess all mandatory environmental permits. The variety of industrial noise sources is wide and the noise generated can be complex, making noise monitoring of large industrial areas not trivial. Low-frequency noise sources in particular are hard to localize using conventional sound measurement equipment, and may give rise to noise annoyance complaints at great distance from the source. This paper reports on the development of a 50m-wide robust microphone array system designed for long-term monitoring of industrial areas. The microphone array system consists of a set of curvilinearly and non-equidistantly spaced microphones, allowing to cover a broad spectral range and to minimize front-back confusions. The audio signals recorded by the microphones are periodically synchronized using a test signal played back through a set of two loudspeakers that are placed at each end of the array. This synchronization method also produces an estimate of the local, instantaneous speed of sound, which serves as a validation of the synchronization process, and which is used within the free-field beamforming algorithm. Finally, a high-end microphone unit is placed near the array to achieve accurate sound power level estimates. Field tests conducted in strong wind conditions with the microphone array demonstrate its effectiveness in detecting the direction of various types and combinations of sound sources with good angular resolution.

Keywords: Array, Beamforming, Industrial area

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1 Introduction

Large industrial areas typically contain a wide variety of industrial noise sources, which may all have their own complex spectrotemporal radiation pattern. An example of such a site is the Europoort/Maasvlakte area in the harbour of Rotterdam. Noise, mostly of low-frequency nature, generated by sources within this area, is often a cause of annoyance for people living in the surroundings, particularly for residents of the nearby city of Oostvoorne, located 3km to the south of the area. Figure 1 shows an aerial photograph of the region; the Maasvlakte area and the city of Oostvoorne are both delimited on the map. Measurements by the local environmental authority, the DCMR Milieudienst Rijnmond, have pointed out that all individual industrial plants located in the Maasvlakte each adhere to the relevant environmental regulations. Nevertheless, (low-frequency) noise can sometimes be heard in the residential area, depending on an adverse combination of industrial activities and meteorological conditions, leading to noise complaints by residents of Oostvoorne. In order not to impede potential future expansion of the Rotterdam harbour, it is important that the sources and mechanisms of this problem are known.



Figure 1: Aerial photograph of the Europoort/Maasvlakte industrial area (red line) and the residential area to the south (blue oval). The locations of the arrays are indicated with a red star.

The project “Geluid in Beeld” (“a view on sound”) [1] was commissioned in 2007 to investigate the influence of meteorological conditions on the propagation of sound from the Maasvlakte area to the city of Oostvoorne. As a result of this project, a detailed meteorological-acoustical model was constructed that allows to estimate the propagation of sound from the industrial area

to the residential area, accounting for the complex effects of the North sea and the lake situated between both areas. However, the “Geluid in Beeld” project was not intended to discern the particular sources of noise that give rise to complaints.

Long-term noise monitoring with the purpose of locating and characterizing the noise sources and meteorological conditions that lead to noise complaints poses some serious problems. First of all, the size of the Maasvlakte industrial area (about 30 km²) makes installing a grid of fixed noise monitoring stations with sufficient spatial resolution infeasible. Furthermore, industrial plants are, in general, not easily accessible, which hinders the installation and maintenance of noise monitoring stations. On top of that, the industrial area is mostly situated on land reclaimed from the North sea, and has a harsh environment for noise measurements, with high wind speeds, dust, salt and the seasonal nesting of protected seagulls. The project “Geluidmeetnet Maasvlakte” (“Maasvlakte sound measurement network”) was commissioned by DCMR in 2014 with the goal to tackle these problems efficiently, using a novel microphone array approach.

The objectives of the Geluidmeetnet Maasvlakte project are to continuously locate all relevant sound sources in the Maasvlakte area, over the course of one year, and to estimate the source power of these sources and their expected noise immission in the residential area. The general approach is to localize all active sound sources periodically using microphone arrays. To this purpose, a system of large microphone arrays is permanently installed along the edges of the area. Using beamforming techniques, the direction of arrival at each microphone array of the sound originating from all active noise sources within the area that have a sufficiently high source power can be estimated. Subsequently, similar information from all arrays can be combined using triangulation methods, taking into account the propagation of sound within the industrial area, to determine the location of these sources with reasonable spatial accuracy.

In this paper, the development and testing of the microphone arrays used in this project is described. We refer to other work for a discussion on the applied sound propagation model [2], details on the beamforming algorithm and the method used to combine the information of all arrays to estimate the active noise source locations and their source power [3], and the method used to assess the noise immission in the residential area [4]. In Section 2 of this paper, the hardware design of the arrays is presented, together with the synchronization and beamforming technique used; in Section 3, the results of preliminary field tests are discussed, and in Section 4, the large-scale implementation of the microphone array system is briefly discussed.

2 Microphone arrays

2.1 Hardware and layout

The microphone arrays are designed to allow beamforming over a wide frequency range that covers all octave bands with central frequency from 31.5 Hz to 2 kHz. Low-frequency aliasing limits and maximum beam width specifications require that the aperture of the arrays is at least 50m, whereas microphones need to be sufficiently close to each other to allow beamforming at the higher frequencies. A layout that combines 40 microphones into a series of 8 subarrays, each consisting of 4 or 8 microphones, satisfies these requirements. Figure 2 shows the locations of all microphones.

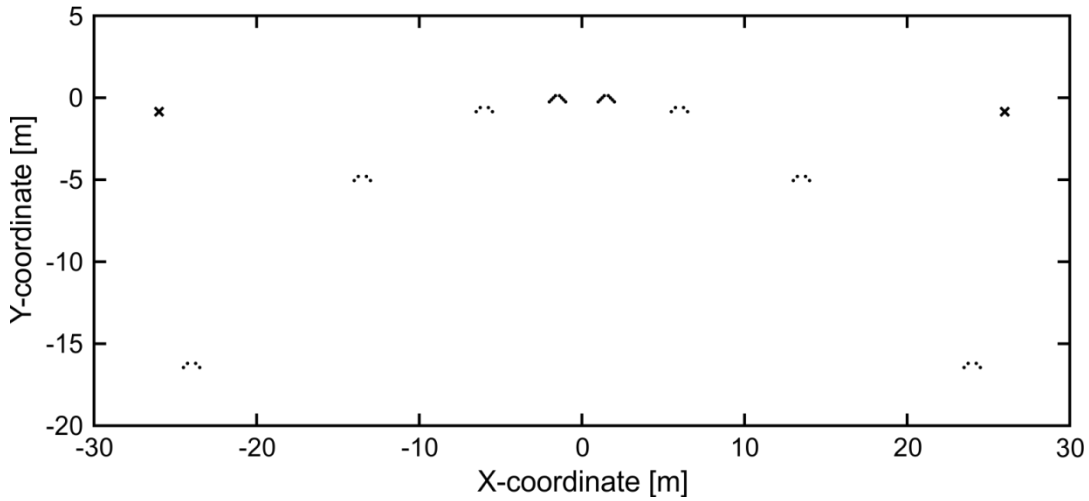


Figure 2: Locations of the microphones (dots) and synchronization loudspeakers (crosses).

This layout allows for some flexibility in placement of the subarrays, such that the shape of each array can be slightly adjusted to the particular terrain conditions at the array location. The main requirement is that the absolute locations of all microphones and synchronization loudspeakers (see Section 2.2) are known with good spatial resolution (to the order of 1cm). Therefore, after placement of each array, the coordinates of all microphones and loudspeakers have to be measured by a land surveyor. Furthermore, the arrays are curved, breaking front-back symmetry, such that a more or less unidirectional beamforming pattern is obtained, and noise originating from outside the industrial area is suppressed. Figure 3 (left) shows a close-up of an installed subarray with 8 microphones (wind screens and bird spikes are visible). One controller (a single-board computer with high quality audio card), connected to a central unit using power-over-ethernet, is installed with each microphone, for audio acquisition and transmission to a central server, where further processing takes place. The use of a separate controller for each microphone introduces a certain level of robustness in the system.



Figure 3: Example of an installed subarray with 8 microphones (left) and an Omnitronic NOH-40R loudspeaker, installed near the ground surface for audio synchronization (right).

2.2 Audio synchronization

The local times of the controllers of all microphones are periodically synchronized using the network time protocol (NTP) that is commonly applied for time synchronization between different computers. This allows audio recordings to be started simultaneously for all microphones within an array, with a temporal accuracy of about 10 ms. However, in order to be able to perform beamtracing at frequencies above 1 kHz, it is essential that the audio signals from all microphones within an array are synchronized with an accuracy much smaller than 1 ms. Such time synchronization accuracy between controllers cannot easily be achieved using NTP.

A possible albeit costly solution would be to connect a GPS receiver to each controller. A more elegant solution, however, is to use an audio signal for time synchronization. This has the added benefit that an estimate of the local, instantaneous speed of sound can be obtained, which is essential for beamforming. Two loudspeakers (Figure 3, right) are installed at opposing sides of the array (see Figure 2 for their locations) and produce, every 10 minutes, a logarithmic frequency sweep from 300 Hz to 3 kHz (the usable frequency range of the audio synchronization loudspeakers) with a duration of 5s. The arrival times of the frequency sweeps at each microphone are obtained using cross-correlation with the original sweep signal.

To illustrate the process, consider the 1-D case without wind. Assume that the first sweep is played back by loudspeaker L, located at position x_L , at time 0. This sweep will then arrive at microphone i , located at position x_i , after a time $(x_i - x_L)/c$, with c the instantaneous, local speed of sound. The controller of microphone i may have a small but unknown time offset Δt_i , such that the sweep is measured at time $t_{L,i}$ with

$$t_{L,i} = \Delta t_i + \frac{x_i - x_L}{c} \quad (1)$$

Now assume that the second sweep is played back by loudspeaker R, located at position x_R , at time T (here 7s). In a similar way, one finds that this sweep will be measured at time $t_{R,i}$ with

$$t_{R,i} = T + \Delta t_i + \frac{x_R - x_i}{c} \quad (2)$$

Equations (1) and (2) together form a system of two equations and two unknowns, Δt_i and c , and can be solved easily; one finds that

$$c = \frac{2x_i - x_L - x_R}{T + t_{L,i} - t_{R,i}}, \quad \Delta t_i = t_{L,i} - \frac{x_i - x_L}{c} \quad (3)$$

The implemented algorithm works in 3-D, and takes into account the influence of the local wind speed and direction on the speed of sound. The estimated local speed of sound can be used in the beamforming process, and can also be compared to the theoretically expected speed of sound, calculated based on the temperature, which allows to detect microphone malfunctioning.

2.3 Beamforming and triangulation

On the basis of the time-synchronized audio signals of each array, the direction of arrival of the sound originating from all active noise sources is estimated using a beamformer specially tailored to the problem at hand. The beamformer scans all potential source locations (located on a non-uniform grid), and estimates the probability that a source is present at that location, accounting for the propagation of sound between each potential source location and each microphone of the array. In the near field, also a rough estimate of the distance to the source is obtained. Two main algorithms, both performed in 1/3-octave bands, are implemented: a native broad-band delay-and-sum beamformer that performs a brute-force cross-correlation of all microphone signals of each array, and an implementation of the MUSIC algorithm [5], based on detected tonal components. This results in the probabilities $P_{b,i}(x, y)$ and $P_{t,i}(f, x, y)$ for every location (x, y) to be the origin of the sound received at array i . A probabilistic approach is then used to combine the information obtained from all microphone arrays; the general idea is that sources need to be detected by at least 3 arrays to be retained. Finally, the sound power of all sources is estimated by minimizing the squared error of sound level estimates with measured sound levels, obtained through a series of class-1 noise measurement stations placed near the arrays and at strategic locations within the study area. For a detailed description of this process, we refer to the paper by Botteldooren *et al.* at Internoise 2016 [3].

3 Field tests

3.1 Overview

A field test with a full-scale array was performed in January 2015 in the harbour of Rotterdam. During the test, 1-minute average wind speeds at 1m height ranged from 4 to 10 m/s. Figure 4 (left) shows the microphone array setup. To test the angular discernment of the array, two separate loudspeaker systems were placed in a field in front of the array, mimicking industrial sources, and a series of test signals was played back, with the goal to estimate their direction of arrival at the array. Figure 4 (right) shows one of these loudspeaker systems, which consists of a trailer with 4 loudspeaker. As an illustration of the beamformer performance, two test conditions will be described below, one with tonal sound and one with band-pass filtered noise.



Figure 4: Photographs of the microphone array setup (left, subarrays are indicated with an arrow) and of one of the trailer-mounted test loudspeaker systems used for the field test (right).

3.2 Detection of tonal sources

The two loudspeaker systems are placed at a distance of about 200m from the array, under an angle of about 10° as seen from the array. A pure tone at a frequency of 251 Hz is played back by the left loudspeaker (as seen from the array), giving rise to a sound pressure level of 54 dB near the array. A pure tone at a frequency of 276 Hz is played back by the right loudspeaker, resulting in 64 dB near the array. Figure 5, left, shows the spectrum measured for each source separately (as the spectra shown here are recorded at two different times, the background noise is slightly different). During the test, the tones are played back simultaneously.

Figure 5, right, shows the beamformer output (using the MUSIC algorithm) at both frequencies, which are automatically detected within the algorithm. The direction of both sources can be clearly discerned, as a line is visible in each panel. An idea about the distance to the source can also be obtained, but it is clear that a second microphone array would have been needed, in order to be able to localize the source along this line more accurately. Note that the dynamic range of the colormaps of both figures has been normalized; as the level of the left loudspeaker is lower, the relative background level is higher.

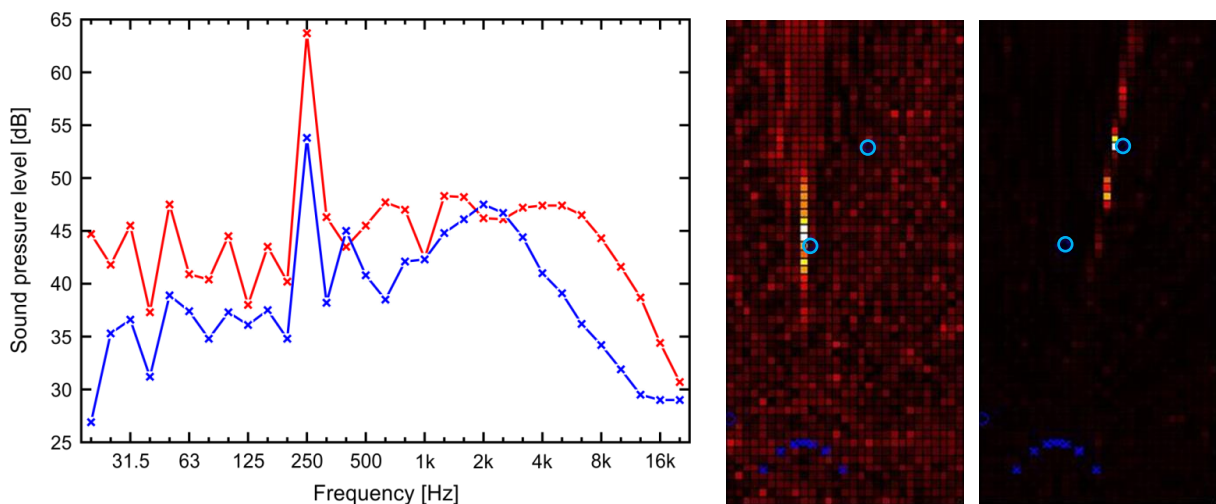


Figure 5: Spectrum of the tonal sound sources (left), and the beamformer output (right). On the right panels, the locations of the two loudspeaker systems are indicated with a blue circle.

3.3 Detection of octave-band filtered noise sources

In a second test, (uncorrelated) octave-band-filtered noise within the 125 Hz octave band is played back through both loudspeakers. The location of the loudspeakers is the same as in Section 3.2. Figure 6, left, shows the spectrum of both sources, measured separately. This time, the left loudspeaker produces a sound pressure level at the location of the microphone array that is about 5 dB lower than the level produced by the right loudspeaker.

Figure 6, right, shows the beamformer output (using the native broad-band beamformer). As both sources produce sound within the same octave band, both detected directions end up in the same map this time. Again, the direction of both sources can be clearly discerned. However, the distance estimates are less accurate using this beamforming algorithm.

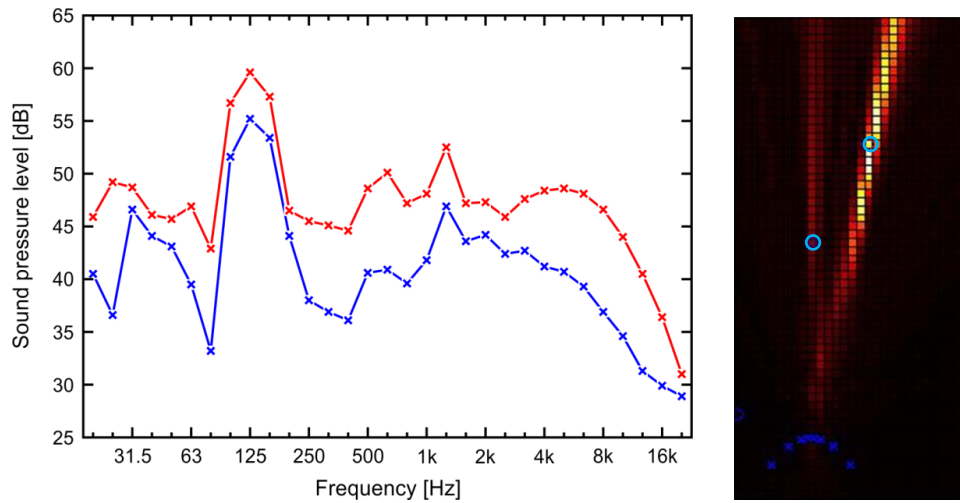


Figure 6: Spectrum of the octave-band sound sources (left) and the beamformer output (right).

4 Large-scale implementation

The Geluidmeetnet Maasvlakte project provided for the placement of 4 microphone arrays along the edges of the industrial area. The selection of optimal locations for placement was guided by good visibility of potential sources from the array locations, avoiding collinearity of arrays with potential sources, relevance for propagation to the residential area, and practical considerations (accessibility, availability of power etc.). Based on these principles, a shortlist of 8 locations was composed. The final selection of the 4 locations was based on numerical simulations. For a number of virtual sources distributed over the industrial area, synthetic audio signals at all array microphones were created using free field sound propagation from sources to microphone locations. Atmospheric turbulence was accounted for by introducing a random distance-dependent time jitter on the audio signals. The combination of arrays that showed the best potential for detecting the virtual sources using the beamformer was selected. The selected locations are shown on Figure 1 with a red star; the locations are conveniently called North, West, South and East. Figure 7 gives an impression of the array placed at the West location.



Figure 7: Photograph of a complete installed array, consisting of 8 subarrays, 2 loudspeakers, a central processing unit, and a 10m high meteo tower in the middle with a class-1 microphone attached at 5m height. The photo also shows the fence built around the structure.

Microphone arrays were installed during the autumn of 2015, have been fully active since January 2016, and are foreseen to be active for 1 complete year. Next to the 4 microphone arrays, the complete measurement network includes 10 conventional class-1 noise monitoring stations placed at strategic locations within the area, and 4 masts for meteorological observations. Data generated by all microphone arrays (about 1 TByte/month) is transferred to a number of internet gateways (optical fiber network) over licensed 3.5 GHz radio links using OFDM technology, with fall-back to 4G LTE or 3G for load balancing, and are then sent to a set of central storage and computing servers for further processing. Maps of the sound power of all sources are calculated in real-time every 10 minutes. An online interface allows the user to select arbitrary time windows (typically related to periods of noise complaints); noise emission maps are then aggregated over this observation time window, and frequency bands are combined, resulting in more stable estimates of the sound power level. Figure 8 shows an example of such an aggregated noise emission map.

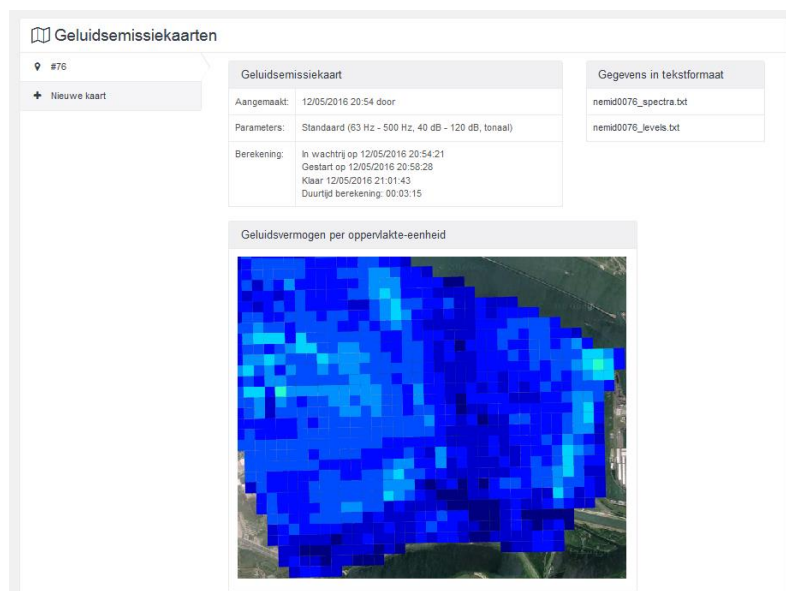


Figure 8: Screenshot of part of the online interface (in Dutch), in which a noise emission map, aggregated over a period of 8h, is visible.

Results on the performance of the microphone array system in locating active sound sources within the industrial area will become available after the end of the project. Nevertheless, the data gathered during the first months of the measurement period already provides insight in the functioning of the system. As an illustration, the speed of sound estimated through the audio synchronization algorithm can be compared to the theoretical speed of sound, based on the temperature. Figure 9 shows the estimated (blue curve) and theoretical (red curve) speed of sound, as a function of temperature, aggregated over a 5 month period (21888 measurements in total) for array West. During this period of time, a wide range of meteorological conditions have occurred. Differences with the theoretically expected speed of sound are typically well below 1 m/s; the variance is mainly caused by those situations when there is a high wind speed with a direction perpendicular to the axis of the array.

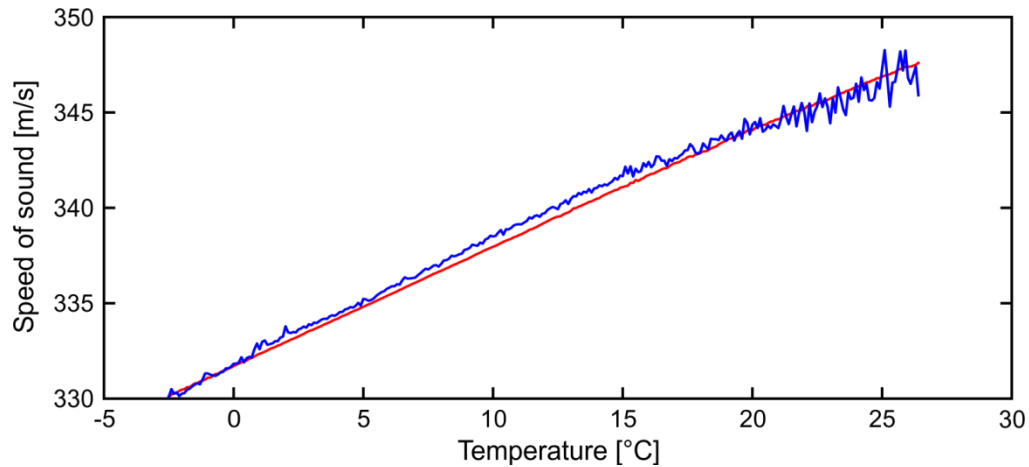


Figure 9: Estimated (blue) and theoretical speed of sound (red) as a function of temperature.

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References

- [1] van der Eerden, F.; van den Berg, F.; Randrianoelina, A. Application and validation of a meteorological-acoustical model for outdoor sound propagation. *Proceedings of the 38th International Congress and Exposition on Noise Control Engineering (Internoise '09)*, Ottawa, Canada, 2009, pp 4332-4339.
- [2] van der Eerden, F.; Wessels, P.; Segers, A.; Basten, T.; De Coensel, B.; Botteldooren, D.; Van Renterghem, T.; Dekoninck, L.; Spruytte, V.; Makovec, A. Time varying sound propagation for a large industrial area. *Proceedings of the 45th International Congress and Exposition on Noise Control Engineering (Internoise '16)*, Hamburg, Germany, 2016. In CD-ROM.
- [3] Botteldooren, D.; Van Renterghem, T.; De Coensel, B.; Dekoninck, L.; Spruytte, V.; Makovec, A.; van der Eerden, F.; Wessels, P.; Basten, T. Fusion of multiple microphone array data for localizing sound sources in an industrial area. *Proceedings of the 45th International Congress and Exposition on Noise Control Engineering (Internoise '16)*, Hamburg, Germany, 2016. In CD-ROM.
- [4] Wessels, P.; van der Eerden, F.; Basten, T.; De Coensel, B.; Botteldooren, D.; Van Renterghem, T.; Dekoninck, L.; Spruytte, V.; Makovec, A. Removing local sound disturbances from industrial noise monitoring at long distance. *Proceedings of the 22nd International Congress on Acoustics (ICA '16)*, Buenos Aires, Argentina, 2016. In CD-ROM.
- [5] Schmidt, R. Multiple emitter location and signal parameter estimation. *IEEE Transactions on Antennas and Propagation*, Vol 34 (3), 1986, pp 276-280.

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