Fusion of multiple microphone array data for localizing sound sources in an industrial area

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ABSTRACT

Locating sound sources that contribute to noise annoyance near large industrial areas under different meteorological conditions is a hard problem. Permanently installed microphone arrays at the edges of an industrial area allow to determine the direction of arrival of the sound at their location. Several algorithms have been proposed for this purpose yet not all of them are robust against changes in effective sound speed and loss of coherence. Therefore algorithm parameters have to be chosen carefully. In addition, in this paper, a probabilistic approach is proposed to combine the information obtained from three or more arrays. The methodology accounts for the effect of wind and temperature on the direction of arrival. It also estimates uncertainty caused by uncertainty in the local meteorological situation, ground impedance, and presence of typical harbor objects such as stacked containers and piles of coal, etc. The proposed methodology is applied to an industrial area of over 10 kilometer squared and the consistency of detected sources under varying weather conditions is investigated. This shows the validity of the approach.

Keywords: microphone array, industrial noise I-INCE Classification of Subjects Number(s): 52.2, 74.7

1. INTRODUCTION

The range of applications of microphone arrays has grown considerably the last years amongst others due to the increased availability of cheap and reliable microphones. Detection of the source of environmental noise is an appealing application that has been proposed by many acoustic array product vendors. However, the devices offered are not applicable for monitoring purposes and long-term outdoor use. For very low frequencies and large distances, permanent monitoring arrays have been deployed since many years (1,2).

In this paper a monitoring system for industrial noise based on microphone arrays is proposed. Development of such a system faces several challenges: The hardware has to be robust and redundant to guarantee a high up-time. Details of the design of the microphone arrays will be given elsewhere (3). The performance of microphone arrays under atmospheric turbulence may degrade if the microphone array has not been over-sized with respect to the CRLB (4). However, meteorological conditions in combination with surface impedance and its changes along the propagation path also change the angle of arrival and amplitude of the sound at the arrays. Changes in propagation affect signal to noise ratio strongly over the large distances envisaged and thus also determine the array accuracy. Detailed meteorological measurements combined with high performance propagation models should be used to

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back propagate the incoming waves to their source. Details of the meteorological measurements and time changing propagation effects will be given by Frits van der Eerden at this same conference (5).

This paper will focus on the fusion of data obtained from several arrays around an industrial area to estimate the location of the main sources of sound and their approximate sound power level. The problem of microphone array data fusion has mainly been addressed for speaker detection (6) and at small distances. In this case it does not face any problems related to atmospheric boundary layer propagation which the current application faces.

2. METHODOLOGY

2.1 Array data processing

Four semi-permanent microphone arrays have been placed around the industrial area. Given the large horizontal distances involved and the relative low height of the sound sources, two dimensional, horizontal arrays have been designed for the purpose. To capture the extremely low frequencies of interest down to 25 Hz, the arrays are about 30 m in diameter. The arrays have 40 microphones spaced non-equidistantly. For robustness, the data from each array microphone is processed individually and transferred to the back office computers for further processing over a dedicated wireless connections towards wired locations (optical fiber) a couple of kilometers away. These wireless connections are based on exclusive usage of licensed spectrum in the 3.5 GHz radio band using OFDM technology with failover sustainability using on site ruggedized 3G/4G/LTE networking radio equipment for load balancing as full redundant solution. At every array test signals are periodically emitted to estimate speed of sound, synchronize sensors and detect failure or temporary malfunction (e.g. due to the presence of birds or wind bursts).

For detection direction of arrival (DoA), a robust native beam former is combined with a MUSIC beam forming algorithm. The latter outperforms the robust beam former in particular when the sound contains tonal components however, its performance significantly degrades under turbulent atmosphere (7).

Near large industrial areas, low frequency noise [25Hz-600Hz] is a main issue as higher frequencies get attenuated considerably at larger distances where neighbors are usually found. This frequency range is nevertheless still rather broad. Hence, before array processing, microphone signals are band-pass filtered in 1/3 octave band spectra. All beam forming is performed on a 1/3 octave band basis.

2.2 A probabilistic approach for fusion of DoA

For every possible source location in the industrial area, the probability for this location to be the origin of the sound received by the array, $P(x,y)$, is calculated. Several approaches for calculating this probability distribution are combined. The main approach, $P_b(x,y)$, consists in combining DoA at the four arrays. Rather than using a plane wave approximation, the travel time for sound emitted at each location to each of the array microphones is calculated for performing the beam forming. This allows to account for non-straight propagation paths caused by wind and wind gradients relatively easily. For information fusion a combination of intersection and union operators is used:

$$P_b(x,y) = \bigcup_c \left(\bigcap_{i \in c} P_i(x,y) \right) \qquad (1)$$

where is the probability distribution obtained from a single array and the union operator runs over all possible combinations of three arrays out of four and the intersection operator over all microphone arrays within that combination. This logic assumes that a source has to be detected by three arrays to be real (AND-operation), but detecting it by any of the combinations of three arrays is sufficient (OR-operation). Allowing combinations of two arrays would lead to ambiguities for sources lying near the line connecting the two arrays, relying on detection by four arrays is unrealistic in large study areas. To implement intersection and union, any t-norm and t-conorm can be used. We opted for the product norm as it is rather soft in transitions and is congruent with the interpretation of the distributions as probabilities, and the matching probabilistic sum.

2.3 Tonal sound and TDoA for time fluctuating sounds

Additional information can be added to the fusion operation. Often industrial sound although broadband in general terms, consists of a number of tonal components and their harmonics. Fans,

blowers, engines, electromotor, etc. all generate sound that contains tonal components. Within the uncertainty of spectral shift and spectral broadening caused by refraction in turbulent atmosphere, these tonal components may be observed simultaneously at several of the arrays. If this is the case, there is a high probability that they might emerge from the same source. Hence a second probability distribution is derived:

$$P_t(f,x,y) = \bigcup_c \left(\bigcap_{i \in c} P_i(f,x,y) \right) \tag{2}$$

where $P_i(f,x,y)$ is the probability distribution at frequency f obtained using the MUSIC algorithm. If a frequency is not detected as a prominent tone at array i, the corresponding probability distribution is set to zero.

In sensor networks, time difference of arrival (TDoA) is often used to identify the location of a source. However in the application at hand with distances of several kilometers between arrays and an industrial terrain where temperature and wind may change considerable over the area of interest, differences in speed of sound lead to uncertainties in propagation time of the order of several tens of milliseconds. Added to that are synchronization issues between distant arrays that in themselves cause similar uncertainties in arrival time. One should also note that the period corresponding to the frequencies of interest is of the order of 10 ms or less. Straightforward TDoA thus would lead to periodic patterns with typical distance of 30 m and would thus be completely useless. For both of these reasons, a new algorithm was designed that analyses TDoA on amplitude modulation of the observed signals and leads to an additional estimate of source probability distribution P_{am}. Note however that this probability distribution will give no information on the location of sources with a constant amplitude such as tonal sounds.

2.4   Sound power estimation using probability weighting.

Sound power estimation algorithms usually assume a very limited number of sound sources. The first step in these algorithms consists in uncovering all sound source positions using all available beam forming mechanisms. The second step consists in assigning a sound power level to these sources (6). In industrial areas the number of small sound sources can be huge, yet together they can have a significant impact on the environment. Therefore, the above procedure is inverted. In a first step, the sound power $W_{b,i}(x,y)$ is estimated for every location in the area of interest using a basic beam former algorithm and an estimate of the meteorology-dependent amplitude attenuation function between this location and the ith array, $A_i(x,y)$:

$$W_{b,i}(x,y) = B(R_i,x,y) / A_i(x,y) \tag{3}$$

where B is the beam former relating the matrix of cross correlation between the sensors of the ith array, R, to the location (x,y). The estimates obtained from each array are then aggregated to a single sound power level estimate $L_{W,b}$ using an OWA (ordered weighted average) operator (9) that focusing on the lower levels.

$$L_{Wb}(x,y) = OWA(L_{Wb,i}, \mu_i) \tag{4}$$

The weights, μ_i, are an estimate of the accuracy of each estimation of the power. This accuracy depends on the accuracy of the estimation of the attenuation function which in turn is governed by meteorological conditions. Typically arrays upwind of the source will have less influence as the accuracy of the estimation of the transfer function is low. The choice for focusing on the lower sound power levels is inspired by the observation that a true source power will be detected by all of the arrays (provided that they can make a good estimate as expressed by the weight μ_i).

For an ideal noise-free beam former, B, the above procedure would give perfect estimates of the source power levels. However, the finiteness of the array as well as sensor noise – wind induced noise amongst others – result in broadening of the beams and an over-estimate of source power levels. At this step the probability of sound locations, $P(x,y)$, is considered. Translating probability to the presence of a sound source requires a crisp decision on the threshold of probability that is considered. As propagation conditions change, the probability distribution will also change. Hence at one moment in time a potential source location can become more or less probable. Thus keeping also source locations with lower probability partly assigned, in the long run some information can still be gained

on sources at these locations. For this reason rather than to select possible source locations in a crisp manner, a probability enhancement function, *E*, is introduced that transforms *P(x,y)* to a more peaked spatial distribution. This eventually results in a sound power distribution over the industrial area:

$$L_W(x, y) = E(P(x, y), \alpha)L_{Wb}(x, y), \qquad (5)$$

where the enhancement parameter, α, is obtained by minimizing the squared error between the sound pressure level obtained from this sound power level distribution and the attenuation, *A*, at the one hand and the measured sound pressure level at a set of microphones inside the industrial area at the other.

3. SIMULATION RESULTS

The microphone array fusion method is first validated on simulations. In order to explore the efficiency of all of the algorithms for sound source localization 5 sources are introduced at randomly chosen locations (Figure 1). Source 1 and 2 emit white noise at constant amplitude in a low frequency range. Source 3 is a tonal noise source with constant amplitude, source 4 is a white noise source with low frequency modulated amplitude, and source 5 is a tonal source with low frequency modulated amplitude.
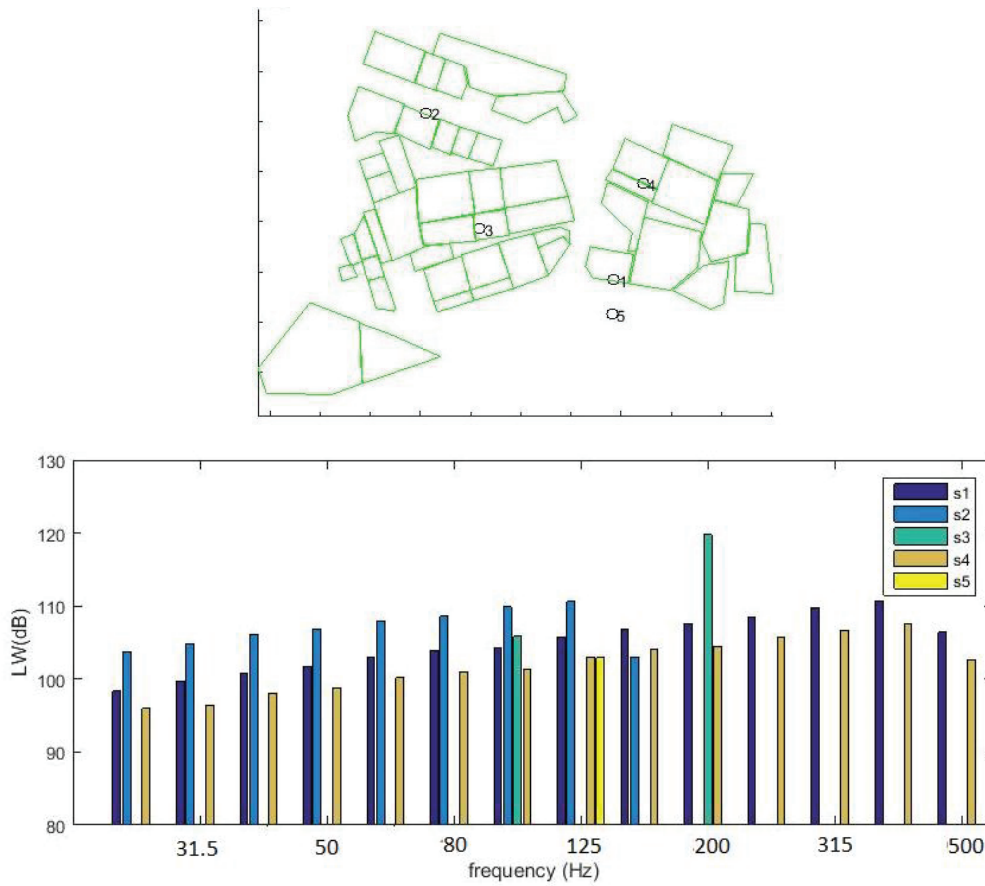

Figure 1 – Location of the 5 simulated sound sources in an industrial area (upper, 1km tick marks) and their respective sound power levels (lower).

Sound pressure at the array microphones was simulated using the attenuation functions, *Aᵢ(x,y)*, that were obtained using PE simulations (5). These simulations were conducted for a representative meteorological situation: a 3m/s northeastern wind (80 degrees) and a temperature of 6 degrees.

Figure 2 shows the probability distributions obtained from the fusion of information from the four arrays in this simulation. In the left column, showing broadband probability estimation, the location of source 1 is always visible although its presence may be obscured by tonal sources nearby in the 100 Hz and 125 Hz 1/3 octave band. The middle column giving the probability distribution for matching tones reveals the location of the 98 Hz tone (source 3) and the 123Hz amplitude modulated tone (source 5). The tonal detector also seems to detect the very strong 196 Hz tone (source 3) in the 250 Hz 1/3 octave band although it theoretically lies outside this 1/3 octave band. However as standard octave band

filters are not perfect, this is reasonable. Finally, the third column giving the probability of amplitude modulated sound, reveals the location of source 4 in the 250 Hz 1/3 octave band and source 5 in the 125 Hz band. Although not explicitly amplitude modulated, source 1 also pops up in these probability distributions. Note that source 2, although relatively strong is not at all appearing in the probability distributions. This is due to the adverse meteorological conditions for detecting this source. It is upwind with respect to array 1 and at large distance from the other arrays.

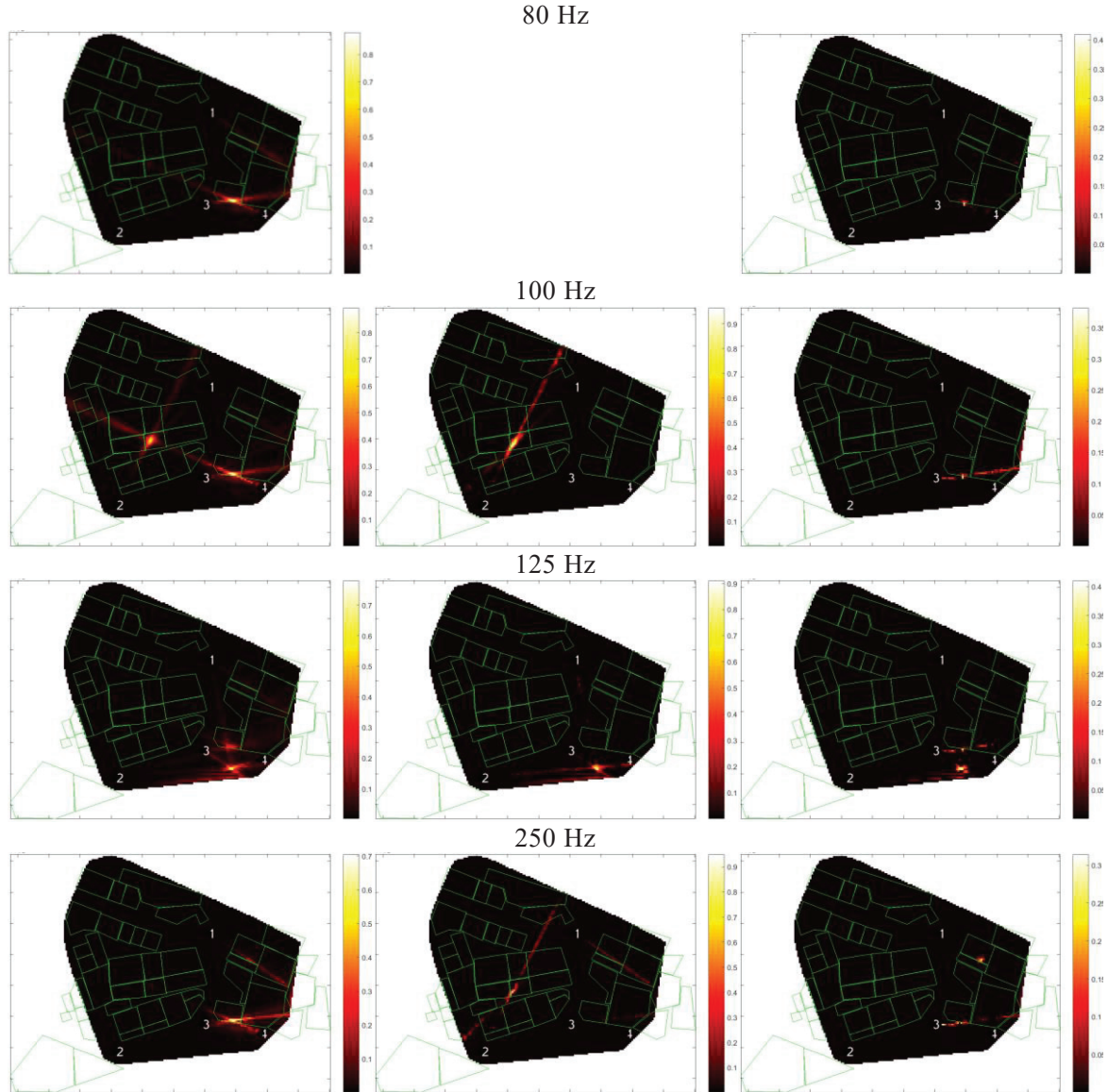80 Hz

100 Hz

125 Hz

250 Hz

Figure 2 – probability distribution for finding sources in the simulated industrial area in 4 different 1/3 octave bands (rows) for: broadband $P_b(x,y)$ (left column); matching tones $P_t(x,y)$ (middle column); sound with amplitude modulation $P_{am}(x,y)$ (right column).

4. EXPERIMENTAL RESULTS

The experimental setup includes 4 microphone arrays and 10 more microphones placed at strategic locations, 4 masts for meteorological observations. The area is the same as the one used for the simulations shown above. More details on the measurement setup can be found in (5). A few examples from the several months of continuous measurements illustrate the practical applicability and limitations of the multiple array fusion technique. Figure 3 shows three snapshots taken every ten minutes of the instantaneous estimate (1 minute) of sound power level distributions across the industrial area. During the observation period wind speeds are low and mostly northwestern but variable; temperature gradients are slightly upwardly refracting. The location of the dominant sources

and the estimate of their sound power level changes significantly over time. Obviously, there is no guarantee that the dominant industrial sound sources in the area are not varying in time (e.g. loading and unloading, moving vehicles, …). Yet this also illustrates two known effects of meteorological conditions on array processing: shifting of source positions due to turbulence or short term fluctuations in effective sound speed and instantaneous fading due to upward refraction (8). However, also some trends become visible. When averaging over longer observation times and combining frequency bands results in more stable estimates of sound power levels as can be seen in Figure 4.
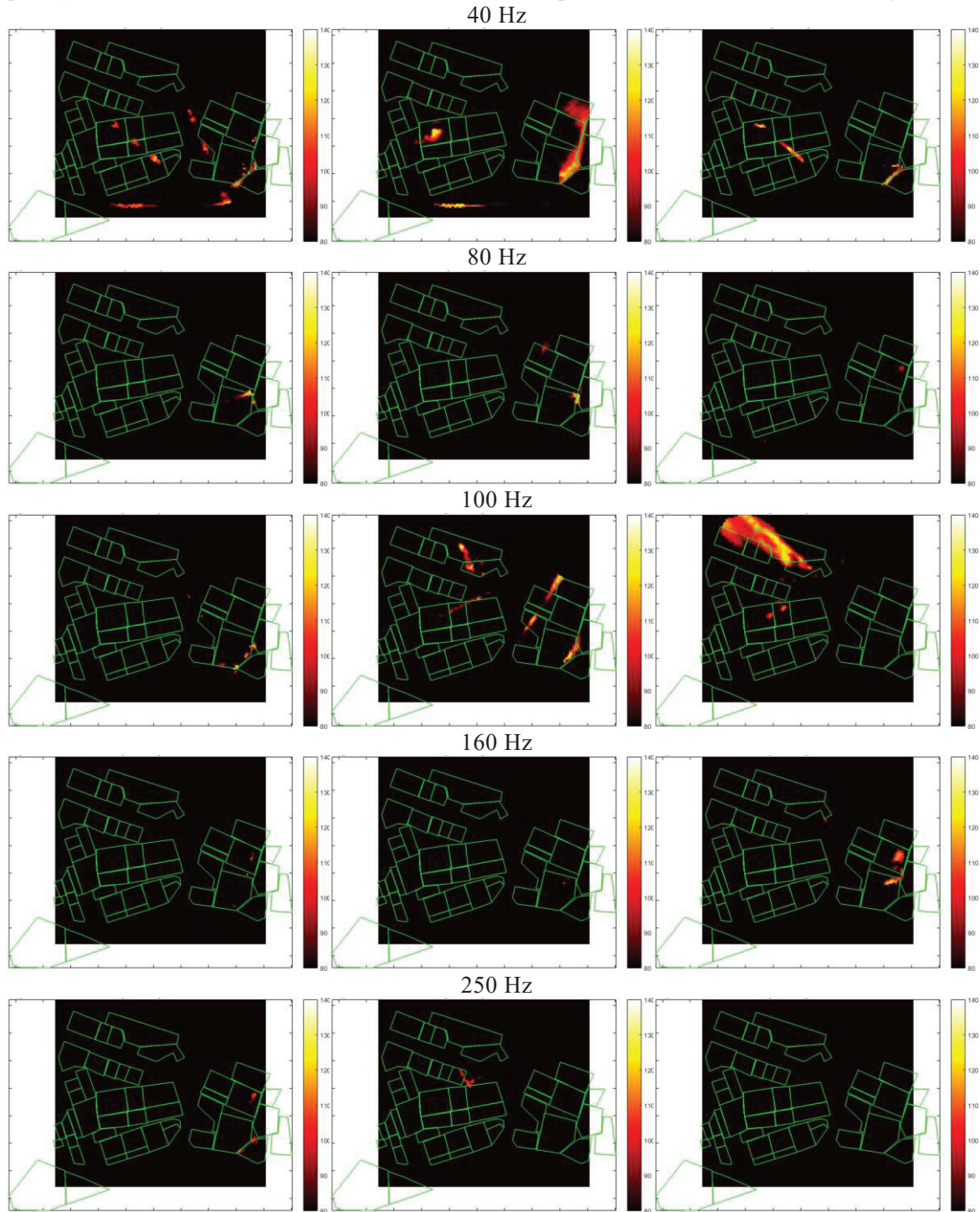
40 Hz

80 Hz

100 Hz

160 Hz

250 Hz

Figure 3 – Instantaneously estimated sound power level (dB) in different 1/3-octave bands; from left to right at 10:10, 10:20, and 10:30 local time.
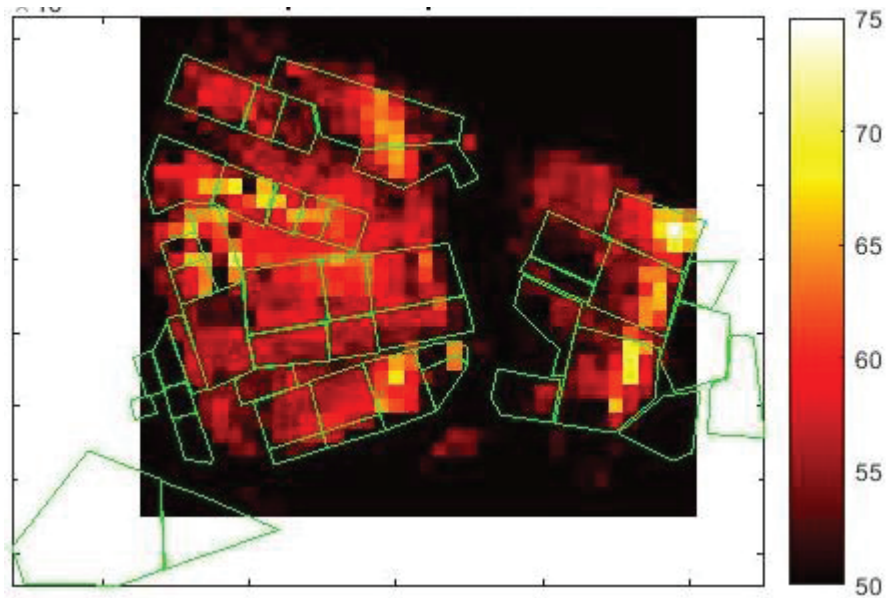
Figure 4 – Average sound power level per square meter in 200mx200m grid [63Hz,500Hz] averaged between 2:00 and 10:00, same day as Figure 3. Green colors indicate higher levels than blue colors.

5. CONCLUSIONS

This paper introduces new ideas for fusion of the information obtained from microphone arrays placed around an industrial area. It shows that even at extreme distances and facing complex meteorological conditions, useful information on the locations of the industrial noise sources that contribute to the impact on the environment can be detected. The permanent character of the measurement network as a whole allows to analyze instances where noise complains arise and deduce their main cause. Choosing the location of arrays and validation microphones is crucial to identify exactly those conditions where these complaints may emerge as wind and temperature gradients have a very significant influence at the long distances considered.

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