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NOISE CONTROL FOR A BETTER ENVIRONMENT

Noise source localization in industrial facilities

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ABSTRACT

Measuring and analyze noise at industrial facilities is a challenging task. Not only high levels but also a large number of different sound sources can appear. It is difficult and time consuming to identify and analyse different sound sources separately with conventional methods. This publication presents a case study using microphone arrays and beam-forming algorithms to localize, identify and quantify several noise sources inside industrial facilities.

Keywords: Beamforming, noise source localization, microphone array
I-INCE Classification of Subject Number: 71

1. INTRODUCTION

Noise pollution is a problem in modern societies. Awareness about the relationship between noise exposure and health is increasing in recent times, regardless of whether this problem affects a worker inside of a industrial facility or a neighbour suffering from the noise of the same facility in his house [1&2]. Several standards and laws around the world set the value limits for different areas and the procedures fto conduct the noise measurements correctly [3&4]. In addition, in several countries, penalties due to low frequencies, impulsiveness and tonality conclude adding 3, 6 or 9 dB to the final results when comparing them to the value limits established. This is a very good reason for localizing the noise sources that produce these noise levels and identify the frequency range for each one in order to go deeper in the situation analysis and the noise reduction solutions proposal.

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From another point of view, sound localization is a useful tool for analyzing the workflow of the important machines inside a industrial facility; pumps, engines, electric inverters or fans can be the targets for noise reduction but also their noise footprint can be analyzed in order to compare between different workflows or product life status.

The Acoustic Camera system can improve the noise information in these scenarios providing noise localization information that confirms which noise source is emitting the main level and in what frequency range. The system is consist of a microphone array, there are several ones with different geometries and features, depending on the application; the microphone array is connected to a multichannel data recorder for feeding the microphones and recording all the signals, all managed by a computer running the software for acquisition and processing the data; this means that you are able to have acoustic pictures results on site.

2. LOCALIZATION PRINCIPLE

Beamforming is the most widely known sound localization technique. There are other localizations techniques like Intensity or Nearfield Acoustic Holography, more indicated for low frequencies and very short distances. This publication is focused on Beamforming, this technique will provide us the whole picture of a complete facility. With the same technique it is also possible go inside the facilities and take measurements from closer points of evaluation.

The basic algorithm is named Delay-And-Sum-Beamformer in time domain (TDBF), the principle can be described by decomposing the signal processing into four main steps. Figure 1 illustrates an easy example using two point sources situated in front of the microphone array with 4 channels [7].

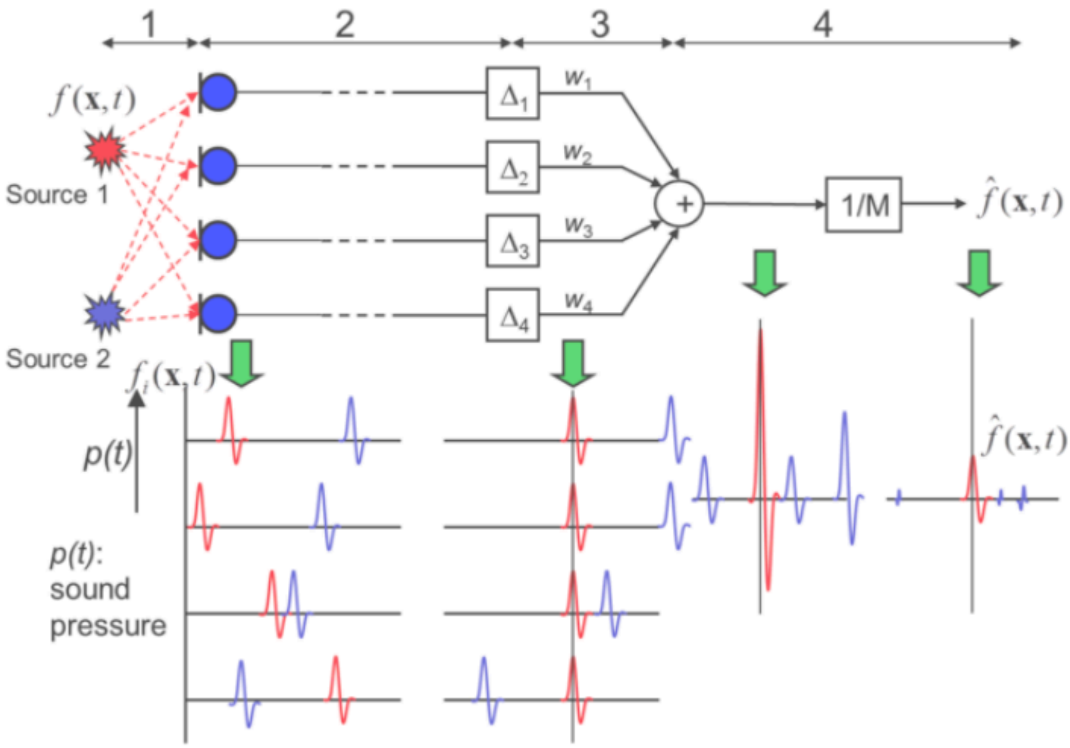

Figure 1 – Delay and sum Beamforming in time domain block diagram [7]

1. The sound of each source travels to every microphone along different paths.

2. The signals acquired by the microphones are similar in wave form, but show different delays and phases. Both are proportional to the travelled distances. The delays can be determined from the speed of sound, the distance between the microphones, and the sound sources. The Beamformer targets the point where source 1 is situated (Figure1).

3. The signal of each microphone is shifted by a corresponding runtime difference depending on the focus point. As a result, the signal components of source 1 (red impulses) in all channels are in phase, whereas the signal components originating from source 2 (blue impulses) are out of phase.

4. The signals of all channels are summed together and finally, the summed signal is normalized by the number of microphone channels. The amplitude of the signal component of source 1 (red) in the sum signal is as strong as the original amplitude of source 1 and the signal components originating from source 2 (blue) are negligible. The RMS or the maximum value can be calculated from the time signal f_{BF}(x,t) and visualized in the acoustic map.

The Delay-And-Sum-Beamformer in frequency domain is based on a similar principle as in time domain. The block diagram in Figure 2 illustrates the same easy example like in time domain, using two point sources [7].

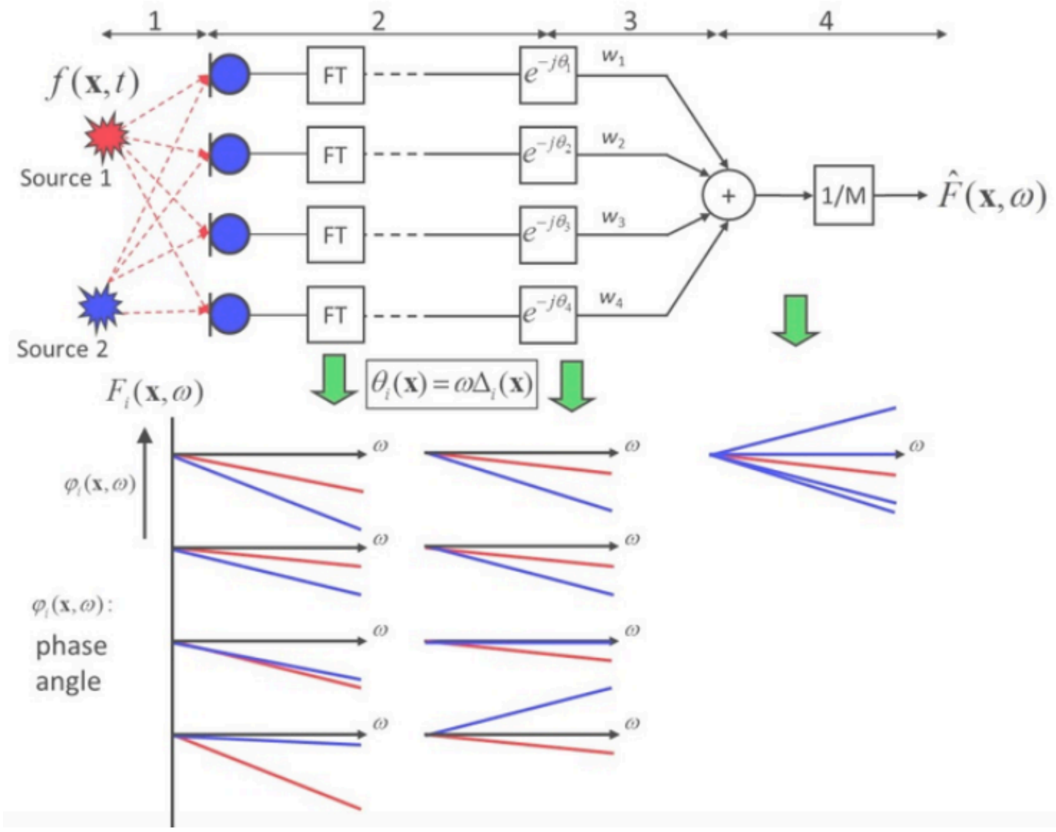

Figure 2 – Delay and sum Beamforming in frequency domain block diagram [7]

1. The first step is the same as in TDBF, the sound of each source travels to each of the microphones along different paths. This leads to delays and phases in the measured signals, which are both proportional to the travelled distances. The delays can be determined from the known distance between the microphone array and the measurement plane and the speed of sound.

2. After performing the Fourier transformation of each microphone signal, the spectra are available as amplitude and phase. Now, the phase of each individual microphone signal can be corrected with respect to a particular delay. It is multiplied by a complex phase term without influencing the amplitude. Here, the phasing of the signal parts is carried out to the target source 1. As a result, the signal parts of source 2 are out of phase. It is important to note that a constant time delay results in a frequency dependent phase correction term.

3. The resulting complex spectra are added up. In this process, the signal parts of source 1 add up constructively and the signal parts of source 2 diminish.

4. Finally, the sum signal is normalized by the number of microphone channels. From the sum spectrum the RMS and the maximum value can be calculated and visualized in the acoustic map. To auralize the sound signal at a focus point, it is possible to perform the inverse Fourier transform resulting in the local time signal f_{BF}(x,t) at the focus point.

2.1 2D mapping

The process for developing the final result as an acoustic photo set a virtual plane, parallel to the microphone array plane, with the distance between them introduced by the user. Using the process described in the previous point, the calculation is done for each point from the virtual plane. The higher the number of pixels set in the virtual plane, the higher the resolution of the acoustic photo, and the time processing.

It is important to have in mind that the results are always related in a parallel plane to the microphone array. Because of this, it is recommended to try to set the microphone array parallel aligned to the targets sources when possible. The results are represented as a two dimensional photo. Reflexions will be visualize as well and will be mapped on the photo, too. For analysing the any set-up optimally it is recommendable to map on a 3D model of the scene.

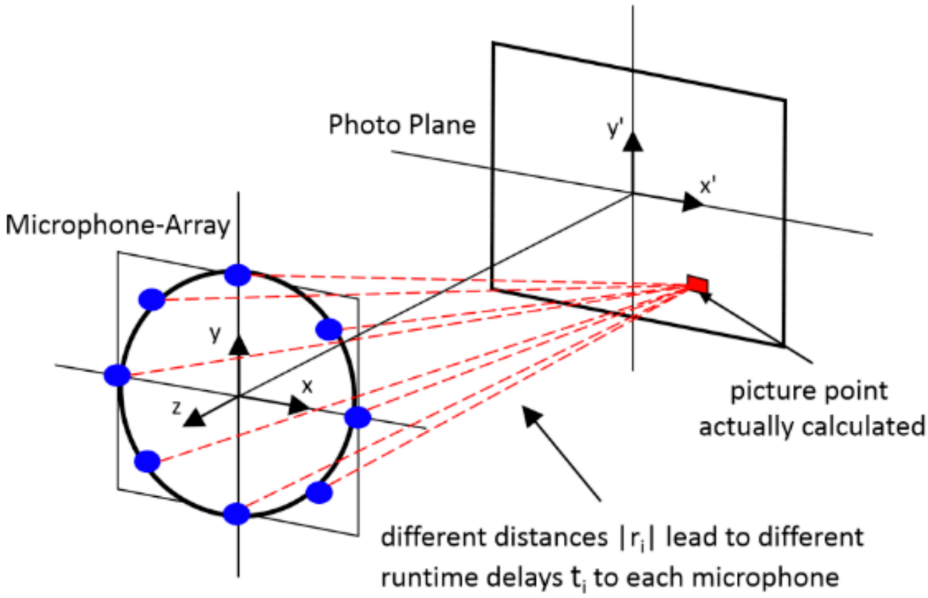

Figure 3 – Virtual plane calculation in 2D

2.2 Microphone arrays

Two main keys for selecting the correct microphone array: distance to noise sources and expected frequency range. With these two variables you can select the proper microphone array for the specific sound localization application. If localizing low frequencies is on the target, we need to use big arrays because of the relationship between the frequency and the wavelength. It is not possible to have precise results in low frequencies using microphone arrays with small diameters.

Another important decision is whether to develop the measurement in 2D or 3D. Sphere geometries arrays with microphones in all the directions can localize noise sources no matter where the source is; but for 3D applications the 3D model is required for presenting the noise results on it. The 3D model can be acquired using a laser-scanner or created by modelling software.

For the two measurement series developed for this publication, a Star array with 48 microphones was used. This array is specially designed for outdoor applications and large distances to the noise sources, furthermore it is the right microphone array for low frequencies because the diameter of 3,4 meters [Figure 4, centre].

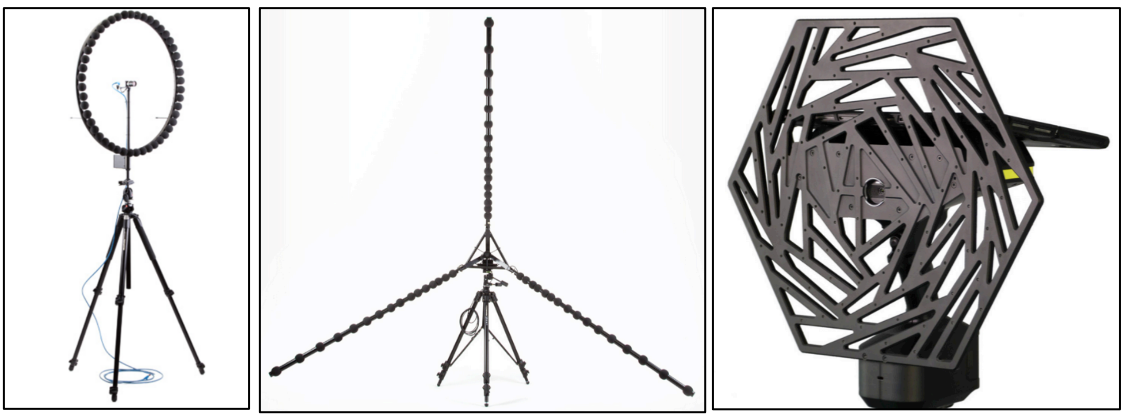

Figure 4 – Microphone arrays, from left to right: Ring, Star and Portable

3. MEASUREMENTS AND RESULTS

3.1 Scenario 1: Electric plant

In this first scenario, the aim is to localize the different noise sources in an electric facility during normal operation. The microphone array was situated 70 meters from the first electric units. This area is surrounded by roads and highways, it is important to check that no external noise from traffic pollutes our measurement. The spectrogram is a good tool for evaluating if some sound event out of our aim is on the measurement and take a first look at the frequency component. In this case (Figure 5), the spectrogram shows continuous frequencies with no variations.

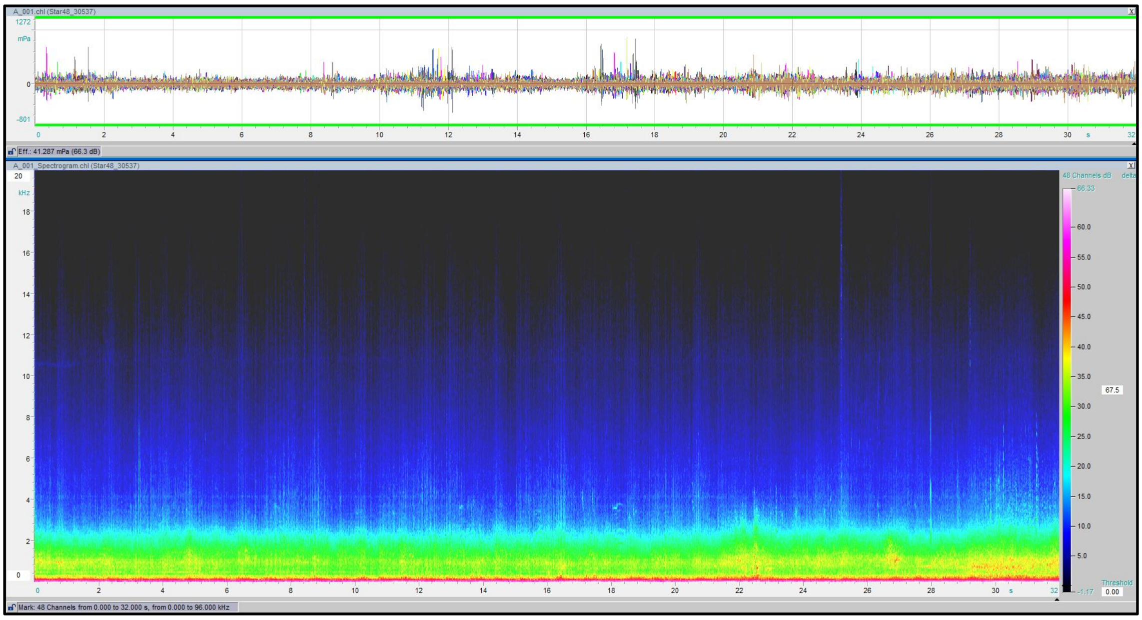

Figure 5 – Spectrogram

After checking the spectrogram, we can use the frequency spectrum generated from the time signal to evaluate the whole spectrum by third octaves bands, frequency ranges or focused frequency peaks. Using this processing technique, several noise sources are clearly identified. Two noise sources in two electric cable junctions are localized between 2.500 Hz and 7.000 Hz (Figure 6). The frequency peak on 1.125 Hz comes from the cabin situated on the left side (Figure 7). The range between 280 Hz and 1.030 Hz is very focused on the electric units from the right side (Figure 8). Analysing the very low frequency, the source is out of our view, on the right side of the facility (Figure 9).

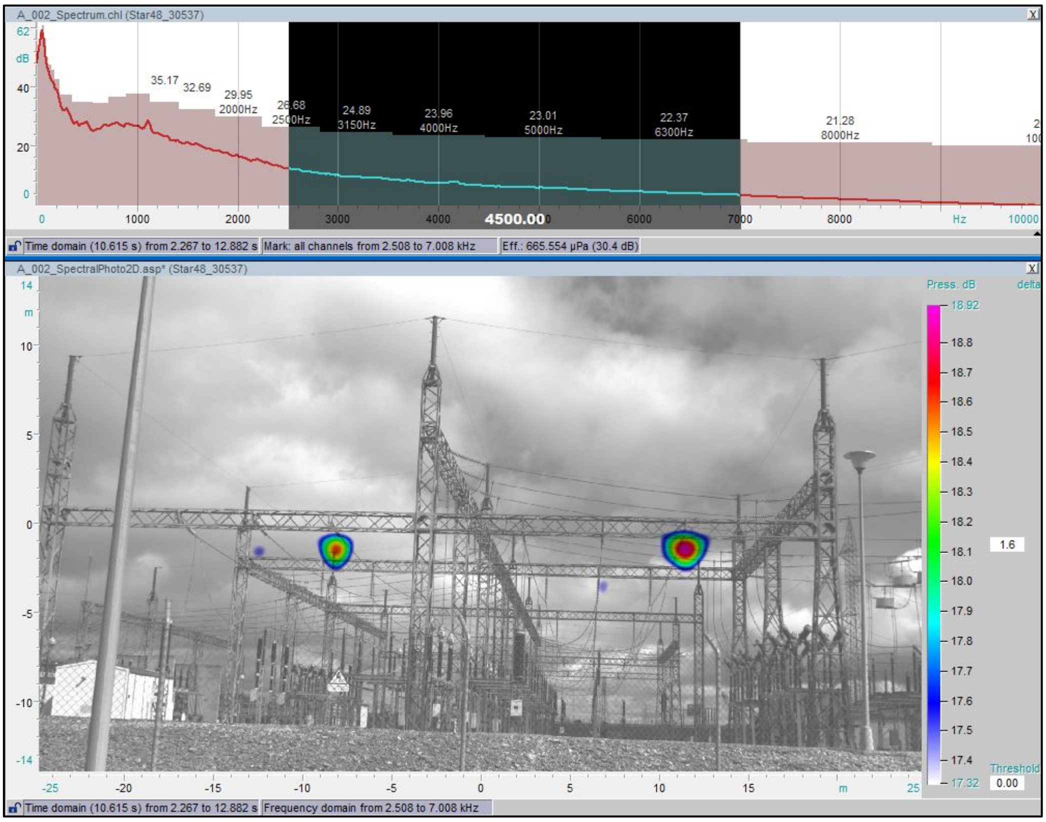

Figure 6 – Frequency range between 2.500 and 7.000 Hz

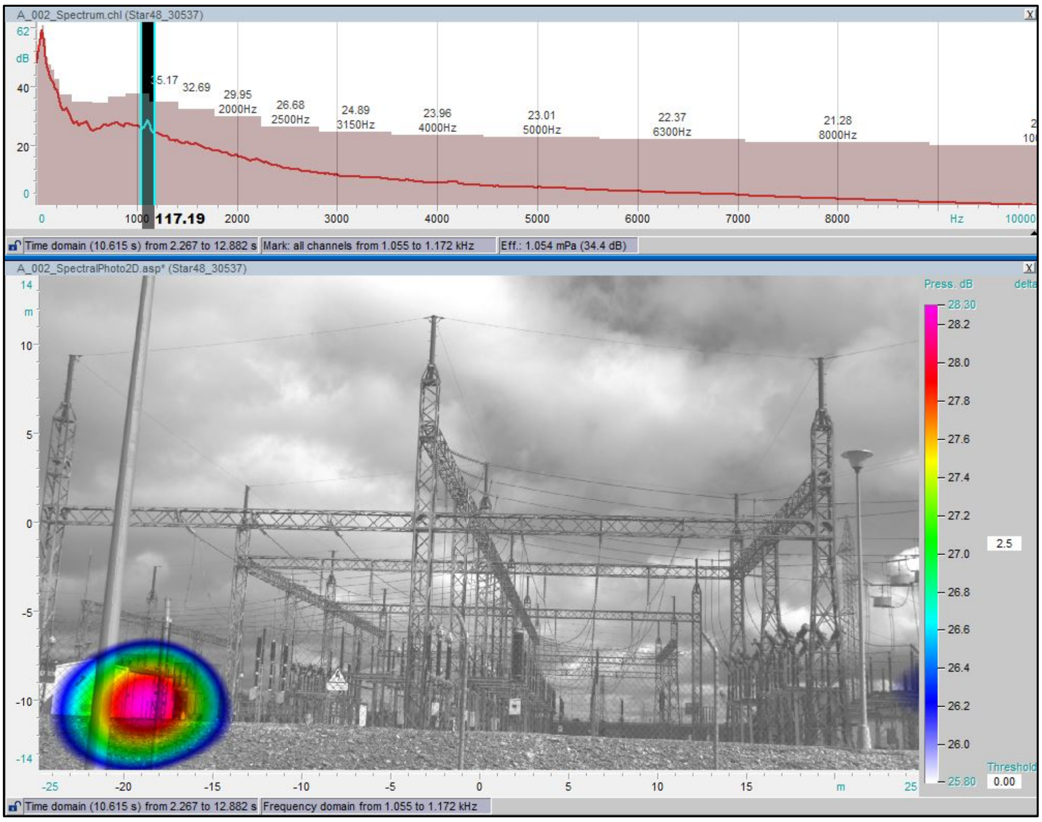

Figure 7 – Frequency peak on 1.125 Hz

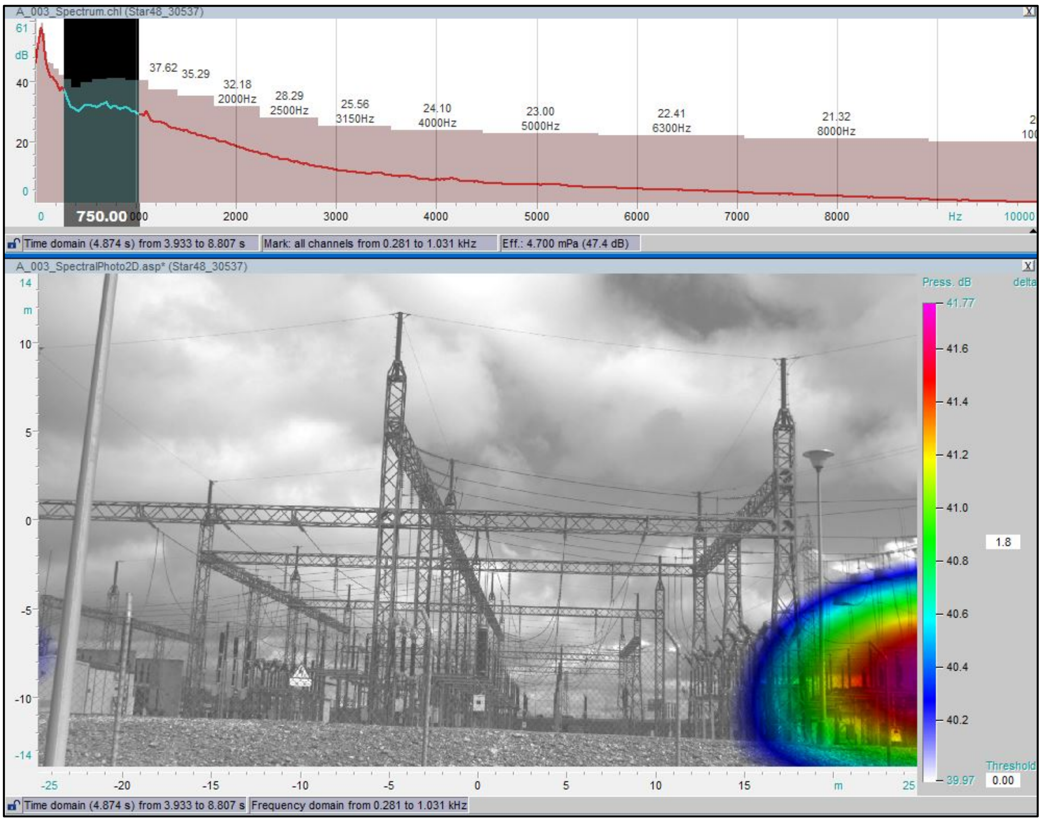

Figure 8 – Frequency range in low-medium frequencies

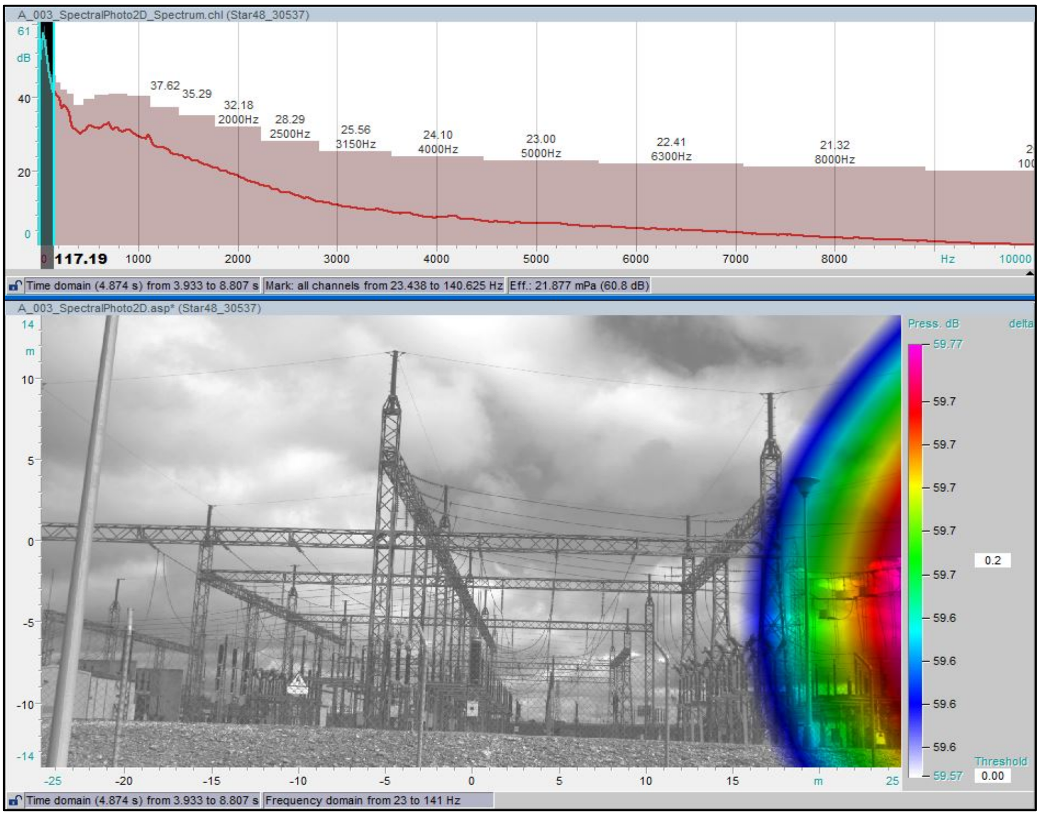

Figure 9 – Low frequency

3.2 Scenario 2: Industrial facility

The second scenario is a common industrial facility, with several noise sources outside like air compressors, ventilators, chimneys, pipes or big industrial refrigerators. The microphone array was situated 115 meters from the facility perimeter and, like in the other scenario, one road is very close; in this case, between the target and the microphone array and with a lot of traffic. Again, we use the spectrogram (Figure 10) to be sure that there are no trucks or cars contaminating our measurements. In addition we have all the photos taken during the acquisition, so we can also check if some vehicle was captured in the photo.

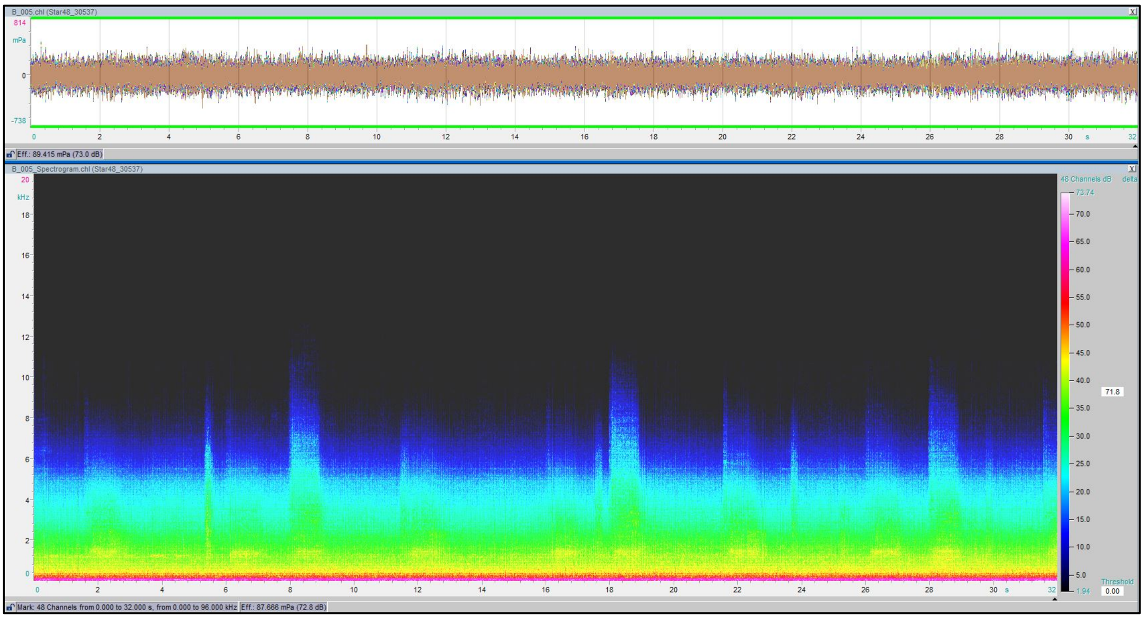

Figure 10 – Spectrogram

The spectrogram shows a continuous noise, according with the facility workflow, except some repetitive events visible in the spectrogram between 5 kHz and 10 kHz. These events are sounds from air liberations and are very difficult to hear from the acquisition position because the high background noise. Selecting these areas from the spectrogram, this means integrating the time selections between the frequencies range set, the noise source is clearly identified near to a big air compressor (Figure 11). Analysing the frequency range between 750 Hz and 5.600 Hz the main source is focused on one of the chimneys from the right side (Figure 12). The frequency peak on 590 Hz comes from one of the chimneys air extraction unit from the left area of the facility (Figure 13).

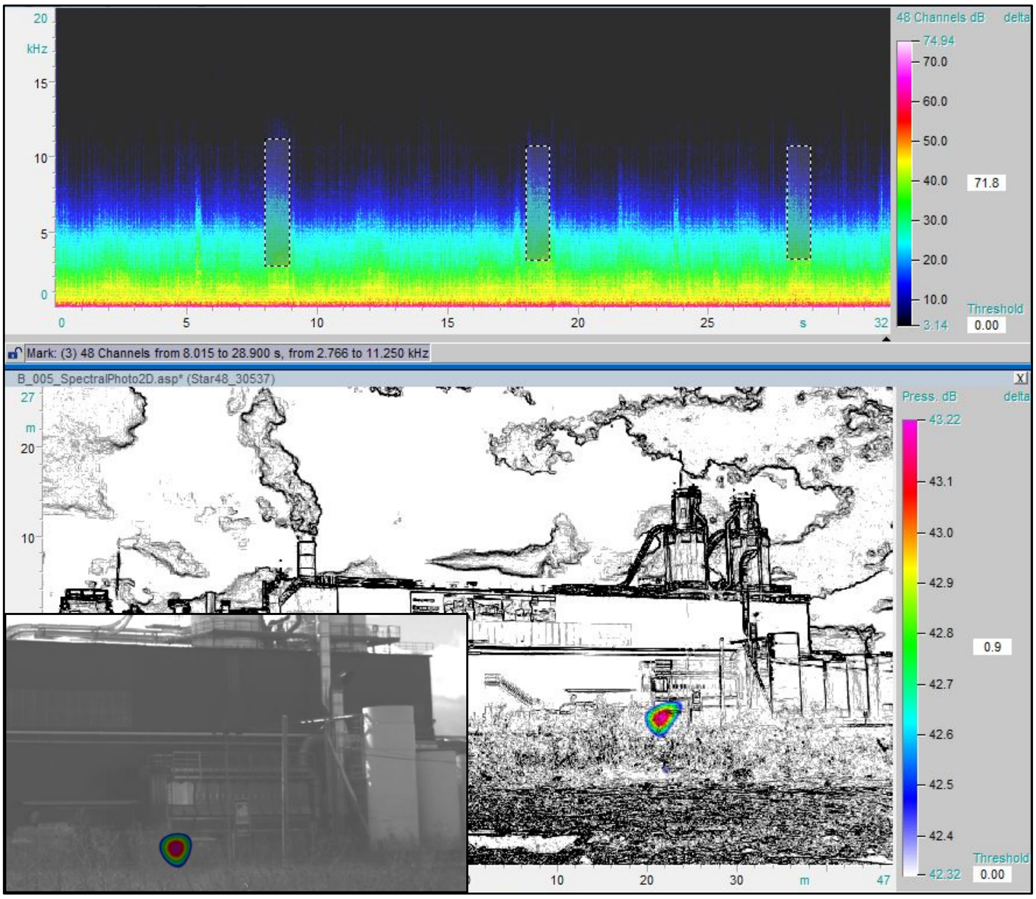

Figure 11 – Events selection in the spectrogram.

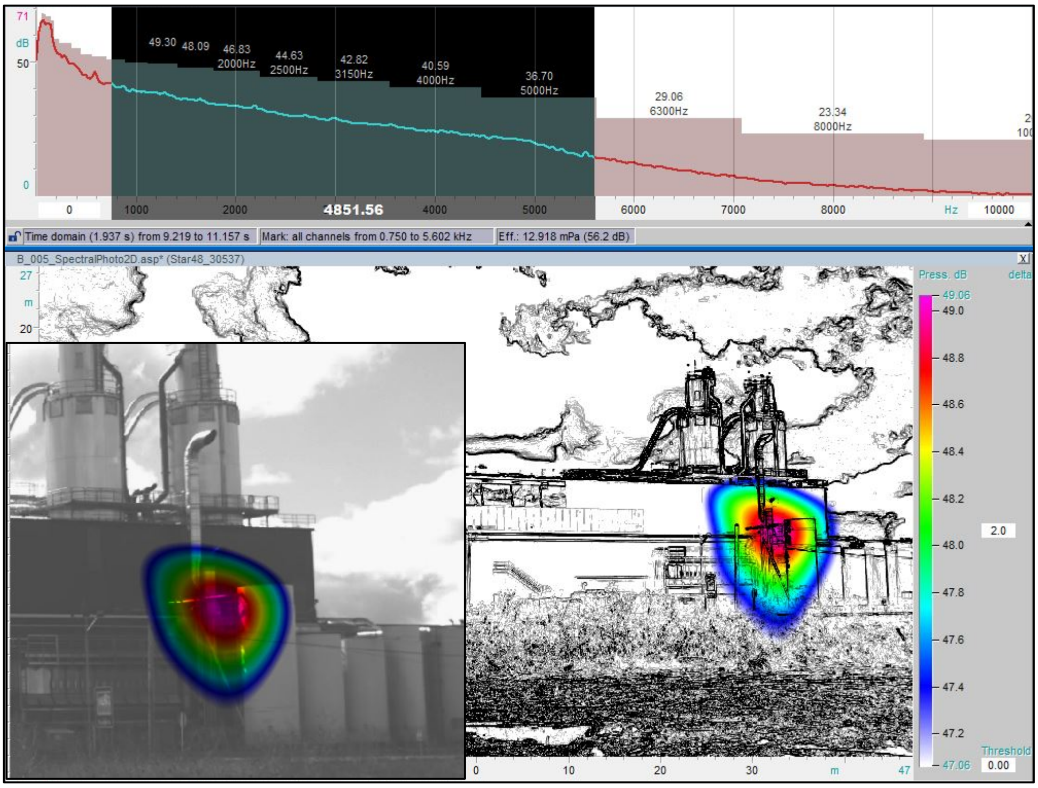

Figure 12 – Frequency range focus on one chimney

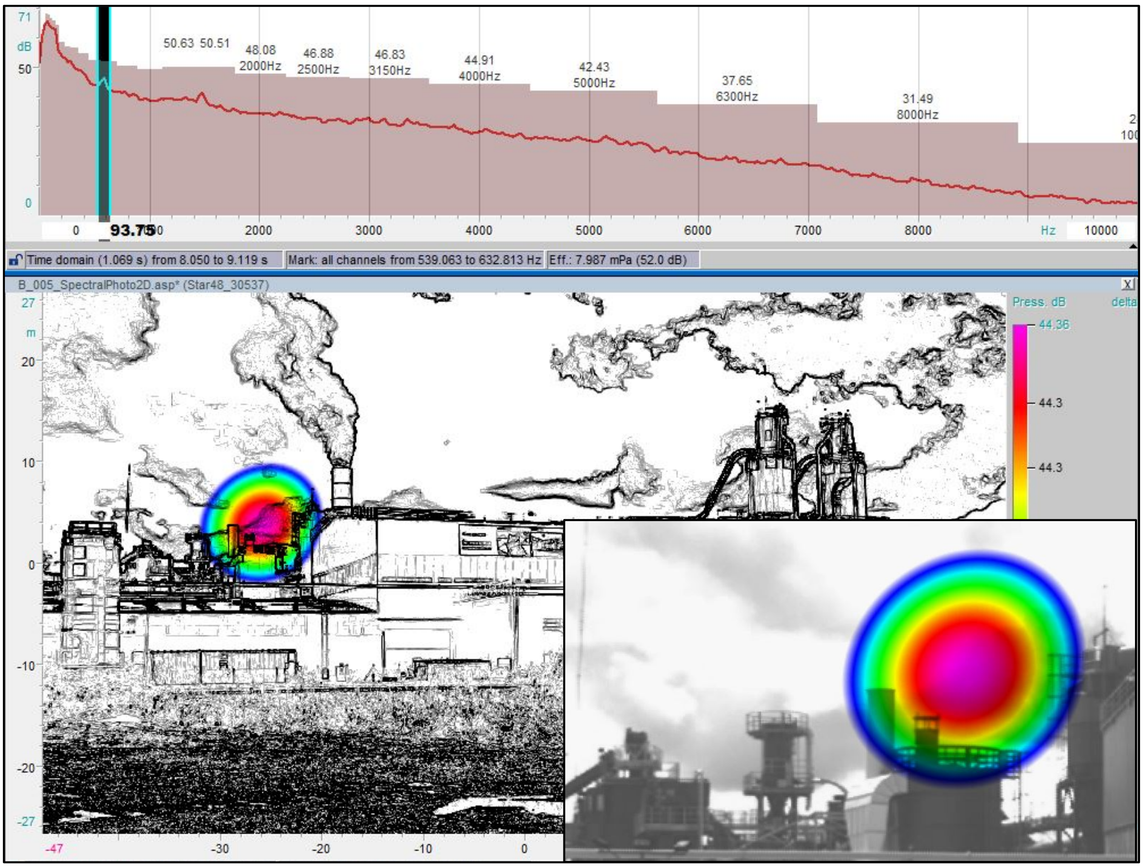

Figure 13 – Frequency peak from air extractor unit

4. CONCLUSIONS

Noise analysis in industrial facilities is not an easy matter, several noise sources, different workflows and external noise sources out of our interest make the interpretation a difficult task. The large amount of work time related to traditional methods is reduced when using the Acoustic Camera: in a short time the main and secondary noise sources become visible. Not only the levels, even more importantly, the frequency footprint that characterizes each one.

The results provide useful information for taking decisions about noise control when required. It is easy to identify the main noise problems and plan accordingly. When noise reduction solution is necessary, the frequency information of the noise source provide the key for developing or implementing and adequate solution.

This frequency characterization can also be used for maintenance, predicting catastrophic failures or control and monitoring by comparing different noise footprints OK/NOK.

Two complex scenarios are checked on this publication with several noise sources, high background noise and traffic noise around the area; using noise localization and frequency analysis in the acoustic photos, the noise sources can been localized analysed with high accuracy.

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