New method for measuring sound absorption coefficients in an industrial hall

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Predicting the sound pressure level at a workplace requires in-situ characterization of the facings. This work describes a new method for the measurement of the sound absorption coefficient of flat panels present in industrial halls. In such room, it is necessary to separate the echo coming from the studied panel from the others due to the entire reverberation. This separation has been achieved in space by an acoustic array and in time by an impulse sound source. The array processing uses a multi-polar weighting to achieve a directivity constant with frequency and with attenuated side lobes. This weighting requires a limited number of microphones. The impulse source has been designed using the inverse impulse response from the emission system (equalizer, amplifier, and loudspeaker). This inverse filtering technique allows equalizing the response of the emission system in order to radiate very short pulses. Sound absorption coefficient of several flat facings have been evaluated by mean of this new device in a semi-anechoic chamber and in an industrial hall designed for testing. The results show a good agreement with others techniques except at low frequencies for which the array length is too small and the absorption coefficients too low.

1 Introduction

To improve the acoustic treatment of facings and provide appropriate solutions for noise control at workplace, the development of methods of acoustic characterization walls present in industrial halls is necessary. This work concerns the development of a new experimental system to measure the acoustic absorption coefficient of these facings. The system must be transportable and avoid the problem of acoustic reflections due to the reverberant acoustic field in workplaces. It contains an impulse source and an acoustic array (receiving system) which has been optimized so that the major part of the received acoustic energy arises from a sufficient representative surface of the studied facing. This array directivity is independent of frequency with a narrow main lobe and attenuated rear lobes. It uses a multi-polar weighting since the associated directivity can be carried out with a limited number of microphones [1,2]. The sound source is impulsive enough to properly distinguish the echoes received by each sensor of the array. This must closely approximate a point source and have a fixed position with respect to the centre of the acoustic array during the measurement [3]. In order to test the experimental device (impulse sound source and acoustic array) in its entirety, measurements of sound absorption coefficients have been carried out in a semi-anechoic room and in an industrial hall.

2 Receiving system

The acoustic array is constituted by sensitive microphones which are assembled to localize the incident signal. To obtain a constant directivity diagram according to the frequency, it is possible to align several differently spaced out microphones. The most spaced out sensors give a constant directivity in low frequencies whereas those who are close guarantee a constant directivity in high frequencies. If the number of sensors increases, the main lobe is constant in frequency and narrow in width [4]. We notice however that in very low frequencies, the main lobe of this type of antenna finally widens. One of the methods allowing obtaining very narrow directivity in low frequencies uses the Schelkunoff weighting [5]. For this kind of weighting [1], secondary lobes are very weak in low frequencies but less weak in high frequency. The main lobe widens slightly in high frequencies. A linear array using multi-polar weighting has been designed to produce directivity with a narrow main lobe which is constant in frequency and side lobes rejected [1].

2.1 Multi-polar weighting

In the acoustic far field hypothesis of a point source, the spatial gradient of acoustic pressure is :

$$\frac{\partial p}{\partial x} = -jk\frac{A}{4\pi r}\cos(\theta).e^{j(\omega t - kr)} \qquad (1)$$

The spatial gradient estimation of the acoustic pressure can be obtained by using the acoustic pressure difference of two sensors :

$$\frac{\partial p}{\partial x} = \frac{p_2 - p_1}{d} = -jk\frac{A}{4\pi r}\cos(\theta).e^{j(\omega t - kr)} \qquad (2)$$

p_1 and p_2 are, respectively, the acoustic pressure received at each unidirectional sensors spaced by d. ω is the cyclic frequency and k is the wave number.

The first order polarized pressure obtained for a dipole antenna output is :

$$p_{dip} = \frac{p_2 - p_1}{-jkd} = \frac{A}{4\pi r}\cos(\theta).e^{j(\omega t - kr)} \qquad (3)$$

Eq.3 shows that the directivity of an antenna with a first order multi-polar weighting vary like cos(θ). To obtain a more narrow directivity, it is necessary to increase the order i of the spatial derivation to reach a directivity evolving in $\cos^{i-1}(\theta)$. If the antenna is constituted by 5 sensors, then 4 spatial derivatives can be made to obtain a directivity in $\cos^4(\theta)$. The directivity becomes narrower as the number of spatial derivatives increases. However, if the number of spatial derivatives becomes too large, the resulting signal can be strongly affected by the increasing phase noise in sound pressure differences measured by the different sensors. The weaker the pressure gradients, the more should the relative phase between the sensors be precise. One must therefore be quite cautious when using higher order multi-polar weighting, in order to obtain narrower directivity.

To eliminate the back lobe in such kind of directivity, it is possible to use the cosine function properties. Cosine to odd powers is positive from $-\pi/2$ to $+\pi/2$ and negative from $-\pi$ to $-\pi/2$ and from $+\pi/2$ to $+\pi$. Whereas cosine to even powers is positive from $-\pi$ to $+\pi$. Since the absolute values of these two functions are close, their sum from $-\pi$ to $-\pi/2$ and from $+\pi/2$ to $+\pi$ is very weak. The N^{th} and the $N\text{-}1^{th}$ derivatives resulted from the successive spatial derivations (Fig. 1) are, respectively, odd and even function. By summing these two functions, the values, in the directivity diagram, corresponding to the angles in $-\pi$ to $-\pi/2$ and $+\pi/2$ to $+\pi$ ranges are almost suppressed (see Fig. 2).

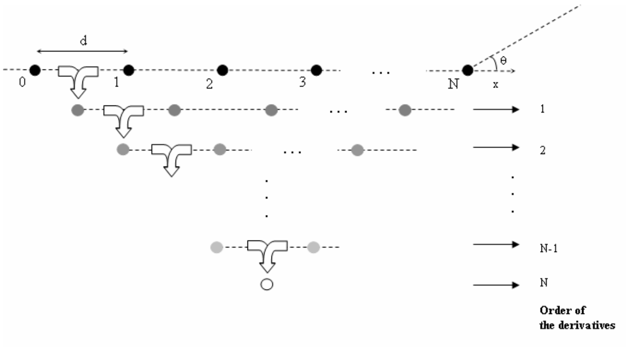


Fig.1 Principle of the successive spatial derivatives for N+1 sensors

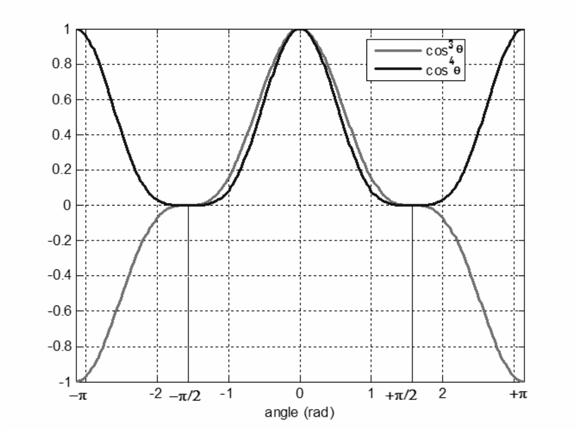


Fig.2 Cosine to 3 and 4 powers functions

But in this range however, the moduli associated to the N^{th} derivative and one of the $N\text{-}1^{th}$ derivatives are not exactly identical. A weak residue resulting from the sum will constitute the back lobe. The successive derivation technique leads to two derivatives of order N-1 (Fig. 1). In that case, we choose to add the arithmetic average of the two derivatives of order N-1 with the last derivative of order i. Fig. 3 shows the theoretical directivity diagram resulted from this technique, using a 5 sensors antenna with 2.5 cm spacings.

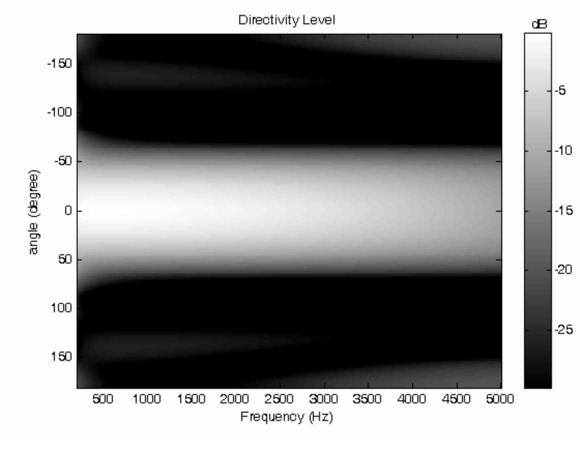


Fig.3 Theoretical directivity diagram for a linear antenna constituted of 5 microphones spaced out by 2.5 cm – Multi-polar weighting

Weak secondary lobes are present in the case of the multi-polar weighting. We notice that the main lobe remains narrow but however its level decreases with the frequency.

2.2 Improvement of multi-polar weighting

The error ε committed by the finite difference between both sensors is thus a cardinal sine function :

$$\varepsilon = \operatorname{sinc}\left(k\frac{d}{2}\cos(\theta)\right) \quad (4)$$

This error is maximum in the microphones alignment direction ($\cos(\theta) = \pm 1$). It is thus necessary to correct the spatial pressure gradient (Eq. 3) by taking into account in the calculation this corrective function ε. This error is also noticed in the principle of the acoustic intensity measure between two microphones [6].

Using this technique, we found that the error committed on the i^{th} derivative is generally a cardinal sine in the power i [7]. For a multi-polar weighting of order 4 corresponding to an acoustic antenna with 5 microphones, the error committed on the last derivative estimation is :

$$\operatorname{sinc}^4\left(k\frac{d}{2}\cos(\theta)\right) \quad (5)$$

We have demonstrated that the error committed on the estimation of the third spatial derivative of the acoustic pressure to the central sensor is [7] :

$$\operatorname{sinc}^3\left(k\frac{d}{2}\cos(\theta)\right) \times \cos\left(k\frac{d}{2}\cos(\theta)\right) \quad (6)$$

We took into account these corrective functions (Eq. 5) and (Eq. 6) for the microphones alignment direction (θ=0°) into the directivity calculation. To improve the secondary lobes attenuation, we constituted from the two $i\text{-}1^{th}$ derivatives and the i^{th} derivative an odd function. By summing this odd function with its absolute modulus we obtained canceled values for angles included between π/2 and 3π/2. We noted that this technique involves a widening of the main lobe level in the theoretical directivity diagram. This technique allows improving considerably the multi-polar weighting directivity. Fig. 4 shows the theoretical directivity diagram obtained for a linear antenna constituted of 5 sensors spaced out by 2.5 cm with this technique. The secondary lobes are rejected to more than 30 dB.

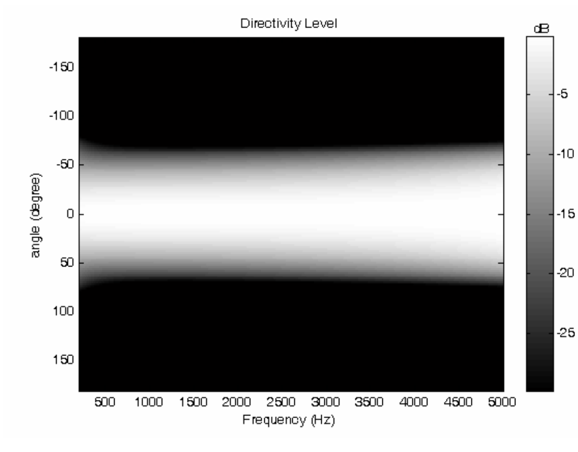


Fig. 4 Theoretical directivity diagram for a linear antenna constituted of 5 microphones spaced out by 2.5 cm – Improved Multi-polar weighting

2.3 Experimental validation of the directivity obtained with the improved multi-polar weighting

The receiving linear antenna is composed of calibrated microphones with mutual phase mismatches that do not exceed 0.05 degrees. This phase difference is very weak but still enough to cause, at low frequencies, instabilities at the main lobe level. To avoid this problem and to produce a constant level in a wide frequency range, several sub-arrays are used. The final receiving system contains 4 sub-antennae, each using 5 sensors whose spacings are multiples of 2.5 cm. These arrays use microphones in common and have exactly the same centre. Such a system has been developed and enables to cover a wide frequency range extending, approximately from 150 Hz to 5000 Hz. It contains, on the whole, 13 sensors. The microphone spacings for each sub-array are 2.5 cm, 5 cm, 10 cm and 15 cm. The array is 60 cm long (Fig. 5).

The experimental validation of directivity has been carried out in a semi-anechoic room. In the final device, the array will be used for measuring the sound absorption coefficient of flat panels subject to normal sound incidence. Consequently, the directivity validation has been achieved with the point source position close to the array centre. Fig. 6 shows respectively the theoretical (a) and the experimental (b) directivity diagram produced by the receiving system using multi-polar weighting with a point source placed at 1.5 m of the array centre. The experimental diagram is comparable to the theoretical one. The multi-polar weighting provides a constant main lobe versus frequency. The secondary lobes are efficiently attenuated by more than 30 dB.

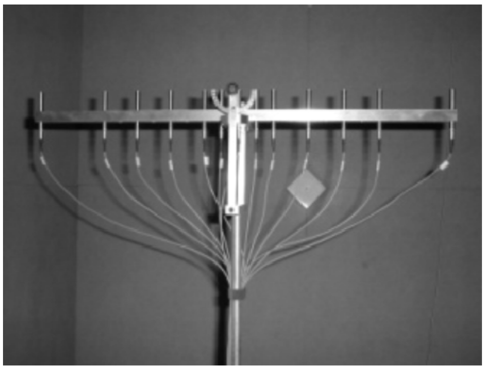
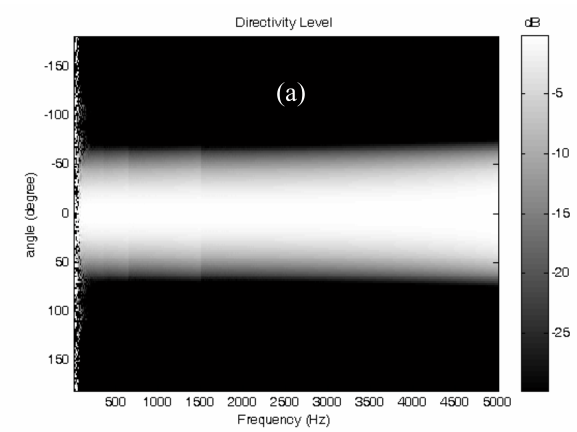


Fig. 5 Picture of the receiving system



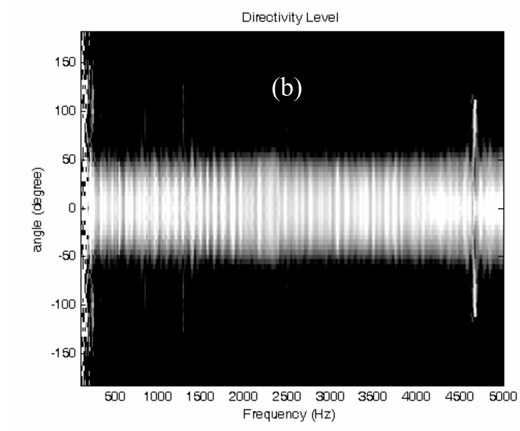


Fig.6 Theoretical (a) and measured (b) directivity diagram for the receiving system – Improved Multi-polar weighting

3 Development of an impulse sound source

The impulse source has been designed with the inverse filtered impulse response of an emitting system. This inverse filtering technique has been used to calculate the driving signal necessary to equalize the response of the emission system in order to radiate short pulses [8]. The transfer function H(f) of the emitting system has been measured in free field conditions with a pseudo-random binary Sequences of Maximum Length as driving signal ($16383=2^{14}-1$ samples). The microphone (B&K type 2669) has been positioned at 80 cm from the loudspeaker (Fig. 7). The impulse response h(t) of the emitting system is shown in Fig. 8 (black line). We can observe a long oscillating impulse response which is not short enough to allow several echoes separation. In a first time, the MLS driving signal has been filtered with the inverse of the emitting system's proper response. But the loudspeaker can not radiate sound energy in low frequencies, consequently the magnitude of the inverse impulse response possesses too high values in this frequency domain. To avoid destroying the loudspeaker in low frequencies because of these high values, the inverse impulse response has been filtered with a high-pass filter (frequency cut-off fixed at 100 Hz) in a second time. Fig. 8 (grey line) shows the short impulse response of the emitting system obtained with the filtered inverse impulse response of h(t) inserted at the output of the MLS signal source.

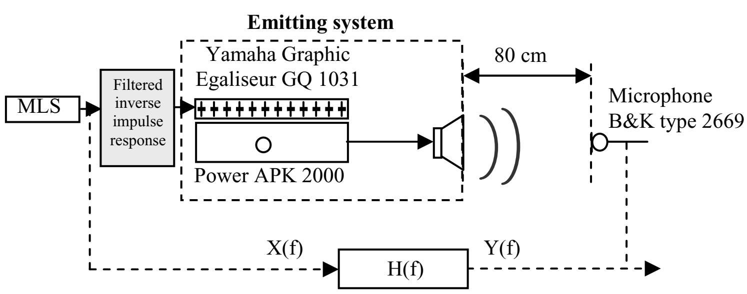


Fig. 7 Measurement of the transfer function of the emitting system in free field with the filter

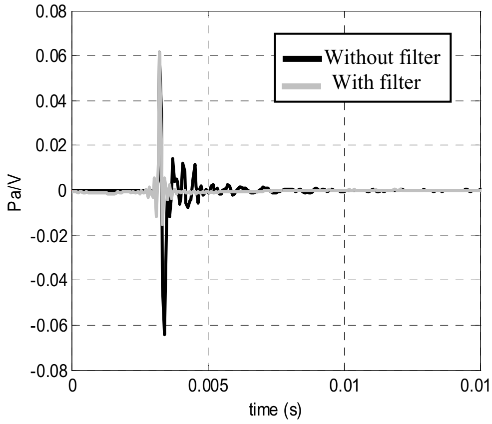


Fig. 8 Impulse response of the emitting system with and without filtering

4 Measurement of sound absorption coefficient of flat panels with the receiving system and the impulse sound source in an industrial hall

Measurements of sound absorption have been carried out, using the new device, in an industrial hall designed for testing. The reverberant conditions make the measurement more difficult. The loudspeaker is placed at d_2=1.5 m of the centre array and at d_1=3m of the studied panels (Fig. 9). Different flat facings of total sample size of 3×3 m^2 were investigated :

- mineral wool panels of 50 mm in thickness,
- fibralith panels of 30 mm in thickness.

The array directivity allows isolating the incident signal from the impulse source, by rotating the main lobe of the directivity by 180°. For that purpose, the sign of the finite difference of the sound pressure in the multi-polar weighting technique must be inverted [3] (Fig. 9 position 1). The echo coming from the investigated cover is observed by the array using the positive finite differences in the multi-polar weighting (Fig. 9 position 2).

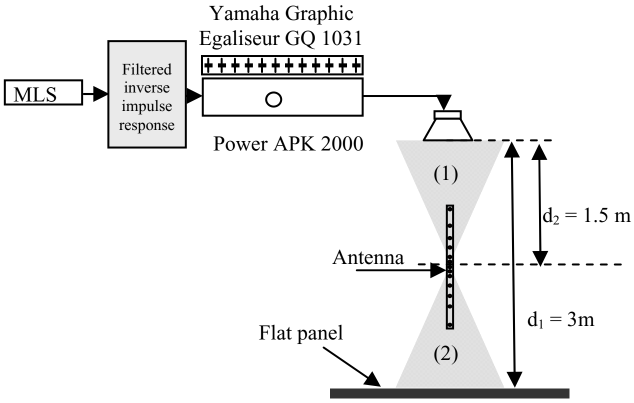


Fig. 9 Measurement of sound absorption of flat panels

The measurements have been carried out near several reflecting structures (3 or 4 m). All the planers which happened to be found in a radius of 3-4 m around the cover were turned off.

Fig. 10 (a) shows the impulse response of the signal received by the array central microphone. We can clearly see the incident impulse signal, the echo coming from the studied facing and the reverberation. In Fig. 10 (b), we notice that the directivity of the acoustic array insures a spatial filtering that leads to an attenuation of the reverberation.

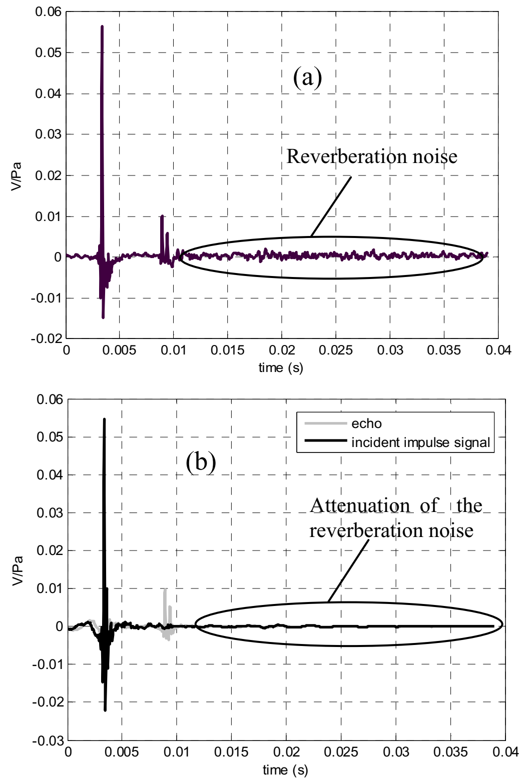


Fig.10 (a) Impulse response received by the central microphone of the array – (b) Impulse response received at the array centre with multi-polar weighting – Fibralith sample

This spatial filtering technique enables to separate the different echoes (Fig 10 (b)). Consequently, a time window W(t) with a sufficiently large width can be used [9].

Fig. 11 shows the sound absorption coefficients α(f) of the fibralith panels (a) and the mineral wool (b). These measurements have been compared with those carried out previously in a semi-anechoic room.

These coefficients have been deduced from the measured sound reflection coefficients R(f).

$$R(f) = \frac{F[p_r(t).W_r(t)]}{F[p_i(t).W_i(t)]} \cdot \frac{d_1 + d_2}{d_1 - d_2} \tag{2}$$

$$\alpha(f) = 1 - |R(f)|^2 \tag{3}$$

$F[p_r(t).W_r(t)]$ is the Fourier transform of the measured reflected impulse $p_r(t)$ weighted by its time window $W_r(t)$.

$F[p_i(t).W_i(t)]$ is the Fourier transform of the measured incident impulse $p_i(t)$ weighted by its time window $W_i(t)$.

The results show that it is difficult to correctly measure the weak values of sound absorption. Indeed, in low frequencies (below 250 Hz) we obtain erroneous results because the estimated sound absorption coefficient can present negative values. Whatever the frequency domain, the absorption measurement is error sensitive when the studied cover is very reflecting. In this case, the frequency response of the echo, weighted by the distances ratio, is similar to that of the incident signal source : the cross-spectrum becomes very sensitive to the measurement noise.

For absorbent materials, the difference between the incident and the reflected signals is quite large, the cross-spectrum and consequently, the module of the acoustic reflection coefficient are less sensitive to the measurement noise.

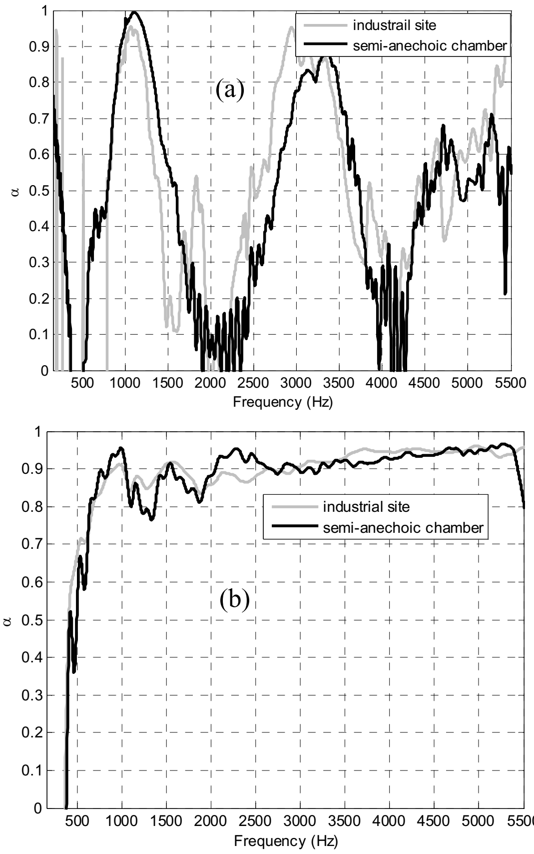


Fig. 11 : Measured values of the sound absorption coefficient for fibralith panels (a) - mineral wool of 5 cm in thickness (b)– Semi-anechoic chamber

As a matter of fact, the sound absorption in low frequencies cannot be correctly estimated because of the small dimension (60 cm) of the array compared to the wavelength. In other words, it would be suitable in this case, to use a longer array. This, however, would require an array too long to be considered as portable. It is also worth to mention that the impulse system includes a loudspeaker that cannot efficiently radiate sound at frequencies below 100 Hz.

5 Conclusion

Measurements results of the acoustic absorption coefficient in the industrial site come out to be quite similar to those obtained in the semi anechoic room. The problem of the noise due to the reverberation is solved with the array directivity, the incident impulse signal and the reflected one are perfectly separated.

In a another work, the device has been implemented in unfavourable conditions : in the presence of reverberation and a powerful disturbing source near the array. These conditions and the results are described and commented in a future published article [3]. The spatial filtering and the cross-correlation function, used to determine the impulse response at the array centre, lead to a considerable attenuation of the noise emitted by spurious sources near the device.

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