

# Characterisation of car equipments by Real-Time Nearfield Acoustic Holography

M.-H. Moulet<sup>1,2</sup>, J.-C. Le Roux<sup>1</sup>, J.-H. Thomas<sup>2</sup>

1: Centre de Transfert de Technologie du Mans, 20 rue Thalès de Milet – F – 72000 Le Mans

2: Laboratoire d'Acoustique de l'Université du Maine, rue Aristote – F 72085 Le Mans

**Abstract:** Localisation of non stationary or moving sources can be performed using Real-Time Nearfield Acoustic Holography (RT-NAH). This technique consists in reconstructing the acoustic fields using a time domain description of the problem. In this paper, a 110 microphone array is used to study the characteristics of the method: firstly, in terms of SNR, it is possible to localise correctly sources beyond a threshold of approximately 8dB. Secondly, the method allows us to localise correctly two close sources as long as the distance between them is greater than an array step. In the case of a smaller distance, two sources are found but not at the exact real positions. Finally, the study of moving sources in a reference case (moving speakers) shows that sources are correctly localised during the motion. Moreover in a real car equipment case (rear wiper noise), a source is localised on the wiper in the forward motion but not in the backward one.

**Keywords:** Real time acoustic holography, moving sources, rear wiper noise.

## 1. Introduction

Acoustic holography is a measurement technique used to localise acoustic sources from pressures measured with a microphone array nearby the source plane. This technique, called Near-field Acoustical Holography (NAH) was introduced in the 80's [1], and is mainly dedicated to stationary sources which are static. In order to study non-stationary sources or moving sources, some derived methods have been developed: the Time Domain Holography [2], [3] and the Transient Method [4]. These two methods are "time domain" extensions of the standard NAH. In this paper, another formulation is used: the real-time acoustic holography (RT-NAH). This technique consists in the reconstruction of the acoustic field using a time domain description of the problem [5]. This method was firstly validated for fixed reproducible sources in [6]. A 6x6 microphone array was then used in four steps.

In this paper, after recalling the method, its characteristics in terms of signal to noise ratio and discrimination of two close sources are studied. Then the method is applied to two cases of moving sources: a reference configuration with known

sources and a real car equipment configuration (a rear wiper).

## 2. Recall of the method

### 2.1 Direct problem

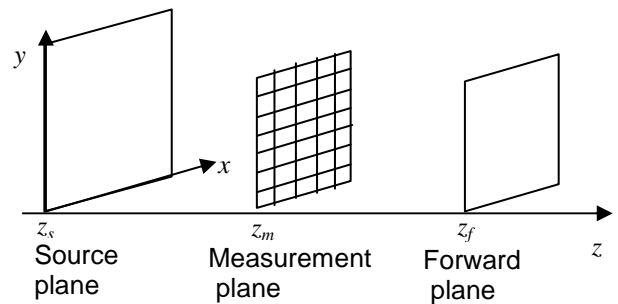


Figure 1: Geometry of the problem

The direct problem in acoustic holography consists in propagating the acoustic field  $p(x, y, z_m, t)$  measured from the plane  $z=z_m$  towards the forward plane at  $z=z_f$  ( $z_f > z_m$ , see figure 1). It has been shown [7] that this propagation can be done by convolving each component of the instantaneous wavenumber spectrum  $P(k_x, k_y, z_m, t)$  with a time domain impulse response  $h$ :

$$P(k_x, k_y, z_f, t) = P(k_x, k_y, z_m, t) \otimes h(k_x, k_y, \Delta z, t) \quad (1)$$

where  $P(k_x, k_y, z_m, t)$  is obtained from the two-dimensional spatial Fourier transform of the pressure field  $p(x, y, z_m, t)$  along  $x$  and  $y$  axis,  $k_x$  and  $k_y$  are the wavenumbers along  $x$  and  $y$  axis, and  $\Delta z = z_f - z_m$ .

The expression of the impulse response  $h$  is reached by first considering the acoustic propagation described by the wave equation (2):

$$\Delta p(x, y, z, t) - \frac{1}{c^2} \frac{\partial^2 p(x, y, z, t)}{\partial t^2} = 0 \quad (2)$$

and secondly by solving the two-dimensional Fourier transform of this equation (3):

$$\frac{\partial^2 P(k_x, k_y, z, t)}{\partial z^2} - \frac{1}{c^2} \frac{\partial^2 P(k_x, k_y, z, t)}{\partial t^2} - (k_x^2 + k_y^2) P(k_x, k_y, z, t) = 0 \quad (3)$$

where  $P(k_x, k_y, z, t)$  is the instantaneous wavenumber spectrum obtained by spatial Fourier transform of the signal on the microphone array at each discrete sample time, and  $c$  is the sound velocity in  $\text{m}\cdot\text{s}^{-1}$ .

The expression of the impulse response  $h$  is then given by (4):

$$h(k_x, k_y, \Delta z, t) = h(\Omega_r, \tau, t) = \delta(t - \tau) - \tau \Omega_r^2 \frac{J_1(\Omega_r \sqrt{t^2 - \tau^2})}{\Omega_r \sqrt{t^2 - \tau^2}} \Gamma(t - \tau) \quad (4)$$

where

$\tau = \Delta z / c$  is the delay corresponding to the propagation distance  $\Delta z$ ,

$\Omega_r = 2\pi f_r = ck_r = c\sqrt{k_x^2 + k_y^2}$  is the transition pulsation,

$J_1$  is the first order Bessel function,

$\Gamma(t)$  is the heaviside function defined by

$$\Gamma(t) = \begin{cases} 0 & \text{if } t < 0 \\ 1 & \text{if } t \geq 0 \end{cases}$$

In (1), by replacing  $h$  with its expression from (4), the time dependent wavenumber spectrum in the forward plane  $P(k_x, k_y, z_f, t)$  is obtained. Computing the two-dimensional inverse Fourier transform of this wavenumber spectrum yields the instantaneous spatial pressure in the forward plane  $p(x, y, z_f, t)$ .

For a practical implementation, several methods to sample the impulse response were investigated [7]. The method, which gives the best result, numerically computes the convolution of the impulse response by a Kaiser-Bessel window using the trapezoidal formula.

## 2.2 Inverse problem

In the context of real time holography, the objective is to calculate the pressure field on the source plane  $p(x, y, z_s, t)$  from measurements made by the microphone array. In order to back-propagate this pressure field, it is therefore necessary to solve the inverse problem, that is to find the inverse impulse response  $h^{-1}$  operating in the time domain and fulfilling the following condition:

$$P(k_x, k_y, z_s, t) = P(k_x, k_y, z_m, t) \otimes h^{-1}(k_x, k_y, \Delta z_s, t) \quad (5)$$

where  $\Delta z_s = z_m - z_s$ .

The Fourier transform  $H^{-1}(f) = TF[h^{-1}(k_x, k_y, \Delta z_s, t)]$  shown in figure 2 allows us to explain more clearly the reconstruction process for a given point  $\{k_x, k_y\}$  of the wavenumber spectrum. The components with a frequency greater than the transition frequency  $f_r$  are propagative waves (their amplitude remains unchanged). The others components, which frequency is lower than  $f_r$ , correspond to the evanescent waves.

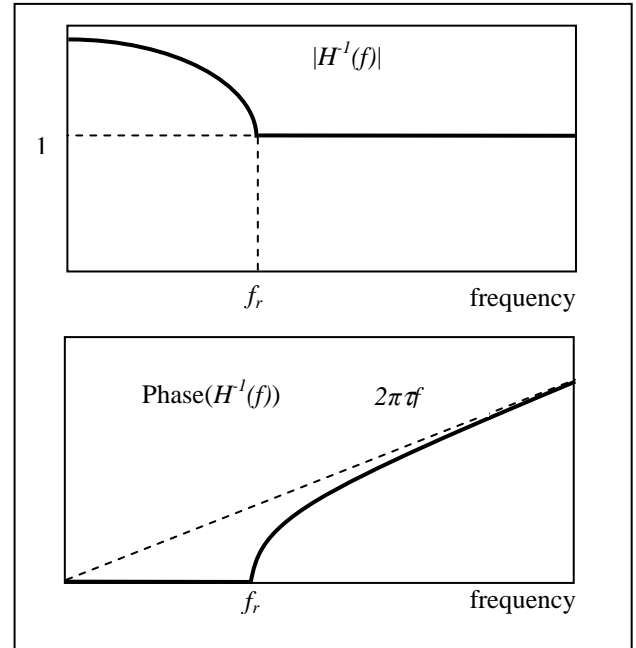


Figure 2: Amplitude and phase of TF  $[h^{-1}]$

A critical point in the reconstruction process concerns the amplification of evanescent waves. To limit the distortion due to the presence of noise, some techniques of filtering and regularization in the wavenumber domain are used in the case of NAH [8]. Concerning RT-NAH, some studies have been carried out on this topic [9]. For the experimental study presented here, the impulse response given by (4) is inverted using Wiener filtering [5].

## 3. Preliminary studies

### 3.1 Description of the experimental set-up

A large array of 110 (11x10) microphones is used to acquire simultaneously the time acoustic signals (figure 3). The distance between adjacent microphones is 8.5cm so that the total size of the array is 85cmx76.5cm. With such an array, frequencies included in the range [400-2000Hz] can be studied. The distance between the source plane and the array depends on the application. For all

measurements, an 8kHz sampling frequency is used.



Figure 3 : position of the array for rear wiper noise measurements

### 3.2 Influence of the SNR (Signal to Noise Ratio)

For this study, the configuration consists in a small speaker excited with a broadband noise filtered between 200Hz and 5kHz. The distance between the source plane and the array is 9cm. Measurements are performed for different levels of excitation. The SNR is calculated with the following relation :

$$SNR(dB) = 20 \log\left(\frac{L_1}{L_0}\right) \text{ with } L_k = \sqrt{\sum_{i=1}^{N_t} \sum_{j=1}^{N_m} p_j^2(t_i)} \quad (6)$$

where  $L_0$  is calculated for the background noise and  $L_1$  is calculated for the signal. In relation (6),  $N_t$  is the number of time samples and  $N_m$  is the number of microphones.

In figure 4, pressure fields calculated on the source plane and measured on the array plane, at an instant  $t$ , are presented for two SNR. The source position is indicated by the "x" symbol. The microphones lines and columns are represented by the dotted lines.

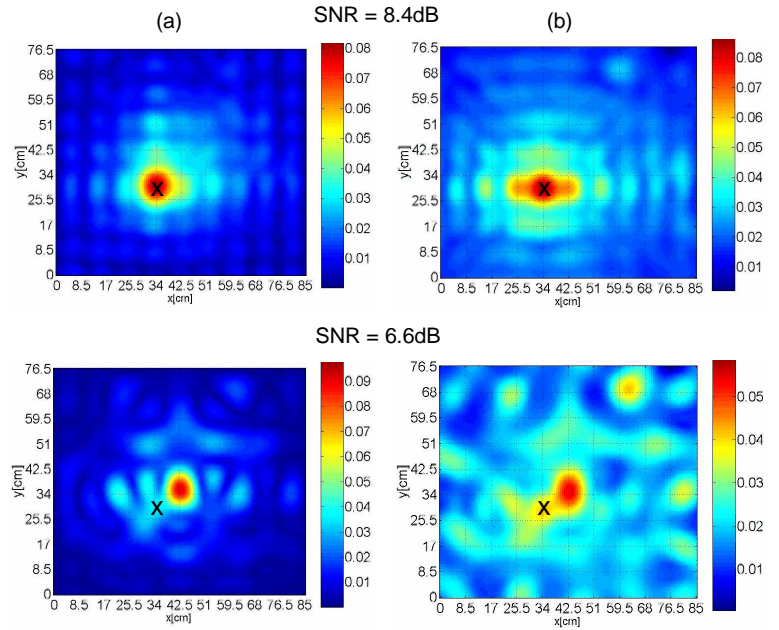


Figure 4: Reconstructed (a) and measured (b) pressure fields for  $\Delta z=9\text{cm}$  and for different SNR at a time event.

For a 8.4dB SNR the source position is localised correctly in the reconstructed pressure field. On the contrary, for a 6.6dB SNR, a maximum of pressure appears at a different position than the source location. In these conditions, SNR must be beyond a threshold of approximately 8dB to localise sources correctly.

According to the acoustic fields directly acquired from the array in the case of an 8.4dB SNR, backpropagating the measured fields to the source plane improves the localisation of the source.

### 3.3 Influence of the distance between two sources

For this study, the configuration consists in two small speakers (3cm diameter) which are excited by the same signal: it is a broadband noise filtered between 400Hz and 2kHz. The level of excitation is fixed.

In the case of industrial applications, it is not always easy to set-up the array close to the source plane. In order to characterise the influence of the distance between the source plane and the array, two distances are studied: 12cm and 24.5 cm. The 24.5cm configuration is more often used in industrial measurements.

For the 12cm configuration, the space between the centres of the two speakers, denoted  $d$ , varies from 4.5cm to 6cm. For the 24.5cm configuration,  $d$  varies from 6cm to 10cm. SNR levels are between 13dB to 18dB which satisfy the threshold of 8dB defined previously.

Results are presented in figure 5 for the 12cm configuration and in figure 6 for the 24.5 cm

configuration. The “x” symbols represent the positions of the sources.

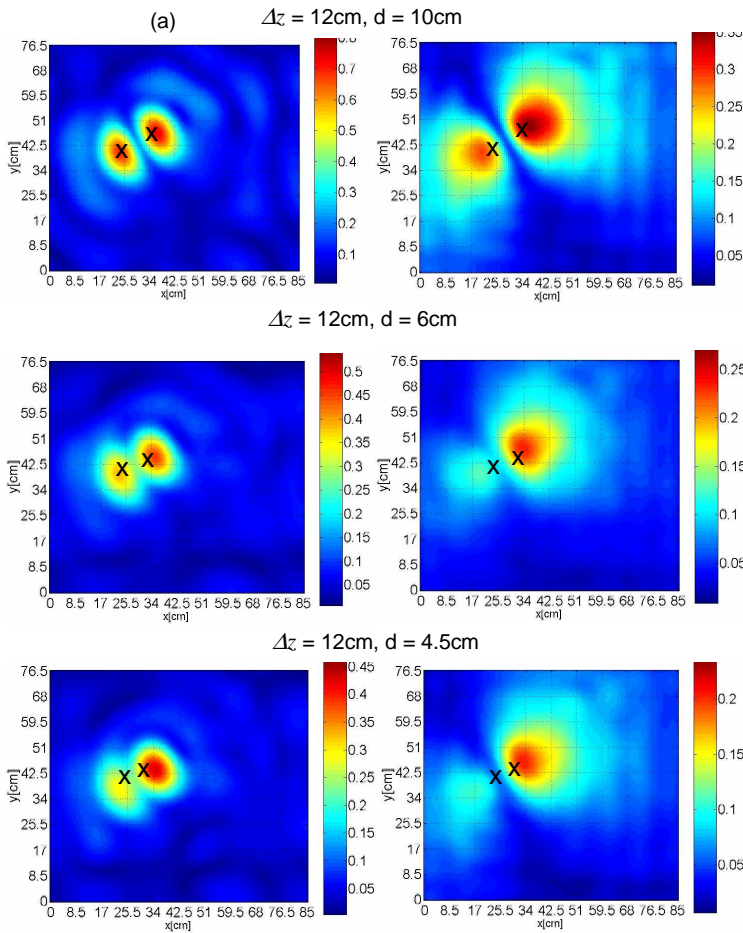


Figure 5: Reconstructed (a) and measured (b) pressure fields for  $\Delta z=12\text{cm}$  and different interspeakers distances at a time event.

Firstly, comparisons between pressure fields on source planes (figures (a)) and on array planes (figures (b)) show that sources are better discriminated on reconstructed pressure fields. It is especially true for a 24.5cm distance. It shows the importance and the interest of backpropagation.

For the 12cm configuration, when the distance between the two speakers is 10 cm long, source positions are localised correctly in the reconstructed pressure field. On the other hand, for distances of 6 and especially of 4.5cm, even though two sources are actually found, their positions are not the exact real ones. Because of the 8.5cm space between two adjacent microphones, even if the interpolation technique allows us to find two sources in the reconstructed pressure field, it does not permit to localise them correctly.

For the 24.5cm configuration, for different distances between the two speakers, two sources are

identified but their sizes are bigger than real sizes and locations are not the exact ones. It shows the difficulty of separating two close sources when the array is placed further from the source plane.

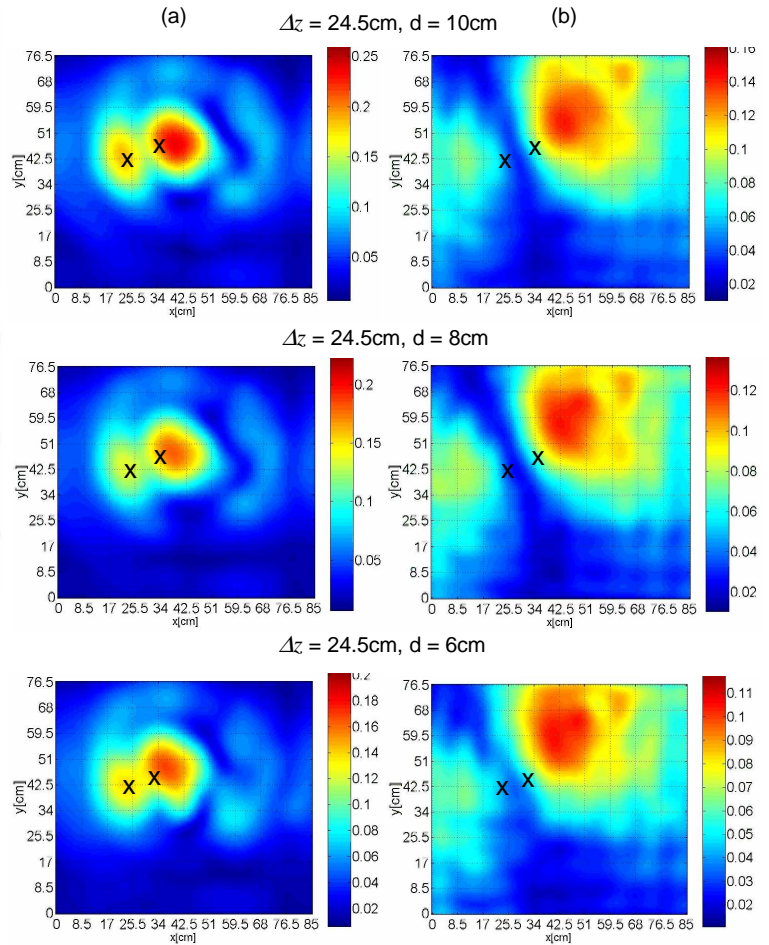


Figure 6: Reconstructed (a) and measured (b) pressure fields for  $\Delta z=24.5\text{cm}$  and different interspeakers distances at a time event.

### 3.4 Conclusion

In terms of SNR, it is possible to localise correctly sources beyond a threshold of approximately 8dB. About the discrimination of two sources, when the array is placed close to the source plane, the technique allows us to identify two sources even if the distance between them is less than an array step. However, sources are localised correctly if their distance is greater than an array step. When the array is placed further from the source plane, two sources are still reconstructed but their locations are not exactly defined.

#### 4. Study of moving sources

Until now, only fixed sources have been studied. Their excitation were varying in time (chirp or sweep type signals, for instance) but not their positions. The aim of this paragraph is to characterise moving sources with the RT-NAH technique.

##### 4.1 Reference configuration

###### 4.1.1 Description

The reference configuration consists in two speakers mounted on a rear wiper. Speaker S1 is mounted at 21cm from the rotation center of the wiper. Then, the distance between the two speakers is 13 cm (figure 7). The motion of the speakers is the rear wiper motion. The window is dry for this reference configuration so that the rear wiper noise is negligible compared to speaker noise. The excitation signal is a broadband noise filtered between 400Hz and 2kHz. This signal is identical for both speakers. The distance between the source plane and the array is 9cm in order to be in good backpropagation conditions (see 3.4).

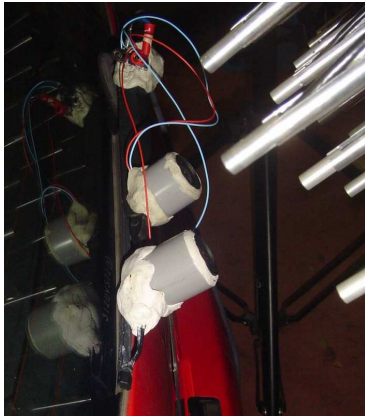


Figure 7: Positions of the two speakers on a rear wiper.

###### 4.1.2 Results

Results are presented in figure 8 for a level of excitation of 1.5Volt (SNR=29dB) and in figure 9 for a level of 0.12Volt (SNR=11.2dB). For each level of excitation, two different times are shown: the first one corresponds to a movement of the wiper from left to right and the second one corresponds to a movement from right to left. The arrows represent the sources trajectories.

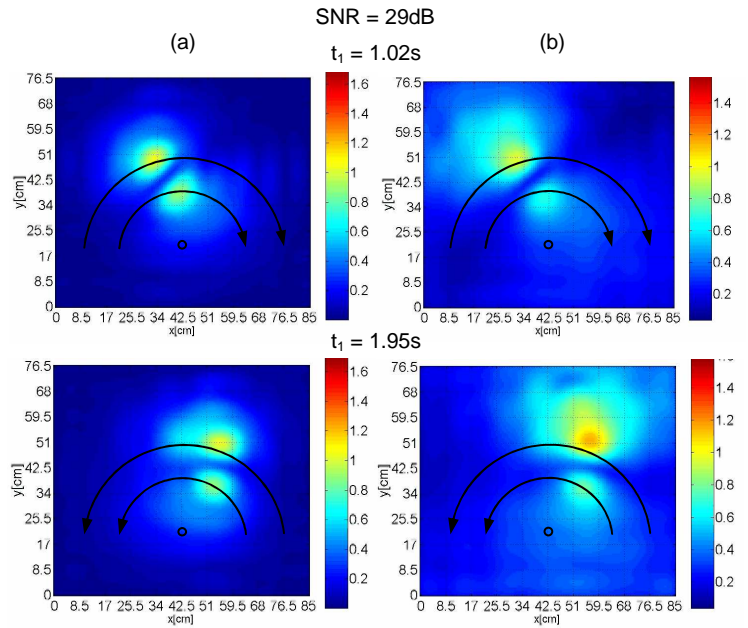


Figure 8: Reconstructed (a) and measured (b) pressure fields for two moving sources at two time events (SNR = 29dB).

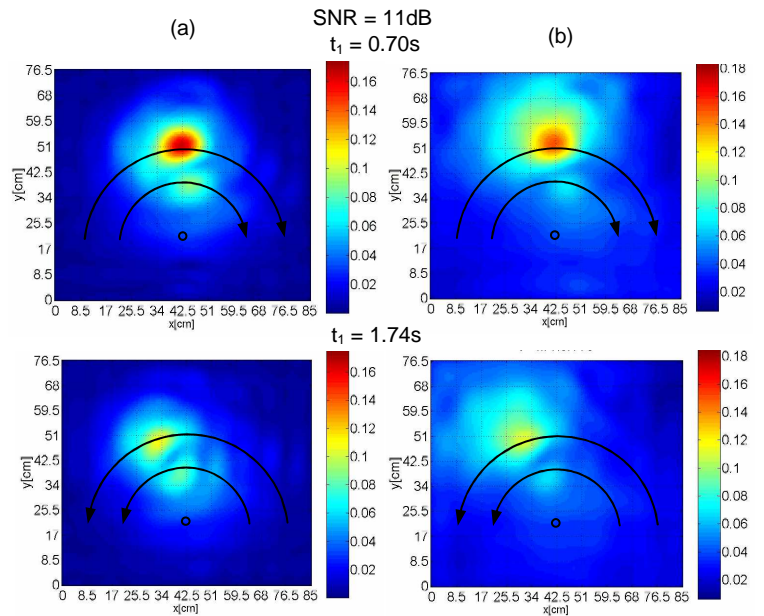


Figure 9: Reconstructed (a) and measured (b) pressure fields for two moving sources at two time events (SNR = 11dB).

For both SNR examples, both moving sources are clearly identify on reconstructed fields for backward and forward movements. Sources are also visible on the measured pressure field but their positions and sizes are worse defined.

## 4.2 Real car equipment configuration : rear wiper

### 4.2.1 Description

Finally in this paragraph, we focus on the measurement of the noise generated by a rear wiper of a car. This noise depends on the level of wetness of the window and is clearly not reproducible. A cycle takes 1.9s. An example of a cycle is given in the following.

### 4.2.2 Results

The pressure measured by the microphone in front of the centre of the wiper is presented in figure 10. The forward movement is clearly noisier than the backward one.

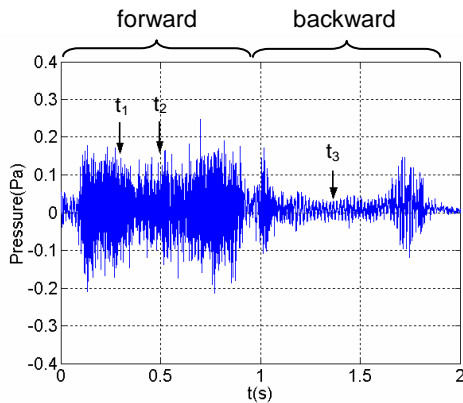


Figure 10: Pressure measured by one microphone of the array.

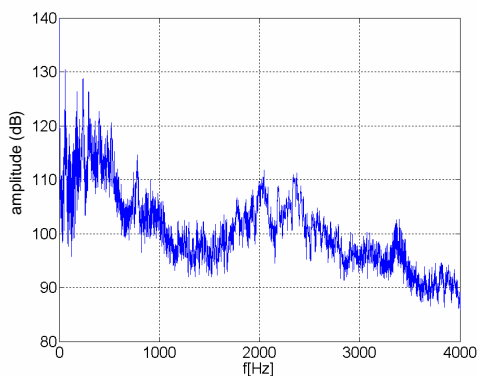


Figure 11: Quadratic average spectrum of the pressures measured by all the microphones of the array.

A Fast Fourier Transform on time acoustic signals on all the microphones of the array (see figure 11) shows that pressure levels are particularly high in the following frequency ranges [0-550Hz], [750-800Hz] and [2000-2400Hz].

Reconstructed and measured pressure fields are presented in figure 12 for the three instants of the cycle defined in figure 10. The first two correspond to

the forward movement and the third one corresponds to the backward movement.

When the wiper is going forward, a preponderant source can be localised at approximately 2/3 length of the wiper on the backpropagated field. No source can be identified on the direct measured pressure field. Concerning the backward movement, the SNR is too small to allow us to identify preponderant sources.

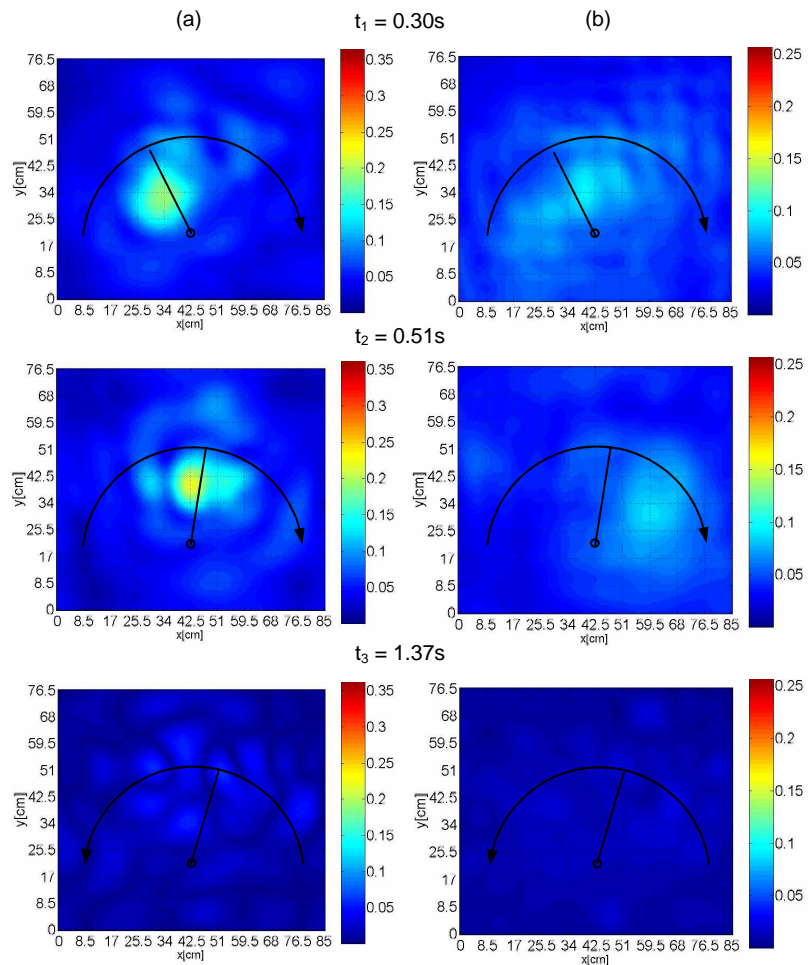


Figure 12: Reconstructed (a) and measured (b) pressure fields for a rear wiper motion at three different instants.

## 5. Conclusion

The Real-Time Nearfield Acoustic Holography (RT-NAH) technique is used to study different kinds of sources : fixed sources and moving sources. The study of fixed sources shows that in terms of SNR, it is possible to localise correctly sources beyond a threshold of approximately 8dB. Then the method allows us to localise correctly two close sources as long as the distance between them is greater than an array step. In the case of a smaller distance, two

sources are found but not always at the exact real positions. Concerning moving sources, they are accurately identified in the source plane. And for a rear wiper noise, we can clearly localise a source on the wiper in the forward motion and no source in the backward one. More experiments on industrial non stationary or moving systems will be performed in the following months.

## 6. References

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