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A LIGHT SOLUTION FOR ACOUSTIC SOURCE LOCALIZATION

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ABSTRACT
The article presents a spatial localization technique applied to MADRAS methodology. MADRAS is a software measurement assistant for automatic noise source recognition. It is based on a tree classifier using neural networks and spectral shape learning process. The localization module was developed and tested on a PC-based 4 channels analyzer. Complementary to signal processing approach, the spatial localization module is using a tetrahedral antenna, coupled with a commercial and versatile acquisition board ("Harmonie" platform). A possible application area of automatic localization of acoustic noise sources in a confined environment (localization of "squeaks & rattles" inside a vehicle cabin) is presented and discussed.

1 - INTRODUCTION
Since several years, noise source localization techniques have been developed at METRAVIB R.D.S. for military applications such as acoustic localization & identification of snipers ("PILAR" systems) or acoustic tracking of vehicle. Within this frame, some original methods have been set-up to accurately estimate the position of a noise source using only a restricted number of acoustic sensors (typically, four sensor disposed on a tetrahedral antenna). In parallel, 01dB has developed new tools for automatic classification of the acoustic noise signatures using various signal processing techniques. Due to the merging of both companies within the MVI Technologies Group, new application fields can now be envisaged for these methods as e.g. in the automotive industry area. Indeed, it turns out that large efforts have been devoted by car manufacturer in the recent years for the improvement of acoustic comfort inside vehicle cabin. Up to now, the problem was mainly focused on the reduction of the global noise level perceived by the driver or the passengers, and considerable improvement of commercial vehicles have been reached. The attention of acoustic designers of car manufacturer is now devoted to the elimination of transient noises, usually referred to as "squeaks and rattles", that may appears under some particular driving situations and that are synonymous of non-quality for the customers. For this purpose, some new processing techniques are required to accurately determine the origin of this transient noise from measurement inside the vehicle. A combined approach is proposed in the present article. The perceived noise source is characterized in terms of both temporal signature (using signal recognition primitives embedded in MADRAS) and spatial localization (using acoustic localization processing derived from PILAR systems).

2 - MADRAS METHODOLOGY
The MADRAS concept (Methods for Automatic Detection and Recognition of Acoustic Sources) has been developed within a European project (CEE DG12) led by 01dB [1]. It provides new tools for automatic classification of the acoustic noise signatures using various techniques such as: time-frequency & time-scale analysis, mathematical morphology, factorial data analysis and neural networks. All these tools have been embedded in a user friendly environment (PC-based software) so as to simplify and to ease their routine use. Acoustic localization module have been integrated in this environment, and can

be used in parallel to recognition facilities. This parallel use of both spatial localization techniques and noise source recognition primitives is a decisive advantage for the characterization of "squeaks & rattles", since it will ease the identification of the component responsible for the noise and make it more robust.

3 - DESCRIPTION OF THE SYSTEM

The hardware system is composed of a four channels ("Harmonie") PC-Card acquisition board connected to a set of four microphones. The sampling frequency may be adjusted up to 48 kHz for each channel. Once digitized, the four signals provided by the microphones are processed by the PC. Two strategies may be envisaged for sensor deployment, as depicted in fig. 1.

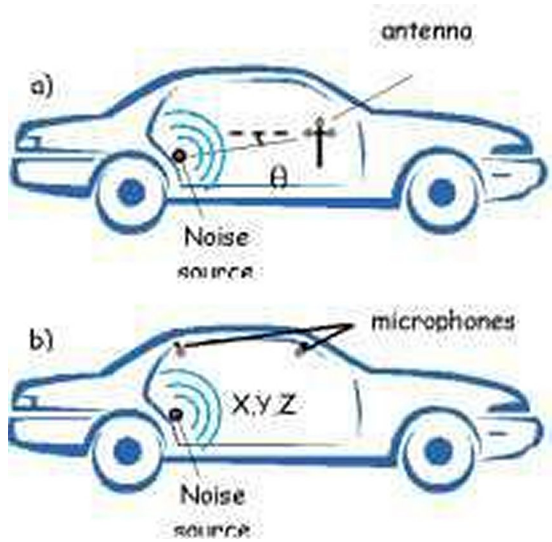

Figure 1: Possible sensor deployment inside car cabin.

In the first configuration (fig. 1a), the microphone are arranged in a compact antenna similar to the one shown on fig. 2. The compactness of this configuration is a decisive advantage for onboard measurements. However, due to the limited size of the antenna, only the direction of the source (defined by azimuth and elevation angles) can be estimated. In order to ease the interpretation of the results, the result of acoustic localization can be superposed on a digital picture provided by a video camera fixed to the antenna.

Figure 2: Example of acoustic antenna.

In the second configuration (fig. 1b), four sensors fixed to the roof of the car cabin are used for localization. The advantage is that the large spacing between microphones allows a complete estimation of noise source position (x, y and z co-ordinates). The counterpart is a less compact solution (although some solutions may be proposed to ease the deployment).

4 - ACOUSTIC LOCALIZATION MODULE

The principle of acoustic localization module is depicted on fig. 2. Due to acoustic propagation between the noise source and the antenna (or the set of distributed microphones), the signals recorded by each microphones are delayed copies of the transmitted signal. The delays are different for each microphone because of their different spatial position.

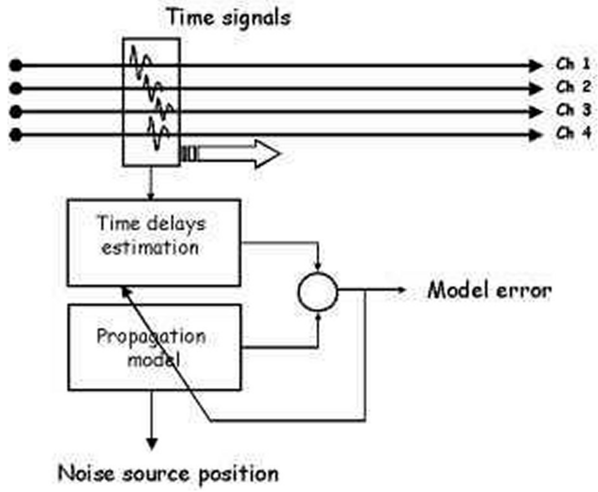

Figure 3: Principle of acoustic localization.

The localization algorithm is in fact very similar to acoustic beam-forming used in sonar systems, because it makes use of the measured delays to estimate the position of noise source. From the signals measured at each microphone, time delays are estimated and compared to theoretical values provided by a propagation model. Adjustments of model parameters are then performed in order to minimize the error between estimated and measured time delays. Once the model is tuned, the estimated position of noise source is returned. It should be noted that only one noise source position is returned by the proposed estimation scheme and, when several noise sources are present in the measured signal, only the most energetic one will be detected. Transient noise sources may however be separated by performing spatial localization on a sliding window as shown on fig. 3. Conversely, if the localization of stationary sound source is considered, the proposed spatial processing may be applied in the frequency domain as well.

5 - RESULTS ON SIMULATED DATA

Because onboard measurement have not been performed at the time the article is written, experimental results cannot be presented. However, some preliminary tests of the localization algorithm have been performed using simulated data. The microphone outputs are synthesized by delaying an original waveform according to the propagation distance from the source to the sensors. Effects due to multiple reflections inside the cabin are also taken into account, so as to evaluate the influence of a confined environment on the performance of localization.
Table 1 summarizes the average error on the estimate of noise source positions for both a free field propagation and a confined environment using either a compact microphone antenna or a set or distributed microphones.

	Compact antenna (fig. 1-a)	Distributed microphones (fig. 1-b)
Free field propagation	$\Delta\theta \approx 2$ °	$\Delta x \approx 10$ cm
	$\Delta\varphi \approx 4$ °	$\Delta y \approx 10$ cm
		$\Delta z \approx 10$ cm
Confined environment	$\Delta\theta \approx 10$ °	$\Delta x \approx 10$ cm
	$\Delta\varphi \approx 10$ °	$\Delta y \approx 10$ cm
		$\Delta z \approx 30$ cm

Table 1: Localization error on simulated data.

From this table, it appears that distributed microphones correspond to the most robust sensor configuration for confined environment, since localization error remains almost the same as for free field propagation. However, simulation results obtained for the compact antenna show that localization is also feasible using this kind of sensor deployment.

6 - CONCLUSION

A spatial localization technique, coupled to a hardware and software environment capable of automatic noise source recognition has been presented. Although the performance estimated on simulated data remain to be assessed on real applications, it is thought that such a combined approach of temporal and spatial characterization of acoustic noise sources may be of great help in a wide range of applications. In particular, this technique may be advantageously used for the monitoring of environmental noise, because spatial localization will provide unambiguous identification of the measured noise source.

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