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Time domain localization technique with sparsity constraint for imaging acoustic sources

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ABSTRACT

This paper addresses source localization technique in time domain for broadband acoustic sources. The objective is to accurately and quickly detect the position and amplitude of noise sources in workplaces in order to propose adequate noise control options and prevent workers hearing loss or safety risk. First, the generalized cross correlation associated with a spherical microphone array is used to generate an initial noise source map. Then a linear inverse problem is defined to improve this initial map. Commonly, the linear inverse problem is solved with an l_2-regularization. In this study, two sparsity constraints are used to solve the inverse problem, the orthogonal matching pursuit and the truncated Newton interior-point method. Synthetic data are used to highlight the performances of the technique. High resolution imaging is achieved for various acoustic sources configurations. Moreover, the amplitudes of the acoustic sources are correctly estimated. A comparison of computation times shows that the technique is compatible with quasi real-time generation of noise source maps. Finally, the technique is tested with real data.

1. Introduction

Many workers are exposed to high sound levels that may be harmful and lead to hearing loss or safety risk. Passive solutions have been developed to reduce noise emitted by acoustic sources based on acoustic panels, curtains, enclosures or damping materials. However, the first step in an acoustic diagnosis is to accurately localize the position of the noise sources in order to act at the right place. The goal of this study is to develop an acoustic tool to accurately and quickly localize acoustic noise sources and reflections.

Commonly, the dimensions of an industrial hall are large and the workers undergo the direct sound field and multiple reflections. Therefore, the source localization technique has to correctly identify all the source positions and reflections in order to adequately design and implement noise control solutions.

Acoustic intensimetry is a technique to localize noise sources [1]. The sound field around an object is scanned with a two-microphones probe in order to estimate the acoustic intensity. Then, the radiated acoustic power can be computed and can be used as input to ray tracing software to predict the sound field in a closed environment. In a workplace the noise sources are multiple and distributed, therefore it is impossible to scan all the volume. The main source positions have to be known *a priori*. Moreover, this technique is time consuming when the dimensions of the source are large.

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An alternative technique is to use an array of microphones associated with a source localization algorithm [2]. The goal is to compensate the time or phase delay between microphones in relation to a virtual scan point. The processing is performed either in the time or frequency domain. Frequency techniques use the cross spectral matrix of the microphone signals. The most common technique is beamforming [3]. Its main disadvantage is the poor spatial resolution at low frequencies. Deconvolution techniques have been developed to improve the resolution of the noise source map [4–8]. Recent works based on inverse methods with a l_1-regularization have shown good performances [9,12,13]. However, in a workplace, the noise sources are generally broadband so that the computational cost is large since the processing has to be done for each frequency. Alternative strategy has been proposed based on the average of the output of beamforming obtained from different microphone array locations [14]. Despite promising results, this strategy is difficult to implement in real situations.

The most common technique in time domain is the Generalized Cross Correlation (GCC) method which is based on the time delay between a microphone pair [10]. This time delay can be used to generate a hyperbola for the possible source positions over the scan zone. The intersection of all the hyperbolas (for all the microphone pairs) provides the source positions.

Noël et al. [11] have used the GCC associated to an inverse problem to localize source positions in an industrial hall. The solution of the inverse problem minimizes the difference between theoretical and measured cross-correlation functions. They obtained a noise source map with the angular energy flow received from each direction relative to the microphone array. The results are promising despite a small number of scan points and large computational cost due to the computation time of the global matrix.

The objective of this study is to propose a fast source localization technique which is able to detect the main source and reflections. Therefore, a minimization problem based on the GCC is proposed but with a different theoretical formulation and solver from Noël et al. [11]. Two different sparse representations with a l_1-norm minimization are used to solve the minimization problem. Section 2 describes the theoretical background of the proposed source localization technique. The performances of the proposed technique are compared in terms of source position detection, source level estimation and computation time with synthetic data in Section 3. Finally, the source localization technique is validated with experimental data in Section 4.

2. Source localization technique

2.1. Microphone array signal

An acoustic point source at location $\mathbf{r}_s$ generates a signal $s(\mathbf{r}_s, t)$ (with t the time) recorded by a set of M microphones ($m = 1, \ldots, M$) at location $\mathbf{r}_m$. Throughout the paper, bold letters denote matrices or vectors. The acoustic pressure signal p_m recorded by a microphone m in free field conditions is given by

$$p_m(t) = \alpha_m(\mathbf{r}_s)s(\mathbf{r}_s, t - \Delta t_{ms}) + v_m(t), \tag{1}$$

where $\alpha_m(\mathbf{r}_s)$ is the geometrical attenuation due to the propagation between the source and the microphone and $v_m(t)$ is an uncorrelated additive noise due to background or sensor noise. The Time of Flight (ToF) Δt_{ms} between the source s and the microphone m is defined from the Euclidean distance

$$\Delta t_{ms} = \frac{1}{c_0}\|\mathbf{r}_m - \mathbf{r}_s\|_2, \tag{2}$$

where c_0 is the sound velocity and $\|\cdot\|_p$ is the p-norm of a vector or matrix. The microphone array signal $y(\mathbf{r}_s, t)$ is given by the arithmetic mean of the microphone signals

$$y(\mathbf{r}_s, t) = \frac{1}{M}\sum_{i=1}^{M} p_i(t). \tag{3}$$

2.2. Time domain Beamforming

Classically, acoustic source localization or imaging is performed using the output power of the microphone array signal $y_e(\mathbf{r}_s)$ defined for a continuous signal by

$$y_e(\mathbf{r}_s) = \mathbf{E}\{y(\mathbf{r}_s, t)^2\} = \int_{-\infty}^{+\infty} \frac{1}{M^2}\sum_{i=1}^{M}\sum_{j=1}^{M} p_i(t)p_j(t)dt, \tag{4}$$

where $\mathbf{E}\{\cdot\}$ is the expectation value. The output power of the microphone array signal can also be written as

$$y_e(\mathbf{r}_s) = \frac{1}{M^2}\sum_{i=1}^{M}\sum_{j=1}^{M}(p_i \star p_j)(\tau), \tag{5}$$

where the product $(p_i \star p_j)$ corresponds to the cross-correlation function of two microphone signals at time lag $\tau = (\Delta t_{js} - \Delta t_{is})$ defined by