



Genetic optimisation of a plane array geometry for beamforming. Application to source localisation in a high speed train



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ABSTRACT

Thanks to its easy implementation and robust performance, beamforming is applied for source localisation in several fields. Its effectiveness depends greatly on the array sensor configuration. This paper introduces a criterion to improve the array beampattern and increase the accuracy of sound source localisation. The beamwidth and the maximum sidelobe level are used to quantify the spatial variation of the beampattern through a new criterion. This criterion is shown to be useful, especially for the localisation of moving sources. A genetic algorithm is proposed for the optimisation of microphone placement. Statistical analysis of the optimised arrays provides original results on the algorithm performance and on the optimal microphone placement. An optimised array is tested to localise the sound sources of a high speed train. The results show an accurate separation.

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1. Introduction

Noise pollution is a major concern for authorities. The World Health Organization has stated that daily exposure to noise leads to various health issues, such as stress increase, insomnia or cardiovascular problems [1]. Traffic noise is one of the major parts of noise pollution. A deeper knowledge of its sources is required to provide effective acoustic protection and noise reduction solutions. The sound sources are often characterised by their position on the vehicle, their spectra and their relative power.

Array processing has been widely used for imaging acoustic sources in the industrial context. It consists, essentially, in delaying and summing microphone signals to estimate the power coming from one point [2]. The obtained estimate is a convolution of the source distribution and a spatial filter, i.e. the array impulse response or the beampattern. Thanks to its easy implementation and its robustness compared to high resolution methods, the beamforming method is applied commonly for source localisation on trains [3,4], cars [5], airplanes [6] and also wind turbines [7].

In recent years, research in acoustic array processing has proposed many algorithms to improve the imaging results, such as deconvolution algorithms [9,10], CLEAN method [11] and Functional Beamforming [12]. These methods manage to reduce the sidelobes and increase the resolution of the localisation maps.

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For imaging moving vehicles or rotor blades, the beampattern is then focused on a plane moving with the vehicle to suppress the Doppler effect [8]. The array response is often considered as isotropic, i.e. it does not depend upon the focusing direction. This hypothesis allows deconvolution algorithms to be applied conveniently. This method has been successfully applied on airplanes [6] and wind turbines [7], where the sources are far from the array. In a context of near field propagation (e.g. for imaging trains and cars), the tracking of the sources implies large variation of the beampattern which isotropic property cannot be verified [13,14].

The performance of beamforming depends mainly on the sensor placement (i.e. on the array geometry) [15]. For a given frequency range, wide arrays are known to achieve accurate spatial localisation whereas closely spaced sensors provide low sidelobes [16]. Since these two properties lead to opposite designs, a trade-off has to be made to obtain optimal performance. A relevant placement for linear 1D array was achieved by minimising the redundancy of the inter-microphone distances [17]. The redundancy cannot be expressed for 2D arrays using this method. The study of classical geometry shapes, such as cross [18], star [4], circles [19] or spirals [20,21], are often proposed. The random placement of microphones is considered as a generalisation of the classical shapes [22]. The statistical properties of the mainlobe width and the peak sidelobe level are then investigated.

In this paper, a genetic algorithm (GA) is proposed to solve the problem of finding the geometry that minimises cost functions on the beampattern. The goal is to optimise well known array properties such as the beamwidth, the Maximum Sidelobe Level (MSL) and more complicated criteria that are derived from the classical ones. In particular, the invariance of the beampattern with the focusing direction is studied through an application on moving sources. GA are known to quickly provide good solutions. From a more general point of view, global optimisation algorithms have been suitable to find appropriate geometries [23,24].

The document is organised as follows. In Section 2, the beamforming algorithm is introduced. Array aperture and maximum sidelobe levels are defined. The variation of the beampattern properties with source motion is discussed. In Section 3, the GA used for optimising the sensor position is developed. Cost functions are defined considering the beampattern properties. In Section 4, the results of the optimisation are investigated to validate the convergence of the algorithm and to define the properties of new arrays. In Section 5, Monte-Carlo (MC) simulations are performed to compare the performance of the optimised array with more classical geometries. An application on a high speed train illustrates in Section 6 the localisation improvement of the new geometry.

2. Beamforming

For N_s uncorrelated sources $s_n(t)$ and an array composed of M sensors, the pressure acquired by the m th sensor is expressed as the convolution of the source signals by the impulse response (linked to the Green function) of the medium

$$p_m(t) = \sum_{n=1}^{N_s} s_n(t) * h_{mn}(r_{mn}, t). \quad (1)$$

The impulse response $h_{mn}(r_{mn}, t)$ describes the propagation properties in the medium and r_{mn} is the distance between the n th source and the m th sensor. Considering narrowband sources centred on frequency f , the convolution product of Eq. (1) is expressed in the frequency domain as

$$P_m(f) = \sum_{n=1}^N S_n(f) H_{mn}(r_{mn}, f), \quad (2)$$

where $H_{mn}(f)$, $S_n(f)$, $P_m(f)$ are the Fourier transforms of the impulse response of the propagation $h_{mn}(r_{mn}, t)$, the source $s_n(t)$, the pressure $p_m(t)$ respectively. Thus, the received signal on the array can be expressed in the frequency domain as a $[M \times 1]$ vector

$$\mathbf{p}(f) = \mathbf{H}(f)\mathbf{s}(f), \quad (3)$$

where $\mathbf{s}$ is a $N_s \times 1$ source vector and

$$\mathbf{H}(f) = [\mathbf{h}_1(f) \cdots \mathbf{h}_n(f) \cdots \mathbf{h}_{N_s}(f)], \quad (4)$$

is a $M \times N_s$ matrix containing all the impulse response vectors. $H_{mn}(r_{mn}, f)$ is the m th element of $\mathbf{h}_n(f)$. Considering some hypothesis on the medium and the sources, a propagation model is defined. For point sources with spherical propagation in a homogenous free field and isotropic medium, the transfer function describing the sound propagation from the n th source to the m th microphone is expressed as [25]

$$H_{mn}(r_{mn}, f) = \frac{1}{4\pi r_{mn}} \exp\left(-2j\pi f \frac{r_{mn}}{c}\right). \quad (5)$$

c is the sound speed and r_{mn} the distance between the n th source and the m th microphone. $H_{mn}(r_{mn}, f)$ corresponds then to a phase shift (or a delay of r_{mn}/c in the time domain) and a geometrical attenuation of the sources. Beamforming considers the propagation effects by applying a steering vector $\mathbf{w}_n^H$ on the measurements and estimating the n th source amplitude, H denotes the Hermitian operator. The dimension of $\mathbf{w}_n^H$ is $1 \times M$. The m th element of $\mathbf{w}_n^H$ is defined as

$$W_{mn}(r_{mn}, f) = \frac{\exp(2j\pi f r_{mn}/c)}{4\pi r_{mn}} = H_{mn}^*(r_{mn}, f), \quad (6)$$

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