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A Python framework for microphone array data processing

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ABSTRACT

Acoular is an open source object-oriented Python package for microphone array data processing. It supports various methods for sound source characterization and mapping. The background of these methods, which rely on synchronously captured microphone signals, is shortly introduced, and the requirements for a software that implements these methods are discussed. The object-oriented design based on Python allows for easy-to-use scripting and graphical user interfaces, the practical combination with other data handling and scientific computing libraries, and the possibility to extend the software by implementing new processing methods with minimal effort. Built-in result caching and fast C++ based parallelized implementation of core routines is explained. Together with data handling procedures that can accommodate the huge amounts of measured data needed, this makes the application of Acoular to industrial-scale problems possible. Basic examples of Acoular use and extension are given.

1. Introduction

The design of low-noise machinery and vehicles requires analyzing the sources of sound. Information on the characteristics of any sound sources is necessary to find measures to reduce the generation of sound or its propagation. This includes the location and strength of the sources as well as their frequency content. Often, this information is only available through experimental analysis, either on the machine or vehicle itself or on specially designed laboratory setups of noise generating machinery parts. The necessary acoustical measurements can generally be performed using standard equipment such as a microphone, sound level meter or analyzer. However, this approach makes it difficult to reliably characterize sound sources in the case of multiple sound sources, which is a very common scenario.

In such a case, the results are dominated by the strongest source and expensive experimental procedures are needed to get separate results for each source. One solution for this problem is the application of a microphone array, where a number – some ten to some hundred – of microphones is used simultaneously to characterize multiple sound sources at the same time. This is done by computing acoustic source maps (often referred to as acoustic photographs) from the output signals of the microphones. Then, location, strength, and spectrum of the sources can be estimated from these maps.

A number of different methods are available for the computation of acoustic source maps. These methods either rely on the direct simultaneous processing of a large number of time-dependent microphone signals, or they perform the computation in the frequency domain after having transformed the signals accordingly into cross power spectra. Both kinds of methods are computationally demanding and require considerable computer resources. Some of the methods need to solve huge systems of equations, while others need to deal with large-scale optimization problems. The methods have different properties and deliver results of different kind and quality. Depending on the specific acoustic source characterization task, different methods may be appropriate. Consequently, the practical application of microphone arrays benefits from the uncomplicated availability of different methods.

While a larger number of publications on the methods themselves is available, the implementation of the methods is less often discussed. To the knowledge of the authors, no software is publicly available to date that implements more than a few of these methods. The available commercial software products are generally bound to a specific vendor's measuring and data acquisition hardware. Moreover, available software codes are not focused on the extensibility with new or modified methods.

The present contribution introduces Acoular, an open source Python library [1] that was published under the terms of the

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new BSD license and is available for all major operating systems. The library is aimed at applications in acoustic testing where sources of sound need to be characterized. Its design is object-oriented and it is intended to be easily extensible with the incorporation of new methods. Further design goals are computational efficiency and easy application. This concept makes it possible to apply the library for education and for research on methods for sound source characterization. Further, Acoular can be used to efficiently handle applications of industrial size, where larger numbers of measurements need to be analyzed.

The remainder of the paper is organized as follows. First, the theoretical background of the microphone array methods implemented in Acoular is briefly introduced. Second, the object-oriented design and operation of Acoular is addressed. Third, the application of Acoular is demonstrated using some examples. Finally, it is explained how a new method can be introduced into Acoular.

2. Background

Consider the scenario shown in Fig. 1, where a microphone array consisting of N microphones is used to analyze a sound field that is produced by an arbitrary number of sound sources. The sound will need a certain time $\Delta\tau_{mn}$ to travel from source m to microphone n, and its amplitude will be changed by a certain factor a_{mn}. Both will depend on the distance between source and microphone and other factors such as the speed of sound and presence of flow. Because linear superposition can be assumed, the sound pressure at the microphone is the sum of contributions from all sources:

$$p_n(t) = \sum_m a_{mn} q_m(t - \Delta\tau_{mn}). \tag{1}$$

The quantity q_m stands for a measure of source strength such as the flux of a monopole source. As long as both source strength and position of the sources are known, the calculation of all p_n is straightforward. The characterization of sound sources from microphone array measurements represents the inverse problem: to estimate the source strength and position of the sources from the measured p_n.

One possibility to achieve this is to calculate a weighted sum of the delayed and attenuated microphone signals, as shown in Fig. 1:

$$p_m = \sum_n h_{mn} p_n(t + \Delta t_{mn}). \tag{2}$$

Here, the idea is to choose h_{mn} and Δt_{mn} in such a way that the output p_{out} will contain the signal q_m from source m while any other source signals will be suppressed as much as possible. This can be seen as a spatial filter, and there are a number of options to calculate the filter coefficients h_{mn} and Δt_{mn} [2] from a_{mn} and τ_{mn}. Because of the calculation procedure in (2), this approach is called Delay-and-Sum Beamforming. The characterization of multiple sources with unknown positions is possible when the procedure is applied for each possible source position in a grid of source positions (see Fig. 1) in turn. In order to get an acoustic photograph, it is then convenient to calculate the power $\langle p_{out}^2 \rangle_T$ over a certain time interval T and map this quantity onto the grid. This can be done for arbitrary kinds of grids including such that are irregular or three-dimensional.

The principle from (2) can be extended in many different ways. Instead of using fixed filter coefficients that depend on the sound propagation model only, it is possible to use variable coefficients that adapt the spatial filter and depend also on the signals p_n themselves [3]. Another option, that can be used if the sound sources are moving on a known trajectory, is to treat the filter coefficients as functions of time [4,5]. Very often (2) is combined with subsequent frequency filtering. With a bandpass filter applied for each frequency band of interest, the method then also allows to estimate the frequency spectrum of the sound sources. In a practical application the microphone signals are containing additional noise not originating from the sound sources. The influence of this noise on the result can be considerably reduced by the following modification of (2):

$$p_m^2(t) = \max\left(\left\langle \left(\sum_n h_{mn} p_n(t + \Delta t_{mn}) \right)^2 - \sum_n (h_{mn} p_n(t + \Delta t_{mn}))^2 \right\rangle_T , 0 \right). \tag{3}$$

Besides Delay-and-Sum Beamforming in the time domain, a second possibility to estimate the source strength and position of the sources from the measured p_n is analysis in the frequency domain. To this end, the cross spectral matrix of the microphone signals is estimated using Welch's method [6]. All microphone signals are partitioned into n_d blocks of a certain number of samples. These blocks are then Fourier-transformed. An estimate of a matrix containing the N^2 cross power spectra of all possible pairs of microphones is then

$$\mathbf{G}(f_k) = 2\frac{1}{n_d T} \sum_{i=1}^{n_d} \mathbf{p}_i(f_k) \mathbf{p}_i^*(f_k), \tag{4}$$

where $\mathbf{p}(f_k)$ denotes the vector of the values of the Fourier-transform at the frequency f_k for all microphone signals. The cross spectral matrix $\mathbf{G}$ can then be used as a basis to perform the spatial filtering in frequency domain:

$$p_m^2 = \mathbf{h}_m^H \mathbf{G} \mathbf{h}_m. \tag{5}$$

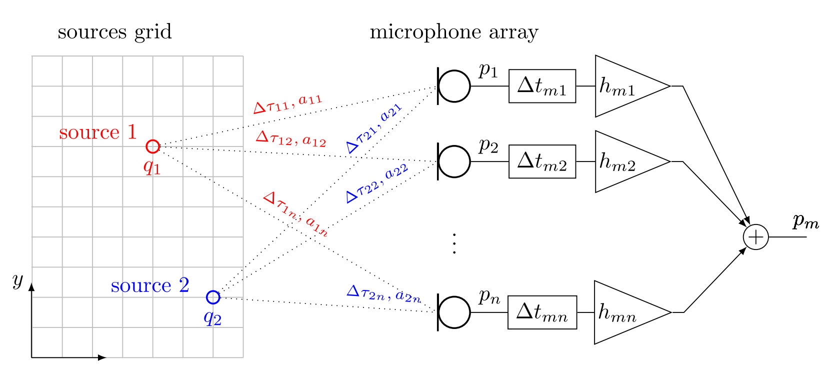

Fig. 1. Working principle of beamforming algorithms.