

Simultaneous Sound Power Measurement of Engine Components

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ABSTRACT

The sound power level is the most important quantity that characterizes the noise emission of machinery. Following standardized procedures, it is usually calculated from sound pressure levels measured at a number of reference positions on a surface enveloping the object. The resulting value does not hold any direct information on the noise contributions of subcomponents. However, effective noise reduction necessitates a prior identification of acoustic sources and their characteristics. Combining the enveloping-surface method with microphone array measurements facilitates the evaluation of synchronously noise-emitting subcomponents. The application of this technique on a water pump with a four-stroke engine as power source is presented in this paper. The microphones are arranged on a cuboid surface surrounding the setup. The measured data is processed so as to yield sound power levels in a defined 3D focus region. As is shown, with this technique it is not only possible to identify and spatially visualize noise sources and the emitted respective sound powers, but also to determine spectral characteristics of noise emitted by chosen components.

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INTRODUCTION

The rising desire for vehicles with low impact on the environment necessitates the further research and development of low-noise machinery. Therefore, noise reduction and the selective manipulation of sound characteristics of machines are major objectives in NVH engineering. Very often, the noise emitted by machines or vehicles is generated by several sources of different strength, positioned at different locations. Furthermore, the contribution of each source depends on the state of operation of the machine.

The exact identification of the acoustic sources, their position, extent, and strength, is a challenge to be met for developing effective noise control methods. The overall sound power is usually calculated from sound pressure levels measured at a small number of reference positions on a surface enveloping the object. Further measurements may include the isolated operation of components of interest or the use of additional sensors close to or mounted on the object. However, this is not always feasible or may require considerable efforts.

In recent years, the use of microphone arrays, which do not necessarily have to be positioned in the immediate vicinity of the measured object, has become state of the art. While array-based methods implement highly directive sensors and thereby allow sound source separation, they deliver only the sound pressure as a result but not the sound power. Combining this technique with the enveloping measurement surface method provides a powerful tool, which allows attributing portions of the overall sound power level to components of the tested object with one single measurement.

The remainder of this paper is organized as follows. First, the working principle of a microphone array method is outlined, as well as the standardized procedure to calculate sound power levels from sound pressure measurements. A hybrid method combining these techniques is introduced. Then, experiments with a water pump driven by an internal combustion engine, which were conducted to verify the applicability of the method, are described. Third, the obtained results are presented and discussed. The findings are then summarized in the concluding section.

THEORY

Beamforming

The objective of microphone array methods is to implement a sensor which can spatially separate noise sources. This is achieved by simultaneously recording sound pressures at distributed sensor positions and then shifting the phase of the signals according to the sound travel times from the sensors to assumed source positions on a "focus" grid. The summation of the shifted signals yields a sound pressure at grid points where a source is present, and cancels out at points without source contribution [1].

Aside from faster computing, performing the calculations in frequency domain has the advantage that uncorrelated noise at the sensors can be subtracted. Furthermore, sophisticated deconvolution algorithms can be applied to achieve a higher spatial resolution of the final result.

The beamforming formulation for one focus point $\mathbf{x}_t$ in frequency domain is:

$$b(\mathbf{x}_t) = \mathbf{h}^H(\mathbf{x}_t) \mathbf{C} \mathbf{h}(\mathbf{x}_t), \quad t = 1 \dots N, \quad (1)$$

with the cross spectral matrix $\mathbf{C}$ and the steering vector $\mathbf{h}$. N is the total number of focus points. The steering vector contains the phase shift according to sound travel times from the focus point to each microphone, as well as the level correction according to a monopole source model:

$$h_m = \frac{1}{r_{t,m} \sqrt{M \sum_{l=1}^M r_{t,l}^{-2}}} e^{-jk(r_{t,m} - r_{t,0})}, \quad m = 1 \dots M, \quad (2)$$

with $r_{t,m}$ being the distance from the focus point to the m -th of M microphones and $r_{t,0}$ the distance from the focus point to an arbitrary reference position, at which the sound pressure levels are evaluated (usually the array center). It has been shown that presently used steering vector formulations consist of a trade-off between correct reconstruction of the location and the source strength [2]. With the formulation in Eq. (2), the location of sources can be computed accurately while the sound pressure levels will be slightly underestimated.

For the approximation of the cross spectral matrix (CSM), the measured time signals are processed using Welch's method. The signal is divided into overlapping blocks, which are multiplied with a von-Hann window and transformed into frequency domain using FFT. The CSM can then be calculated by averaging the cross spectra

$$\mathbf{C} = \overline{\mathbf{p}\mathbf{p}^H}, \quad (3)$$

with $\mathbf{p}$ containing the complex spectral sound pressures at each microphone position.

For further processing, the main diagonal of the CSM (containing the autospectra) can be removed. This suppresses the influence of uncorrelated noise sources, such as pressure fluctuations caused by flow turbulences or noise introduced by the sensors themselves, in the reconstruction.

Deconvolution

Depending on the configuration of the array and the focus grid, as well as on the source characteristics, the classic beamforming result features artifacts. This can be interpreted as the correct source level map being convoluted with a point spread function.

Several algorithms have been proposed to deconvolute the map, one of which is the CLEAN-SC algorithm by Sijtsma [3]. Aside from fast computation, an advantage of this algorithm is that it does not require

the point spread function to be known prior to the calculations, but estimates it itself by evaluating coherent source portions in the convoluted map. The method works iteratively, in each loop looking for the maximum value in the convoluted map, storing it in a new (clean) map, and subtracting it and coherent portions from the first map. In order to account for the possible superposition of sources, usually only a fraction of the maximum value is subtracted.

Enveloping Surface Method

An important quantity to characterize noise emission of machinery is the sound power level. Since determining the sound power with high precision is complex and requires sophisticated measurement tools and facilities, its approximation by using a small number of sound pressure measurements is desirable. Methods for this have been standardized in ISO 3744 (accuracy grade 2) [4] and ISO 3746 (accuracy grade 3) [5]. These norms describe the calculation of sound power levels from sound pressures measured at selected positions on a surface enveloping an object on a reflecting plane. The principle is to average the sound pressures and apply correction factors depending on background noise, reflecting environment and measurement surface.

Following ISO 3746, the sound pressures are measured at the centers of five sides of an cuboid enveloping the object. The sound power level L_{WA} is then calculated via:

$$L_{WA} = \overline{L_{pA}} + 10 \lg \frac{S}{1 \text{ m}^2} \text{ dB}. \quad (4)$$

$\overline{L_{pA}}$ comprises the averaged, A-weighted and corrected sound pressure levels. S is the total surface of the five sides.

Hybrid Method

Aiming to create a tool capable of measuring the sound power of machine components, the enveloping surface method is used in combination with microphone array techniques. For this, the array is constructed such that it implements a highly directive microphone at the center of each of the five cuboid sides, i.e., synchronously measured data are evaluated subsequently with sub-arrays, the reference position for $r_{t,0}$ in Eq. (2) being at the respective side centers.

For each side, the sound pressure levels are evaluated at the same focus points. These are chosen to describe a three-dimensional area containing the measured object. In order to account for reflections on the plane beneath the object, the focus area is extended below the ground to capture the acoustic mirror image.

Similar to the ISO 3746 procedure, the five sound pressure maps are averaged and the corrections to calculate the sound power are applied. Finally, the mirror image is added to the above-ground portion of the map, so that the sound powers can easily be evaluated according to the position of machine subcomponents.