Using Microphone Array for Fault Detection Constant Directivity Beamforming

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Abstract: Sound measurements inherently exhibit high sensitivity to noise. In order to suppress extraneous sounds from the environment, measurements are usually done in an anechoic chamber. In laboratory experiments this can be rather easily carried out, whereas in industrial environment this can be a serious issue. In mass production for instance, the production rate is usually too high to allow for application of the anechoic chamber, for closing and opening of the chamber aperture alone can take too much time to fit the production cycle. One way to avoid the use of the chamber is to deploy a microphone array. This is a set-up of several precisely positioned microphones which simultaneously measure the sound. The number of microphones and their positions are crucial for successful application and are defined according to the frequency range in question. By processing the acquired signals a desired directionality of the array is achieved. Thereby the array actually focuses on sounds coming from the desired space direction, while suppressing the sounds from elsewhere. In the paper an approach of using the microphone array for constant directivity beamforming (CBD) application will be described.

Keywords: fault detection, microphone arrays, signal processing, sound analysis

1 Introduction

Producers of household appliances are under increasing pressure of ever-stricter standards on the quality requirements. The quality requirements common to all fields regard the emitted sound during operation, which generally must not exceed standardized level. In the field of rotating machinery, the level of the emitted sound is usually pretty high and small faults can make the sound pretty annoying. Hence, fault detection is of vital importance for the quality of the sound and necessitates sound measurements.

Sound measurements are highly prone to corruption with extraneous noise from the surroundings and inappropriate measurement conditions can easily drop signal to noise ratio to a level where no useful information is retained. In laboratory experiments, achieving high signal to noise ratio presents no issue, whereas on-line industrial applications often does not allow for simple solutions. The minimum requirement applied is the use of an anechoic chamber. The chamber suppresses the sounds from the surroundings, it eliminates echoes and reverberation and is rather easy to use in laboratory experiments. In general, the same applies to some industrial applications. However, in many industrial applications it is impossible to use the anechoic chamber for either of two reasons. The first reason is that a machine to be monitored is simply too big and cannot be build in the anechoic chamber. The second reason is that in mass production it is rather inconvenient to mount a chamber due to hastened wear and the related costs. Namely, high production rate implies increased wear of moving parts of the chamber such as the hatch. Therefore, the costs incurred by maintenance can quickly soar, plus maintenance itself can interfere production.

An alternative approach is to use multiple microphones and execute the measurements in an open space. In that respect two approaches are available.

The first approach is to use a blind source separation. Here, the microphones are carefully positioned throughout the place to record different sound sources as clearly as possible. This means that each microphone should be placed as close as possible to the respective source or to the place where the source is anticipated to emerge. The sound signal of the monitored (desired) source is then isolated by proper signal processing. Generally, there are several ways to perform blind source separation and all are built upon certain assumptions. Some assume certain time-frequency domain characteristics of the source signals [11, 12] others convolution properties of mixtures [5, 1], the third certain statistical features such as statistical independence of the sources etc. In the latter, the signal processing technique is called (ICA) independent component analysis. The grounds of ICA techniques have been established in last two decades and a lot of algorithms have been developed since then [7, 6, 2, 8]. The issue encountered when using ICA in real measurements is related to delays of the measured signals. Moreover, even if delays are accounted for in mathematical models, ICA still cannot accomplish the signal separation satisfactorily.

The second approach is to use a microphone array. This means that instead of using the anechoic chamber, an array of microphones is carefully positioned and pointed towards the measured object. The microphones pick up all the sounds present at the time of recording. By using proper signal processing the sound coming from desired direction can be isolated and all the extraneous sounds coming from other directions stifled. This approach has been pursued by several authors [10, 9] and has proved successful.

This paper describes one of the basic approaches of using microphone arrays called Constant Directivity Beamforming (CBD). CBD means that microphone array is programmed to isolate sounds coming from certain constant directions while suppressing the sounds from elsewhere. Constant directions means that desired directions do not change in time.

The paper is organized as follows. In the second section a brief description of problem background is given. The third section provides basics of CBD. In the forth section an application of CBD to fault detection is provided. The section five summarises the conclusions.

2 Problem background

A diagnostic system for electrical motors was developed and has been fully operating in industrial application for a year. It employs four modules, i.e. vibration analysis, commutation analysis, parity relations and sound analysis. A detailed description of the prototype version of the system can be found in [13, 3]. As shown by the authors, the system has exceptional diagnostic performance and manages complete diagnostics of electrical motors. However, from the constructional viewpoint it has one drawback. It employs anechoic chamber for acoustic measurements.

In order to consummate the system, further improvements call for substitution of the chamber with one of alternative solutions. In that respect, a preliminary research has been conducted on the use of different ICA methods on one side and microphone array on the other as two possible solutions. ICA methods have proved only partially useful as they rather insufficiently deal with time delays in signals at the moment. On the contrary, microphone array is actually based on time delays since it uses them to achieve the directionality. Given that fact, microphone array seems to be the solution to resort to.

In the following section a basic approach of CBD is presented in short.

3 Constant Directiviy Beamforming - CBD

Beamforming, or spatial filtering, is one of the simplest and most robust means for discriminating between different signals based on the physical locations of the sources [4].

Consider a linear array of $M = 2N + 1$ microphones positioned at $p_n, n = -N, ..., N$. The data received at the $n - th$ microphone, is given by

$$x_n = \sum_k e^{-i2\pi f c^{-1} p_n \cos\theta_k} s_k \tag{1}$$

where k stands for the number of sources and θ_k the respective impinging angles. The beamformer output is formed by applying a weight vector to the received array data as

$$y = \mathrm{w}^H x$$

where ^{H} denotes Hermitian transpose, and $\mathbf{w}$ is the M vector of array weights. Another two important properties of the beamformer are the spatial response defined as

$$b(\theta, f) = \mathrm{w}^H e^{-i2\pi f c^{-1} p \cos\theta} \tag{2}$$

and the beampattern defined as the squared magnitude of the spatial response (2). The problem of CBD design can now be formulated as finding the array weights such that the resulting spatial response remains constant over all frequency bands of interest. This is usually done separately for different frequency ranges of the signal. As shown by [4], frequency invariant beampattern $b(\theta)$ can theoretically be obtained if the sensor weighing function is defined as $w(p, f) = fB(pf)$, where $B(\cdot)$ stands for beamshaping function, and f and p for frequency and sensor position respectively. As seen from the definition the beam shaping function also has the symmetry property with respect to space and frequency, which is also utilized in practical implementation [14].

3.1 Practical Implementation

By using a trapezoidal integration rule, the spatial response reads as

$$\hat{b}(\theta, f) = \sum_{n=-N}^{N} fB(p_n f)\Delta_n e^{-i2\pi f c^{-1} p_n \cos\theta} \tag{3}$$

where $\hat{b}$ denotes an approximation of frequency invariant spatial response $b(\theta)$, $\Delta_n = \frac{1}{2}(p_{n+1} - p_{n-1})$ is the length of the nth subinterval. Thence it follows that the weight on the nth sensor at the frequency f is $w_n = f\Delta_n B(p_n f)$.

3.2 Sensor Placement

Another key thing in microphone array technology is the sensor placement. In order to choose an appropriate sensor array geometry, the most important consideration is to ensure that at any frequency spatial aliasing is avoided. Typically, an equally spaced microphone array geometry with on half-wavelength of the highest frequency spacing is considered. The number of sensors can be reduced if the relative active aperture size is constant as shown by [14].

4 Example

This section provides an example which demonstrates the use of the microphone array for fault detection purposes. In section 2 the diagnostic system for electrical motors is briefly described. This diagnostic system utilizes sound analysis to detect some mechanical faults in electrical motors. One of them is also the bearing fault. Namely, in this particular motor the bearing fault can only be detected by means of sound analysis. As the on-line measurement is done on the production line, an anechoic chamber is used to suppress the surrounding noise. In the following the paper will demonstrate how the chamber could be replaced by the microphone array in this particular case.

In the case of bearing fault, the frequency band of interest ranges from 2.4 kHz to 3.4 kHz approximately. The microphone array should therefore have the spatial response function with amplification 1 in this frequency range and in the desired spatial angle θ and 0 elsewhere. Without loss of generality, we can assume $\theta = 0$. If otherwise, the microphone signals can be appropriately delayed resulting in analogous situation.

For this task the array should consist of 9 microphones positioned as given in table 1.

Table 1: Positions of microphones in meters for detecting bearing faults

p_{-4}	p_{-3}	p_{-2}	p_{-1}	p_0	p_1	p_2	p_3	p_4
-0.1697	-0.1132	-0.0754	-0.0503	0	0.0503	0.0754	0.1132	0.1697

By using simple methods of designing the array weights, the spatial response given in figure 1 is obtained.

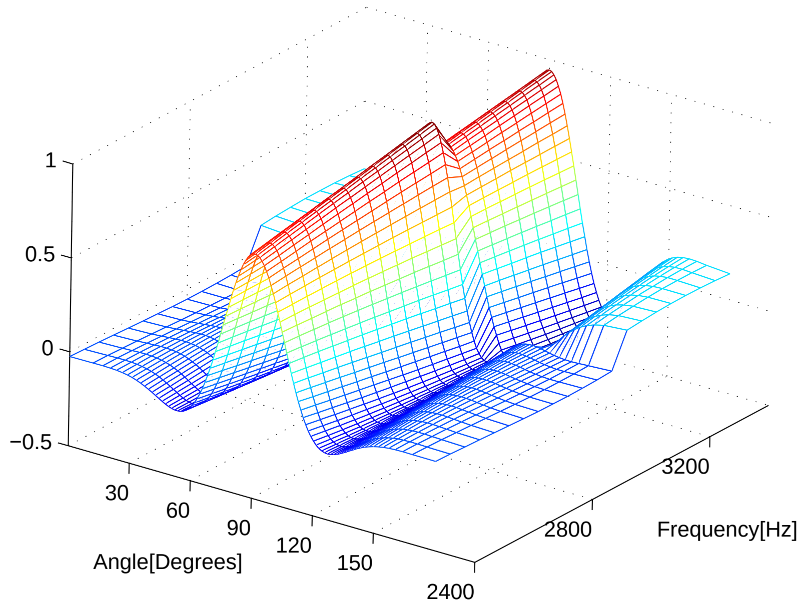

Figure 1: Spatial response.

5 Conclusions

The paper presents a utilization of a microphone array for fault detection purpose. Given the fact that the measurement object (electrical motor) is stationary, constant directivity beam-forming technique is used. The early results give promising results.

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