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Calibration of acoustic vector sensor based on MEMS microphones for DOA estimation

Józef Kotus*, Grzegorz Szwoch

Gdansk University of Technology, Faculty of Electronics, Telecommunications and Informatics, Multimedia Systems Department, Narutowicza 11/12, 80-233 Gdańsk, Poland

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ABSTRACT

A procedure of calibration of a custom 3D acoustic vector sensor (AVS) for the purpose of direction of arrival (DoA) estimation, is presented and validated in the paper. AVS devices working on a p-p principle may be constructed from standard pressure sensors and a signal processing system. However, in order to ensure accurate DoA estimation, each sensor needs to be calibrated. The proposed algorithm divides the calibration process into two stages. First, amplitude calibration is performed in order to compensate for amplitude differences between pairs of microphones situated on each axis. After the pressure and velocity signals are computed from the corrected microphone signals, the second stage is performed in order to correct phase differences between the pressure and velocity signals, which then allows for computing the intensity signals for each axis. In order to validate the calibration method, a reference AVS was constructed from low cost components, namely MEMS microphones and a DSP board. The method was engineered with an assumption that it will be applicable to any AVS working on the same principle. A set of experiments was performed in order to validate the calibration method and to compare the accuracy of the calibrated sensor with a commercial AVS. It was found in the experiments that DoA accuracy of the proposed 3D AVS calibrated with the proposed procedure matches that of a commercial, high cost, factory-calibrated sensor. Therefore, the proposed calibration method fulfills the requirements of accurate DoA estimation, and it is applicable to calibration of custom built, low cost AVS devices, that may be implemented in practical applications for determining the direction of sound sources, such as environmental monitoring, traffic monitoring and public security systems.

1. Introduction

Acoustic vector sensors (AVS) are devices that are capable of measuring acoustic particle velocity as well as pressure, as in a standard microphone. AVS devices working on the ‘p-u’ principle consist of one omnidirectional sensor for measuring pressure, and three directional sensors on the orthogonal axes, measuring acoustic particle velocity [1]. However, AVS can be also practically realized using only acoustic pressure sensors. The majority of available sound intensity measurement systems, including AVSs, are based on the ‘two-microphone’ (*p-p*) principle, which uses two closely spaced pressure microphones and rely on a finite difference approximation of the sound pressure gradient [2]. The IEC 1043 standard on instruments for the measurement of sound intensity deals exclusively with the *p-p* measurement principle [3].

Sound probes based on p-u or p-p principles are used for measurements of sound intensity which is a measure of the flow of acoustic energy in a sound field. The sound intensity $\boldsymbol{I}$ is a vector quantity defined as the time average of the flow of sound energy through a unit area in a direction perpendicular to the area. The intensity in a certain direction is the product of sound pressure (scalar) $p(t)$ and the particle velocity (vector) component in that direction $u(t)$. Typically, intensity metric quantities are used for measuring energy transmission and propagation paths [4,5], as well as for detection of noise source localization [6], determination of acoustic impedance and reflection index of materials [7], although one may find several examples of employing them in audio engineering [8], and in particular in the recording and reconstruction of the acoustic field, e.g. ambisonics [9]. Commercial sound intensity measurement systems were introduced to the market in early 1980s, and the first international standards for sound intensity measurements and related instruments were issued in the mid-1990s [3,7,10,11].

One of the most common applications of AVS devices is direction of arrival (DoA) estimation, i.e. determining the azimuth and elevation angle of the incoming sound, allowing for positioning the sound source

* Corresponding author.
E-mail address: joseph@sound.eti.pg.gda.pl (J. Kotus).

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in 3D space [12]. The described sensors are useful in many practical applications, e.g. in sound field measurements, acoustic detection of security threats in surveillance systems [13,14], environmental noise monitoring, etc. However, the cost of commercial AVS devices that provide a satisfactory level of accuracy, is usually high. This is caused not only by the cost of components, but also by the fact that each individual sensor needs to be calibrated at the factory in order to compensate for differences in signals obtained from individual sensors. Therefore, AVS devices are a kind of measurement equipment, not a solution suitable for large scale implementations, e.g. in urban monitoring systems. However, a custom AVS may be constructed from low cost microphones, supported with a signal processing system, employing the principle of p-p sensors. Previously, the authors experimented with a custom sound intensity probe built from seven microphones [15] and experimented with application of a commercial AVS for DoA estimation [6,16]. In this paper, a novel AVS construction, based on miniature digital microphones, is used. In order to obtain meaningful data from this sensor, with errors below some accepted level, a proper calibration method has to be developed.

There are several known techniques that can be employed to calibrate a sound intensity probe based on the pressure gradient principle [17]. The first one requires using specialized microphones for the sound intensity measurements, having similar properties (i.e. sensitivity, frequency response) and which need to be exposed to the same sound pressure by using a coupler [2,18]. In such case, the microphones should have identical frequency responses. A well-known method of sound intensity probe calibration based on the cross-spectral density approach was described by Krishnappa [19]. The phase and gain mismatch errors are corrected by measuring the transfer function between the two microphone systems while exposing them to signals with identical phase and pressure level, over a wide range of frequencies. Calculation of cross-spectral density between two microphones (p-p probe) was then used to determine the sound intensity. Another kind of calibration technique, proposed by the Nagata et al. [20], is called a rotating microphone method. The calibration in this case is performed in an anechoic chamber and it uses a reference intensity probe which was calibrated with a sound-pressure phase-difference calibrator. During the comparative sound intensity measurement, the bias error is determined. Another approach by Zhang [21] aims to compensate for phase differences by means of instantaneous correlation integral. Other techniques are employed for AVS calibration in special applications, e.g. for underwater measurements [22].

The calibration methods presented above focus mainly on correcting the amplitude and phase responses of microphone pairs, by means of processing the acoustic pressure signals acquired from microphone pairs. In this paper, the Authors take a similar approach and propose a two-stage procedure for calibration and correction of a 3D p-p intensity probe [23]. Since the purpose of the AVS described here is DoA estimation, the calibration procedure does not need to ensure that accurate values of sound intensity are measured. It is sufficient to ensure that relationship between sound intensity measured on the orthogonal axes is maintained. Contrary to the calibration methods presented earlier, a reference intensity probe or a calibrator are not needed to perform the calibration with the proposed approach. In order to validate the calibration method, a custom AVS was constructed from MEMS microphones and a DSP board. The method was engineered with an assumption that it will be applicable to any AVS working on the same principle. A set of experiments was performed in order to validate the calibration method and to compare the accuracy of the calibrated sensor with a commercial AVS.

The rest of the paper is organized as follows. Section 2 presents the algorithm used for DoA detection from signals recorded by the microphones. Section 3 shows the proposed calibration method. In Section 4, a reference hardware implementation of the sensor is presented, and some practical issues of software and hardware implementations of the calibration method are discussed. Section 5 presents the results of experiments related to the calibration of the proposed AVS, the achieved accuracy of DoA detection with the results obtained from a commercial AVS.

2. Acoustic vector sensor

An acoustic vector sensor (AVS) measures velocity of sound particles in a defined direction, as well as pressure [1]. A three-dimensional (3D) AVS described in this paper is able to measure the acoustic particle velocity (referred to as "velocity" further in the text) in three orthogonal directions, as well as pressure in the central point of the sensor. Based on the velocity and pressure measurement results, sound intensity vectors in three orthogonal directions may be obtained. The AVS described here is based on the p-p principle, i.e. on pressure measured at two points, then the velocity vector is computed as a pressure gradient (a difference of pressure values measured at two points), and the pressure scalar at the AVS central point is averaged from these point measurements [3].

In the described AVS, pressure has to be measured at six points located on three orthogonal axes, at identical distances d from the origin. These points are denoted as $x1$, $x2$, $y1$, $y2$, $z1$, $z2$, describing their location in the coordinate system, e.g. point $y2$ is located at $(0, d, 0)$ and $y1$ at $(0, -d, 0)$. Omnidirectional microphones of the same type are used to measure pressure $p_l(t)$ at six locations l. According to the Euler's formula, velocity vectors $\mathbf{u}_i(t)$ alongside axes X, Y, Z may be computed as:

$$\begin{bmatrix}\mathbf{u}_x(t)\\ \mathbf{u}_y(t)\\ \mathbf{u}_z(t)\end{bmatrix} = \begin{bmatrix}a_x & 0 & 0\\ 0 & a_y & 0\\ 0 & 0 & a_z\end{bmatrix}\cdot\begin{bmatrix}p_{x2}(t)-p_{x1}(t)\\ p_{y2}(t)-p_{y1}(t)\\ p_{z2}(t)-p_{z1}(t)\end{bmatrix} \quad (1)$$

where a_i are scaling factors. Magnitude of the $\mathbf{u}_i(t)$ vector will be denoted as $u_i(t)$. Pressure $p(t)$ measured at the origin is averaged from two points at the given axis and it has to be equal on all three axes. In practice, pressure is averaged from all six microphones (as in Eq. (2)):

$$p(t) = \frac{p_{x1}(t) + p_{x2}(t) + p_{y1}(t) + p_{y2}(t) + p_{z1}(t) + p_{z2}(t)}{6} \quad (2)$$

Sound intensity I at a given axis may be then computed as [24]:

$$I = \frac{1}{T}\int_T p(t)u(t)\mathrm{d}t \quad (3)$$

where T is the integration period.

If a single, omnidirectional sound source is put into the system at polar coordinates (r, φ, θ), then the angles of the sound received by the AVS may be computed as:

$$\phi = \arctan\left(\frac{I_y}{I_x}\right) \quad (4)$$

$$\theta = \arctan\left(\frac{I_z}{\sqrt{I_x^2 + I_y^2}}\right) \quad (5)$$

In a calibrated AVS, the following conditions have to be fulfilled:

$$\begin{aligned} p_{x1} &= p_{x2} \text{ if } \varphi = \pm\pi/2,\\ p_{y1} &= p_{y2} \text{ if } \varphi = 0 \text{ or } \varphi = \pi,\\ p_{z1} &= p_{z2} \text{ if } \theta = 0. \end{aligned} \quad (6)$$

3. Calibration procedure

In an uncalibrated AVS, Eq. (6) is usually not fulfilled due to imperfections in microphone positioning in the system and differences in parameters between microphones. Therefore, a calibration procedure is required in order to equalize amplitude and phase differences between the microphones. The proposed algorithm divides the calibration