Elimination of the Phase Mismatch Error in PP Probe Using Synchronous Measurement Technique

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Abstract: The paper presents a modification of the pressure-pressure (PP) sound intensity measurement method. In the proposed solution simultaneous measurement with a pair of microphones (used in the classical PP probe) is replaced by a sequence of measurements taken with a single microphone placed in successive positions. This approach requires an additional (reference) microphone to synchronize the successive measurements. Although, in the process of calculating the sound intensity only the signal from the measurement microphone is used. Thanks of this the errors associated with differences in the frequency responses of the measurement microphones (especially phase mismatch error) that occurs in the classical PP method are eliminated. This approach simultaneously increases the random error and limits the measurements to periodic signals only. The article presents the principle of operation of the classical PP probe and the currently used methods of phase mismatch error elimination based on pre-calibration of the probe. Next, the theoretical basis of the proposed measurement method is described. To verify the effectiveness of phase mismatch error elimination in the proposed method, an experiment was conducted. It consisted in estimation the angle of incidence of an acoustic wave under controlled conditions in an anechoic chamber. The measurement was carried out with the classical PP probe and with the modified method. Measurements were made for different sound sources (a loudspeaker set and a small electrical device). In the final part of the paper, the results are discussed, both methods (classical and modified) are compared and potential applications of the proposed method are indicated.

Keywords: PP-probe, sound intensity, phase mismatch error

1. Indroduction

1.1. Sound intensity and its applications

The theoretical basis of sound intensity was formulated in 1878 by Lord Rayleigh in his work “Theory of sound”. Attempts to measure this quantity took place in the following decades of the 20th century. A breakthrough came in 1977 when Fahy and Chung (independently) described the concept of using Fast Fourier Transform (FFT) and a cross-spectrum method to calculate the sound intensity from signals coming from two microphones [5, 3]. This method allowed the construction and practical use of sound in-tensity probes (so-called intensity probes). Due to the dynamic development, intensity measurements and the probes used in them have been included in international standards [28–31].

Examples of application areas of the sound intensity measurement are:
- determination of sound power of the sources [28–30];
- testing the acoustic absorption of materials [7, 23];
- localization of sound sources (Direction of Arrival estimation) [2, 12, 13, 17, 18];
- testing of acoustic energy flow in waveguides and closed areas [26];
- testing the properties of diffusers and loudspeakers [18, 19];
- scanning of noise sources to indicate the mechanism of noise generation [22, 24, 25];
- measurement of acoustic impedance [15].

A significant advantage of sound intensity measurements (in comparison to the most popular pressure measurements) is the possibility of making the measurements without an anechoic chamber, e.g. on a production hall in the presence of noise generated by many sources (in-situ).

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Measurement of sound intensity is connected with the necessity of using more complicated measuring equipment (in comparison to sound pressure measurements which require the only microphone). To mathematically describe the sound intensity it is necessary to define two terms: sound pressure and particle velocity.

Acoustic (sound) pressure is defined as pressure oscillations around a fixed level (usually atmospheric pressure). The value of the acoustic pressure at the time in the presence of a fixed pressure is described by (1).

$$p_a(t) = p(t) - p_0, \tag{1}$$

Particle velocity defines the instantaneous velocity of an acoustic elementary particle. It is a vector quantity described in the Cartesian coordinate system by (2).

$$\vec{u}(t) = \vec{i}u_x(t) + \vec{j}u_y(t) + \vec{k}u_z(t) \tag{2}$$

The instantaneous sound intensity is the product of the instantaneous particle velocity and the instantaneous sound pressure. It is described by:

$$\vec{I}_{inst}(t) = p_a(t) \cdot \vec{u}(t) \tag{3}$$

A sign of the sound intensity indicates the direction of the acoustic energy flow (to or from the acoustic source). In practice, the average sound intensity described by:

$$\vec{I}_{avg} = \lim_{T\to\infty} \frac{1}{T}\int_0^T \vec{I}_{inst}(t)dt \tag{4}$$

Nowadays, there are two main types of sound intensity probes: PU probes (e.g. Microflown [1]) and PP probes. A PU probe consists of a pressure microphone (P) that allows the measurement of the sound pressure and a velocity sensor (U) that is used to measure the acoustic velocity. In modern PU intensity probes, acoustic velocity measurement is based on two alternative technical solutions: an ultrasonic sensor or a hot-wire anemometer sensor. The velocity sensor allows the determination of one, two, or three orthogonal components of the acoustic velocity vector. The main disadvantage of the PU probe is the high price and the significant failure rate of the velocity sensor.

1.2. PP probe principle

In the PP-type probe, the determination of the acoustic velocity is done indirectly. Based on the linearized Euler equation, the acoustic velocity (in one direction) is given by:

$$\vec{u}(t,x) = -\frac{1}{\rho}\int \frac{\partial}{\partial x} p(t,x)dt \tag{5}$$

The gradient operation is approximated by the differential quotient using the sound pressure recorded simultaneously by two microphones located at a short distance Δx from each other (Fig. 1).

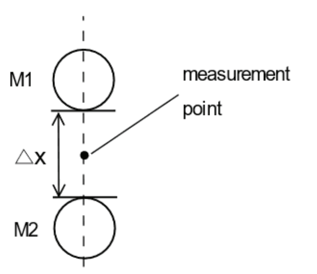


Fig. 1. Measurement microphones used in one-dimensional PP probe; M1, M2 – microphones

Rys. 1. Mikrofony pomiarowe użyte w sondzie pomiarowej (do pomiaru jednej składowej wektora natężenia dźwięku) typu PP. M1, M2 – mikrofony

Then, the acoustic velocity (more precisely: one component lying on the line passing through the acoustic centers of the used microphones) can be calculated using:

$$\vec{u}(t) = -\frac{1}{\rho\Delta x}\int_{-\infty}^{t}\left[p_{a1}(\tau) - p_{a2}(\tau)\right]d\tau \tag{6}$$

The pressure is the average of the pressures recorded by a pair of microphones is done by:

$$p_a = \frac{p_{a1} + p_{a2}}{2} \tag{7}$$

For the harmonic plane wave and a phase shift between signals p_1 and p_2 with RMS values P_1 and P_2 equal to φ_a sound intensity is equal to:

$$I = \frac{P_1P_2\sin\varphi_a}{2\omega\rho\Delta x} \approx \frac{P_1P_2\varphi_a}{2\omega\rho\Delta x} \tag{8}$$

The approximation is correct for small angles: $\varphi_a \ll 1$ rad, which, in the case of the PP probe (typical distance between microphones in PP probe is 8–25 mm) is generally satisfied [6]. The sound intensity level in this case is equal to:

$$SIL = 10\log_{10}\left|\frac{P_1P_2\varphi_a}{2\omega\rho\Delta x I_0}\right| \tag{9}$$

1.3. Phase mismatch error and its elimination

There are several types of systematic errors occurring in sound intensity measurement using PP probe (diffraction error, finite difference error [6]). Especially important is the error resulting from the fact that two pressure signal acquisition channels (microphones, amplifiers, etc.) used to determine a sound intensity component do not have the same frequency characteristics (amplitude and phase). The phase mismatch error of the microphones is particularly significant.

In the real intensity probe, in addition to the shift introduced by the geometric distance between microphones, there is an additional phase shift introduced by the microphones. If we denote this shift by φ_b the error between real (SIL) and measured (SIL_m) sound intensity level is given by the:

$$SIL_m - SIL = 10\log_{10}\left|1 + \frac{\varphi_b}{\varphi_a}\right| \tag{10}$$

Commonly, the assumed acceptable value of the error in sound intensity estimation is 1 dB [6].

The difference between Sound Pressure Level (SPL) and Sound Intensity Level (SIL) obtained when an identical signal is applied to both microphones is called Pressure-Residual Intensity Index [10] denoted by δ_{pI0} and described by:

$$\delta_{pI0} = 10\log_{10}\left(\frac{\Delta x\frac{\omega}{c}}{\varphi_b}\right) \tag{11}$$

International standards [4] defines the minimum values of Pressure-Residual Intensity Index for I and II class PP probes. Phase mismatch error is especially important for measurements performed at low frequencies (≤ 500 Hz) in the near field [33].

The measurement microphones used in a commercial PP probes have a phase mismatch error of 0.05 deg for 250 Hz and proportional to frequency [11, 36]. The use of such microphones results in a high price of the PP probe. Even when using high quality measurement microphones, their characteristics may fluctuate over time. Therefore (and also in the case of probes using a lower class microphones [8]) it is necessary to periodically perform an initial calibration of the probe.

The basic calibration method [6, 14] involves placing both PP probe microphones in the pistonphone and applying a wide-band signal to them. Then, by calculating the cross-spectral signals coming from both microphones, the pressure-residual intensity index is determined.

In [9], a simplified calibration method was presented by taking measurements twice and switching microphones between measurement channels.

Another method [21] requires the use of a reference (previously calibrated) measuring probe and is carried out in an anechoic chamber. Phase mismatch error is eliminated by taking comparative measurements with two probes. This type of approach is used for PP probes used to measure acoustic power [11].

A range of interesting solutions related to frequency response correction of microphones are used in Acoustic Vector Sensor (AVS) devices, which are based on the PP probe principle and are used to determine the angle of attack of an acoustic wave (Direction of Arrival – DOA).

In publication [27], a calibration method is presented that requires carrying out several (at least three) preliminary measurements for different incidence angles of the acoustic wave. Then an all-phase Fast Fourier Transfer is applied to correct the phase shift (and also the gain) between the acoustic velocity and the acoustic pressure.

A similar solution presented in [12, 13] was to use DSP algorithms (cross-correlation, sweep technique, FFT) and perform calibration in two steps. First the amplitude responses are corrected and then the phase responses. This approach has been implemented using popular MEMS microphones and a DSP board. The advantage of this approach is that the properties of the sound field during calibration do not need to be known exactly, but must remain constant throughout the calibration process. The use of an anechoic chamber during calibration seems to be necessary.

A common feature of all the above mentioned methods of phase mismatch error elimination is the necessity to apply an initial calibration procedure carried out in a known (or at least unchanging) acoustic field. For PP probes designed for sound power measurements, additional equipment (pistonphone) or a reference probe is additionally required. Furthermore, the calibration procedure has to be systematically repeated due to variations of the microphone parameters over time.

Elimination of the necessity of the initial probe calibration is possible by use a single-microphone PP method. In this type of PP probe the measurement microphone is placed in positions corresponding to the positions of microphones in the classical PP probe. Similar to a standard PP probe, the microphone must be set in two positions to measure one sound intensity component. For two and three components, the number of positions is 4 and 6, respectively.

Acoustic pressure is measured at each position of the microphone. Of course, these measurements must be synchronized and the acoustic field must be repeatable. Two types of synchronization could be used:

- direct synchronization of the excitation signal and measurement data acquisition process;
- use of an additional (external) synchronization signal.

Direct synchronization of the acquisition and generation processes is only possible in a limited number of applications – generally only in cases where the sound source is a loudspeaker set controlled by a generation system synchronized with the acquisition system. This is impossible to achieve when the sound source is, for example, an electrical machine.

In direct synchronization method, no additional data acquisition channels used to synchronize signals from subsequent measurements are required. This approach can be used, for example, to measure the acoustic parameters of loudspeakers [19], diffusers, reflective elements [18] or to measure the impulse (intensity) response of the room [20].

The second method of synchronization can be used to analyse acoustic fields generated by “autonomous” sources. This means that the acoustic source is not directly synchronized with the data acquisition process. Although, the acoustic field generated by the source must be periodically. In this case, an additional measurement channel used to synchronize the signals recorded at successive microphone positions is required.

The main disadvantage of the single-microphone method is that it can only be used for periodically acoustic signals.

In [16] a method with external synchronization was used to measure the sound intensity distribution in the waveguide. This method required an additional reference signal from generator. The consequence of use of only one measurement microphone was the elimination of any errors (including phase mismatch error) associated with differences in the frequency responses of the microphones. The article discusses the influence of sampling frequency on the accuracy of measurement and how to increase it by oversampling. A high quality measurement microphone was used in the experiment. The results obtained were compared with measurements made with a commercial PP-type probe.

In the works [17–19] direct synchronization of the measurement signal generation and acquisition processes was used, which allowed the elimination of the reference microphone. The measurement method was used to measure the reflective properties of materials and to measure the parameters of loudspeakers. A commercial PU-type probe was used as a reference.

In [20] a comparison was made between two measurement methods using low-cost miniature microphones: a classical PP

Table 1. Comparison of PP methods
Tabela 1. Porównanie metod PP

Classical PP	One-Microphone PP	
	with direct synchronization	with external synchronization
– the necessity to apply a calibration procedure under controlled acoustic field conditions – can be applied to any type of acoustic field (stationary and non-stationary)	– calibration procedure is unnecessary – can only be applied to situations where the sound source is fully synchronized with the data acquisition process (e.g. loudspeaker and microphone controlled by a common generation/acquisition interface)	– calibration procedure is unnecessary – acoustic signal must be periodic (e.g. an electric machine unning periodically) – random error is high and dependent of sampling frequency – additional (reference) microphone (or another sensor is necessary to synchronize measurements in two positions

probe and a single-microphone PP probe using direct synchronization of the signal acquisition and generation process.

Table 1 gives an illustrative overview of the advantages and disadvantages of the classical PP method, the single-microphone PP method with direct synchronization and the single-microphone PP method with external synchronization.

The aim of this paper is to compare the classical PP method and the single-microphone PP method wit external synchronisation using an additional reference microphone. Both probes were based on low-cost microphones. The performed experiment consisting in the estimation of the angle of incidence of the wave on the measuring probe allows for a quantitative comparison of both methods and an assessment of the efficiency of phase mismatch error elimination in the single-microphone method.

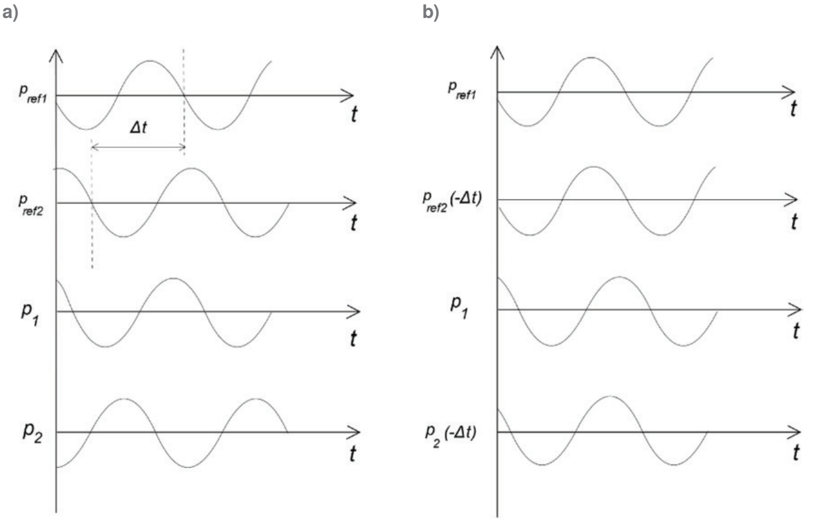


Fig. 3. Signals from the measurement and reference microphone: before (a) and after (b) synchronization. In the final form are the signals: p_1 and $p_2(-\Delta t)$

Rys. 3. Sygnały z mikrofonu pomiarowego (p(t)) oraz mikrofonu referencyjnego (pref(t)) zarejestrowane w pozycji 1. i 2. Sygnały przed (a) oraz po (b) przeprowadzeniu synchronizacji. Sygnały w ostatecznej wersji (gotowej do wyznaczenia na ich podstawie jednej składowej natężenia dźwięku) to: p_1 oraz $p_2(-\Delta t)$

2. One-microphone PP with external synchronization – detailed description

In this measurement method the acoustic pressure is recorded simultaneously by the measurement microphone (placed in successive positions) and an additional reference microphone, located in the constant position during the all measurement process. An example of a measurement sequence in two positions is shown symbolically in Fig. 2.

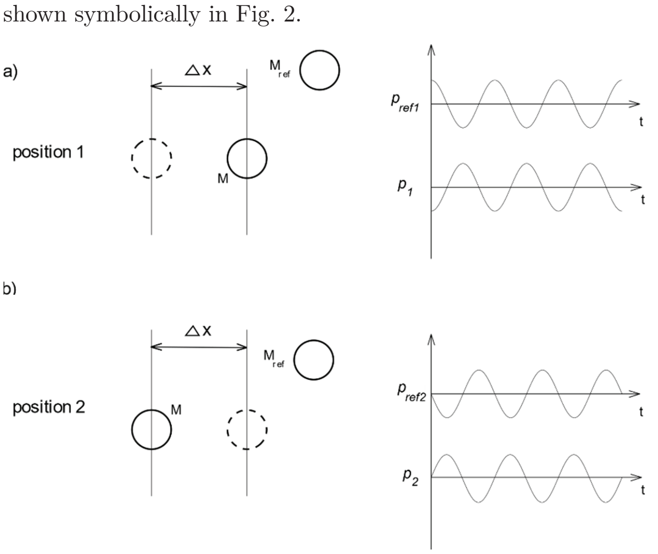


Fig. 2. Position 1 (a) and position 2 (b) of the measurement microphone (M) and reference microphone (M_{ref}) during the measurement. The example waveforms recorded by the microphones are presented on the right

Rys. 2. Pozycja 1 (a) i pozycja 2 (b) mikrofonu pomiarowego (M) oraz mikrofonu referencyjnego (M_{ref}) podczas pomiaru. Przykładowe zarejestrowane sygnały są zaprezentowane po prawej.

The successive measurements are not synchronized, therefore the phases of the recorded signals are random. The application of an algorithm (e.g. cross-correlation) allows determining the phase shift between reference signals recorded in the subsequent measurements.

Then the signal p_2 is shifted by this value in relation to the signal p_1. The result is a pair of signals whose phase shift corresponds to the phase shift between signals acquired by microphones in a standard PP probe. The process of signal synchronization is illustrated in Fig. 3.

An important feature of the synchronous method with external synchronization is that, although two microphones are used (a variable position measurement microphone and a reference microphone), only the signals from the measurement microphone are used to calculate the sound intensity. The reference microphone is used only to synchronize the signals from the measurement microphone. Therefore, differences in the frequency responses of the measurement and reference microphones do not affect the accuracy of the measurement.

The described method of measurement allows theoretically the complete elimination of the phase mismatch error (and other errors resulted from non-identical parameters of PP probe channels). Unfortunately, a significantly higher random error in comparison with the classical PP method occurred in the modified method (it is connected with the sampling frequency). According to the theory, this error can be eliminated using statistical methods.

By sampling a signal of the frequency f_a with sampling frequency f_s maximum value of synchronization error φ_s is done by:

$$\varphi_s = \frac{360° f_a}{f_s} \tag{12}$$

The reduction of the error can be obtained most simply by increasing the sampling rate. For example, for: $f_a = 1$ kHz, $\Delta x = 10$ mm and assumed $\varphi_s < 0{,}1°$, $f_s = 3.6$ MHz. Such a high sampling rate is difficult to achieve with equipment designed for audio recording. Another way to reduce random error is oversampling performed after the data acquisition process. This solution was proposed in [16].

Since the error φ_s has a random character, it is possible to use statistical methods to eliminate it. The simplest method is to make a series of measurements and average the results. According to the theory, performing N averages reduces the standard deviation (whose main source is φ_s) $\sqrt{N}$ times.

3. The experiment

To verify the possibility of eliminating the phase mismatch error by using the one-microphone measurement method with external synchronization (with reference microphone), an experiment was carried out. It was based on the determination of the angle of attack of the acoustic wave on the intensity probe.

There is a group of specialized devices (Acoustic Vector Sensor – AVS [2, 12, 13, 17, 18]) for the determination of Direction of Arrival (DoA) of acoustic wave. In the case of the proposed one-microphone measurement method, it is not possible to apply it to the determination of DoA for real-time random signals. Nevertheless, conducting an experiment consisting in the estimation of the angle under controlled conditions of a steady acoustic field for known values of this angle allows to determine the estimation error. Then the phase mismatch error can be determined.

3.1. Influence of phase mismatch error on the accuracy of acoustic wave incidence angle estimation

In a two-dimensional plane, to estimate angle α (Fig. 4), two components of sound intensity (perpendicular to each other) marked as I_x, I_y (13) must be measured.

$$\alpha_{est} = \mathrm{arctg}\left(\frac{I_y}{I_x}\right) \tag{13}$$

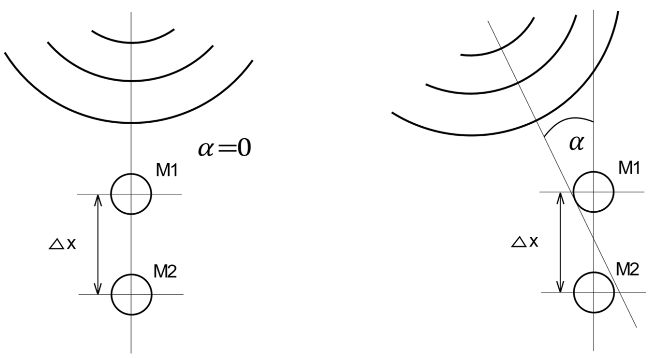


Fig. 4. Interpretation of the incidence angle of the wave on the PP probe. The direction of wave propagation coincides with the probe axis (left), the direction of wave propagation is angled away from the probe axis by angle α (right)
Rys. 4. Interpretacja kąta padania fali akustycznej na sondę pomiarową. Kierunek propagacji fali pokrywa się z osią sondy (po lewej), kierunek propagacji fali jest odchylony o kąt α od osi sondy (po prawej)

In the classical PP probe (assuming identical amplitude responses of both microphones) the angle α is estimated by the equation:

$$\alpha_{est} = \mathrm{arctg}\left(\frac{I_y}{I_x}\right) = \mathrm{arctg}\left(\frac{\Delta x \dfrac{\omega}{c}\sin\alpha + \varphi_b}{\Delta x \dfrac{\omega}{c}\cos\alpha - \varphi_b}\right) \tag{14}$$

where φ_b is phase mismatch error.

Estimation error is given by:

$$\alpha_{err} = \alpha_{est} - \alpha \tag{15}$$

Figure 5 shows the calculated error values of the angle estimation α as a function of the true value of this angle for several values of phase mismatch error for frequencies: 500 Hz and 1000 Hz.

3.2. Experimental setup

The measurement system is shown in Fig. 6. Type of the used microphones was Sonion 8011. Diameter of the microphones was 2.56 mm (0.1 inch). Frequency response was about 150 Hz–12 kHz. Details of the used transducers are described in [32]. The distance between the geometric centers of the microphones was 10 mm. Microphones were placed in a specially prepared plastic, symmetrical disc-shaped head. It is visible in Fig. 7. The positioning inaccuracy (estimated at 0.1 mm) introduces an additional phase mismatch, which for low frequencies is negligible (for 250 Hz about 0.02 degree). For higher frequencies its importance increases (for 5 kHz it is 0.5 degrees).

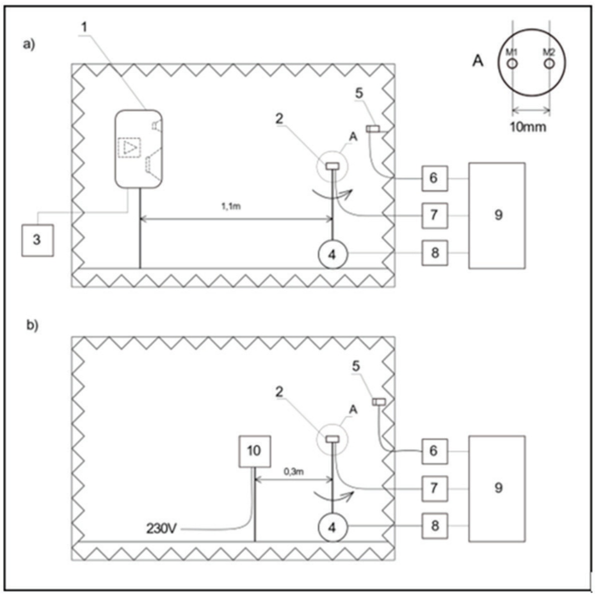


Fig. 6. Measurement setup in the anechoic chamber. Measurement of: loudspeaker set (a), small electrical device (b) Description of the components: 1 – Active loudspeaker set, 2 – Measurement probe head, 3 – signal generator, 4 – Stepper motor, 5 – Reference microphone, 6, 7 – Amplifiers, 8 – Stepper motor driver, 9 – Industrial computer, 10 – Tested electrical device
Rys. 6. Stanowisko pomiarowe w komorze bezechowej. Pomiar z wykorzystaniem: zestawu głośnikowego (a), małego urządzenia elektrycznego (b). Opis elementów: 1 – aktywny zestaw głośnikowy, 2 – głowica sondy pomiarowej, 3-generator sygnału sinusoidalnego, 4 – silnik krokowy, 5 – mikrofon referencyjny, 6,7 – przedwzmacniacze, 8 – sterownik silnika krokowego, 9 – komputer przemysłowy, 10 – badane urządzenie elektryczne

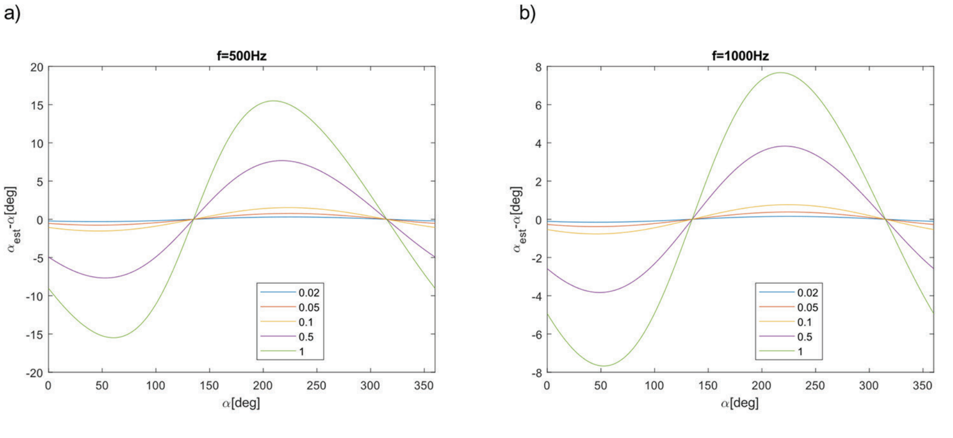


Fig. 5. Values of angle α estimation error for 500 Hz (a) and 1000 Hz (b) for different values of phase mismatch error (values in degrees) as a function of angle α
Rys. 5. Wartości błędu estymacji kąta α dla częstotliwości 500 Hz (a) oraz 1000 Hz (b) dla różnych wartości błędu fazowego (jego wartości podane w stopniach) jako funkcja kąta α

The stepper motor allows the head to rotate in the horizontal plane with a resolution of 1.8 degrees (there are 200 positions per full rotation). Due to the presence of two microphones, the probe can be used as a classic PP probe (using two microphones) or as a single-microphone probe (using one microphone in the probe and reference microphone which is fixed and located outside the probe – Fig. 6). The measurements were performed in an semi-anechoic chamber with a lower cut-off frequency of about 300 Hz. The sound source was a loudspeaker set type Genelec 8040 [33] controlled by a signal generator. Measurement data acquisition is performed by analog to digital converters (ADC) with a 16-bit resolution that are part of the PXIe-6368 [34] card. It cooperates with an industrial computer type PXie-1082 [35].

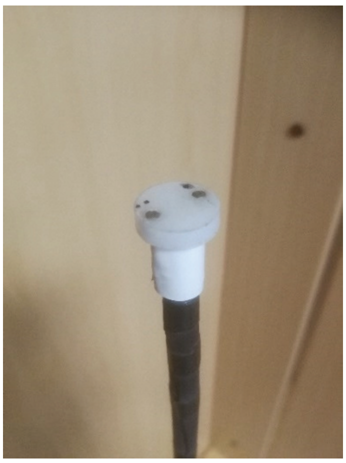


Fig. 7. Probe head
Rys. 7. Głowica sondy

To determine the angle estimation error for the full range of angles the probe head was set in 200 successive positions. At each position, the acoustic pressure was measured with the probe microphones and with an additional reference microphone.

This approach allowed the estimation error of the α angle to be determined using the classical PP method (the signals from both: M1 and M2 microphones, were used for the calculations) and the PP method with external synchronization (the signals from microphone M1 and the reference microphone were used for the calculations).

The successive head positions are shown in Figure 8.

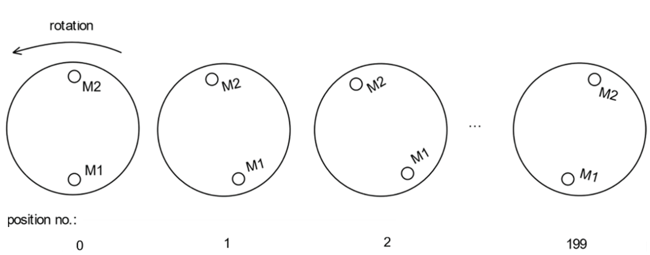


Fig. 8. Successive positions of the probe head during the measurement
Rys. 8. Kolejne pozycje sondy pomiarowej podczas pomiaru

A series of 100 measurements were made at each position of the probe head with a 100 kHz sampling rate. The described measurement procedure was carried out for three different sound sources:
- 500 Hz sinusoidal signal feeding the loudspeaker set;
- 100 Hz rectangular signal feeding the loudspeaker set;
- noise from the small electrical device (blender type SilverCrest SSMS 600 C3 600 W 230 V);

The measurement data were saved in standard LVM format. The head positioning and data acquisition were controlled by an application written in the LabVIEW software.

3.3. Data processing

The signals recorded in the described experiments allow the α estimation in each position of the probe head. The estimation was performed for both classical and modified methods to compare them.

Two methods of calculating I_x and I_y components of the sound intensity vector are presented below.

1. Classical PP method

In the classical PP method, at first, the sound intensity values (using Equation 11 in the discrete version) were calculated for each head position using two microphones: M1 and M2. The intensity at position n is calculated according to the:

$$I_n = f\left(p1_n(t), p2_n(t)\right) \tag{16}$$

The expression $f()$ denotes the function described by Equation (6), $p1_n(t)$, $p2_n(t)$ denote the acoustic pressures from microphone 1 and 2 recorded at position n, respectively. Then, based on the obtained intensities, the α estimations were calculated using the equation:

$$\alpha\_est_n = \begin{cases} a\tan\left(\dfrac{I_n}{I_{n+50}}\right), & \text{for } n \le 150 \\ a\tan\left(\dfrac{I_n}{I_{n-150}}\right), & \text{for } n > 150 \end{cases} \tag{17}$$

2. Modified (synchronized) PP method with external synchronization

In the modified method, a microphone (M1) and a reference microphone (M_{ref}) are used to determine the sound intensity components (I_n). The sound intensity at position n is determined by the equation:

$$I_n = f\left(p1_n(t), p1_{n+100}(t+\Delta t)\right) \tag{18}$$

The symbol $p1_n(t)$ represents the pressure from microphone M1 at position n, and $p1_{n+100}(t+\Delta t)$ represents the pressure recorded by microphone M1 at position $n + 100$ (which means rotating the head by 180 degrees) in reference to position n). This pressure is shifted in time by Δt. The Δt factor is calculated from the signals recorded by the reference microphone M_{ref} while the head was placed at positions n and $n + 100$. It is described by the equation:

$$\Delta t = m\frac{1}{f_s} \tag{19}$$

m takes the value for which the variable *diff* defined by (20) takes the minimum value. This means that the signals recorded by the reference microphone when the probe head was in the position n and in the position $n + 100$ are the best synchronized (in the sense of their squared error):

$$diff = \left(pref_n(t) - pref_{n+100}(t+m)\right)^2, \quad m = 0,1,2,\ldots,M \tag{20}$$

$pref_n(t)$ is a fragment of the signal recorded by the reference microphone when head was at position n, and $pref_{n+100}(t+m)$ is a fragment of the signal recorded by the reference microphone when the head was at position $n + 100$, shifted in time by m samples.

4. Results

Figure 9 shows the calculated values of the α estimation error (in degrees) as a function of the true value of this angle (head position) for a sampling frequency (f_s) of 50 kHz (obtained by the decimation of the original signal). Calculations were performed for several numbers of averages (denoted by N) of sound intensity I_n. Figure 10 shows the same data for $f_s = 100$ kHz.

Figure 11 shows the standard deviation values of the α estimation error for different averages for the classical method (blue color) and the modified method (red color) for $f_s = 50$ kHz (a) and $f_s = 100$ kHz (b).

For the analyses of the rectangular signal, the recorded signals (p_1, p_2, p_{ref}) were filtered with a 2nd order bandpass filter with Butterworth characteristics. The filtering was done for standard 1/3 octave bands with standard center frequency values: 250 Hz, 500 Hz, 1 kHz, 2 kHz, 5 kHz. For each of the described frequencies, the α estimation error values were calculated (only for the maximum number of performed averages $N = 100$). Results are presented in Fig. 12. The same analysis was performed for the recorded signal coming from the electrical device. The results are shown in Fig. 13.

5. Discussion

The conducted experiment performed for three types of acoustic signals allowed verification of how the type of signal influences the accuracy of α estimation error.

The measurement performed for the sinusoidal signal is the most idealized case, in which the factors that could influence the result (e.g. filtering process used for rectangular signal) were eliminated.

Figures 8–10 clearly show that increasing the number of averages N causes a decrease of the statistical error in synchronized PP method, but error in classical PP method is generally independent of the number of averages and sampling frequency. It is still equal about 6.7 degrees. This is obvious, since in the classical PP method the main component of the angle α estimation error is the phase mismatch error (bias error). In the synchronous method, in contrast, the main component of the angle α estimation error is the random error resulting mainly from the sampling frequency (but also from the variation of the acoustic field).

The synchronous method without any averaging ($N = 1$) has a larger α estimation error than the classical method (for both

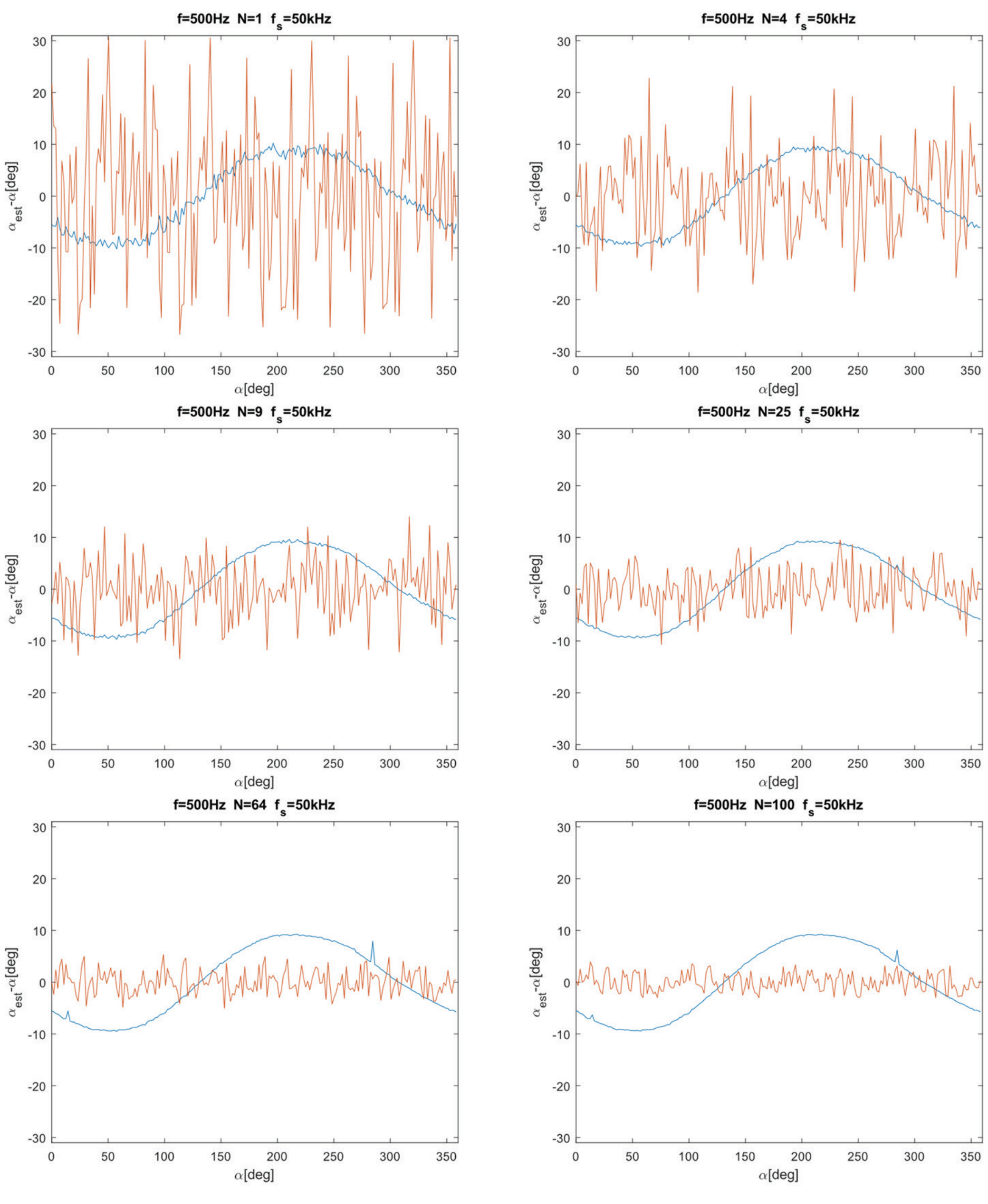


Fig. 9. α estimation error as a function of head position for several numbers of signal averages (N = 1, 4, 9, 25, 64, 100) for f_s = 50 kHz. The blue color indicates the classical PP method, the red color the modified (synchronous) PP method. Values expressed in degrees

Rys. 9. Błąd estymacji kąta α w funkcji pozycji głowicy dla różnych liczb uśrednień sygnału pomiarowego (N = 1, 4, 9, 25, 64, 100) dla częstotliwości próbkowania równej 50 kHz. Kolorem niebieskim oznaczono wyniki dla klasycznej metody PP, czerwonym dla zmodyfikowanej (synchronicznej). Wartości podano w stopniach

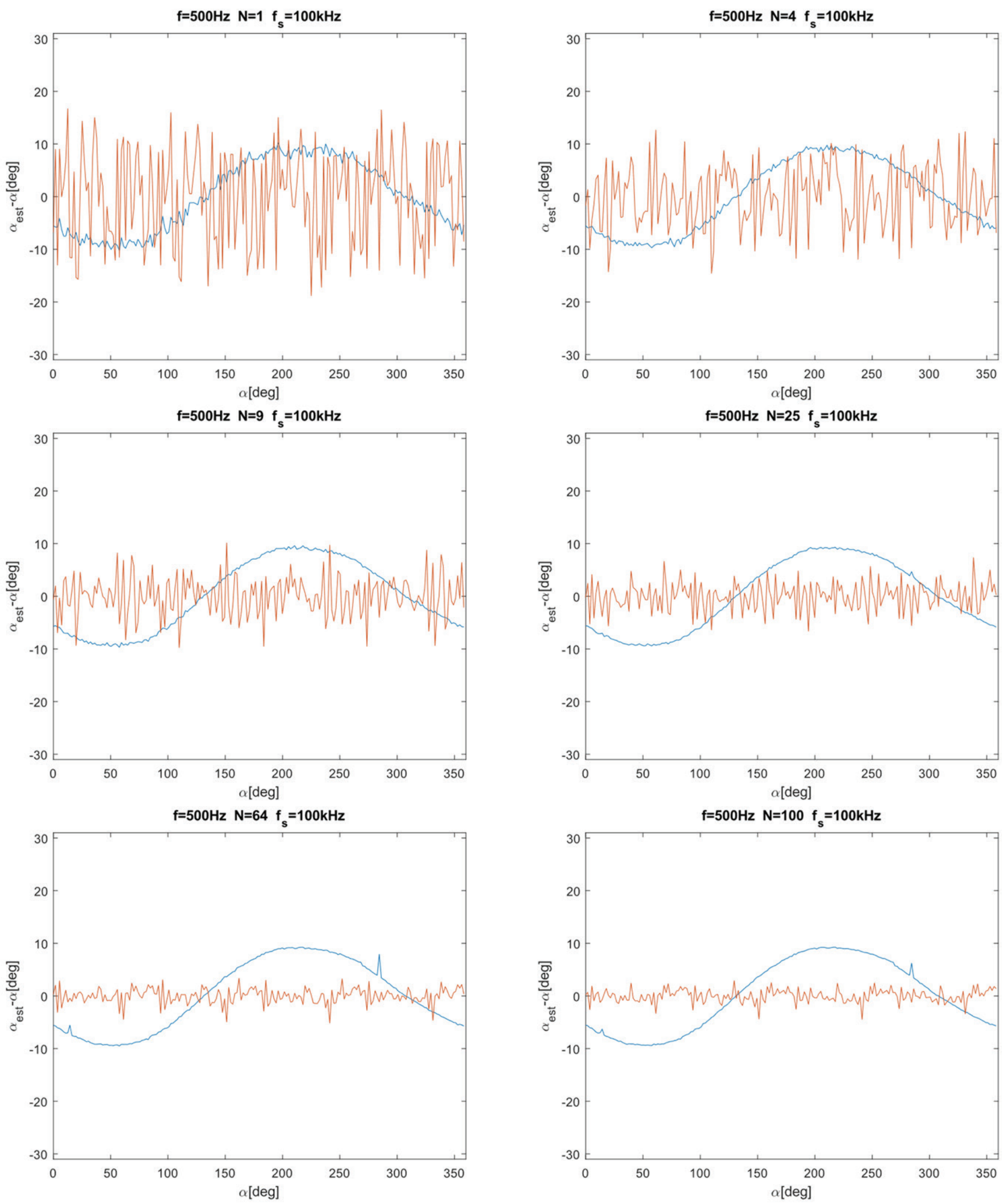


Fig. 10. α estimation error as a function of head position for several numbers of signal averages (N = 1, 4, 9, 25, 64, 100) for f_s = 100 kHz. The blue color indicates the classical PP method, the red color the modified (synchronous) PP method. Values expressed in degrees

Rys. 10. Błąd estymacji kąta α w funkcji pozycji głowicy dla różnej liczby uśrednień sygnału pomiarowego (N = 1, 4, 9, 25, 64, 100) dla częstotliwości próbkowania równej 100 kHz. Kolorem niebieskim oznaczono wyniki dla klasycznej metody PP, czerwonym dla zmodyfikowanej (synchronicznej). Wartości podano w stopniach

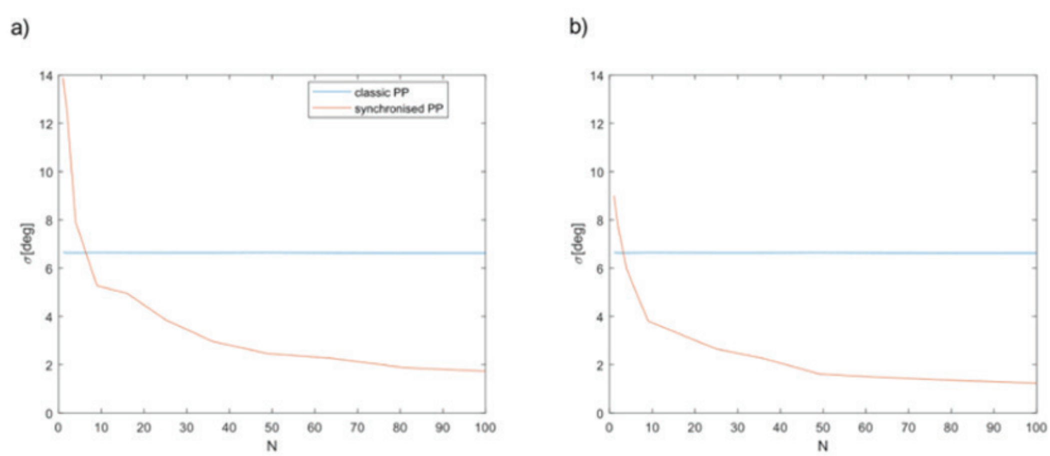


Fig. 11. Dependence of the standard deviation of the α estimation error as a function of the number of averaged measurements for f_s = 50 kHz (a) and f_s = 100 kHz (b). The blue color indicates the classical PP method, the red color the modified (synchronous) PP method. Values expressed in degrees

Rys. 11. Zależność odchylenia standardowego błędu estymacji kąta α w funkcji liczby uśrednień sygnału pomiarowego dla częstotliwości próbkowania 50 kHz (a) oraz 100 kHz (b). Kolorem niebieskim oznaczono wyniki dla klasycznej metody PP, czerwonym dla zmodyfikowanej (synchronicznej). Wartości podano w stopniach

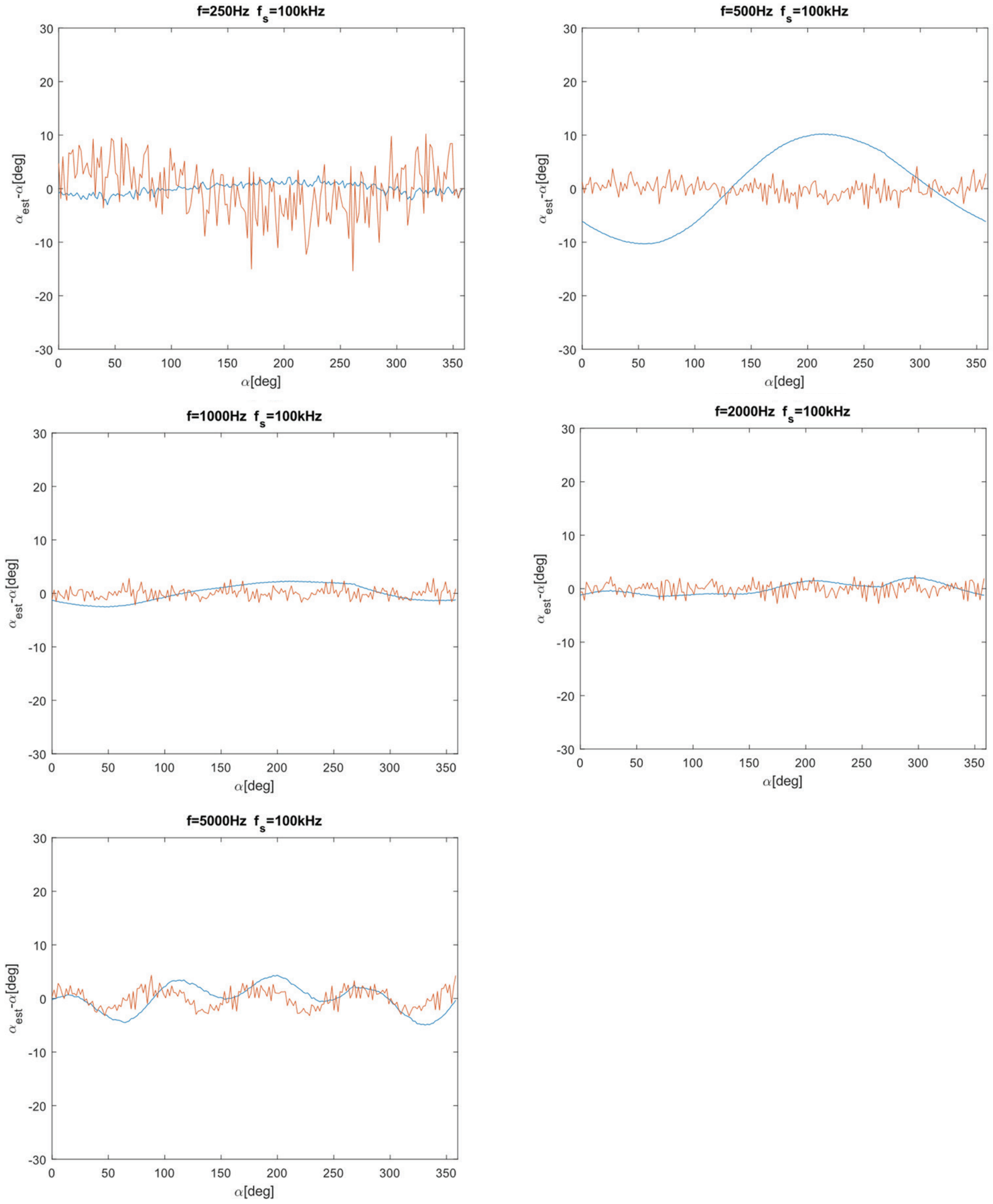


Fig. 12. α estimation α error as a function of head position for a rectangular signal, for several frequency bands. The blue color represents the classical PP method, the red color the modified (synchronous) PP method. The values expressed in degrees, f_s = 100 kHz, *N* = 100
Rys. 12. Błąd estymacji kąta α w funkcji pozycji głowicy dla sygnału prostokątnego dla różnych pasm częstotliwości. Kolorem niebieskim oznaczono wyniki dla klasycznej metody PP, czerwonym dla zmodyfikowanej (synchronicznej). Wartości podano w stopniach f_s = 100 kHz, *N* = 100

50 kHz and 100 kHz sampling rates). Statistical error in the modified method is higher than the phase mismatch error in the standard PP method. For the synchronous PP method (at $f_s = 50$ kHz), the number of N required to obtain a standard deviation of an error equal to the classical method is about 6. For the maximum number of averages $N = 100$, the standard deviation of the estimation error for the synchronous PP method is about 1.8 degrees, which is 3.7 times smaller than for the classical PP method.

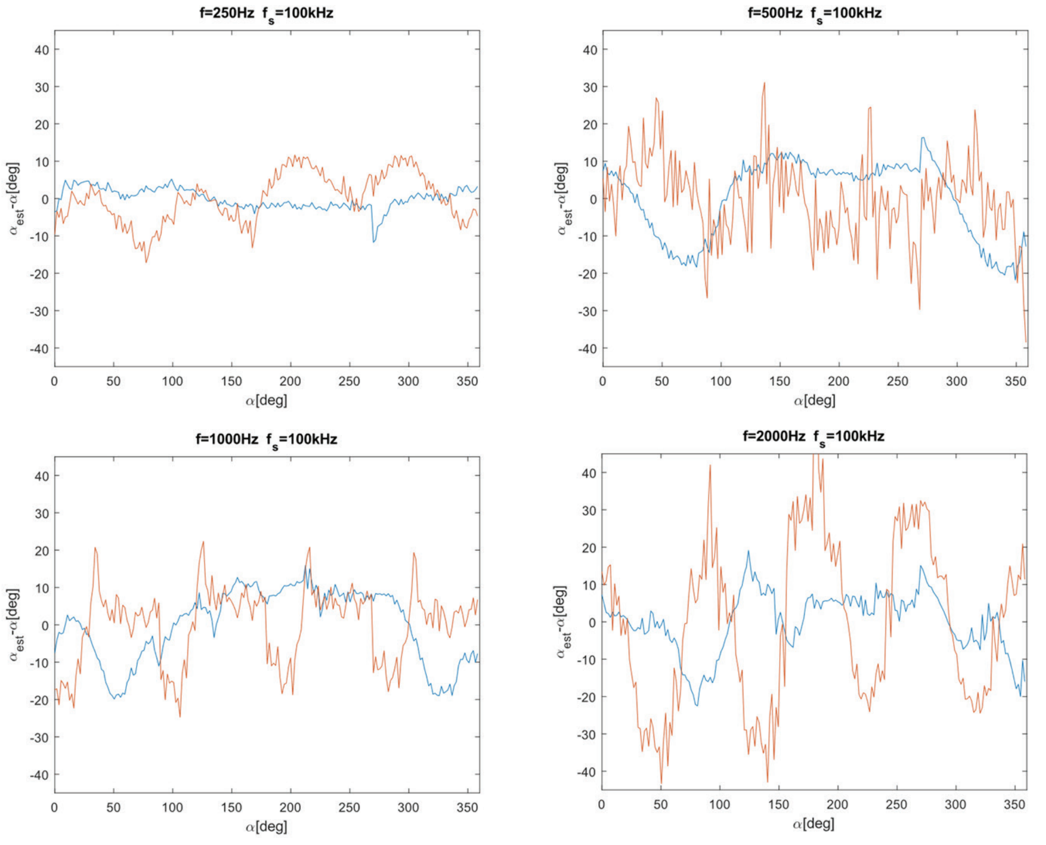
For $f_s = 100$ kHz the synchronous PP method gives a standard deviation of an estimation error equal to the classical PP method at $N = 3$. For $N = 100$ the standard deviation is about 1.1 degrees for the synchronous PP method (6 times smaller than classical PP method).

Based on the results of measurements of the α estimation error (Fig. 8, Fig. 9) and (30), the phase mismatch error between the probe microphones was calculated to be approximately 0.6 degree. This value is significantly higher than the error allowed by the standard [31]: class I probe: 0.05 degree and class II: 0.11 degree at 500 Hz. This means that the use of popular microphones in the classical PP probe does not provide the required accuracy. In the case of the proposed modified synchronous PP method (for $N = 100$), the error, as mentioned above, decreases by 3.7 times (for $f_s = 50$ kHz) and 6 times (for $f_s = 100$ kHz). It means that it is within the limits specified by the standard [31] for a class II probe. Results are presented in Table 2.

For the rectangular signal, the classical PP method gives significantly better results than the synchronous PP method at 250 Hz. However, the results for this frequency may be distorted because the cut-off frequency of the anechoic chamber is about 300 Hz. The advantage of the synchronous method is clearly visible at 500 Hz, while for higher frequencies the results for classical and modified PP method are comparable, with a slight advantage of the modified method.

The obtained α estimation error results are comparable with the results obtained in [20]. In this paper the modified PP method with direct synchronization of the measurement data acquisition and generation process was described. It should be pointed out that this method has a significant advantage over the method with external synchronization described here, as it does not require the use of averaging. However, it is not always technically possible to apply such synchronization.

For acoustic noise generated by an electrical device, both methods give significantly poorer results. It follows from the



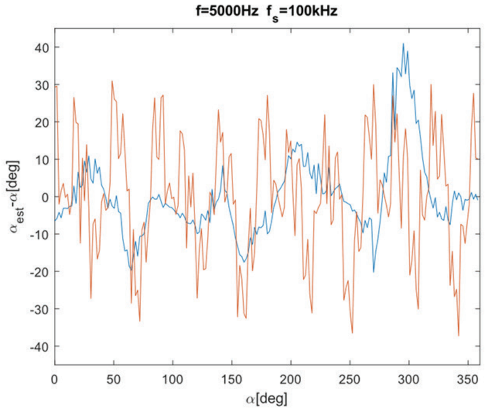


Fig. 13. α estimation error as a function of head position for noise generated by the small electrical device, for several frequency bands. The blue color represents the classical PP method, the red color the modified (synchronous) PP method. The values expressed in degrees, f_s = 100 kHz, N = 100

Rys. 13. Błąd estymacji kąta α w funkcji pozycji głowicy dla sygnału generowanego przez urządzenie elektryczne dla różnych pasm częstotliwości. Kolorem niebieskim oznaczono wyniki dla klasycznej metody PP, czerwonym dla zmodyfikowanej (synchronicznej). Wartości podano w stopniach. f_s = 100 kHz, N = 100

Table 2. Phase mismatch error values obtained for both methods

Tabela 2. Wartości błędu fazowego otrzymane dla obydwu metod

Type of measurement method / f_s – sampling frequency, N – number of averages	Classical PP	One-microphone PP with external synchronization
$f_s = 50$ kHz, $N = 100$	$\varphi_b = 0.6$ deg	$\varphi_b = 0.2$ deg
$f_s = 100$ kHz, $N = 100$	$\varphi_b = 0.6$ deg	$\varphi_b = 0.1$ deg

above facts that the effectiveness of α estimation error depends strongly on the nature of the sound source.

The results obtained for both methods are similar, with a slight advantage for the classical method. The maximum error for the synchronous method exceeds 40 degrees (for 2 kHz bandwidth). A similar error value occurs for the classical method for the 5 kHz bandwidth.

An important aspect regarding the synchronous method is the accuracy of microphone positioning. This is particularly important at higher frequencies (>1 kHz). The periodic nature of the error for the modified method, visible e.g. in Fig. 12 at 1 kHz, is most likely due to the inaccuracy of the microphone positioning. This fact may induce e.g. to use the synchronous method for low frequencies, while the classical method for higher frequencies.

6. Conclusions

The results presented in this paper show how, using single-microphone measurement with external synchronization, phase mismatch error can be minimized (or rather "replaced" by random error, which can be minimized by increasing the sampling rate or performing measurement averaging).

Experiment clearly shows that idea of replacing bias (phase mismatch) with statistical error and then removing it with statistical methods is correct. It allows the use of low-cost microphones in the PP probe while satisfying the requirements of the international standard.

The number of measurements performed in the presented experiment was large (200 probe positions, with 100 measurements in each). This large number of measurements was due to the need to test the full range of angles. In practical applications, the number of measurements can of course be lower and depends on the specific application. In general: The determination of one sound intensity component requires measurements at two microphone positions. The number of measurements taken at a certain position determines the number of averages that can be performed and thus increases the accuracy of the measurement.

The single-microphone method is not capable of replacing the classical PP probe, especially as it can only be used for stationary fields. It is, however, a supplement to it.

The described synchronous method has a significant advantage over other methods of phase mismatch error elimination that use DSP algorithms for phase mismatch error elimination. The proposed method does not require initial probe calibration before measurement under well-known measurement conditions.

The use of the single-microphone method with external synchronization allows an increase in the field of application (e.g. for testing noise generated by electrical machines) compared to the single-microphone method with direct synchronization.

The next research that would allow to better determine the range of usefulness of the synchronous PP method should consist in the measurement of the acoustic fields coming from sources generating a signal of a precisely defined level, spectrum, and signal-to-noise ratio (generated, for example, by an arbitrary generator).

It is also worth exploring the possibility of synchronizing the measurements using a mechanism other than an additional reference microphone e.g. an accelerometer placed on the surface of the test object.

The synchronous PP method has an advantage over the classical PP method for periodic signals with a large signal-to-noise ratio. Therefore, it is effective to use it especially in applications where signals coming from electroacoustic transducers controlled by signals with strictly defined parameters. Such applications are e.g. mapping of acoustic field surface around loudspeakers (this type of approach was already described in [19] and gave positive results), measurement of acoustic properties of materials, investigation of the impulse response of rooms.

References

1. de Bree H.-E., *The Microflown: An acoustic particle velocity sensor*, "Acoustics Australia", Vol. 31, 2003, 91–94.
2. Cao J., Liu J., Wang J., Lai X., *Acoustic vector sensor: reviews and future perspectives*, "IET Signal Processing", Vol. 11, No. 1, 2017, 1–9, DOI: 10.1049/iet-spr.2016.0111.
3. Chung J.Y., *Cross-spectral Method of Measuring Acoustic Intensity*, Research Publication, General Motors Research Laboratory, GMR-2617, Warren, Michigan, 1977.
4. Duan W., Kirby R., Prisutova J., Horoshenkov K., *Measurement of complex acoustic intensity in an acoustic waveguide*, "The Journal of the Acoustical Society of America", Vol. 134, No. 5, 2013, 3674–3685, DOI: 10.1121/1.4821214.
5. Fahy F.J., *Measurement of acoustic intensity using the cross-spectral density of two microphone signals*, "The Journal of the Acoustical Society of America", Vol. 62, No. 4, 1977, 1057–1059, DOI: 10.1121/1.381601.
6. Fahy F.J., *Sound Intensity*, 2nd edition London, England: E&FN Spon, 1995.
7. Farina A., Torelli A., *Measurement of the Sound Absorption Coefficient of Materials with a New Sound Intensity Technique*, "Journal of The Audio Engineering Society", 1997.
8. Gundre K., *Comparative Study and Design of Economical Sound Intensity Probe*, Open Access Master's Report, Michigan Technological University, 2019.
9. Jacobsen F., *A simple and effective correction for phase mismatch in intensity probes*, "Applied Acoustics", Vol. 33, No. 3, 1991, 165–180, DOI: 10.1016/0003-682X(91)90056-K.
10. Jacobsen F., *Handbook of Signal Processing in Acoustics. Intensity Techniques*, Springer: New York, NY, USA, 2008, 1109–1127.
11. Jacobsen F., de Bree H-E., *A comparison of two different sound intensity measurement principles*, "The Journal of the Acoustical Society of America", Vol. 118, No. 3, 2005, 1510–1517, DOI: 10.1121/1.1984860.
12. Kotus J., Czyżewski A., Kostek B., *3D Acoustic Field Intensity Probe Design and Measurements*, "Archives of Acoustics", Vol. 41, No. 4, 2016, 701–711, 2016, DOI: 10.1515/aoa-2016-0067.
13. Kotus J., Szwoch G., *Calibration of acoustic vector sensor based on MEMS microphones for DOA estimation*, "Applied Acoustics", Vol. 141, 2018, 307–321, DOI: 10.1016/j.apacoust.2018.07.025.
14. Krishnappa G., *Cross-spectral method of measuring acoustic intensity by correcting phase and gain mismatch errors by microphone calibration*, "The Journal of the Acoustical Society of America", Vol. 69, No. 1, 1981, DOI: 10.1121/1.385314.
15. Lanoye R., Vermeir G., Lauriks W., Kruse R., Mellert V., *Measuring the free field acoustic impedance and absorption coefficient of sound absorbing materials with a combined particle velocity-pressure sensor*, "The Journal of the Acoustical Society of America", Vol. 119, No. 5, 2006, DOI: 10.1121/1.2188821.
16. Mickiewicz W., Jabłoński M.J., Pyła M., *Calculation of spatial sound intensity distribution based on synchronised measurement of acoustic pressure*, [In:] Proceedings of the 18th International Conference on Methods & Models in Automation & Robotics (MMAR), Międzyzdroje, Poland, August 2013, DOI: 10.1109/MMAR.2013.6669996.
17. Mickiewicz W., Raczyński M., *Mechatronic 3D sound intensity probe and its application to DOA*, [In:] Proceedings of the 23rd International Conference on Methods & Models in Automation & Robotics (MMAR), Międzyzdroje, Poland, August 2018, DOI: 10.1109/MMAR.2018.8486005.
18. Mickiewicz W., Raczyński M., *Mechatronic Sound Intensity 2D probe*, [In:] Proceedings of the 22nd International Conference on Methods & Models in Automation & Robotics (MMAR), Międzyzdroje, Poland, August 2017, DOI: 10.1109/MMAR.2017.8046947.
19. Mickiewicz W., Raczyński M., *Modified pressure-pressure sound intensity measurement method and its application to loudspeaker set directivity assessment*. "Metrology and Measurement Systems", Vol. 27, No. 1, 2020, DOI: 10.24425/mms.2020.131720.
20. Mickiewicz W., Raczyński M., Parus A., *Performance Analysis of Cost-Effective Miniature Microphone Sound Intensity 2D Probe*, "Sensors", Vol. 20, No. 1, 2020, DOI: 10.3390/s20010271.
21. Nagata S., Furihata K., Wada T., Asano D.K., Yanagisawa T., *A three-dimensional sound intensity measurement system for sound source identification and sound power determination by ln models*, "The Journal of the Acoustical Society of America", Vol. 118, No. 6, 2005, DOI: 10.1121/1.2126929.
22. Oswald L.J., *Identifying the noise mechanism of single element of a tire tread pattern*, [In:] Proceedings of Inter-Noise 81, ed. Royster L.H., Hunt D., Stewart N.D., Noise Control Foundation, 1981, 53–56.
23. Prascevic R., Milosevic A., Cvetkovic S., *Determination of absorption characteristic of materials on basis of sound intensity measurement*, "Journal de Physique IV Colloque", Vol. 4 (C5), 1994, C5-159–C5-162.
24. Reinhart T.E., Crocker M.J., *Source identification of a diesel engine using acoustic intensity measurements*, "Noise Control Engineering", Vol. 18, No. 3, 1982, 84–92, DOI: 10.3397/1.2832203.
25. Sani M.S.M., Rahman M.M., Baharom M.Z., Zaman I., *Sound intensity mapping of an engine dynamometer*, "International Journal of Automotive and Mechanical Engineering", Vol. 12, 2015, 2820–2828, DOI: 10.15282/ijame.12.2015.2.0237.
26. Tervo S., *Direction Estimation Based on Sound Intensity Vectors*, [In:] Proceedings of the 17th European Signal Processing Conference, 2009.
27. Zhang Y., Fu J., Li G., *A Novel Self-Calibration Method for Acoustic Vector Sensor*, "Mathematical Problems in Engineering", 2018, DOI: 10.1155/2018/1219670.

Other sources

28. ISO 9614-1:1993 Acoustics — Determination of sound power levels of noise sources using sound intensity — Part 1: Measurement at discrete points; International Organization for Standardization: Geneva, Switzerland, 1993.
29. ISO 9614-2:1996 Acoustics — Determination of sound power levels of noise sources using sound intensity — Part 2: Measurement by scanning; International Organization for Standardization: Geneva, Switzerland, 1996.
30. ISO 9614-3:2002 Acoustics — Determination of sound power levels of noise sources using sound intensity — Part 3: Precision method for measurement by scanning; International Organization for Standardization: Geneva, Switzerland, 2002
31. International Standard. IEC Standard 1043: Instruments for measurement of sound intensity; International Electrotechnical Commission Geneva, Switzerland, 1993
32. Sonion. [www.sonion.com/hearing/microphones/8000-electret] (01.02.2021).
33. Genelec. [www.genelec.com/8040b] (01.02.2021).
34. National Instruments. [www.ni.com/pdf/manuals/374455c.pdf] (01.02.2021).
35. National Instruments. [www.ni.com/pl-pl/support/model.pxie-1082.html] (01.02.2021).
36. Sound Intensity Probe Kit Type 3599, Product Data, Brüel & Kjær Sound & Vibration Measurement A/S DK-2850 Nærum, Denmark, 2020

Wykorzystanie synchronicznego pomiaru do eliminacji błędu fazowego w sondzie typu PP

Streszczenie: W artykule zaprezentowano modyfikację metody pomiaru natężenia dźwięku z wykorzystaniem sondy PP (ang. pressure-pressure). W zaproponowanym rozwiązaniu jednoczesny pomiar ciśnienia akustycznego za pomocą dwóch mikrofonów pomiarowych (używanych w klasycznej sondzie PP) został zastąpiony sekwencją pomiarów dokonywanych jednym mikrofonem pomiarowym umieszczanym w kolejnych pozycjach. Zaproponowana metoda pozwala na eliminację błędu związanego z niejednakowymi odpowiedziami częstotliwościowymi (głównie fazowymi) mikrofonów użytych w klasycznej sondzie PP. Jej zastosowanie jest ograniczone do pomiaru sygnałów okresowych. Jednocześnie wzrasta błąd przypadkowy, który można jednak wyeliminować metodami statystycznymi. Pomimo, że zaproponowane podejście wymaga zastosowania mechanizmu synchronizacji pomiarów i użycia w tym celu dodatkowego mikrofonu pomocniczego, to w samym procesie wyznaczania natężenia dźwięku bierze udział jedynie sygnał z mikrofonu pomiarowego. W artykule zaprezentowano zasady pomiaru natężenia dźwięku za pomocą klasycznej sondy PP oraz obecnie stosowane metody eliminacji błędów związanych z niedopasowaniem charakterystyk częstotliwościowych mikrofonów, bazujące na wstępnej kalibracji sondy. Następnie przedstawione są teoretyczne podstawy zaproponowanej metody pomiarowej. Aby zweryfikować jej skuteczność przeprowadzono eksperyment pomiarowy polegający na pomiarze kąta padania fali akustycznej w ściśle określonych warunkach w komorze bezechowej. Eksperyment przeprowadzono z wykorzystaniem klasycznej metody PP oraz z wykorzystaniem zaproponowanej metody zmodyfikowanej. Dokonano pomiarów dla różnych źródeł dźwięku (zestawu głośnikowego oraz małego urządzenia elektrycznego). W końcowej części artykułu wyniki porównawcze są poddane dyskusji w celu wskazania potencjalnych zastosowań zaproponowanej metody.

Słowa kluczowe: sonda natężeniowa typu PP, błąd niedopasowania fazy, pomiar natężenia dźwięku

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