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MODIFIED PRESSURE-PRESSURE SOUND INTENSITY MEASUREMENT METHOD AND ITS APPLICATION TO LOUDSPEAKER SET DIRECTIVITY ASSESSMENT

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Abstract

Sound intensity measurements using special sensors in a form of pressure-velocity and pressure-pressure probes are becoming more and more often the method of choice for characterization of sound sources. Its wider usability is blocked by the probes' costs. This paper is on a possible modification of the well-known pressure-pressure sound intensity measurement method. In the proposed new approach a synchronized measurement procedure using only single microphone is used. The paper presents the basics of the sound intensity theory, a review of currently used methods of intensity measurement and requirements and limitations of the new method. In the proposed approach one microphone and a properly designed positioning system is used. The application of the method to study the directional characteristics of an active loudspeaker system have been described in detail. The obtained results were compared with those of measurements performed with a commercial p–u probe. The paper contains conclusions indicating advantages of the applied method in comparison with standard pressure measurement methods.

Keywords: sound intensity, pressure-pressure intensity probe, loudspeaker set measurements.

1. Introduction

The development of modern sensors, signal processing methods, multi-channel data acquisition systems and automatic positioning systems means that we are able today to measure physical quantities that have been known for many years, but technical limitations made it impossible to measure them. We could not measure them completely or the required costs and time consumption meant that the measurement of these quantities were not applied in the common industrial activities. Such physical quantities include vector quantities describing the acoustic field: acoustic velocity and sound intensity. The concept of sound intensity was brought by Lord Rayleigh into the modern acoustics in 1878 in his work The Theory of Sound [1]. But just 99 years later due to the independent work of F.J. Fahy and J.Y. Chung the measurement was possible with a digital sound meter system using a pressure-pressure probe and a dual-channel digital signal analyser using the fast Fourier transform [2, 4]. Since the beginning of the 1990s, the method has become a common practice in determining the acoustic power of sources and its application has been included in the framework of international standards [3, 5, 15]. It is also used to localize acoustic sources in complex machine systems.

Despite its stabilized position in the above applications, its use is more complicated compared with pressure measurements using a single microphone. The main problem is the need to use a special intensity probe: Fahy and Chung used pairs of measuring microphones very carefully selected in terms of frequency amplitude and phase characteristics, which directly affects the high cost of intensity pressure-pressure probe. The problem of high similarity of used microphones is still not fully resolved [16]. The measurement itself usually consists of scanning the measuring surface with either a single probe or multiple probes and requires the use of a specialized data processing system [12]. These facts contribute to the still low popularity of using the method in technical measurements of intensity and to the repressive opinion that, despite its advantages, it is reserved for users with sufficiently large budgets for the measurement equipment. On the other hand, this state of affairs causes a certain stagnation in widening knowledge about the acoustic phenomena, to explain which observation of acoustic energy flows, and not just effects in the form of acoustic pressure distribution would be very helpful.

This paper is the authors' contribution to extending the use of current methods by a sound intensity measurement method which does not require the use of expensive probes and can be used in research, among others, on electroacoustic transducers – *e.g.* loudspeakers. In the paper the authors present an idea of measuring sound intensity using one microphone and compare metrological properties of the proposed method with those of a commercial p–u probe. In the literature one can find solutions based on the use of cheaper microphone matrices with different geometry combined with a complex block correcting the mismatch between the converters using digital signal processing methods, *e.g.* [18]. Such an approach generally has at least 2 drawbacks. Firstly, the amplitude-phase characteristic corrections in the form of a digital filter corrects discrepancies between the transducers only with a certain approximation. The higher the accuracy of fitting, the higher the computational complexity of the algorithms used. Secondly, long-term changes in the parameters of the transducers require complicated broadband calibration. Thus, in the opinion of the authors, the possibilities of multi-microphone matrix systems seem attractive from the point of view of *e.g.* surround sound technology, but their use as a measurement instrument with specific metrological characteristics remains questionable. The proposed method is, in principle, free from these drawbacks, as the position of the transducer is changed, but the transducer itself and the measurement signal path remain the same. The problem of matching many measurement channels disappears. The basic problem in the new method is the evaluation of the uncertainty of measurement resulting from the assumption of the required stationarity of the measured object and the repeatability of the measurement process in the case of sequential measurement. The results of the research presented in the paper give an opinion on this issue.

In the second part of the paper the advantages using the intensity method in the case of measuring the directional characteristics of loudspeakers are shown. In our opinion, the prevalence of cheap but sufficiently trusted methods of measuring vector properties of the sound field can help to revolutionize technical acoustic measurements.

2. Scalar and vector measurements in acoustic field

The basic physical quantity recorded and measured in acoustics is sound pressure $p_a(t)$. It is a component varying in time and representing the disturbance of the medium as a result of the propagation of acoustic waves. If the medium is in the state of equilibrium (silence) under the pressure p_0, the sound pressure p_a is defined as the difference between the instantaneous pressure value in the medium with the acoustic wave $p(t)$ and the pressure p_0:

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

As a propagating sound wave causes momentary compression and rarefaction of the medium, the instantaneous values $p(t)$ take higher or lower values than p_0, so the instantaneous values of $p_a(t)$ can be either positive or negative. An alternating signal $x(t)$ is usually described using an *RMS* value x_{RMS} defined in general as:

$$x_{RMS} = \sqrt{\frac{1}{T}\int\limits_{t_0}^{t_0+T} x^2(t)\,\mathrm{d}t}. \tag{2}$$

The *RMS* value of acoustic pressure is given by (3) and is so called equivalent value [17]. Time T can be arbitrary chosen be the operator. A special model of *Sound Level Meter*, so called integrating SLM is used for such measurements. The SLMs which use time constants SLOW and FAST are called SLMs with exponential time weighting. The measured instantaneous *RMS* acoustic pressure is given by the formula [17]:

$$p_{aRMS}(t) = \sqrt{\frac{1}{T}\int\limits_{-\infty}^{t} p_a^2(\tau)e^{-\frac{t-\tau}{T}}\,\mathrm{d}\tau}, \tag{3}$$

where T value is the detector's time constant and in sound pressure meters it can be chosen from two values: FAST (125 ms) and SLOW (1 s). The result of the *RMS* value evaluation is always a positive number.

Due to the properties of human hearing, the logarithmic measure is commonly used in acoustics to express the sound pressure value. Thus, the *sound pressure level* (SPL) is defined as:

$$\mathrm{SPL} = 20\log\left(\frac{p_{aRMS}}{p_{ref}}\right). \tag{4}$$

From the above considerations, it is clear that the sound pressure measured at a single point as a scalar quantity does not carry any directional information about the flow of acoustic energy. A vector quantity describing the flow of acoustic energy is called the *sound intensity* (SI). The instantaneous value of the sound intensity $I_{inst}(t)$ is defined as the product of the sound pressure $p_a(t)$ and the velocity of the sound particle $\vec{u}(t)$:

$$\overrightarrow{I_{inst}}(t) = p_a(t) \cdot \vec{u}(t). \tag{5}$$

The mean value of the instantaneous sound intensity is called the active component of the sound intensity and is determined as follows:

$$\overrightarrow{I_{act}} = \frac{1}{T}\int\limits_{0}^{T} \overrightarrow{I_{inst}}(t)\,\mathrm{d}t. \tag{6}$$

The unit of sound intensity is $\frac{\text{W}}{\text{m}^2}$.

The above definitions imply important sound intensity properties. The instantaneous sound intensity is a vector quantity and its orientation is consistent with the vector of acoustic velocity. The magnitude and sense of this vector, however, depends on the sound pressure value. Since, as mentioned earlier, the instantaneous sound pressure values can be either positive or negative, the intensity vector sense can be either the same as velocity vector or opposite to it. It is also worth mentioning that an element of acoustic field called acoustic particle can vibrate with a simple trajectory of its mass centre in the form of straight-line segment, but in general in the form of ellipsoid. In the second general case when transitioning from momentary to mean values, the orientation and value of active part of sound intensity is influenced by the phase relation between the pressure wave system and the acoustic velocity wave system. In the free far field, where the observed waves can be considered as flat, the phase shift between the pressure wave and the velocity wave is zero and the direction of the sound intensity vector coincides with the propagation direction and the sense indicates the energy flow from the source to the surrounding space. The matter becomes more complicated in the acoustic field near real sources and / or in the presence of waves reflected from obstacles. In the near field, the real source should be treated as an extended source and – in the frequency range of human auditory system – a correct model must consider the frequency-dependent amplitude and initial phase of individual elementary sources as a function of frequency. The operation of such a system causes interference effects in the surrounding field and the appearance of phase shifts between pressure waves and velocity waves. Acoustic energy ceases to propagate only on straight lines radially from the source, the resonant and non-linear phenomena can lead to the formation of acoustic vortex fields.

There are two types of sound intensity probes which vary in the method of particle velocity measurement. The first type called p–u probe uses different physical phenomena to measure sound pressure and particle velocity. Pressure is measured using a condenser microphone. Particle velocity is obtained using a special sensor, *e.g.* an anemometric sensor with hot wires (as in Microflown [9]) or an ultrasonic sensor. This type of probe was used during the experiments as a reference probe.

The second type of sound intensity probe called p–p probe uses a pair of matched microphones. The principle of operation of this probe is the base for the method developed and presented in the paper. Particle velocity is evaluated from the gradient of acoustic pressure using a linearized Euler equation:

$$\vec{u}(t) = -\frac{1}{\rho} \int \frac{\partial p(t)}{\partial \vec{x}} \, \mathrm{d}t, \tag{7}$$

where u – particle velocity; ρ – density of environment; p – acoustic pressure; x – space variable.

After discretization of (6) it takes the following form:

$$\vec{u}(t) = -\frac{1}{\rho} \int_{-\infty}^{t} \frac{p_{a1}(\tau) - p_{a2}(\tau)}{\Delta x} \, \mathrm{d}\tau. \tag{8}$$

The gradient of acoustic pressure is substituted by the finite difference of two acoustic measurements performed by a pair of microphones which are placed next to each other. The lower integration limit is the beginning of the test signal. To measure one, two or three components of the sound intensity vector, there are necessary one, two or three pairs of microphones, respectively. It is symbolically presented in Fig. 1.

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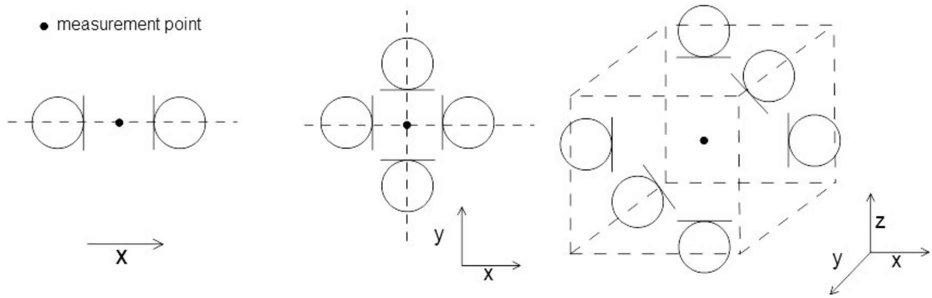

Fig. 1. Alignment of microphones in 1D, 2D and 3D p–p intensity probes.

SPL is calculated as an average result of measurements performed by two microphones. The upper measurement frequency range is limited by a proper approximation of derivative by finite difference. The low frequency range is limited by the phase mismatch error and noise of the measured signal. A typical distance is 12 mm and it provides measurements in the frequency range: 125 Hz – 5 kHz. A general disadvantage of p–p probes is the necessity of using microphones which have identical amplitude and phase frequency characteristics. In practice it is impossible to obtain perfect matching. Thus, there are necessary complex calibration procedures and/or correction of characteristics by analogue or digital filters. Furthermore, the numerical derivative operation is very sensitive to noise.

3. Modification of pressure-pressure method

In some cases where the analysed acoustic field is generated by a repetitive and known excitation and under one assumption that the parameters of the measured system are time invariant can be valid, the pressure-pressure approach to evaluating sound intensity can be modified and simplified. In the modified method proposed by the authors only one microphone and one data acquisition channel have to be used [10, 11]. The simultaneous measurement of sound pressure by two microphones placed at some distance between them to find a one-directional component of sound intensity vector is substituted by a sequence of two measurements per direction taken by the same microphone moved to a new position by the automated positioning system. The measurement positions of one microphone correspond to positions of microphones in p–p probe.

From the metrological point of view, the measurement errors associated with the mismatch between the transducers have been eliminated in the proposed method. However, there were errors related to the non-stationarity and uniqueness of the measurement conditions: parameters of the measured acoustic field and arrangement of the transducers in sequenced measurements. Budget microphones are also usually characterized by a higher level of self-noise, which can also affect the accuracy of measurement. To reduce this drawback, pressure pulse responses are determined indirectly during measurements using a logarithmic sweep signal, which can be emitted repeatedly to improve the signal-to-noise ratio. Careful consideration and modelling of the above error sources has not yet been done. In this stage of research, the measurement uncertainty was estimated on the basis of the distribution of the measured value in subsequent attempts to measure the same object in the established conditions.

In order to show the sense of searching for a simple and cheap method of sound intensity measurement, the following part of the paper shows also its advantages in the case of measuring the directional characteristics of loudspeakers.

4. Loudspeaker directional characteristic – classic and modified approach

The directional characteristic of a loudspeaker or a loudspeaker-cabinet system is an important parameter describing the properties of a given sound source and its suitability for specific applications. The standard procedure for measuring the directional characteristic of a loudspeaker is limited to determining the sound pressure distribution on a circle around the loudspeaker and, as a rule, has remained unchanged for at least 30 years [6, 7]. The measurement is carried out in an anechoic chamber in free field conditions and the radius of the measuring circle should be big enough to guarantee far-field conditions. Depending on the technical solution, the measurements in subsequent points are performed either by turning the loudspeaker on a rotating platform in a fixed position of the measuring microphone or by moving the microphone on an appropriate arm at a fixed position of the loudspeaker. With the development of computer measurement techniques, this procedure has now been practically completely automated. In each measurement point, a test signal is supplied to the loudspeaker. Currently, as the test signal there is widely used a logarithmic sine sweep signal [8]. The acoustic signal generated by the loudspeaker is recorded by the measuring microphone, and the output voltage signal is supplied to the data acquisition card placed in the computer measurement system. Knowing the excitation signal and the registered response of the loudspeaker system, the measurement system applies convolution algorithms and determines the pulse response and frequency response of the system for a given measurement point. Then, using the mechanical actuators (turntable), the measurement system moves to the next measurement point and the whole procedure is repeated. After registering a certain number of points on the circle around the loudspeaker, the measurement system is able to generate pressure directional characteristics, which are usually presented as a family of curves for individual frequency bands on a common polar chart. The individual curves are normalized to the value of pressure measured on the loudspeaker axis. Despite the widespread use of these data and their undoubted informative value, the obtained characteristics do not contain detailed information on the direction of acoustic energy propagation in each measurement point. By default, it is assumed that the direction of energy propagation coincides with the direction of the radius for a given measurement point. It is assumed that the measurement takes place in the far field, where the size of the source in relation to the distance is so small that it can be assumed that it is the point source. This does not apply to the case when large loudspeakers are measured. In this case, one cannot ignore the dimensions of the source, which, as a result of interference, create a field with the reactant component around the loudspeaker and the acoustic energy does not propagate radially. Some researchers proposed a solution to this problem in a form of *near-field acoustic holography* (NAH) [13, 14], which is interesting but still use only the scalar pressure measurements and model-based approach.

Another way to obtain more detailed information about the directional characteristic of the radiation of a given sound source, which is proposed in this paper, is observing a vector quantity, which is the sound intensity. The microphone should be substituted by an intensity probe, which in the best way defines the vector of sound intensity in a three-dimensional space. As we suggest, it can be done using still the same classic automatized measurement system with one microphone. The pressure distribution has to be scanned on two circles around the loudspeaker whose radii differ by a length of ca. 1 cm. Thanks to the synchronization of all measurements, it is possible to determine, besides the standard sound pressure distribution, also the distribution of the sound intensity on the measured circle in its plane (normal and tangential components). The drawing in Fig. 2 shows the method for determining the tangent and radial components of the sound intensity. To calculate the intensity vector on the plane I_{xy}, we take into account pressure measurements from four points: a point for a given angle on a smaller circle (n), a point for a given angle on a larger circle (N), and similarly two points for the next angle values (n + 1 and N + 1). As can be seen in the figure, the points form a trapezium, whose side arms have a length equal to the difference between the radii of the measuring circles, and the bases can be approximated by an arc length of 1.8° and two radius lengths. As can be seen, the decreasing spatial resolution of the measurements should be taken into account as the frequency increases due to the indirect determination of the resultant vector on the basis of average values from four neighbouring measurement positions.

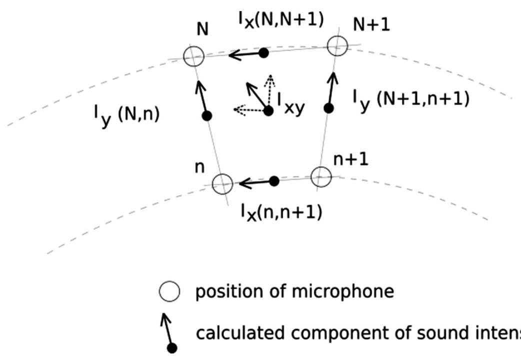

Fig. 2. Pressure measurement points for determining the tangent and radial components of the sound intensity vectors.

5. Measurement setup and experiments

To test the presented ideas in practice, a measurement stand consisting of the following equipment was constructed: the *device under test* (DUT) – an active Genelec 8040 loudspeaker suspended on a tripod, a microphone positioning system based on a bipolar stepper motor and an arm with a nominal length of 56 cm with the possibility of increasing the length by 10 mm using a precise manual feed system with a linear gear, an acquisition and generation system based on a National Instruments industrial computer PXIe-1082 type with PXIe-6368 (excitation signal generator, DAC card, 16-bit resolution) and PXIe-4499 (acquisition ADC card, 24-bit resolution, noise flor < −110 dBFS) ADC and DAC converter cards controlled by software developed in the LabView environment and a Microflown p–u measurement probe [9]. A photo of the measurement stand in an anechoic chamber is shown in Fig. 3a, and a schematic diagram of the principle of operation – in Fig. 3b.

In an anechoic chamber that enables to obtain free field conditions for frequencies above 300 Hz, the DUT and measuring probe on the positioning system are included. The industrial computer with measurement cards is located next to the anechoic chamber in the control room. The process of measuring the whole directional characteristic was carried out as follows. The measuring probe was placed at the starting point (the arm angle equal to 0) and the length of the arm was set to 56 cm (the shorter one). A test signal – a sinusoidal sweep signal from 100 Hz

a) b)

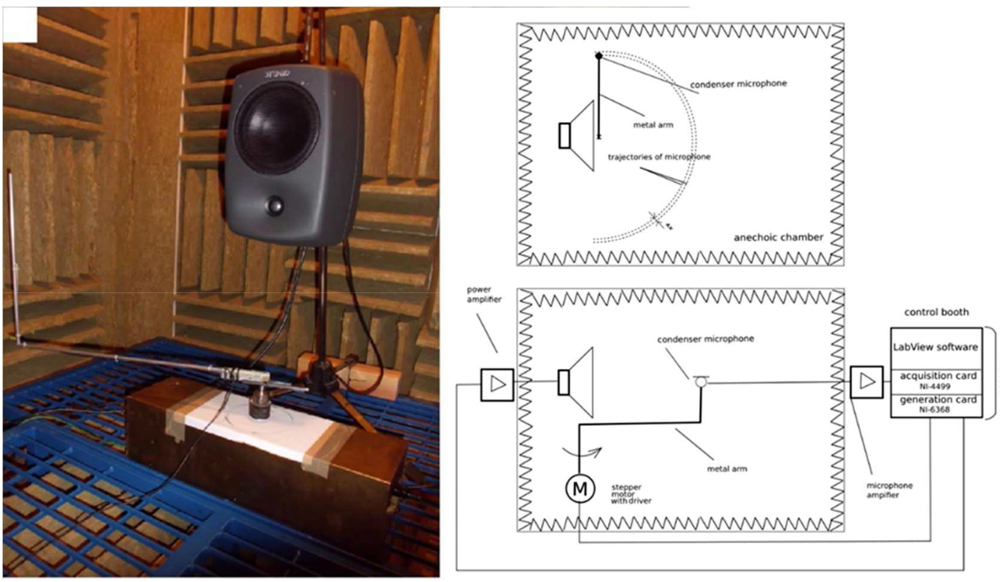

Fig. 3. The measurement stand. a) A photo; b) a schematic diagram.

to 10 kHz with a length of 10 s – was supplied to the active speaker set from the generator card. Synchronized with the generation process, the acquisition of signals from three velocity channels of the p–u probe (components x, y, x) and one pressure channel was started. The sampling rate was 100 kHz. After it was completed, a pulse sequence was given to the stepper motor, causing one step and twisting the arm by 1.8 degrees, which set the probe in the next measurement position. From this moment the generation and acquisition processes are repeated. In this way the probe was set in 100 positions evenly spaced on a semicircle with a radius of 0.56 m. The data collected in this way were sufficient to determine the sound intensity using the p–u probe. Additionally, to test the operation of the modified method based only on the pressure measurement, it was necessary to repeat all 100 measurements with the arm extended by 10 mm. It is worth noting that although the p–u probe was involved in the measurement, only the pressure measurements were used to determine the acoustic velocities using the modified measurement method. The acoustic particle velocities determined by this method were later compared with the velocities obtained directly from the velocity channels of the p–u probe. In order to examine the stationarity of the proposed measurement method and to evaluate the statistical spread of error for the same position, another series of measurements was carried out. The probe was placed in position no. 50, located directly in front of the loudspeaker set, and the arm was set to a shorter position. In this unchanged position, the measurement was repeated 100 times. Then, the length of the arm was increased by 10 mm and the measurement was repeated again 100 times. Then, on the basis of the measured values, the sound intensity values were calculated for subsequent data sets and a histogram was constructed. To shorten the stabilization time, the stepper motor control was optimized by introducing a micro-step control.

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6. Results

In Fig. 4 a pair of pressure pulse responses is shown obtained for two microphone positions: without displacement and after shifting by 10 mm on the radius. The pulses were calculated from the registered answers of the DUT to the sweep sine excitation according to the procedure described in [8]. We can also see a correlation of these two signals. The time gap between pulse responses corresponds to the physical radius length difference of the measuring arm.

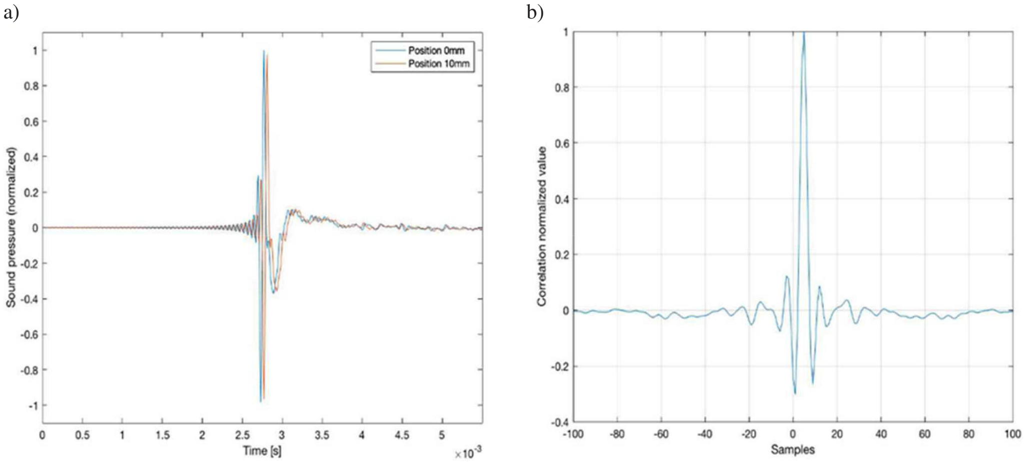

Fig. 4. Pressure pulses from a microphone in two measuring positions (a) and their correlation (b).

In Fig. 5 we can see a histogram showing the distribution of sound intensity values obtained by a combination of 100 measurements of pressure performed in the first microphone position (the shorter measuring arm) and 100 measurements carried out in the second position (the longer arm). The error spread of only about $+/-0.2$ dB proves a good repeatability of the measurement system and stationarity of the measurement conditions.

In Fig. 6. the differences in levels of sound intensity, acoustic velocity and sound pressure obtained with the p–u probe and p–p modified method are shown. The limits of the measurement frequency range are clearly visible. At low frequencies (< 300 Hz) there is a clear influence of the near field at high (> 4 kHz) physical limitations of transducers.

In the following drawings in Fig. 7 the values as a function of measuring position are shown. In the left column we can see directional characteristics (normalized sound intensity level values, reference 0 dB value for position no. 50) obtained using the reference and modified methods and the difference in values between the methods for 3 one-third octave bands: 500 Hz, 1 kHz and 5 kHz (rows). In the right column we can see angle values between the resultant sound intensity vector and the radial component as a function of measuring position. The difference in angle values between the methods is also shown.

Polar drawings in Fig. 8. show the obtained resultant values of sound intensity vector for individual measurement points. The vectors are presented in the form of arrows that come out from points spread equally at 1.8 degrees on a semi-circle. This corresponds to the actual measurement points. The diagram shows both actual orientation (red) and radial components (black) of the sound intensity vectors. Their mutual deviation makes it easy to see the presence of a tangent component in the sound intensity vector. Thus, the graph presents an extended form of the classic directional characteristics obtained when measuring only scalar values of pressure.

The presented characteristics are constructed for 500 Hz, 1 kHz and 5 kHz (rows), both with the reference p–u probe (left column) and with the modified p–p method (right column).

7. Discussion and conclusions

The paper presents an idea of measuring the sound intensity vector without the need of using an expensive intensity probe. As the experiments showed, the presented method is not universal, but it can be successfully used in cases where we can control the excitation signal and the conditions of measurement are stable. In the presented case of testing an electroacoustic transducer in laboratory conditions, the statistical uncertainty of measurement resulting from the time invariance of the system is 0.2 dB, which is a very good result. Also, the frequency range of the method's usability is satisfactory. However, an unequivocal estimation of the low frequency limits requires better field conditions and reference. The upper limit corresponds to the theoretical value resulting from the spatial sampling of the acoustic field. The proposed method was used to study the loudspeaker's directional characteristics. Considering the fact that the proposed method achieves a high uncertainty of measurement of amplitude of vector components of the sound intensity, its sensitivity and resolution in measurement of the parameters of the whole intensity vector (amplitude and angles of spatial orientation) used to characterize the directional properties of the speakers should be of the same order as in the case of classical p–u and p–p methods. This is particularly true for the frequency range of speech, where the difference in intensity vector estimation between the proposed method and the reference p–u probe is less than 1 dB (level) and 5° (localization angle in the measurement plane). The vector representation of

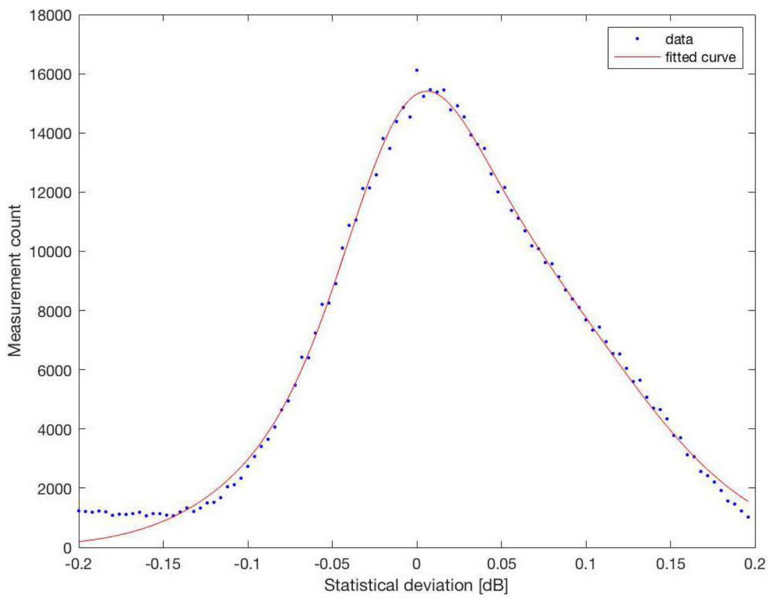

Fig. 5. A histogram showing the distribution of sound intensity values in consecutive measurements in the same measurement point.

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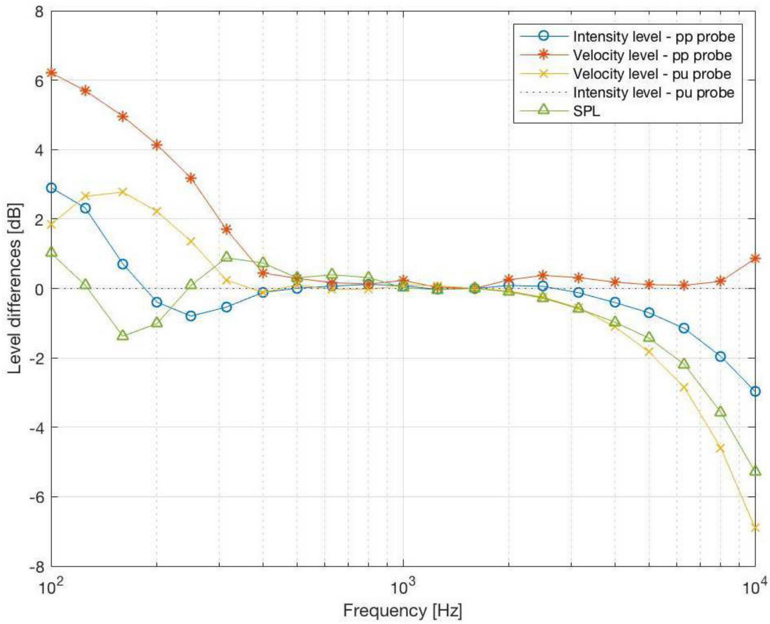

Fig. 6. Differences in sound intensity levels and acoustic velocity levels for a measurement point located in front of the loudspeaker (position no. 50).

the intensity distribution around the source enables a deeper analysis of its operation and more accurate prediction of the distribution of sound generated in the far field. In order to fully confirm this point, further research related to field imaging in more points around the source is required. Further work will be also devoted to the use of the intensity method to measure directional properties of sound sources that do not require the use of free-field conditions. In addition to an appropriate windowing in the time domain, information on the direction of acoustic energy flow will be used.

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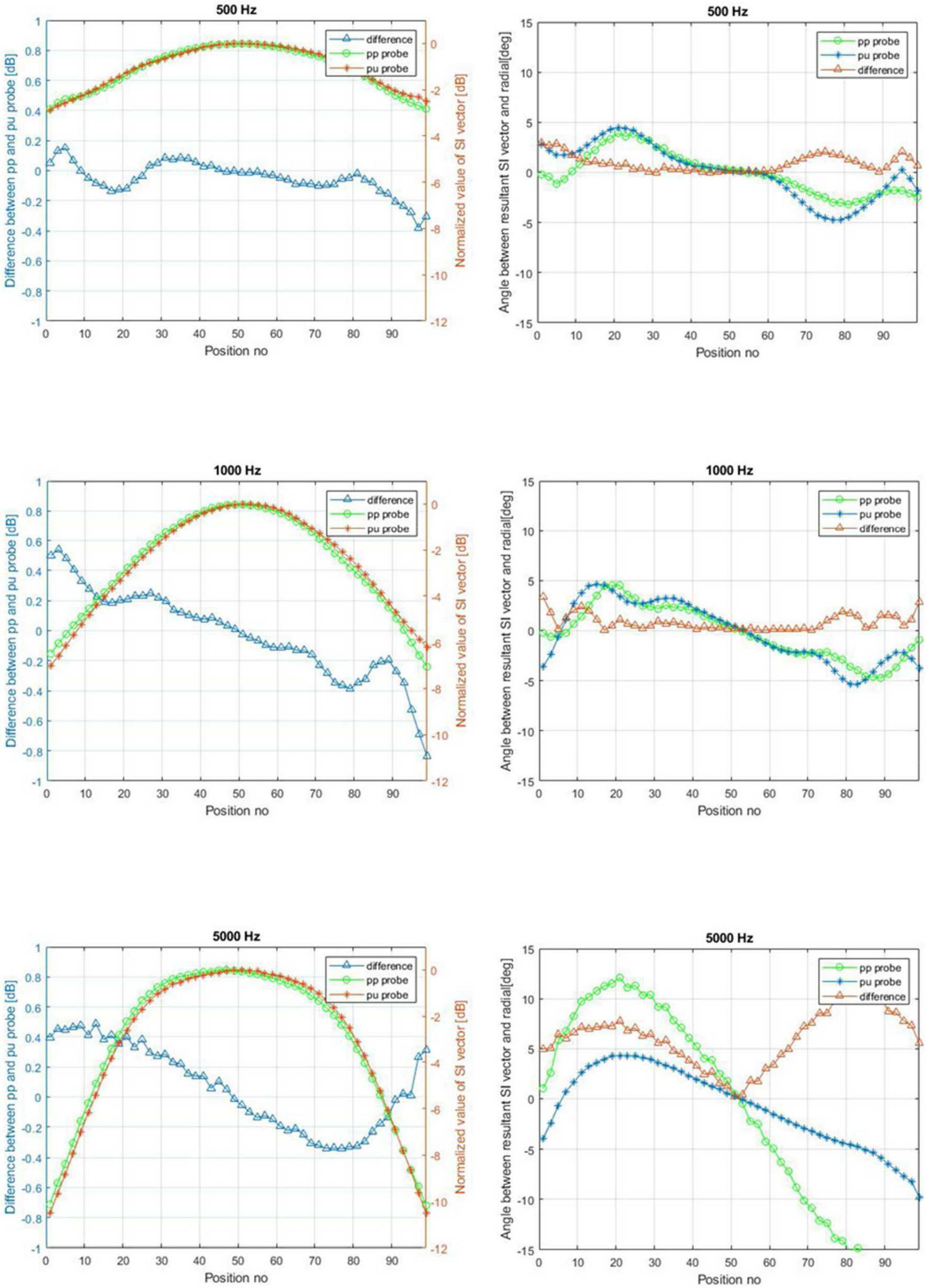

Fig. 7. Measurement results of levels (left column) and angles (right column) of sound intensity vector in 500 Hz, 1 kHz and 5 kHz one-third octave bands.

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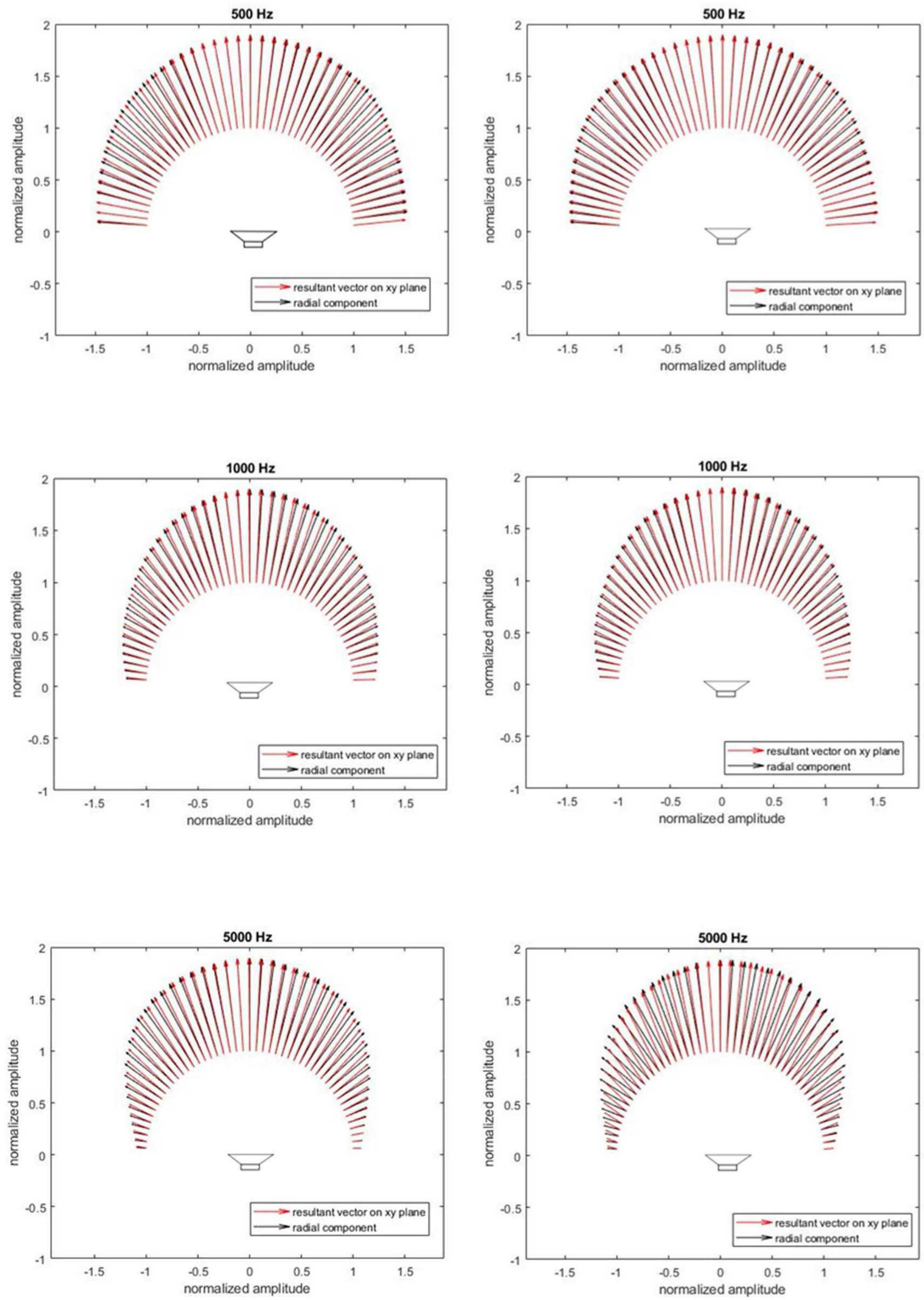

Fig. 8. Polar drawings showing the directional characteristics of the loudspeaker in the form of sound intensity vectors for individual measurement points.

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