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SOUND INTENSITY AND ITS MEASUREMENT

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ABSTRACT

The sound intensity method is one of the most significant developments of the past twenty years in the technology of noise control engineering. The method is now generally recognised as a 'mature technology', which is reflected in the fact that several international and national standards for sound power determination using sound intensity and for instruments for such measurements have been issued in the past few years.

The paper summarises the basic theory and gives an overview of the state of the art with particular emphasis on recent developments in the field. It describes and discusses the sources of error in measurement of sound intensity and the resulting limitations imposed on various applications of such measurements. Finally, some unresolved problems are mentioned, and the possibility of improving the instrumentation is discussed.

1. INTRODUCTION

Sound intensity measurements are now routinely used in the determination of the sound power of operating machinery *in situ*. Other applications of this measurement technique include, in descending order of successfulness, visualisation of sound fields, identification and rank ordering of partial sources, determination of transmission losses of panels and partitions, determination of the radiation efficiency of vibrating surfaces, and measurement of the sound absorption of materials. The purpose of this paper is to give an overview of the status of sound intensity measurements. The second edition of Fahy's book *Sound Intensity*[1] summarises most of the important work in this field up to 1994. The emphasis in this paper is therefore on developments from the past few years.

2. WHY SECOND-ORDER QUANTITIES?

'Energy acoustics', which is central in noise control engineering, makes use of the energy-based concepts of absorption and transmission coefficients, and relies on simple energy-balance considerations. However, from a physical point of view it is not at all selfevident that the acoustic second-order quantity sound power should be a useful measure of the noise emission from a source, inasmuch as the sound power output of a pure-tone monopole source depends on its environment, including the possible presence of other sources that emit sound of the same frequency.[2,3] Nor is it obvious that the ratio of transmitted and incident sound power for a given partition should be independent of the particulars of the rooms it separates - and indeed it is not, at low frequencies.[4] In fact, *none* of the (tacit) assumptions of energy acoustics are strictly true. However, the approximations of 'the energy approach' are often very good approximations in most of the audible frequency range - and most alternative methods would be vastly more complicated. For example, although the sound power of a source in principle depends on its surroundings, such effects are very small in practice except in very small rooms or at very low frequencies, simply because the physical size of real sources of noise prevents them from being placed very near a reflecting surface. By comparison, the sound pressure produced by a given source is far more dependent on its environment. (The usefulness of the energy approach is less obvious in structural acoustics. For example, the vibrational power output of a source of structure-borne noise may be a reasonable measure of its adverse effect on its environment, but this power depends significantly on the dynamic properties of the receiving structure.[5])

Figure 1 gives an idea of the influence of reflecting planes on the sound power output of small but realistic sources. The figure shows the radiated sound power levels of two different sources, one of high acoustic impedance (Brüel & Kjær's Sound Power Source of type 4205, a loudspeaker) and one of low acoustic impedance (a centrifugal fan, Airap A 14, manufactured by Électricité de France) under a variety of conditions. (These and other experimental results presented in this paper have been obtained with an intensity probe of type B&K 3548 with half-inch microphones of type B&K 4181 separated by a 12-mm spacer, in combination with an analyser of type B&K 3550 or 2133.) Apparently, the environment has an influence on the power output, but it is also clear that the effect of even a drastic change in environment is moderate, and quite negligible at medium and high frequencies. Note that reflections affect the power output of the two sources quite differently. (Similar results have been presented by Tashibana and Yano.[6])

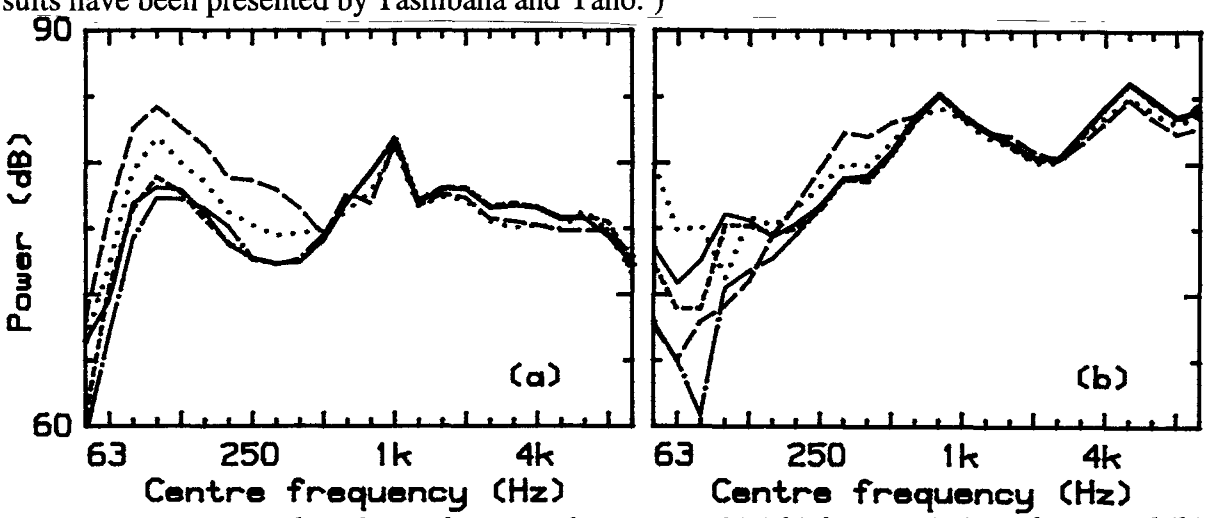

Figure 1. Radiated sound power of a source of (a) high acoustic impedance and (b) low acoustic impedance in one-third octave bands. ——, Source on the floor of an ordinary room, far from the walls; - - - , source on the floor of a reverberation room, far from the walls; · · · , source as near as possible to a wall in a reverberation room; — — , source as near as possible to a corner in a reverberation room; – · · – · , source on a 1.6-m² plate in an anechoic room.

The limitations of energy acoustics should be kept in mind. For example, predictions or measurement procedures that essentially are based on Sabine's 'large room' model of the sound field in a room are not reliable at frequencies where the room is no longer large compared with the wavelength. Quite apart from the influence of measurement errors one should be very careful in interpreting sound intensity contour maps recorded in near fields of vibrating structures, as pointed out by several authors;[1,7] and trying to locate sources in small cabins from sound intensity data cannot be recommended.[1]

3. CONSERVATION OF SOUND ENERGY

The sound intensity is a vector that expresses the flow of sound energy through a unit area. It is defined as the time-averaged product of the sound pressure and the particle velocity,

$$\mathbf{I} = \overline{p(t)\mathbf{u}(t)} \ , \tag{1}$$

or, expressed in the usual complex notation,

$$\mathbf{I} = \tfrac{1}{2} \ \mathrm{Re}\left\{p\mathbf{u}^*\right\} . \tag{2}$$

The usefulness of sound intensity measurements is closely related to the equation

$$\nabla \cdot \mathbf{I}(t) + \frac{\partial w(t)}{\partial t} = 0 \ , \tag{3}$$

in which $\mathbf{I}(t) = p(t)\mathbf{u}(t)$ is the *instantaneous* sound intensity and $w(t)$ is the instantaneous sound energy density.[8] Integrating this equation over an arbitrary volume V enclosed by the surface S and applying Gauss's theorem on the first term leads to the following energy conservation law:

$$\int_S \mathbf{I}(t) \cdot \mathrm{d}\mathbf{S} = -\frac{\mathrm{d}}{\mathrm{d}t}\left(\int_V w(t)\mathrm{d}V\right) \ . \tag{4}$$

Equation (4) expresses the fact that the total net outflow of sound energy through a given surface not enclosing a source equals the rate of decrease of the sound energy within the surface. If the surface encloses a source there is an additional term on the right-hand side: the instantaneous sound power emitted by the source.

In practical applications it is the time-averaged sound intensity and the time-averaged sound power of a source that are of interest, not the instantaneous quantities. Analysis of eq. (4) leads to the conclusion that the integral of the normal component of the time-averaged sound intensity over an arbitrary closed surface is zero unless the surface encloses a source or a sink; in that case it equals the time-averaged sound power of the source (or sink):

$$\int_S \mathbf{I} \cdot \mathrm{d}\mathbf{S} = P_\mathrm{a} \ . \tag{5}$$

A consequence of eq. (5) is the extremely useful result that one can determine the sound power radiated by a stationary source by integrating the normal component of the sound intensity over an arbitrary surface that encloses the source, irrespective of the presence of other stationary sources outside the surface.

The alternatives to sound intensity measurements are the traditional methods based on measurement of sound pressure. These methods require special facilities (anechoic, hemi-anechoic or reverberation rooms), rely on far field approximations or diffuse field considerations, and cannot be used in the presence of extraneous noise.

4. REACTIVE SOUND INTENSITY

It takes four quantities to describe the distributions and fluxes of sound energy in a sound field completely: potential energy density, kinetic energy density, active intensity (usually simply referred to as the intensity) and reactive intensity.[9,10] The last mentioned of these quantities represents the non-propagating, oscillatory sound energy flux that is characteristic of a sound field where the pressure and the particle velocity are in quadrature, as for instance the near field of a small source. The reactive intensity is a vector defined as the *imaginary* part of the product of the complex pressure and the complex conjugate of the particle velocity,

$$\mathbf{J} = \tfrac{1}{2}\,\mathrm{Im}\{p\mathbf{u}^*\}\,. \tag{6}$$

More general time-domain formulations based on the Hilbert transform are also available.[11] Unlike the usual active intensity, the reactive intensity remains a controversial issue (although it was introduced as early as in 1951[12]), perhaps because the quantity has no obvious physical meaning,[13] perhaps because describing an oscillatory flux by a time-averaged vector seems peculiar to some.[14] It has been attempted to locate sources with reactive intensity,[15,16] but really convincing results have never been presented, and, as pointed out by Fahy,[1] the reactive intensity 'tends to be strongest in the vicinity of weakly radiating vibrating surfaces.' However, although the quantity is of no obvious practical use, it nevertheless *is* quite convenient that we have a quantity that makes it possible to describe and quantify the particular sound field conditions found in the near field of a small source in a precise manner. The usefulness of the concept can be illustrated by the fact that the effect of various technical deficiencies on measurement of the (active) intensity is an error that is proportional to the corresponding component of the reactive intensity (see sections 6.3 and 6.6).

For more than forty years researchers have been concerned with studying the vectorial properties of the active and reactive intensity.[12,15,17,18] Alternative definitions of the various instantaneous and time-averaged 'intensities' have also been suggested.[19,20] The most significant difference between the definition of the reactive intensity mentioned above[9-11] and the one proposed by Stanzial *et al.*[19,20] is that vectorial components that correspond to different frequencies are simply added with the Hilbert transform formulation; this would seem to be an advantage. However, the fact remains that the only 'intensity' of established practical use is the conventional time-averaged intensity given by eq. (1).

5. MEASUREMENT PRINCIPLE

By far the most successful measurement principle is the 'two-microphone' or 'p-p' principle, which employs two closely spaced pressure microphones and relies on a finite difference approximation to the pressure gradient.[1] Sound intensity probes based on the alternative p-u principle, which employs the combination of a pressure transducer and a particle velocity transducer, are no longer in commercial production, but attempts to develop p-u probes have occasionally been reported in the literature, also recently.[21] A 'multi-microphone' probe employing no less than twenty electret microphones has also been described lately.[22] However, no convincing experimental results obtained with alternative measurement principles have been presented so far.

A three-dimensional sound intensity probe employing only four microphones has recently been described.[23] This probe is based on a variant of the p-p principle: the four microphone signals are weighted differently in order to obtain a simple relation between the orientation of the coordinate system and the geometry of the probe.

6. ERRORS AND LIMITATIONS IN MEASUREMENT OF SOUND INTENSITY

A surprisingly large fraction of the sound intensity literature has been concerned with identifying and studying the sources of error in measurement of sound intensity, and the study of errors seems to continue to attract the attention of researchers in this field. This preoccupation with errors and limitations is not the result of a particularly gloomy disposition among the members of the 'intensity community'; it results from the disturbing observation that the accuracy of sound intensity measurements depends strongly on the sound field under study, in combination with the fact that small local errors under certain conditions can be amplified into large global errors when the sound intensity is integrated over a surface.

Whoever makes sound intensity measurements should have a certain knowledge of the limitations of the technology. On the other hand, the mere existence of a variety of sources of error should not discourage anyone from using sound intensity measurements. Their influence on various applications is discussed in section 7 of this paper.

In the following exposition of the sources of error in measurement of sound intensity the emphasis is on recent developments, but it is attempted to put the contributions in perspective.

6.1 Phase mismatch

Phase mismatch between the two measurement channels has been recognised as the most serious source of error in measurement of sound intensity with p-p probes since the late 1970s. The influence of phase mismatch is negligible if $\varphi_e << \Delta\varphi$, where φ_e is the phase error of the measurement system and $\Delta\varphi$ is the actual phase angle between the pressures at the microphone positions, and since the latter can be very small indeed, this condition imposes strong restrictions on the range of measurement. The phase angle $\Delta\varphi$ depends on the ratio of the intensity component and the mean square pressure,[24]

$$\Delta\varphi \simeq -\left(I_r \rho c / \overline{p^2}\right) kd \, , \tag{7}$$

where $k = 2\pi f/c$ is the wavenumber, d is the microphone separation distance, and ρc is the characteristic impedance of air. The phase error φ_e is caused by differences between the lower limiting frequencies of the microphones (see section 6.3), between the resonance frequencies of the microphones, between the preamplifiers, and between analogue filters in the two channels. In practice one must, with state-of-the-art equipment, allow for phase errors from 0.05° at frequencies below 250 Hz to 1° at 5 kHz.[25] Both the IEC standard on instruments for the measurement of sound intensity and the corresponding North American ANSI standard specify requirements that ensure that the phase error is within certain limits.[26,27] The 'performance verification' procedure prescribed in the IEC standard[26] has been analysed by several authors.[28-30] The phase error is often expressed in terms of the so-called residual pressure-intensity index, which should be as large as possible. With high-quality instrumentation and a separation distance of 12 mm this quantity is at least 18 dB above 250 Hz. One can increase the residual pressure-intensity index by using a larger microphone separation distance, but this conflicts with the high-frequency optimisation described in section 6.2.

It is easy to show from eq. (7) that the resulting bias error in measurement of sound intensity is proportional to the product of the phase error and the mean square pressure and inversely proportional to the microphone separation:

$$\hat{I}_r \simeq I_r - \frac{\varphi_e}{kd} \frac{\overline{p^2}}{\rho c} \, . \tag{8}$$

This expression shows that the normalised error depends on one single property of the sound field: the ratio $(p^2/\rho c)/I_r$. In sound power measurements the error depends on the ratio between the corresponding surface-integrated quantities:

$$\hat{P}_a \simeq P_a\left(1 - \frac{\varphi_e}{kd}\left(\int_S (\overline{p^2}/\rho c)\,dS \Big/ \int_S \mathbf{I}\cdot d\mathbf{S}\right)\right) . \quad (9)$$

Accordingly, it can be recommended to keep an eye on the pressure-intensity index of the measurement (the ratio in logarithmic form); errors of more than 1 dB occur if it exceeds the residual pressure-intensity index minus 7 dB.[1] The measurement procedure prescribed in the recently issued ISO standard for sound power determination guarantees that the error is within certain limits.[31]

If the phase error of the instrumentation is known from a measurement of the residual pressure-intensity index one can simply subtract the error term of eq. (8) or eq. (9); this is the so-called pressure correction technique (since the mean square pressure should be measured concurrently with the intensity). This method of compensating for phase mismatch has been examined by several authors.[24,30,32-35] The alternative probe reversal technique (which involves measuring twice) is favoured by Guy and Luchian.[35] Unfortunately, most commercially available sound intensity probes are not suited for the reversal technique.

It should finally be mentioned that there are still dissenting opinions about the validity of the global pressure-intensity index as an indicator of the influence of phase mismatch.[36] A discussion of the many different indicators suggested in the literature for this purpose has been presented in reference 37.

6.2 The high-frequency performance of sound intensity probes

The most fundamental limitation of the two-microphone measurement principle is due to the approximation of the pressure gradient by a difference of pressures at two closely spaced points: this finite difference approximation obviously imposes an upper frequency limit. In principle the finite difference error depends on the sound field in a complicated manner,[1] but practice has shown that the error is acceptably small if $kd < 1$,[1,38] and the corresponding upper frequency limit has therefore been widely accepted as inherent in the measurement principle - although attempts to compensate for the finite difference error have been described in the literature.[39] One cannot expand the frequency range by using a very small separation between the microphones since the influence of several other measurement errors is inversely proportional to the frequency.[40] However, a recent numerical and experimental study of sound intensity probes with the microphones in the usual face-to-face arrangement with a solid 'spacer' between them has shown that the upper frequency limit can be extended by more than an octave ($kd < 2.2$) if the length of the spacer between the microphones is about one microphone diameter.[41] The reason is that the resonance of the cavities in front of the microphones (which essentially is determined by the cross-sectional dimension) to a large extent compensates for the finite difference error with this particular probe configuration, irrespective of the nature of the sound field. A shorter spacer cannot be recommended, and with a longer spacer the pressure increase occurs at too high frequencies to compensate for the finite difference error. Figure 2 shows the resulting error of a sound intensity probe with two half-inch microphones in the face-to-face configuration calculated for a plane wave of axial incidence. As can be seen, the most favourable length of the spacer is about 12 mm; an intensity probe with this geometry performs well up to about 10 kHz. It should be mentioned that the IEC standard for sound intensity instruments[26] actually *requires* that the sound intensity response in a plane progressive wave agree with the theoretical expression for the finite difference approximation (which is based on the assumption that the intensity probe measures the pressure as it would be in the absence of the probe) within a certain tolerance. The standard also states that 'a probe only meets the requirements of this standard in the frequency range where the nominal response [the above-mentioned theoretical expression] relative to 250 Hz is 0 ± 1 dB.' Accordingly, a probe with the optimised configuration mentioned above meets the requirements of the standard only up to 5 kHz!

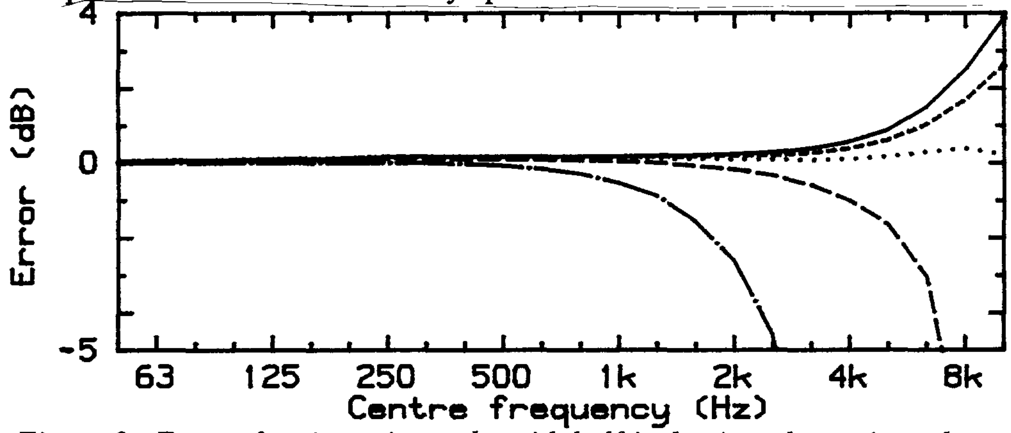

Figure 2. Error of an intensity probe with half-inch microphones in a plane wave of axial incidence for different spacer lengths: ——, 5 mm; - - -, 8.5 mm; · · · ·, 12 mm; — —, 20 mm; — · — ·, 50 mm.

There are no reasons to expect a similar fortunate cancellation of two errors with other probe geometries. In the general case diffraction errors should be reduced by minimising the bulk of the probe, and one has to submit to the upper frequency limit imposed by the finite difference principle.

It is worth mentioning that Brüel & Kjær's intensity microphones of type 4181 (and the microphones they replaced, of type 4177/4165), which are rather dominating in this field, are overdamped so-called free field microphones, designed to have a flat frequency response in a plane wave of axial incidence. It is therefore necessary to compensate for the high-frequency drop in pressure sensitivity. Figure 3 shows the correction, determined by measuring the electrostatic actuator response of microphones of type 4181.[41]

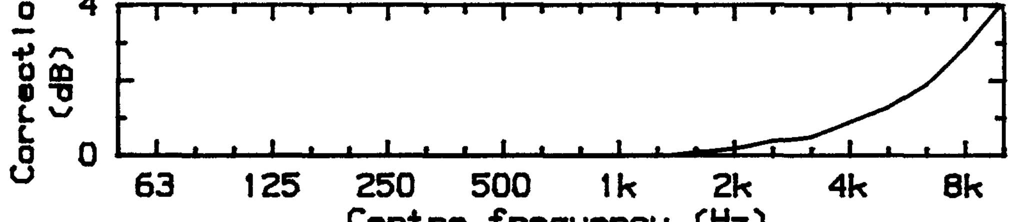

Figure 3. High-frequency correction for the sensitivity drop of sound intensity microphones of type B&K 4181.

6.3 The low-frequency performance of sound intensity probes

Several factors contribute to limiting the performance of sound intensity probes at low frequencies. The low-frequency performance is determined by phase mismatch (see section 6.1), random errors due to the inherent noise of the microphones in combination with a finite averaging time (see section 6.4), and the possible presence of turbulent airflow (see section 6.6). Yet another problem is caused by the microphone vents.

The lower limiting frequency of a condenser microphone is determined by the pressure equalisation system, which is usually a capillary tube from the internal cavity to the outside.[42] The equalisation vents of conventional condenser microphones have no influence on the amplitude response of the microphones above, say, 20 Hz. However, if two such microphones are used for measuring sound intensity with a p-p probe the vent systems give rise to a bias error that is proportional to the reactive intensity.[43] The investigation described in reference 43 has recently been supplemented by Li and Pascal, who showed that the microphone vents also lead to another error term that is proportional to the mean square pressure.[44] The first term is determined by the mean value of the lower limiting frequencies of the microphones, and the second one is determined by the difference in lower limiting frequency. The latter error, which in effect is due to p-p phase mismatch caused by the difference between the microphones, can be corrected (see section 6.1). However, the first term can cause bias errors of up to 10 dB in near field measurements at low frequencies.[43] If such measurements were attempted with an intensity probe fitted with electret microphones with diaphragms of polymeric foil the error would be larger still; this is one of the reasons why cheap, phase-corrected electret microphones are not suitable for sound intensity measurements. Various correction procedures have been suggested.[30,43,44] One manufacturer's (patented) solution to the problem is microphones with additional acoustic low-pass filters in their equalisation system.[45] The problem is altogether eliminated with such microphones.[43,44]

6.4 Random errors associated with a finite averaging time

Random errors due to incomplete time averaging reveal themselves by poor repeatability in certain frequency bands. The random error in measurement of sound intensity does not depend on the measurement principle, and it can be much larger than the theoretical minimum value of $1/\sqrt{BT}$ (known from mean square estimation), where B is the bandwidth and T is the averaging time.[9,46,47] On the other hand, since the global random error in intensity-based sound power determination depends on the *total* averaging time (irrespective of the spatial averaging procedure), the BT-product in *sound power determination* will usually be so large that the resulting error is acceptably small.[40,48] However, random errors may well be of concern in other applications of intensity measurements. Using a very long averaging time at each point is obviously inconvenient if the sound intensity is to be mapped with a hand-held probe in front of a large, complicated source of noise - and even if the measurements are carried out with an automated measurement system, limiting the averaging time can be important, say, because the sound source under study is not completely stable over a long period of time.

Loyau and Pascal have recently published an analysis of the random errors in measurement of the size and direction of the sound intensity vector.[49] Both the active and reactive intensity were examined. For the case of simultaneous measurements of the components of the active intensity vector they found that the random errors of the estimated magnitude and angle of orientation do not depend on the orientation of the intensity probe, whereas they do depend on the orientation if the measurements are carried out in succession. In addition, Loyau and Pascal's theoretical results confirm Dyrlund's observation[50] (from computer simulations based on Seybert's expression[46]) that, given a certain structure of the sound field, the random error depends essentially on the pressure-intensity index, as also shown earlier by Pascal.[51] Unfortunately, the relation between the error and the pressure-intensity index depends strongly on the nature of the sound field. For example, if the direct sound wave from a source is disturbed by diffuse background noise, the normalised random error is (to first order) increased by a factor of $((\overline{p^2}/\rho c)/I_r)/\sqrt{6}$,

$$\varepsilon\{\hat{I}_r\} \simeq \frac{1}{\sqrt{6BT}}\left(\frac{\overline{p^2}/\rho c}{I_r}\right), \tag{10}$$

but if a measurement under free field conditions is disturbed by an independent wave from a direction that is perpendicular to the probe, then the influence of the pressure-intensity index on the random error is much weaker:

$$\varepsilon\{\hat{I}_r\} \simeq \frac{1}{\sqrt{2BT}} \left(\frac{\overline{p^2}/\rho c}{I_r} \right)^{½} . \tag{11}$$

In other words, there is *no* general relation between the random error and the pressure-intensity index, as also concluded in references 1 and 47.

It is interesting that Loyau and Pascal's result for diffuse background noise, in which the coherence of the two pressure signals is

$$\gamma_{pp}^2(\omega) = \left(\frac{\sin(\omega d/c)}{\omega d/c} \right)^2 , \tag{12}$$

is identical with a prediction based on the assumption that the background noise is reverberant, 'diffuse' noise produced by the source itself,[52] because in the latter case the coherence is unity.[47,53] This seems to indicate that, contrary to all expectations,[47,54] the biased 'frequency-band coherence'[10] can be used in predicting random errors. If this holds in the general case, it may be possible to develop a method for on-line prediction of the random error and thus the necessary averaging time in point measurements with ordinary filter bank analysers. A preliminary investigation has not been encouraging, though.

Figure 4 shows an example of the random error of sound intensity measurements two metres from a source in a reverberation room, estimated by repeating measurements with an averaging time of 4 s one hundred times. For comparison a prediction based on eq. (10) and the measured pressure-intensity index is also shown. There is fair agreement.

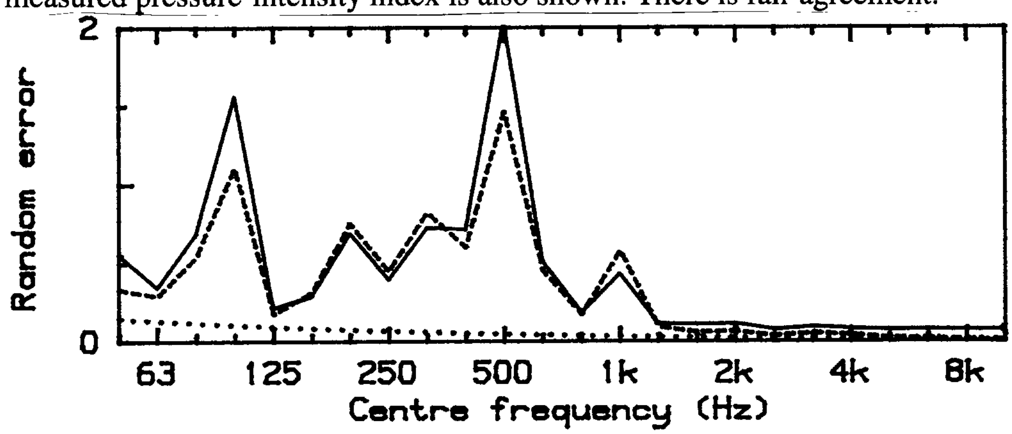

Figure 4. Random error of sound intensity measurements with an averaging time of 4 s in a reverberation room two metres from the source. ——, Measured random error; - - - , predicted random error; · · · , theoretical minimum value, $1/\sqrt{BT}$.

The inherent noise of the microphones causes an *additional* random error that can be considerable at low frequencies unless the signal-to-noise ratio of the microphone signals is *very* large, say, more than 40 dB.[40,49,55] Poor repeatability in the lowest frequency bands is a certain indication of this error. In sound intensity measurements at relatively low levels this is a more serious problem than the general, system-independent random error mentioned above.

6.5 Spatial sampling

Most applications of sound intensity measurements involve determining the surface integral of the normal component of the intensity. In practice the integral is approximated either by measuring at a number of discrete points or by 'scanning', that is, by moving the intensity probe continuously over the measurement surface along a suitable path, manually or with a robot. Today the superiority of the scanning procedure is generally recognised. Evidently, any sampling procedure is approximate, and it is obvious that the accuracy of the approximation depends on the particulars of the sound field. One can get an indication of possible problems by measuring several times using different scanning paths, but there is *no* way to estimate the random error associated with approximating a surface integral by a curve integral (or with approximating a surface integral by a sum of discrete sample values) from the measurement itself;[56] recommendations must be based on empirical observations. Such empirical observations are found eg in references 57-60. They all confirm the usefulness and reliability of the scanning procedure. Shirahatti and Crocker conclude that scanning with a hand-held probe is at least as accurate as using a large number of discrete points,[57] Keith *et al.* observe that the scanning procedure prescribed in ISO 9614-2[31] is much faster and more convenient than the procedure using discrete points described in ISO 9614-1[61] and gives comparable accuracy,[59] and Pettersen *et al.* find that different scanning patterns give almost identical results.[60]

It should be mentioned that a 'scanning field non-uniformity indicator' has been proposed.[36] This quantity is defined as the normalised standard deviation of a large number of sound intensity values corresponding to segments of the scanning measurement. Each value is determined with a short averaging time. The underlying idea, that the random error associated with the scanning should be proportional to the normalised spatial standard deviation of the intensity over the measurement surface, is adopted from statistical considerations originally developed for measurement at discrete points.[61-63] However, the basic assumption of statistically independent samples has been shown to be in error.[56]

The scanning method is not suitable for determining the sound power of sources that operate in cycles.

6.6 Airflow and windscreens

Even a moderate airflow disturbs sound intensity measurements seriously at low frequencies, because the microphones cannot distinguish between sound and turbulence. The result is an additive 'false' intensity that, according to the empirical observations reported in reference 64, is reproducible but unpredictable. For given flow conditions the false intensity depends on the microphone spacer.

Windscreens of porous foam are often used to suppress flow noise (and occasionally they are used even in the absence of flow, simply because they offer some mechanical protection of the probe). However, the losses of the foam give rise to a phase error between the pressure and the particle velocity that leads to very large bias errors in strongly reactive sound fields at low frequencies.[65] One should therefore avoid measuring in near fields with windscreened sound intensity probes; the problem is negligible if the probe is at least, say, 15 cm from the source. Apparently, the high-frequency performance of windscreened intensity probes has never been investigated, and windscreens might give rise to bias errors from other mechanisms at higher frequencies. Figure 5 demonstrates that there are indeed systematic deviations between measurements with and without a windscreen at medium and high frequencies. The figure shows the results of measuring the sound power of a source of high acoustic impedance without and with two different windscreens under fairly difficult measurement conditions. Similar results have been observed under a variety of other conditions. These results are in good agreement with the well-known influence of windscreens on sound pressure measurements.[66] (The low-frequency problem with windscreened probes mentioned above is negligible here because the near field of the source has been avoided.)

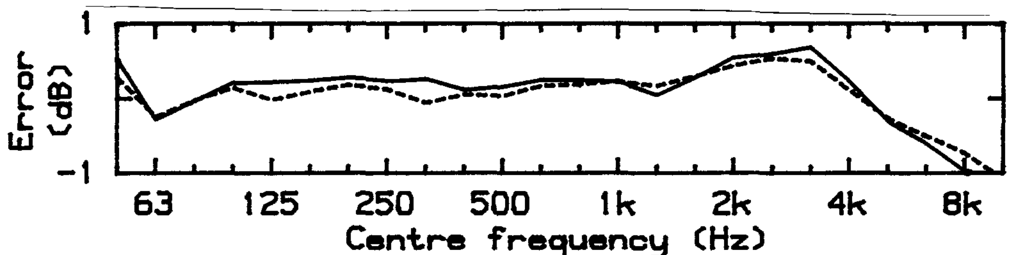

Figure 5. The influence of windscreens on sound power measurements using sound intensity. ——, Spherical windscreen; - - - , elliptical windscreen.

7. APPLICATIONS

Sound intensity measurements have found many applications, and the following discussion is by no means complete. Readers are referred to chapter 10 of Fahy's book *Sound Intensity.*[1]

7.1 Sound power determination

Sound power determination is undoubtedly the most established application of sound intensity measurements. The many sources of error notwithstanding, it is concluded in reference 40 that the intensity method of determining sound power is convenient, fast and reliable provided that a few simple 'rules' are observed. However, the complicated relation between the sound field conditions and the resulting measurement accuracy is reflected in the measurement procedures prescribed in the various international and national measurement standards; they are generally more complicated than the procedures prescribed in the sound power standards based on measurement of mean square pressure (eg the ISO 3740 series), which, by contrast, specify the environment of the measurement.

The two international standards for sound power determination using sound intensity, ISO 9614-1 and 9614-2, rely on measured field indicators that must satisfy certain conditions.[31,61] Ameliorative actions are prescribed if the indicators fail to satisfy the specified conditions. Fahy has recently described the rationale of these standards.[67] The idea is that the total estimation error should be within certain limits when the requirements are met, irrespective of the environment. Several acousticians have expressed the opinion that the requirements specified in the standards are unnecessarily stringent.[38] This criticism is parried off by Fahy,[67] who explains that the fact that the operating environments of sources may vary widely has 'necessitated the incorporation of safeguards against excessive measurement error under extremely unfavourable conditions;' and therefore 'these safeguards may appear to be over-conservative to those making measurements under favourable conditions.'

Paine and Simmons have recently described the results of an investigation in which four laboratories examined the repeatability and reproducibility of measurements determined as specified in various sound power standards.[68] The reproducibility standard deviation for ISO 9614-1 was found to around 1 dB, well within the estimate stated in the standard.

The ANSI standard for intensity-based sound power determination[69] is more pragmatic and less ambitious than the ISO standards. Nevertheless, twenty-six optional field indicators are described; it is left to the user to interpret the data and decide what to do. A critical examination of most of these quantities (and many other field indicators) has recently been published.[37] The conclusion of this overview of the literature on field indicators (and the conclusion in reference 40) is that the *only* essential indicator in sound power determination with p-p intensity probes is the global pressure-intensity index, and this quantity should be compared with the residual pressure-intensity index of the equipment as described in section 6.1 whenever the measurement conditions are unfavourable.

7.2 Visualisation of sound fields: source identification

With the computer graphics that is now available it has become common practice to determine contour maps from point measurements of sound intensity by means of interpolation. The most important application of such graphical representations is to source identification. A typical noise reduction project starts with identification and ranking of the various partial sources, and sound intensity measurements would seem to be an obvious tool – provided that the sources are independent. However, according to Ffowcs Williams,[7] 'energy measures in sound and vibration ... have no predictive significance,' but are 'easy to illustrate in pictorial form and can certainly catch a sponsor's imagination.' Therefore they have 'produced a boom in instrument sales,' even though they generally 'have little use.' These harsh words are not entirely wrong if we interpret 'energy measures in pictorial form' as contour maps of *reactive* intensity data, but the successfulness of source identification from active intensity data seems to be indisputable. Recent examples have been described eg in references 70-72.

Usually, the radiated sound power of the component sources is determined with the scanning method. Mapping requires point measurements and is therefore much more time-consuming, and since near fields of vibrating structures can be very complicated, there is no guarantee that the sound field has been sampled adequately. A more refined mapping procedure that tests the reliability of the data has recently been presented by Klein and Guigné.[73]

Visualisation of sound fields may also serve the purpose of helping us interpreting or understanding complicated phenomena. Advanced time-frequency analysis methods of visualising instantaneous sound intensity data are presented in reference 74.

7.3 Measurement of the transmission loss of partitions

In standardised measurements of transmission loss the incident sound power is determined from an estimate of the spatial average of the mean square pressure in the source room on the assumption that the sound field is diffuse, and the transmitted sound power is determined from a similar measurement in the receiving room where, in addition, the reverberation time must be measured. Alternatively, one can measure the transmitted sound power directly using the sound intensity method. This will usually be more time-consuming, but it has some important advantages: i) the sound field in the receiving room does not have to be diffuse - in fact, there does not even have to be a receiving room; and ii) measuring the transmission loss of individual parts of the partition is possible.

Many authors have found systematic deviations between the results of the traditional method and the method based on sound intensity measurements.[75] Possible reasons for such deviations have recently been analysed.[76] It was found that the various sources of bias error described in section 6.1, 6.2 and 6.3 of this paper together with the known systematic errors in conventional sound power determination in reverberation rooms can easily explain the discrepancies reported in the literature, and it was concluded that the intensity method will give accurate results if the usual precautions from sound power determination are observed.

The intensity method of determining sound insulation is not yet standardised, but an international standard that gives the method as an alternative is under development.[77] Hongisto *et al.*[78] have recently described a test procedure for intensity-based sound insulation measurements developed from ISO 9614-1.

The inherent electrical noise of the microphones is likely to be a problem in measurement of flanking transmission or sound insulation of façades, where the signal-to-noise ratio can be relatively low and the pressure-intensity index can be fairly high. It is likely to have been a problem in the study recently reported in reference 79.

7.4 Measurement of the sound absorption of materials

By definition the absorption coefficient of an acoustic material is the ratio of absorbed and incident sound power. *In situ* measurements of the sound absorption of acoustic materials would be very useful, and such measurements can in principle be carried out using sound intensity. The *incident* sound power must be deduced from a measurement of the stationary mean square pressure on the assumption that the nature of the sound field is known, though; in practice this restricts use of the intensity method to diffuse field conditions.

However, measurement of sound absorption is probably the *least* successful application of sound intensity. The reason is that, unless the material has an absorption coefficient of more than, say, 0.5, the pressure-intensity index of the sound intensity measurement will be very high, more than most intensity analysers would be able to cope with, and the result will be unacceptable errors due to instrumentation phase mismatch.[1]

In spite of this problem it should be possible to measure absorption coefficients in the range from, say, 0.1 to unity with acceptable accuracy if compensation for phase mismatch is applied. The related 'transfer-function method' of determining plane wave incidence sound absorption coefficients from 'two-microphone' measurements in impedance tubes incorporates a correction procedure, and accordingly this method is fairly accurate; an international standard for this method is under development.[80] However, the successfulness of the intensity method for measuring absorption coefficients has yet to be demonstrated in practice.

8. FUTURE RESEARCH DIRECTIONS

There are still some unresolved problems in sound intensity measurements. For example, more research on the influence of turbulent airflow is obviously needed. It may be possible to develop a simple test that warns the operator against the influence of turbulence. Such an indicator has been suggested,[37] but its usefulness remains to be demonstrated in practice. Another problem that has never really been investigated is the influence of non-stationary background noise; it would be useful if guidelines could be derived.

There is certainly still a need for improvements of the instrumentation for sound intensity measurements. The limitations on the range of measurement imposed by phase mismatch are the most serious problem; under unfavourable conditions it forces the operator to evaluate a quality indicator. One solution to this problem could be microphones with im proved production tolerances of the phase characteristics, another solution could be instruments that incorporate compensation for phase errors as described in section 6.1, and yet another possibility could be intensity probes that are suited for the probe reversal technique. Less noisy microphones would also extend the range of measurement.

A sound intensity probe based on the p-u measurement principle would have the important advantage that the measurements would be less affected by extraneous noise. (Noise from sources outside the measurement surface increases the pressure-intensity index of the measurement and thus the influence of a given p-p phase error; it does not exacerbate the measurement error due to a given p-u phase error.[56]) However, particle velocity transducers of sufficient stability are not available at the moment.

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