B

"The sound intensity application"

The sound intensity measurement is the first application that comes in mind if one examines the unique particle velocity sensitive capabilities of the Microflown. This is mainly due to the effort that is taken to determine sound intensity in the traditional way.

Particle velocity is traditionally determined by measuring sound pressure at two closely spaced places. The so-obtained pressure gradient is used to calculate particle velocity (see chapter 2). The sound pressure is calculated by averaging the two pressure signals.

Since nowadays apart from the sound pressure, particle velocity can be determined instantly by the Microflown, acoustic quantities as sound intensity, acoustic impedance and sound energy density can be measured instantaneous and on one position in space. Although one can measure apart from the intensity also the impedance and sound energy, this part is focussed on sound intensity.

Note: Above the 135dB sound is not linear so p-p and **u-u** method are not valid anymore! So at that levels particle velocity cannot be determined from two closely spaced microphones.

Sound intensity is the average rate at which sound energy is transmitted through a unit area perpendicular to the specified direction at the point considered. It is a vector quantity (it has a magnitude and direction); defined as the time averaged product of the sound pressure (scalar) and the corresponding particle velocity (vector) at the same position.

$$\boldsymbol{I}[W/m^2] = \lim_{T\to\infty} \frac{1}{T}\int_0^T \boldsymbol{u}(t)\cdot p(t)dt$$

The measurement enables to determine the amount of radiated sound power of a source without the need for a special acoustic environment (such as a reverberant room or an anechoic room).

It is quite a work to determine the radiated sound power of a source since a number of separate measurements on an imaginary surface around the source must be taken. On the other hand the use of reverberant or anechoic rooms is expensive and not always possible. If for instance the source is too large or if it is not possible to transfer the source the sound intensity measurement is the only option.

The phase between sound pressure and particle velocity contains important information for the sound intensity is determination, see Fig. 46.

A quite loud sound has for example 1Pa pressure variation, if the sound wave is relatively far from the source and when no reflections are present the accompanying particle velocity then is 2.3mm/s. In such case the phase difference between sound pressure and particle velocity is zero degrees. The sound intensity is the mean value of the lower left plot of Fig. 46. As can be seen, it values 2.3mWm^{-2}. If the phase shift alters in to 60 degrees, the sound intensity will be half compared to the zero-degrees situation. If the phase shift is 90 degrees, the sound intensity will be zero and if sound pressure and particle velocity are out of phase (phase shift equals 180 degrees) the sound intensity values -2.3mWm^{-2}. So, even when the amplitude of sound pressure and particle velocity is not changing, the can sound intensity vary strongly.

A comparison between the traditional and novel intensity method is presented paragraph A1. Both types of probes have been tested in critical situations.

To get a better understanding of the p**u** method all steps to determine the emitted sound intensity of a loudspeaker has been presented in paragraph B2.

A one-dimensional sound intensity probe can be extended to a three-dimensional probe by using one microphone and three Microflowns. All the probes are still in one plane, see Fig. 72. The advantage of this is that the 3D sound probe can be positioned near to a sound source (the novel method can also be used in the near field). A broad band three-dimensional sound probe will offer many advantages. The magnitude and direction of the acoustic energy flow can be measured directly and this will improve the effectiveness of the measurement (the traditional probe can only measure in certain bandwidths so several measurements are needed to cover the complete audio band). A measurement will provide vector-based information. If a one-dimensional probe is used only the projection of an energy flow is measured. The 3D measurement allows measuring non-stationary sound fields as for example the noise of moving objects, like aeroplanes. The three-dimensional sound probe is capable of measuring the sound intensity vector in the full band sound but also the total sound energy E and three-dimensional acoustic impedance can be measured with this probe. Only the 3D-sound intensity is investigated in chapter A2.

Due to the sensitivity decay at higher frequencies, the signal to noise ratio at these frequencies is one of the subjects that is questioned often when debating the Microflown's capabilities. Regarding sound intensity this subject is of lesser importance. Selfnoise problems are strongly reduced when measuring sound intensity because all non-correlated noise sources (electrical noise etc) are eliminated by the method itself. This is explained in paragraph A3.

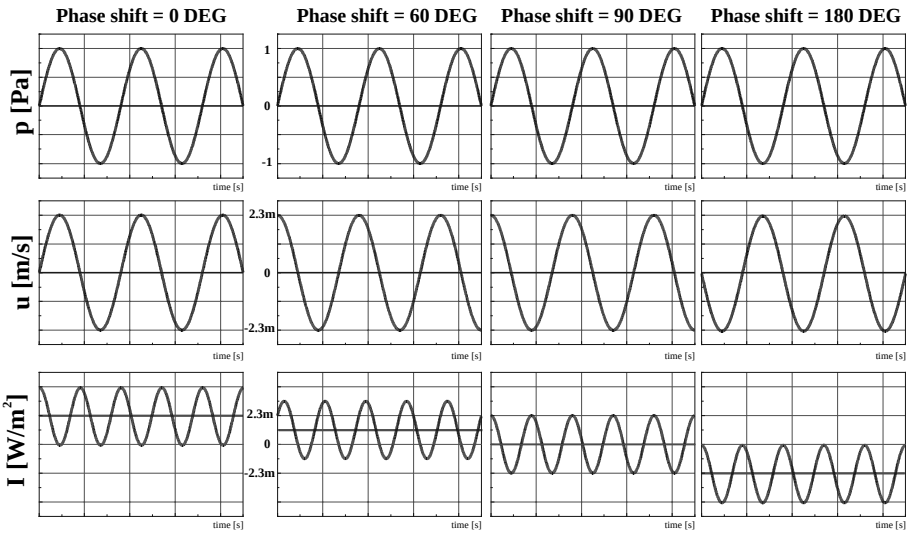

Fig. 46: Upper plots: sound pressure and particle velocity. Lower plots: instantaneous sound intensity and sound intensity (straight lines) as function of the phase shift between particle velocity and sound pressure.

The unwanted sensitivity for wind could cause problems when measuring outdoors. This sensitivity is measured with and without a standard windshield. It shows that regular windshield can be used to eliminate wind induced problems. This subject is discussed in paragraph A4.

A method that is complementary to the traditional sound intensity method is discussed in paragraph A5. The traditional method uses two microphones whereas this method uses two Microflowns. The result is a sound intensity probe that has a higher directivity.

Fig. 47: Photo of a p-u sound intensity probe (2000).

B1 THE P-U PROBE; COMPARISON TO THE BRUEL & KJAER P-P PROBE

The performance of a p-**u** sound intensity probe, a probe consisting a ½" sound pressure probe and a Microflown, has been compared to the commonly used Bruel & Kjaer p-p probe. The B&K p-p probe that has been used for this comparison consists of two ½" matched pressure probes and belongs to the best probes available on this moment. Both types of probes have been tested in critical situations. This chapter is based on the work that is presented in [3].

Introduction, why sound intensity?

Sound intensity measurements are quite useful as an acoustic measuring technique, since in a reverberant environment the free-field as well as the diffuse-sound field are determined. The measurement of the sound pressure only gives the sum of the free-field *and* the diffuse- sound field. The free-field properties are obtained from the (time averaged) product of the instantaneous pressure *p(t)* and the corresponding instantaneous particle velocity ***u(t)*** at the same position:

$$I = \frac{1}{T}\int_0^T p(t)\cdot u(t)dt \tag{116}$$

Where the intensity ***I*** and the velocity ***u*** are vectors. (A simple explanation for the property that the intensity determines the free-field properties is that for an acoustic wave travelling in the negative x-direction the phase difference between *p(t)* and ***u****(t)* is 180°, while for a wave in the positive *x*-direction *p(t)* and ***u****(t)* are in phase.) The measured intensity corresponds to the net flow of acoustic energy at a given position. If the intensity around a sound source is measured at a number of positions, the radiated sound power can be determined. Even in the presence of reverberation or background noise one can determine in this way the (free-field) radiated power of a sound source. Since the intensity provides information about the strength and the direction, it can also be

used for localisation of complicated sound sources and for determining their directional characteristics.

Sound intensity probes

p-p probe

The standard, commercially available, sound intensity probe consists of two closely spaced-identical pressure microphones; in most cases the microphones take the form of short cylinders, which are placed “face to face”, see Fig. 2.20. From the difference in measured pressure, the pressure gradient is determined, which, by Newton’s law, is proportional to the time derivative of the velocity (see chapter 2). The average pressure is determined from the sum of the two microphone signals. The intensity can be calculated from the **imaginary** part of the cross-spectrum between the two microphone signals using a dual-channel audio analyser [1]. Although this probe works well in most practical cases, there are some drawbacks:

- For different frequency ranges one should use different spacers (e.g. 12mm for 125Hz<f<5000Hz and 50 mm for 31.5Hz<f<1250Hz; using f as the frequency [1]). The measurement time increases due to the changing of the spacers.
- The properties of the two microphones, and especially the phase characteristics, should be the same. Therefore a pair of carefully matched microphones are used (Bruel & Kjaer supplies e.g. microphones with a phase matching smaller than 0.3° [1]). This mismatch causes the low frequency limit.
- The quality (or bandwidth) of the sound intensity measurement depends on the reactivity of the sound field. The low end of the bandwidth is rising if the reactivity increases.
- The p-p sound intensity probe is relative large, measurements in small cavities or cabinets will therefore be impossible;
- The sound intensity probe can not be used in the near field of a sound source, when intensity changes along the probe, i.e. if the intensity is different at the two microphone positions.

p-u probe

A more direct approach would be to combine a pressure microphone with a particle velocity microphone, e.g. the Microflown. It should be encapsulated so that it has the same external configuration as the used microphone, see Fig. 48. The microphone and the particle velocity sensor can be positioned quite close to each other, so that the pressure and velocity are determined at almost the same point. Using the same dual channel audio analyser as is used for the p-p probe, the intensity is now calculated from the **real** part of the cross-spectrum between the microphone and the particle velocity signal.

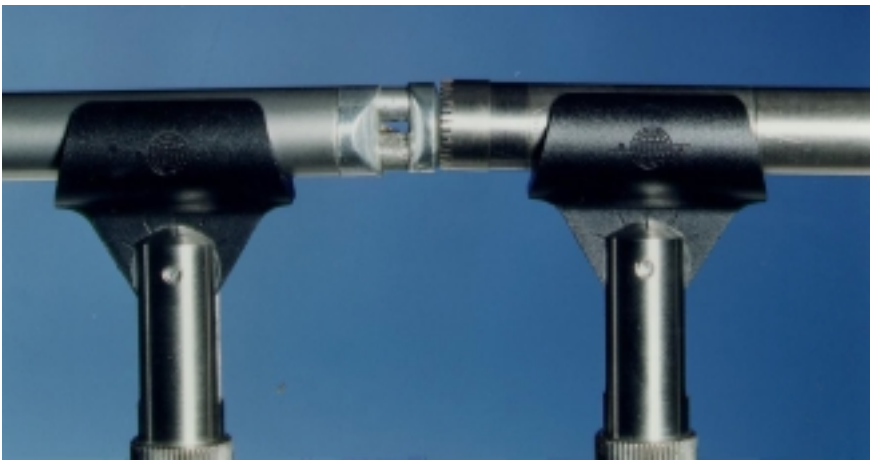

Fig. 48: Photo of the first operational p-**u** intensity probe consisting of a ½" Bruel & Kjaer microphone and a particle velocity sensor, the Microflown that was tested in the Philips Natlab in 1997.

u-u probe

It is possible to calculate the pressure and create a sound intensity probe from two particle velocity probes. This method is somewhat similar to the p-p method and is based upon the linearised equation of mass conservation, see chapter 2. Here the p-p method and the p-**u** method are compared, the **u-u** probe will be the subject in chapter B5: "The u-u sound intensity measurement method".

Directional characteristics

Assuming that the sound pressure microphone has omnidirectional properties the p-p and the p-**u** should have the same directional characteristics, i.e. **I** ∝ cos(φ); referring to the photograph of the p-**u** probe (Fig. 48), φ=0 is defined as the direction normal to the photo; for the p-p probe φ=0 is the direction along the line joining the centers of the microphones (see Fig. 2.20).

Effect of the phase mismatch of probes

The calculation of the sound intensity of the p-p method is very sensitive to a phase mismatch of the two pressure microphones. The particle velocity is derived from the difference signal of both the pressure microphones. Two large signals are subtracted and the (small) result is used to calculate the particle velocity. A phase difference of 0.3° is considered to be large.

The calculation of the sound intensity of the p-**u** method is not very sensitive to a phase difference since the signals of both pressure microphone and particle microphone (the Microflown) are multiplied and not subtracted. A phase difference of 0.3° is considered negligible.

Critical situations

When the distance between sound source and intensity probe is larger than the reverberation distance (the reverberation distance is defined as the distance to the source where the direct field and the reverberant field are equal) it becomes difficult to determine the intensity accurately. The same holds when a high-background noise level is present or when the intensity is directed in a oblique direction (φ ≈90°) or the sound field consists of more uncorrelated parts of equal strengths. A quantity for the validity of intensity measurements is the reactivity index, defined as: $L_k = L_I - L_p$, where L_I and L_p are the intensity and the pressure in dB's (reference to 1pW/m^2 and 2x10^{-5} N/m^2 , respectively). The ideal condition is of course when L_k=0 thus L_p=L_I (free-field condition). When L_k becomes strongly negative the measured pressure (in dB) is much higher than the free-field pressure; the latter quantity is then difficult to determine accurately.

Experimental

The intensity has been measured using a p-p Bruel & Kjaer intensity probe and a p-**u** probe consisting of a ½" Bruel & Kjaer microphone and a Microflown (see Fig. 48). A dual channel Bruel & Kjaer audio analyser (type 3550/2035) is used for calculating the intensity in 1/3-octave bands. If the p-p probe is used $\mathbf{I} \propto Im(G_{AB})$, where $Im(G_{AB})$ is the imaginary part of the cross-spectrum between the two microphone signals. If the p-**u** probe is used, then in principle the intensity can be calculated from the real part of the cross-spectrum between the microphone- and the Microflown signal. However, since the two sensors differ in their physical behaviour and the preamplifiers of both pressure microphone and particle velocity microphone are quite different, there may be an extra phase difference. This phase difference has been determined in an experiment in the anechoic room, using one sound source (a loudspeaker) and the p-**u** probe, connected to the dual channel audio analyser. Suppose that in that experiment a phase difference I_0 between the p- and **u** sensor was found, then the intensity measured in another experiment (more sources or reverberant room) is found from:

$$I \propto cos(I_0) \cdot Re(G_{AB}) + sin(I_0) \cdot Im(G_{AB}) \qquad (117)$$

G_{AB} being the cross-spectrum between the microphone- and the Microflown signal.

Critical experiments have been performed using two loudspeakers. In the first experiment the two loudspeakers were positioned on one line, with the intensity probe in the middle, see figure 2^A. The two loudspeakers were excited by two uncorrelated noise sources. When the excitation strength of the two loudspeakers are equal, the net energy flow should be zero, but both the pressure as the particle velocity level should be 3dB higher than when one loudspeaker is excited.

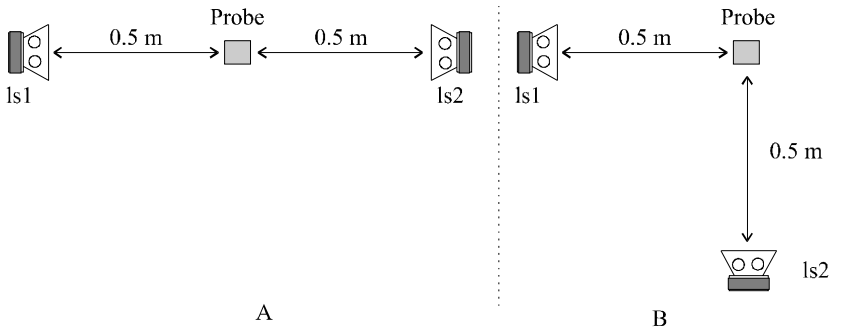

Fig. 49: Measurement set-ups: (A) two loudspeakers positioned on one line, (B) two loudspeakers perpendicular to each other.

In order to make the reactivity index more negative the same experiment was performed in a reverberation room; the reverberation time T_{60} was about 3-8s. A second experiment has been performed using two loudspeakers at right angles to each other. The intensity probe is placed such that the contribution to the intensity from the second loudspeaker should be zero (φ=90°), see figure 2^B. The distance between loudspeaker and intensity probe was about ½ meters.

Experiments

Two sound sources in line, figure 2^A

For the configuration as sketched in figure 2^A the intensity was measured. Some results are given in Figures 3,4..7; figures 3 and 4 refer to the anechoic room for the 1/3 octave bands of 125 and 3150Hz respectively. Figures 5,6 and 7 refer to the reverberation room for the 1/3-octave bands of 125, 315 and 3150Hz.

The loudspeakers are excited with (uncorrelated) noise, where the excitation of loudspeaker 1 is fixed (s_1) and of loudspeaker 2 variable (s_2). In the figures $(s_2/s_1)^2$ is plotted horizontally. The relative intensity is plotted vertically; I_0 is the measured intensity when loudspeaker 2 is switched off and only loudspeaker *1* is excited.

The straight line in the figure corresponds to the expected behaviour, i.e. $I/I_0 \propto (s_2/s_1)^2$ with I=0 for $s_2 = s_1$. For the 1/3-octave bands with centre frequency below 1250Hz three intensity probes were used: p-**u** and the p-

p probes of Bruel & Kjaer with spacers of 12mm and 50mm; for higher frequencies the p-**u** probe and the p-p probe with a spacer of 12mm were used. The abbreviations in the figures refer to these probes. In the experiments in the reverberation room the reactivity index (L_k) when s_2 =0 (I=I_0) was about -7dB.

Almost all experimental results, represented in figures 3,4..7 are in good agreement with the expected theoretical results. The only exception is when in the reverberation room the p-p probe with a spacer of 12mm is used for the 1/3-octave band of 125Hz. This is not unexpected since for the lower frequencies a spacer larger than 12 mm should be used (1). For a frequency of 125Hz, with a spacer of 12 mm and a reactivity index of -7 dB, the actual phase difference across the two microphones is only 0.3°; whereas in the specifications of Bruel & Kjaer for this pair of microphones a phase difference of about 0.1-0.3° is given.

Two sound sources at right angles to each other, figure 2^B.

For the configuration as sketched in figure 2^B the measured intensity should be independent of the excitation of loudspeaker 2, since the direct acoustic field is perpendicular to the direction of the intensity probe (⊥ = 90°) and the contribution of the reverberant field is also zero.

The following experimental steps were carried out:

1) The excitation level of the (uncorrelated) noise to the two loudspeakers were adjusted such that the pressures at the microphone were almost equal; the excitation of the first loudspeaker is denoted as s_0.
2) The intensity was measured with loudspeaker 2 switched off and loudspeaker 1 being excited with the signal s_0; the measured intensity is I_0.
3) The intensity I was measured as a function of the excitation s_1 of the first loudspeaker, with loudspeaker 2 switched on or off.

One expects to measure a linear relation between I/I_0 and $(s_1/s_0)^2$.

Some experimental results are given in figures 8, 9 and 10; I/I_0 is plotted vertically and $(s_1/s_0)^2$ horizontally; a straight line is the expected relation. Similar to the previously mentioned measurements three

intensity probes were used: p-**u** and p-p probes from Bruel and Kjaer with spacers of 12mm and 50mm.

The abbreviations in the figures refer to these probes and to whether loudspeaker 2 is switched on or off; e.g. 's2off p-u' means that loudspeaker 2 is switched off and the intensity is measured with the p-**u** probe and 's2on 12mm' means that loudspeaker 2 is switched on and the intensity is measured with the p-p probe with a spacer of 12mm.

The reactivity index for $(s_1/s_0)^2$=0.04 and s_2 'on' was about –15dB to -20dB.

The figures show a good agreement between the experimental results and the expected (theoretical) results; the only exception is the p-p probe with a spacer of 12 mm for the 160 Hz 1/3 octave band. This is not unexpected since the actual phase difference between the two microphones of the intensity probe was about 0.1-0.2°, whereas Bruel & Kjaer specifies that the phase matching of the microphones is about 0.1-0.3°.

Conclusions

The intensity probe, consisting of a pressure microphone and a particle velocity probe, p-**u** probe, is a good alternative for the existing p-p intensity probe. Critical experiments show that the behaviour of the p-**u** probe is similar to the p-p probe. Advantages of the p-**u** probe above the p-p probe are that for different frequency ranges the same configuration can be used (no spacers), that no accurate matching of the sensors is necessary, the dimensions of the p-**u** probe can be smaller, so that also near-field measurements could be performed.

References.

[1] Bruel & Kjaer Tech. Review no 3,4 (1982), 4 (1985), 4 (1986), 1 (1996).

[2] H.E. de Bree et al, The Microflown, Sensors and Actuators: A, Physical, volume SNA054/1-3, pp 552-557, (1996).

[3] W.F. Druyvesteyn and H-E. de Bree, A new sound intensity probe; comparison to the Bruel & Kjaer p-p probe, J. audio Eng. Soc., Vol. 48, 2000 No. 1/2 January/February.

Measurement results

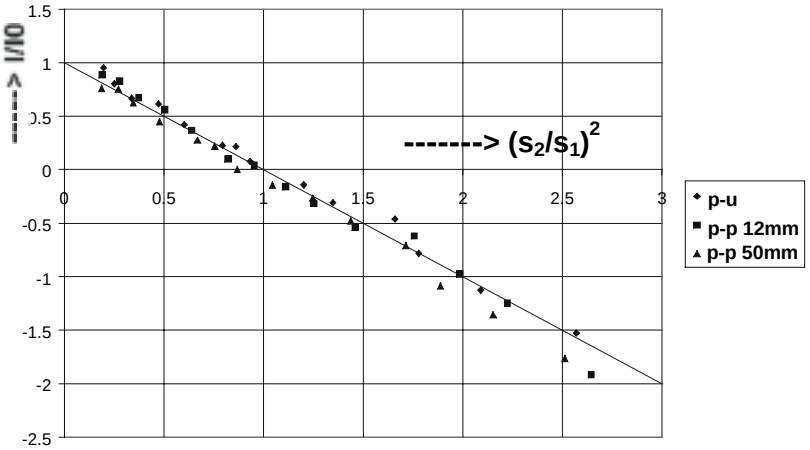

Fig. 50: Measurement performed in the anechoic room at 125 Hz in a 1/3-octave band. The measurement results should be on a line from (0,1) to (3,-2). The 12mm and the 50mm p-p probe have the expected results. The p-**u** probe performs also well.

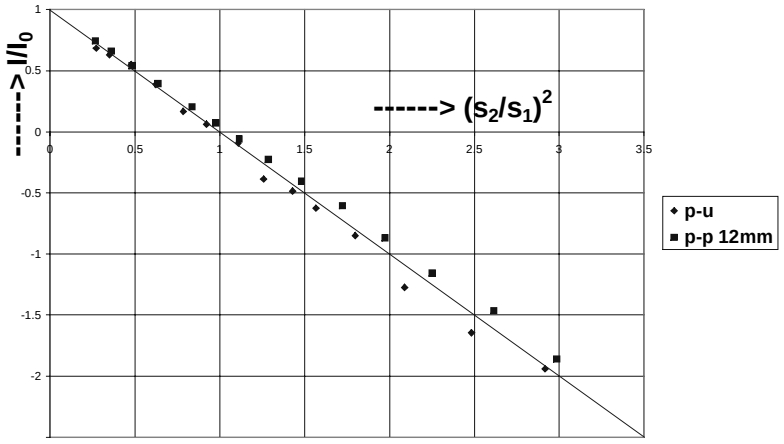

Fig. 51: Measurement performed in the anechoic room at 3150Hz in a 1/3-octave band. The measurement results should be on a line from (0,1) to (3,-2). At this frequency it is not possible to use a 50mm p-p probe. The 12mm p-p probe and the p-**u** probe are performing well.

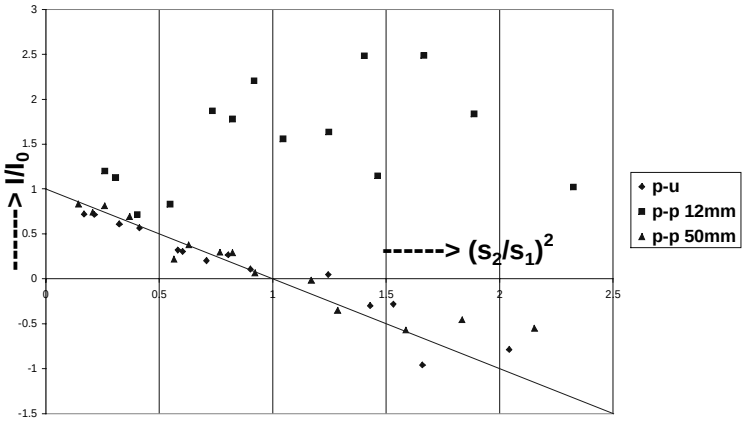

Fig. 52: Measurement performed a reverberant room at 125Hz in a 1/3-octave band. The measurement results should be on a line from (0,1) to (2,-1). As can be seen it is not possible to use a 12mm p-p probe, this is due to the reactivity of the sound field. The 50mm p-p probe and the p-**u** probe are performing well.

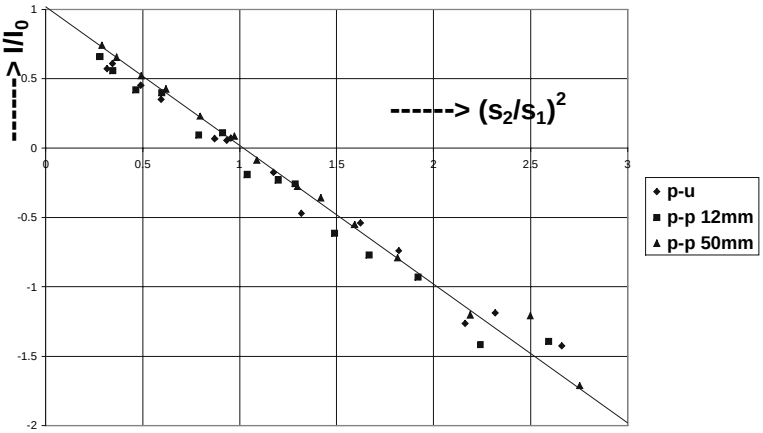

Fig. 53: Measurement performed a reverberant room at 315Hz in a 1/3-octave band. The measurement results should be on a line from (0,1) to (3,-2). For higher frequencies it is possible to use the 12mm p-p probe and the 50mm p-p probe. The p-u probe is performing well.

pu intensity probe

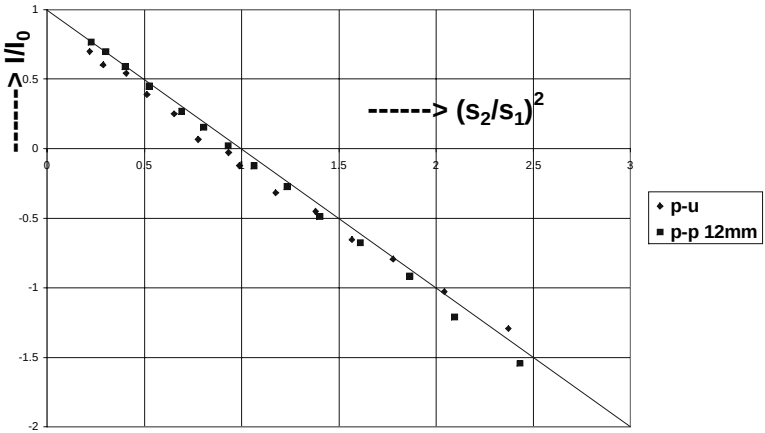

Fig. 54: Measurement performed a reverberant room at 3150Hz in a 1/3-octave band. The measurement results should be on a line from (0,1) to (3,-2). At this frequency it is not possible to use the 50mm p-p probe. The 12mm p-p probe and the p-**u** probe are performing well.

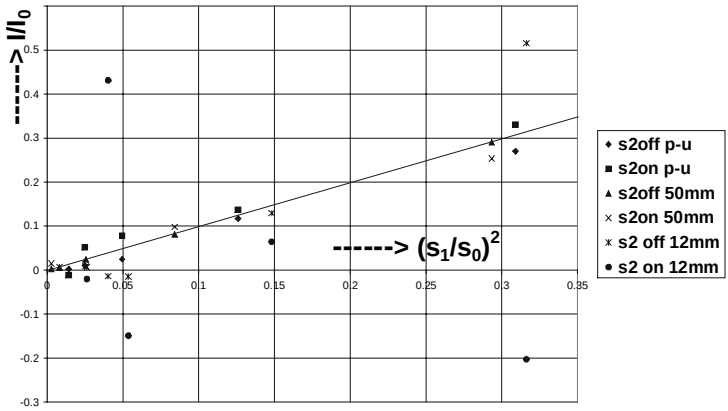

Fig. 55: Measurement performed a reverberant room at 160Hz in a 1/3-octave band to determine the linearity of the probes. The measurement results should be on a line from (0,0) to (0.35,0.35). The 12mm p-p probe is performing poorly if the perpendicular loudspeaker is switched off and even worse if it is turned on. The 50mm p-p probe and the p-**u** probe are performing well.

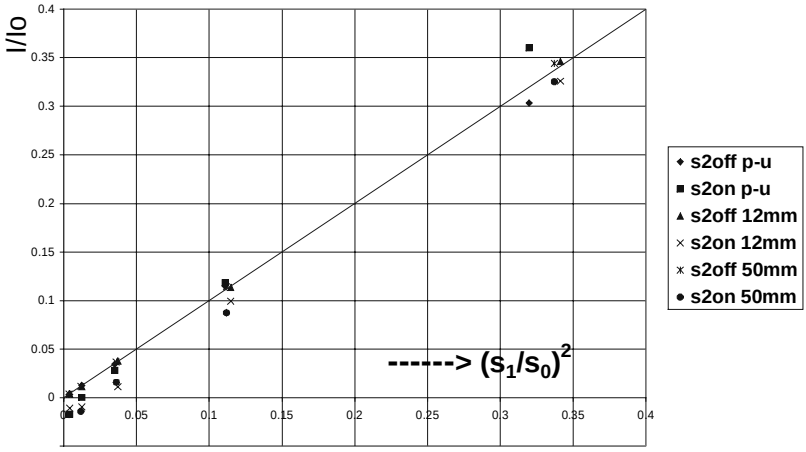

Fig. 56: Measurement performed a reverberant room at 800Hz in a 1/3-octave band to determine the linearity of the probes. The measurement results should be on a line from (0,0) to (0.4,0.4). At these frequencies the 12mm, the 50 p-p probe and the p-**u** probe are performing well.

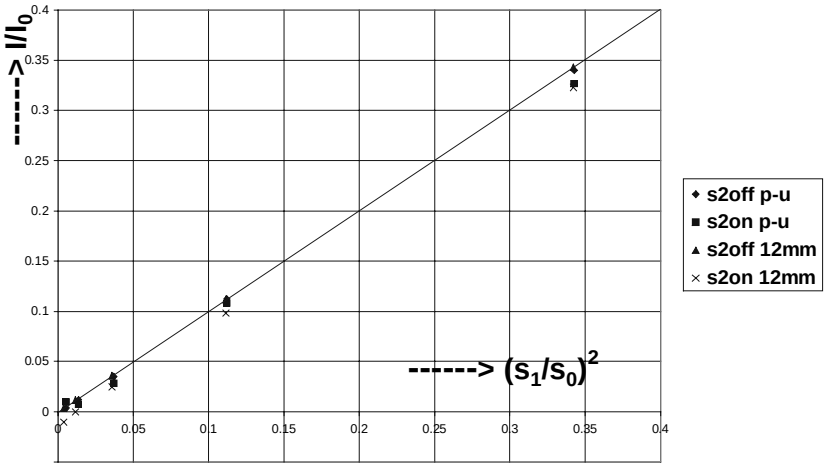

Fig. 57: Measurement performed a reverberant room at 2500Hz in a 1/3-octave band to determine the linearity of the probes. The measurement results should be on a line from (0,0) to (0.4,0.4). At this frequency the 50mm p-p probe can not be used. The 12mm p-p probe and the p-**u** probe are performing well.

B2 A PU INTENSITY MEASUREMENT

The half-inch p**u** (sound) intensity probe is developed to have a far better price performance than the normally so expensive sound intensity measurements. An optimal total cost of ownership is based on the following reasons:

(1) Reduced measurement time (personnel cost), the complete acoustic spectrum can be measured at once.
(2) No special analyser is needed (soundcard of a PC can be used).
(3) The same accessories (mountings, windscreen etc.) as a half-inch microphone can be used (it is simply one ½" probe).
(4) The (open source) software for data acquisition is available as freeware.
(5) Simple and low cost calibration possibilities are demonstrated.

This paragraph is based on the work that is presented in [1].

Introduction

Sound intensity is a useful measure in acoustics because in a reverberant environment free-field measurements can be done. The low cost intensity probe combines a sound particle velocity sensor and a microphone, which can be used as separate devices or in one housing so the velocity and the pressure are measured at almost the same location. Since the intensity is calculated from the cross spectrum of the velocity and pressure, a very accurate phase matching, as for the p-p method, is not necessary; and its signal to noise ratio (SNR) is higher than for the separate sensors. The data-acquisition and processing is implemented on a standard personal computer, combined with a simple calibration, and thus creating a very powerful intensity-measuring device.

Apart from the sound intensity this probe determines other acoustic properties like sound energy and acoustic impedance. However in this paragraph the focus will be de determination of sound intensity.

What is sound intensity?

Sound intensity measurements are quite useful as an acoustic measuring technique, since in a reverberant environment, even with background noise, the free-field as well as the diffuse sound field are determined. The measurement of the sound pressure directly gives the sum of the free-field and the diffuse- sound field, the free-field properties are obtained from the (time averaged) product of the instantaneous pressure *p(t)* and the corresponding instantaneous particle velocity ***u(t)*** at the same position:

$$I = \frac{1}{T}\int_0^T p(t)\cdot u(t)dt \qquad (118)$$

where the intensity ***I*** and the velocity ***u*** are vectors. The measured intensity corresponds in fact to the net flow of acoustic energy at a given position. If the intensity around a sound source is measured at a number of positions, the radiated sound power can be determined. Even in the presence of reverberation or background noise one can determine in this way the (free-field) radiated power of a sound source [2].

Design considerations

At first instance only a specialised half-inch Microflown was used to perform the p-u measurements. The combination of this (bowed version of) a ½” velocity probe and a standard ½” microphone seemed to be the most cost-effective realisation of a low cost intensity probe, see Fig. 47. The idea was that the person that is interested in sound intensity has most likely a ½” microphone so only an additional ½” Microflown is needed to create an intensity probe. Although this sounds logical it seemed not to be the right way to go. It showed that the realisation of the holder to position the Microflown and microphone introduced mechanical problems and standard accessories like windscreens would not fit anymore.

So a ½” probe that contained a Microflown and a miniature microphone was realised. Although the miniature microphone introduced new costs (it must be fitted into the ½” Microflown) and new calibration efforts (now both Microflown as miniature microphone need to be calibrated) this realisation proved to be most convenient.

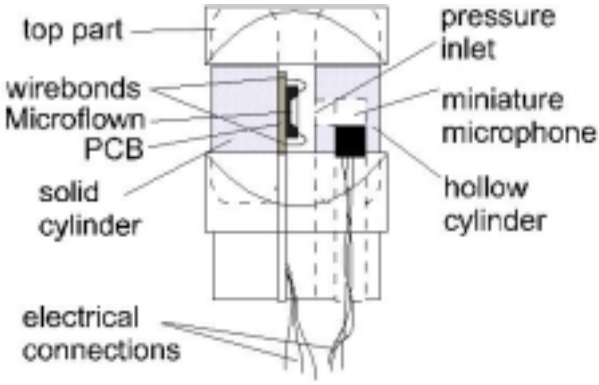

Fig. 58: Cross section of the half-inch sound intensity probe.

The standard ½" accessories can be used and the calibration of the miniature microphone can be done at the same time the Microflown is calibrated. Furthermore this type of realisation determines sound pressure and particle velocity at the same position.

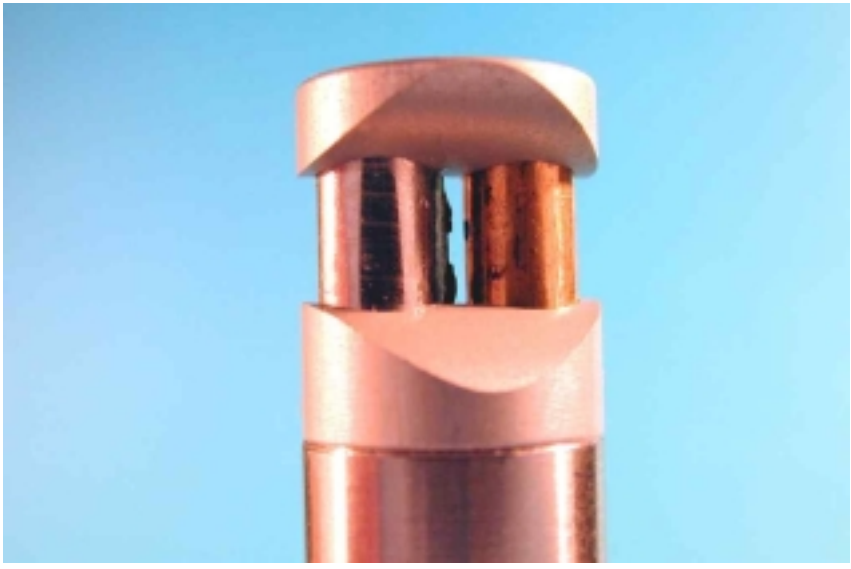

Fig. 59: The half-inch sound intensity probe.

Calibration of the p-u probe

Calibration of acoustical sensors is difficult because the acoustical environment is often not known, and a reference particle velocity sensor is not available. Several calibration techniques can be used, all with its pro's and contra's. These calibration techniques are:

(1) Standing Wave Tube
(2) Anechoic environment
(3) Open window calibration
(4) Reverberant environment.

The first two calibration methods are dealt with in chapter A, the open window and the reverberant method will be discussed in this paragraph.

Open window method

It seems obvious that a setting without any reflections creates an anechoic environment. So as an alternative for the anechoic room, we carried out measurements with the sound probes situated outside an open window. The results where remarkably consistent.

Reverberant room

With use of a single sound source in a reverberant room and the p-u probe it is possible to determine how much sound is direct, and how much belongs to the reverberant sound field (diffuse sound), see also chapter B7 and [12]. From measurements of the particle velocity in the direction of the direct sound wave, combined with measurements of the same velocity in the perpendicular direction (only diffuse sound is measured with the particle velocity sensor), we can calculate the ratio of direct and reverberant sound, knowing that only a third of the power in the diffuse sound field is measured with the particle velocity probe due to the directivity (cos(θ)) of the Microflown.

$$u_{\parallel}^2 = u_{\mathrm{dir}}^2 + \tfrac{1}{3} u_{\mathrm{rev}}^2 \tag{119}$$

$$u_{\perp}^2 = \tfrac{1}{3} u_{\mathrm{rev}}^2 \tag{120}$$

where $u_{\parallel}^2$ and $u_{\perp}^2$ are the auto spectra of the measured particle velocities in the two directions not in the direction of the sound source. Note that the units are not important here since ratios are used. We can rewrite this in terms of the ratio of the direct (free field) and reverberant sound as:

$$\left(\frac{u_{\text{dir}}}{u_{\text{rev}}}\right)^2 = \frac{u_{\parallel}^2 - u_{\perp}^2}{3u_{\perp}^2} \tag{121}$$

and

$$u_{\text{dir}}^2 = u_{\parallel}^2 - u_{\perp}^2 \tag{122}$$

Taken that the reverberant sound field to be uniformly diffuse and the microphone has a omni-directional sensitivity all power in both direct and reverberant field will be measured.

$$p^2 = p_{\text{dir}}^2 + p_{\text{rev}}^2 \tag{123}$$

where p^2 is the auto spectra of the measured pressure and $p_{\text{dir}}{}^2$ and $p_{\text{rev}}{}^2$ of the direct and reverberant pressure.

The combination of the pressure due to the direct sound field, and the particle velocity due to the direct sound field yields the sensitivity of the particle velocity sensor.

$$p_{\text{dir}}^2 = p^2\left(1 + \left(\frac{u_{\text{dir}}}{u_{\text{rev}}}\right)^{-2}\right)^{-1} \tag{124}$$

Taken that the microphone is unidirectional, we can use this ratio for calculation the pressure on site of the p-u probe caused by the direct sound. Assuming all sound waves to behave as plain waves, we can derive easily that the ratio between the direct and reverberant particle velocity equals the ratio between the direct and reverberant pressures.

$$\frac{p_{\text{dir}}}{p_{\text{rev}}} = \frac{u_{\text{dir}}}{u_{\text{rev}}} \tag{125}$$

The combination of the pressure due to the direct sound field, and the particle velocity due to the direct sound field yields the sensitivity of the particle velocity sensor. The pressure sensor can be calibrated for phase

and sensitivity against a known (calibrated) pressure sensor, if the directivity is comparable.

Combining the pressures and particle velocities of the derived direct sound we are able to calibrate the particle velocity sensor sensitivity S_u in terms of [V/Pa*] or S_{u*} in [V/(ms^{-1})] by dividing by $\rho_0 c$.

$$S_{u^*} = \sqrt{\frac{u_{\mathrm{dir}}^2}{p_{\mathrm{dir}}^2}} S_p \tag{126}$$

and

$$S_u = \frac{1}{(\rho_0 c)} \sqrt{\frac{u_{\mathrm{dir}}^2}{p_{\mathrm{dir}}^2}} S_p \tag{127}$$

Calibration of the phase between the pressure and particle velocity sensor is more difficult in a reverberant environment because the sound field is not known. The method above only yields the auto spectra of the free field pressure and particle velocity.

In a pure diffuse sound field, there is no acoustical power in the plane perpendicular to the direct sound field because the a purely diffuse sound does not contain net power, and thus the pressure and particle velocity are out of phase by 90 degrees. Secondary acoustical paths, however, like floors, ceilings, walls and furniture can cause some acoustical power in the plane perpendicular to the direct sound field. Avoiding these secondary paths as much as possible it should be possible to measure the phase between the pressure and particle velocity sensors. Use of a pulse and time-window measurement it is possible to measure the direct sound anechoic also, but in most practical situations the distances are short leaving not much time for measurement and thus resulting in a very poor frequency resolution.

The phase may be determined in the following manner. Because with the previous method the free-field auto spectra of the sound pressure and particle velocity are determined, the product of the free-field pressure and velocity can calculate the sound intensity.

On the other hand, the product of the auto spectrum of the measured pressure and velocity *and* the cosine of the phase of the sound field obtains the sound intensity. Combining the phase of the sound fielded with the phase between the measured pressure and particle velocity we

should be able to calculate the cosine of the phase between the two acoustical sensors. This way at least the active part of the sound intensity can be measured using these sensors.

Calibration Data Acquisition Hardware

Calibration is most likely to be the most essential part of this measurement technique because the results will differ strongly depending on the calibration.
Several parts in the system should be calibrated in order to be able to measure accurately. We should calibrate the pressure- and particle velocity sensor and the A/D converter in the soundcard or an external A/D converter. It is also possible to introduce a preamplifier so that we benefit most of the resolution of the used A/D converter.

The A/D converter can easily be calibrated with the use of a known signal because we can assume them to be almost perfectly linear in the frequency domain. A preferably white or pink signal can be connected to the inputs with known power. With use of little data acquisition and processing, sensitivity and a phase difference between the channels can be easily calculated and implemented. We should note that the gain and panning of the input should not be altered on the personal computer since this will be applied on the acquired signals. If a preamplifier is being used it should also not be adjusted after calibration. The preamplifier can be calibrated as where it part of the Soundcard. Depending on the quality the phase between two channels can be severely large, for example up to 140 degrees at 20kHz for a standard PC Soundcard or only 2.5 degrees for an external A/D converter.

Software

The software is programmed in MATLAB® around a Graphical User Interface for convenience in usage, and because MATLAB® is widely used and is an easy program language so that users are able to modify the software to their demands, see also appendix II. The program is capable of working in three modes: Instantaneous, a measurement for 30 seconds and data processing from a wave file.

The instantaneous mode is very useful in examining the acoustical situation. In the instantaneous mode it is possible to compute and show various windows containing:

- the auto spectra of the pressure and particle velocities.
- the coherence between the two signals.
- the intensity (active, reactive, and the phase).
- the sound energy.
- the active and reactive intensity divided by the energy (indicating a net power transport velocity and reflection).
- the acoustical impedance (magnitude, real and imaginary part).

All axes can be chosen as linear or logarithmic scales and the vertical axis are by default in dB. Both the frequency spectrums in terms of the frequency as the spectrum in 1/3-octave bands can be displayed. Calculated spectra's can be saved to the workspace, or as MATLAB® .mat files and as ASCII files for further processing. It should be noted that the 1/3 octave band of 20kHz is an underestimation in case a sample frequency of 44.1kHz is used. This underestimation is due to the fact that the power is not know in the whole 1/3 octave band, and the A/D converter will not be less accurate above 20kHz. If sample frequencies of 48kHz (or higher if possible) are used, the 20kHz 1/3 octave band will be computed correctly.

In the instantaneous mode it is not convenient to use large frames because of computing time and power, resulting in a small frequency resolution, the default size of the frame is set at 2048. For higher resolution it is advisable to save the signals into wave files so that they can be processed later with higher frequency resolution. We should note that a good frequency resolution is necessary if we would like to use the advantage of low frequency measurements, which is possible due to the physics of the p-u probe. The size of frames for the Fast Fourier Transform can be altered according the wanted frequency resolution, and computation time (default 16834). By default a Hanning window is used for increasing the spectral resolution.

It is possible to record measurements using external hardware like external A/D converters connected to digital Soundcards, CD-Recorders, or DAT-recorders. This is also very convenient since most of these are rather mobile while measuring becomes rather easy and post processing can be automated with use of the available software.

Data processing

The data processing is performed on the raw data according to the processing block (see Fig. 60). The left signal is supposed to contain data from the pressure sensor while the right signal is supposed to contain the data from the particle velocity sensor.

(1) First the signals are converted to the frequency domain because in this domain it is much easier to correct for the sensitivity and phases in the signals than it would be in the time domain.
(2) Secondly, the data is converted from arbitrary units to Volts with the use of the calibration of the soundcard or external A/D converter, amplifier combination.
(3) Note that also the phase shift due to the soundcard is implemented here so that the signals, know in Volts, are in the right phase.
(4) If the signals are known in Volt, the program converts them to the pressure and particle velocity signals using the calibration of the sensitivity and phases of the acoustical probes. The signals are converted to Pascal's for the pressure signal, and to ms^{-1} for the particle velocity signal using values for ρ_0 and c which can be altered in the parameter settings (default 1.219 kgm^{-3} and $343ms^{-1}$).
(5) From the pressure and particle velocity signals the various auto- and cross-spectra are calculated, G_{pp}, G_{uu}, G_{pu}.

The sound Intensity in a sound wave is the product of the pressure and particle velocity, this product is called the instantaneous Intensity. Sound Intensity is defined as the mean of the integral of the instantaneous sound intensity integrated over all times. In practical cases it is never possible to integrate over all times and it is convenient to integrate from $t = -T$ to $t = T$ or from $t = 0$ to $t = T$.
The intensity is defined by:

$$I = \lim_{T\to\infty} \frac{1}{2T} \int_{-T}^{T} p(t)u(t)dt \tag{128}$$

The above equation can be rewritten as a cross correlation:

$$I = \lim_{T\to\infty} (1/T) \int_{0}^{T} p(t)u(t)\,\mathrm{d}t = R_{pu}(0) \tag{129}$$

Note that the subscripts of the cross spectra are not used uniquely. In this text a definition often used in signal processing is used, see [24]. The advantage of this usage is that the reactive part of the intensity equals the imaginary part of the cross spectrum while otherwise a factor –1 to keep up with the convention [2], [21], [9].

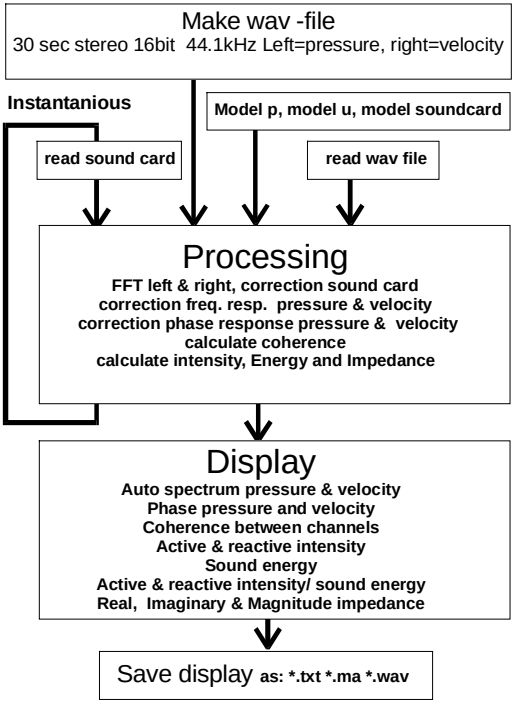

Fig. 60: Schematic representation of the software.

The distribution in frequency is given by the Fourier transform of the cross-correlation function, which is termed the Cross-spectral Density S:

$$S_{\mathrm{pu}}(\omega) = \frac{1}{2\pi}\int_{-\infty}^{\infty} R_{\mathrm{pu}}(\tau)\exp(-\mathrm{i}\omega\tau)\,\mathrm{d}\tau \tag{130}$$

For practical reasons it is convenient to redefine the spectral density functions S as single-sided functions of frequency G. Because the signals of the pressure and particle velocity sensors are both real we can

represent only half the frequency span without losing any information. The advantage of the single sided spectral density function is the fact that less data has to be stored and computing time and effort is decreased.

$$G_{\mathrm{pu}}(\omega) = 2S_{\mathrm{pu}}(\omega) \quad \text{for } \omega > 0 \tag{131}$$

From this cross spectrum the intensity spectrum is easily obtained because the active intensity equals the real part of the cross spectrum between the pressure and particle velocity, while the reactive intensity equals the imaginary part.

$$I_{\mathrm{active}} = \mathrm{Re}\left(G_{pu}\right) \tag{132}$$

$$I_{\mathrm{reactive}} = \mathrm{Im}\left(G_{pu}\right) \tag{133}$$

By integration over the frequency spectra we can calculate the intensity over frequency spans.

$$I(f_1, f_2) = \int_{f_1}^{f_2} G_{pu}(f)df \tag{134}$$

The program calculates both the intensity spectrum in the frequency domain as the intensity spectrum in 1/3 octave bands.

The energy density is defined as [2]:

$$E(t) = \frac{p^2(t)}{2\rho_0 c^2} + \tfrac{1}{2}\rho_0 u^2(t) \tag{135}$$

Using the earlier calculated cross spectra, the time averaged energy density can be calculated as:

$$E = \frac{G_{pp}}{2\rho_0 c^2} + \tfrac{1}{2}\rho_0 G_{uu} \tag{136}$$

The acoustical impedance is defined as the ratio between the pressure and the particle velocity. The program computes the acoustical impedance as:

$$Z = \frac{P}{U} = \frac{G_{pu}}{G_{uu}} \tag{137}$$

Measurements

The half-inch sound probe is calibrated in a short standing wave tube and a small 1m^3 anechoic chamber. The standing wave tube can be used for frequencies up to 4kHz, the anechoic chamber can be used for frequencies higher than 1 to 2kHz. As estimation for the frequency response a simplified model is used. At 4kHz the model has 4dB under-estimation and for frequencies higher than 6kHz it has a 3dB over-estimation. Our opinion is that we can correct for the resulting errors later manually.

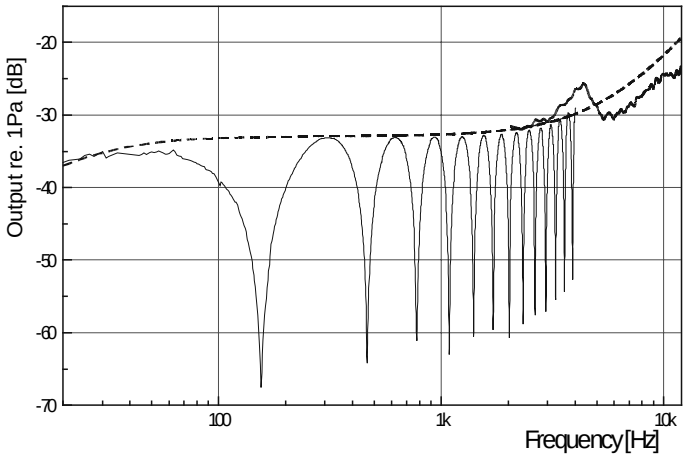

Fig. 61: Calibration measurement of the miniature microphone in a standing wave tube and a small anechoic room. The dotted line represents a simplified model.

The model that we used to estimate the frequency response of the microphone is:

$$\left|H_p\right| = 22.3 \frac{\sqrt{\frac{f^2}{3kHz^2}+1}}{\sqrt{\frac{f^2}{8kHz^2}+1} \times \sqrt{\frac{25Hz^2}{f^2}+1}} \quad [{}^{mV}\!/_{Pa}] \tag{138}$$

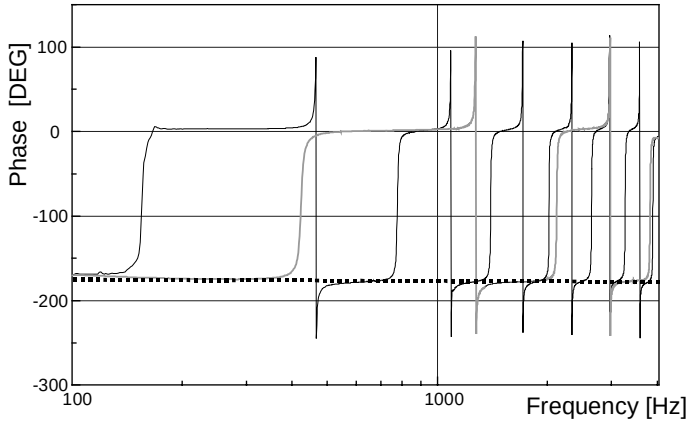

Fig. 62: Calibration measurement of the phase of the miniature microphone in a standing wave tube (gray and black lines). The dotted line is the model.

We use a constant value of -177DEG as a model for the phase response of the miniature microphone.

The model that we used to estimate the frequency response of the Microflown is:

$$|H_u| = \frac{35}{\sqrt{\frac{f^2}{650^2}+1}\times\sqrt{\frac{f^2}{2500^2}+1}} \quad [mV/Pa^*] \tag{139}$$

A Microflown is sensitive for particle velocity rather than sound pressure so the sensitivity cannot be given in mV/Pa. As a reference we use the sensitivity of a Microflown in mV/Pa* with Pa* is 1m/s/ρc.

The phase of the Microflown can also be determined in the standing wave tube. Two measurements are done were the microphone was rotated by 180 degrees so that the mean phase of the two sound fields in the standing wave tube equals zero (see Fig. 64). The phase was fitted by:

$$Arg\{H_u\} = -1.52\times\tan^{-1}\frac{f}{914} \tag{140}$$

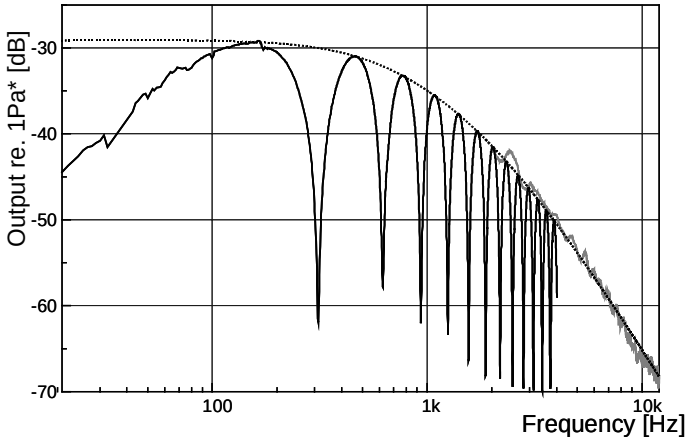

Fig. 63: Calibration measurement of the Microflown in a standing wave tube and a small anechoic room (gray line). The dotted line represents the model.

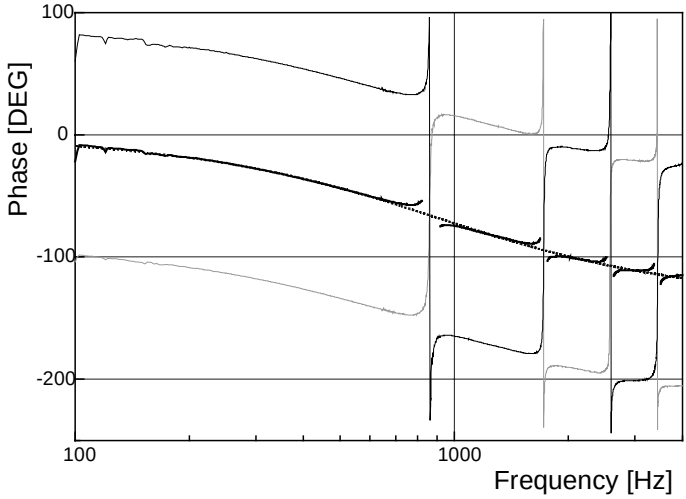

Fig. 64: Calibration measurement of the phase of the Microflown. The black and gray line represents the phase response in the standing wave tube for the two Microflown orientations. The black line in between is the average phase response, the dotted line represents the model.

Calibration measurements, open window and reverberant room

Although the frequency and phase response are known due to the previous calibration measurements, the alternative methods are presented here. The advantage of these alternative methods is that no special facilities are needed.

With the acoustical sensors placed outside an open window we can create an anechoic sound field. Depending on the weather conditions this can be very effective. In the measurement shown a breeze was present, this is visible below 100Hz in the measurements. The use of windscreens will account for this, but the distance between the p-u probe and the reference microphone should not be that large.

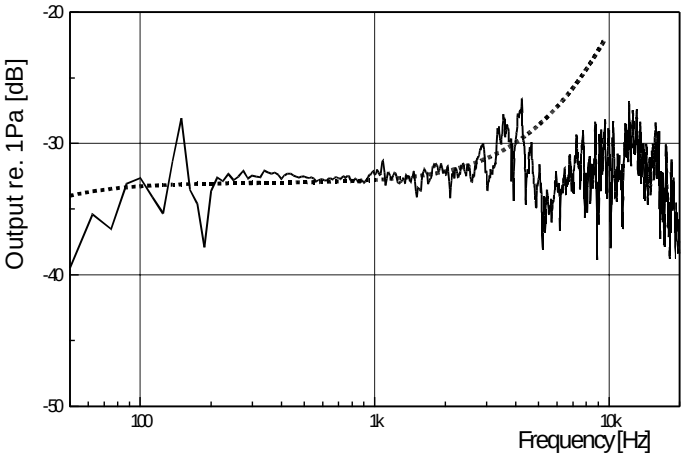

Fig. 65: Calibration measurement (and gray line the model) of the amplitude response of the microphone. Measured with the open window method.

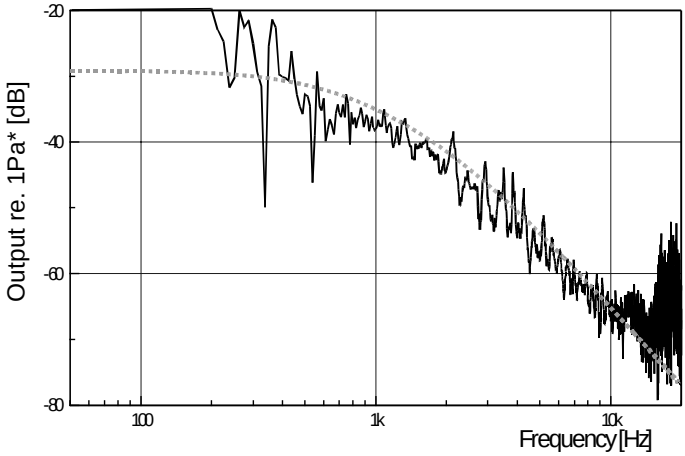

Fig. 66: Calibration measurement (and gray dotted line the model) of the amplitude response of the Microflown. Measured with the open window method.

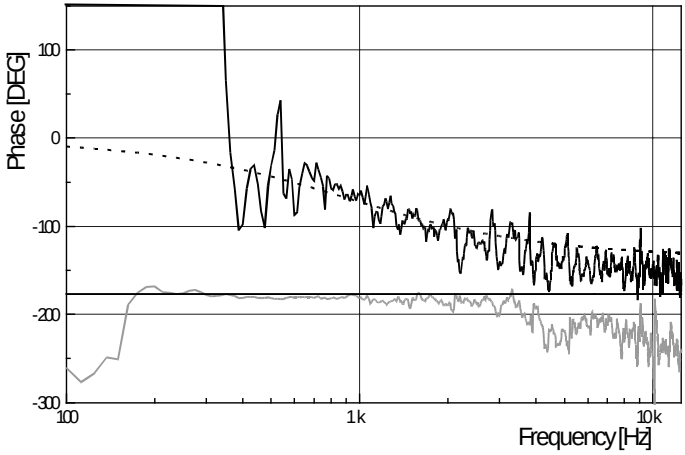

Fig. 67: Calibration measurement of the phase response of the Microflown (black line, and dotted line the model), and of the miniature microphone (gray line, and the model) measured with the open window method.

We also applied the so-called reverberant room calibration technique. The two signals of the p-u probe are measured while a reference microphone was positioned as close as possible. The sensitivity of the pressure is calculated using the transfer function between the two

microphones and the sensitivity of our reference microphone. The sensitivity of the particle velocity probe is obtained using the calculated auto spectra of the free field pressure and free field particle velocity, according to the presented method, combined with the sensitivity of the used microphone.

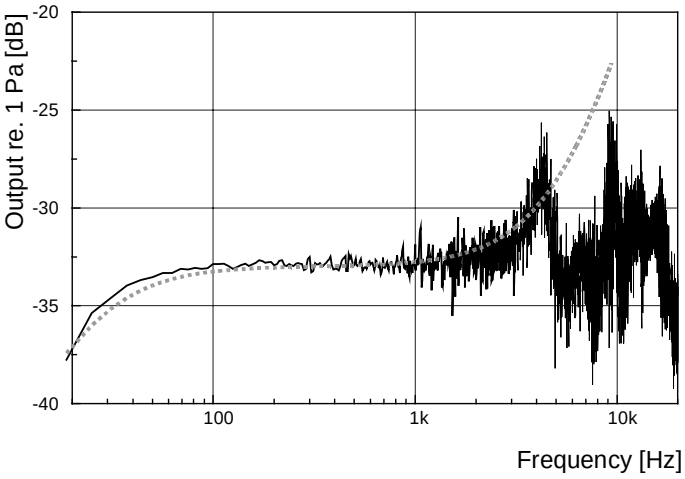

Fig. 68: Calibration measurement (and dotted line the model) of the amplitude response of the miniature microphone. Measured with reverberate room method.

The calculated sensitivities of both the pressure (see Fig. 68) and the particle velocity sensor (see Fig. 69) are consistent with the models that are obtained from the standing wave tube and the anechoic measurements. Above 10kHz the calculated sensitivities out of the reverberant measurements are not valid anymore (see Fig. 69) because no acoustical power is available in the used sound field due to the used loudspeaker (see Fig. 70).

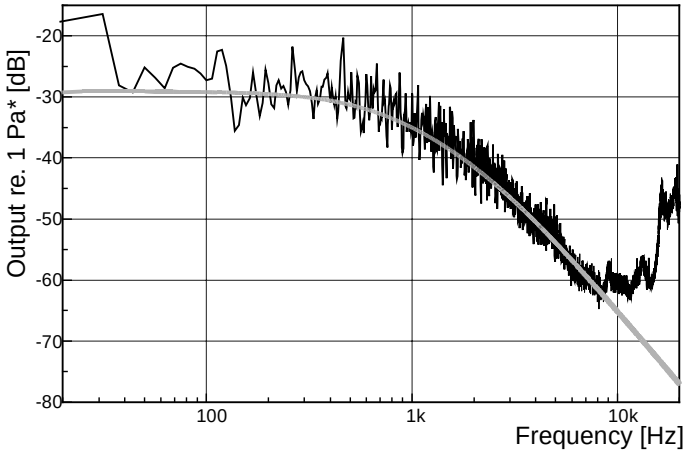

Fig. 69: Calibration measurement (and gray line the model) of the amplitude response of the Microflown. Measured with reverberate room method.

Intensity measurements

Intensity measurements are performed in a reverberant room using a p-p probe and the described p-u probe. The p-p probe is a Brüel & Kjær intensity probe, type 2260 Investigator, using two ½" microphones (type 4181). The measurement was executed twice using a spacers of 50mm (type UC 5270) and of 12mm (type UC 5269) so that we could measure up to 1/3-octave center frequencies of respectively 1.25kHz and 5kHz. The intensities in the 1/3-octave frequency bands are calculated using the Investigator B&K 2260.

The signals of the p-u probe were measured using a personal computer combined with a digital soundcard combined with an external A/D converter and a preamplifier set at about +40dB for both channels. The signals were acquired using Cool Edit Pro. A white noise of 5.1mVrms was applied in order to calibrate the hardware. The created wave file was read into MATLAB® and the power was calculated so that we are able to determine the signal in Volts for our calculations. With use of the presented software we calculated the active intensity in 1/3-octave bands presented in the figures below.

First the Intensity of a small loudspeaker was measured using the p-p and the p-u technique. In the figure below we can notice that the Intensity

measured with the p-u method slightly over estimates the measurements of the p-p measurement in all frequency bands. The cause of this over estimation is very likely to be found in the calibration of the acoustical sensors, and/or preamplifier/soundcard combination. More accurate calibration of the acoustical sensors and data acquisition hardware, including the p-p probe, should minimise this discrepancy.

It should be noticed that below 150-250Hz the loudspeaker radiates almost no acoustical power and the measured intensities are mainly from the background. The noise levels in the p-p measurement mainly cause the discrepancies in the measurements for frequencies below 100Hz. Noise levels can be found in paragraph B4.

The intensities calculated using the p-u method for frequencies above 10kHz do overestimate the actual intensity because the model for the microphone sensitivity is not valid anymore (see Fig. 61 and Fig. 68).

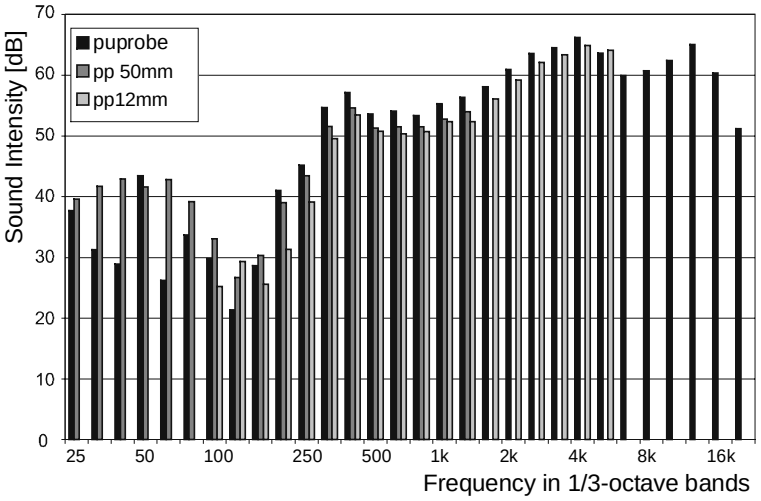

Fig. 70: Measured sound intensity of a simple loudspeaker (p-u probe (black), p-p probe 50mm (dark gray) and p-p probe 12mm (light gray)). The pu probe is used in the complete frequency range, the 50mm p-p probe form 25Hz up to 1.25kHz and the 12mm p-p probe from 100Hz up to 5kHz. No measurements where executed with a 6mm, ¼" p-p probe (possible up to 10kHz).

Secondly we performed intensity measurements in between two identical loudspeakers, which were the same as describe above. The power to loudspeaker 1 (S1) is kept at a constant level, while the power to

loudspeaker 2 (S2) is varied. A linear relationship is expected between the active intensity and the ratio between the acoustical power of both loudspeakers [11]. The minus sign indicates that the power is coming from the direction of loudspeaker 2. From the measurement at for example 1kHz (see Fig. 71) we can note that this is the case for both measurement techniques. Again we notice a small discrepancy between the p-p and p-u measurement.

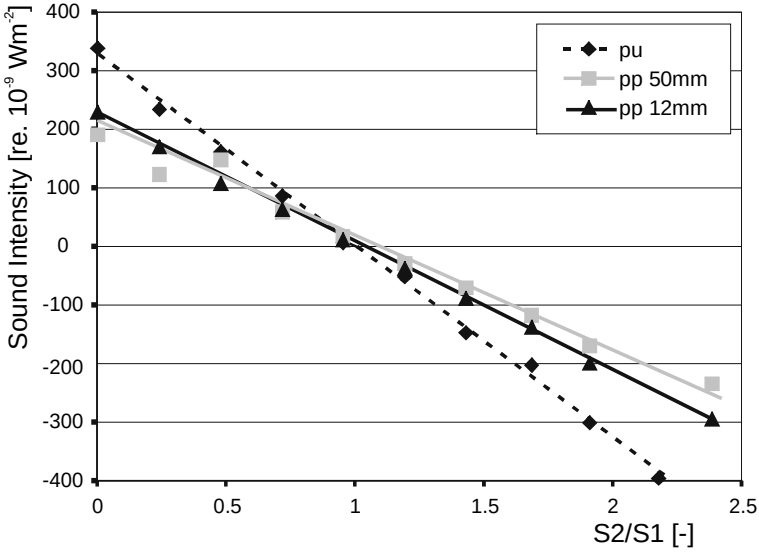

Fig. 71: Sound intensity measurements between two identical loudspeakers at 1kHz 1/3-octave band versus the ratio of the electrical power sent to the loudspeakers.

Conclusions

Intensity can be measured using the p-u probe and a conventional soundcard as is shown in the measurements performed. The measured active intensity is somehow overestimated compared to the p-p measurements, but adequate hardware and calibration should account for this.

The main advantage of the p-u probe versus the p-p probe is its low cost of ownership. Measurements over the whole frequency range can be made at once without changing the probe itself. The main disadvantage is

the calibration of the p-u probe. Because the two signals measure two different acoustical properties it is not possible to use the same method for both sensors unless the sound field is known well. A few calibration methods are presented all with their own accuracy and simplicity.

The combination of the p-u probe and the software creates an affordable and easy to use intensity measuring device. Also other acoustical parameters like the reactive intensity, phase of the sound field, the energy density, and the acoustical impedance are being calculated without extra effort because they are also obtained out of the auto and cross spectra of the pressure and particle velocity signals.

References

[7] Ron Raangs et al. A low cost intensity probe, AES Amsterdam 2001.

[8] F.J. Fahy, Sound Intensity, ISBN 0419198105, second edition 1995.

[9] Bruel & Kjaer Tech. Review no 3,4 (1982), 4 (1985), 4 (1986), 1 (1996).

[10] H.E. de Bree et al, The Microflown, Sensors and Actuators: A, Physical, volume SNA054/1-3, pp 552-557, (1996); thesis University of Twente 1997.

[11] W.F. Druyvesteyn, H-E de Bree. A new sound intensity probe; comparison to the Bruel & Kjaer p-p probe. J. audio Eng. Soc., Vol. 48, 2000 No. 1/2 January/February)

[12] W.F. Druyvesteyn, H-E. de Bree, M Elwenspoek. A new acoustic measurement probe, the microflown. Proc. IOA Vol. 21, part 3, pp 139-149 (1999)

[13] H-E. de Bree, Lammerink, Elwenspoek, Fluitman, Use of a fluid flow-measuring device as a microphone and system comprising such a microphone, Patent PCT/NL95/00220, 1995.

[14] J.W. van Honschoten, G.J.M. Krijnen, V.B. Svetovoy, H-E. de Bree, M.C. Elwenspoek, Optimisation of a two-wire thermal sensor for flow and sound measurements, MEMS Interlagen 2001

[15] V.B. Svetovoy, I.A.Winter, Model of the μ-flown microphone, Sensors &Actuators 86 (2000), 171-181.

[16] Farina, A. Torelli - "Measurement of the sound absorption coefficient of materials with a new sound intensity technique" - Pre-prints of the 102nd AES Conference, Berlin, 23-26 March 1997.

[17] P. Fausti, A. Farina , R. Pompoli Transmission loss measurements: validity of the sound intensity technique in laboratory and in the field Proc. of 11th. International FASE Symposium, Valencia 15-17 November 1994

[18] S.W. Kang, Y.H. Kim. Active intensity control for the reduction of radiated duct noise. Journal of Sound and Vibration (1997) 201(5), 595-611.

[19] T. Loyau, J.C. Pascal, P. Gaillard. Broadband acoustic holography reconstruction from acoustic intensity measurements. I: Principles of the method. J. Acoust. Soc. Am. 84(5), 1988.

[20] H.E. de Bree. The Microflown (E-book)

[21] Eerden, F.J.M. van der. Noise reduction with coupled prismatic tubes. PhD Thesis, University of Twente, The Netherlands, November 2000.

[22] H-E. de Bree, T. Korthorst, P. Leussink, H. Jansen, M. Elwenspoek, A method to measure apparent acoustic pressure, Flow gradient and acoustic intensity using two micromachined flow microphones, Eurosensors X, Leuven, 1996.

[23] J.W.van Honschoten, H-E. de Bree, F.J.M. van der Eerden, G.J.M. Krijnen, The influence of viscothermal effects on calibration measurements in a tube, presented at the LA AES 2000.

[24] J.A. Cadzow. Foundations of digital signal processing and data analysis. ISBN 0-02-318010-2, 1987.

B3 3D P-U SOUND PROBE

A one-dimensional sound intensity probe can be extended to a three-dimensional probe by using three p-p probes (six microphones) or one microphone and three Microflowns (p-$\mathbf{u}^3$). All the probes are still in one plane, see Fig. 72. The advantage of this is that the p-$\mathbf{u}^3$ sound probe can be positioned near to a sound source (the p**u** method can also be used in the near field).

The realisation of a three-dimensional p-p probe is much more difficult. It now consists on six closely spaced sound pressure microphones. The prize of such probe will be very high and the frequency range is limited even more than the 1D p-p probe. This is because for measuring higher frequencies, the microphone spacing should be small and this stands in the way in a three-dimensional realisation: the six microphones will be too close to each other. This chapter is based on the work that has been presented in [1] and [2].

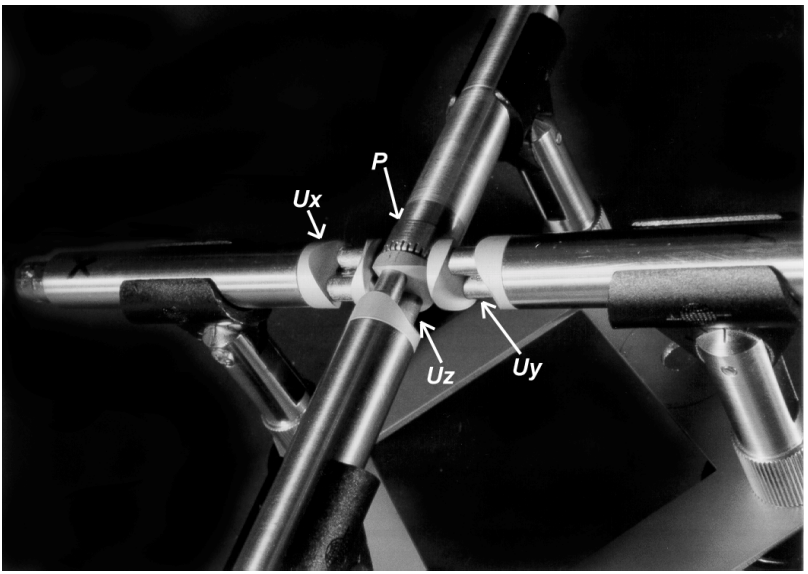

Fig. 72: Realisation of the 3-D probe that consists on three half-inch Microflowns and a half-inch sound pressure microphone.

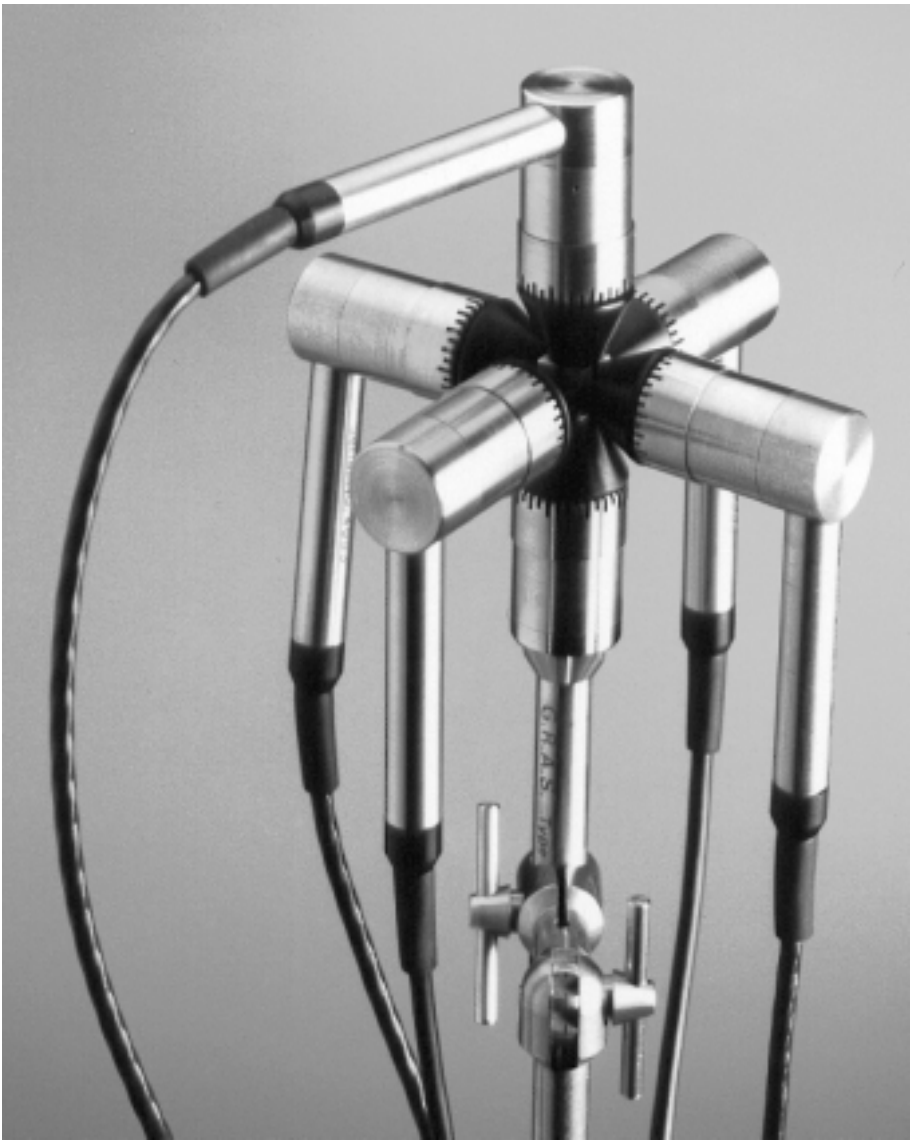

Fig. 73: Realisation of the 3-D ½" p-p probe that consists on three matched pairs of pressure microphones (G.R.A.S.).

A broad band three-dimensional sound probe will offer many advantages. The magnitude and direction of the acoustic energy flow can

be measured directly and this will improve the effectiveness of the measurement. A measurement will provide vector-based information. If a one-dimensional probe is used only the projection of an energy flow is measured. The 3D measurement allows measuring non-stationary sound fields as for example the noise of moving objects, like aeroplanes. The three-dimensional sound probe is capable of measuring the sound intensity vector in the full band sound but also the total sound energy *E* and three-dimensional acoustic impedance can be measured with this probe. Only the 3D-sound intensity is investigated in this chapter.

Realisation

The realisation that is acoustically examined in this chapter is composed from three half inch particle ICP velocity probes and a B&K half-inch pressure microphone. The four probes are arranged by four microphone clips and a brass frame. The possible next step in order to miniaturise the 3D sound probe is an on-chip three dimensional Microflown. The realisation of this 3D Microflown is described in the discussion of this chapter.

Experimental

The experiments were performed in two rooms, in an anechoic room and in a reverberation room. The dimensions of the reverberation room are about 8.5x7.2x5.1m^3, (a volume of 228m^3); it is a nonrectangular room. The reverberation time was about 7 seconds; the corresponding reverberation distance is about 0.5 meter.

As sound source a small loudspeaker box was used. The excitation was noise in the frequency range of 20Hz - 5000Hz. As pressure microphone a ½” Bruel & Kjaer, type 4134, was used. The Microflowns were encapsulated such that the external configurations are the same as the configurations of the pressure microphone, see Fig. 74. The pressure microphone is almost omni-directional, the Microflowns do have a polar pattern like a figure of eight. The signals from the sensors are connected to a Bruel & Kjaer two-channel analyser (type 2035) or to a Siglab four-channel analyser. An averaging over 1/3-octave frequency bands is performed.

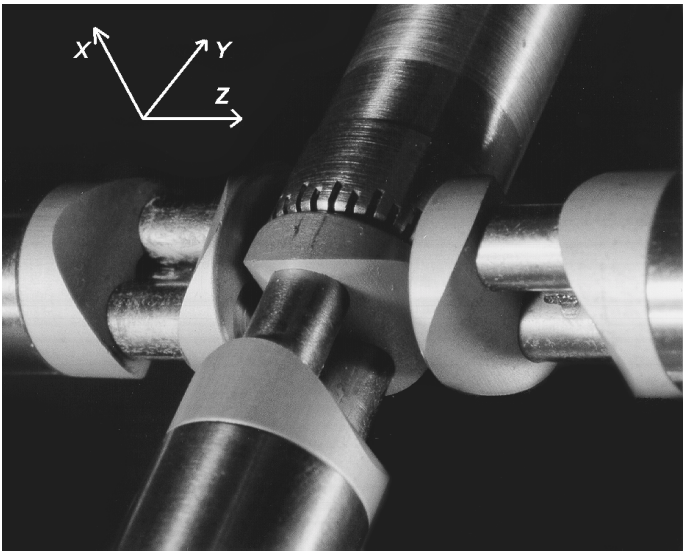

Fig. 74: Detail of the 3-D probe.

First the sensors are calibrated in the anechoic room. The $\boldsymbol{u_y}$ and $\boldsymbol{u_z}$ sensors are rotated 90^0 such that the three sensors are sensitive in the x-direction. The sensitivity and phase differences, as compared to the ½" pressure microphone, were measured, using as a sound source a loudspeaker at the *x*-axis.

The second step was to rotate the three sensors back to the position as shown in Fig. 74. The sound source, a loudspeaker, was positioned at a distance of 1m from the probe; various directions were used, which will be indicated as $[x_i,y_i,z_i]$. A constant excitation and the frequency spectrum of the noise was taken. The intensity was calculated as $\sqrt{(I_x^2+I_y^2+I_z^2)}$, where I_x, I_y and I_z are determined in the same way as in chapter B1 "*A comparison of the p-u method with the p-p method*".

The experiments were performed in the anechoic- and the reverberation room. Some results are shown in Fig.75. The measured intensity is compared with the intensity I,An,[1,0,0], being the intensity as measured in the anechoic room with the sound source in the x-direction and a distance of 1m between sound source and probe. The difference $I[x_i,y_i,z_i]$ – I,An,[1,0,0] is plotted vertically in the figure (in dB). Horizontally the various directions for the sound source are given, where the following abbreviations are used:

An,[110] means: anechoic room, source in [1,1,0] direction at 1m from the probe; R,[111] means: reverberation room, source in [1,1,1] direction at 1m from the probe; R^* refers to the case where the distance source → probe in the reverberation room was diminished from 1 m to 0.55m, thus: R^*,[101] means: reverberation room, source in [1,0,1] direction at 0.55m from the probe.

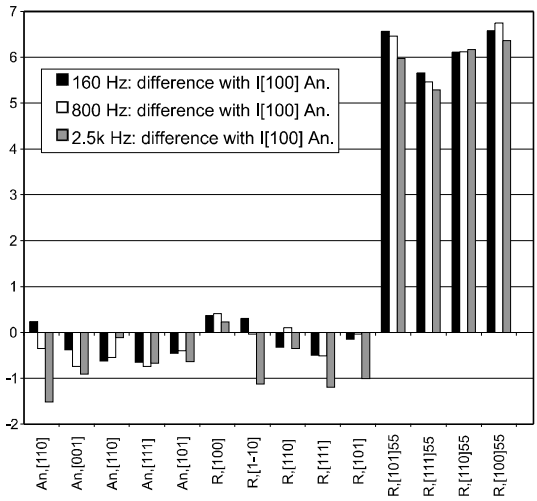

Fig.75: Measurements with a 3D p-**u** sound intensity probe. The first five values (An) should be zero: these are the measured intensities in a certain direction in an **anechoic** room compared with intensity [1,0,0] direction anechoic room.

The three columns for each direction/room refer to the three 1/3-octave band of 160, 800 and 2500Hz. When the distance between sound source and probe is 1m the differences in intensity, as compared to the measured intensity in the anechoic room with the source in the [1,0,0] direction at 1m distance are small; for a distance of 0.55m the differences are, as expected about 5-6dB.

The second five values (R) should also be zero: these are the measured intensities in a reverberation room in a certain direction compared with intensity [1,0,0] direction in an anechoic room.

The last four values should be about +5.2dB: these are the measured intensities in a reverberation room in a certain direction at 0.55m compared with intensity [1,0,0] direction anechoic room at 1m.

Ultra miniature 3D sound (intensity) probe

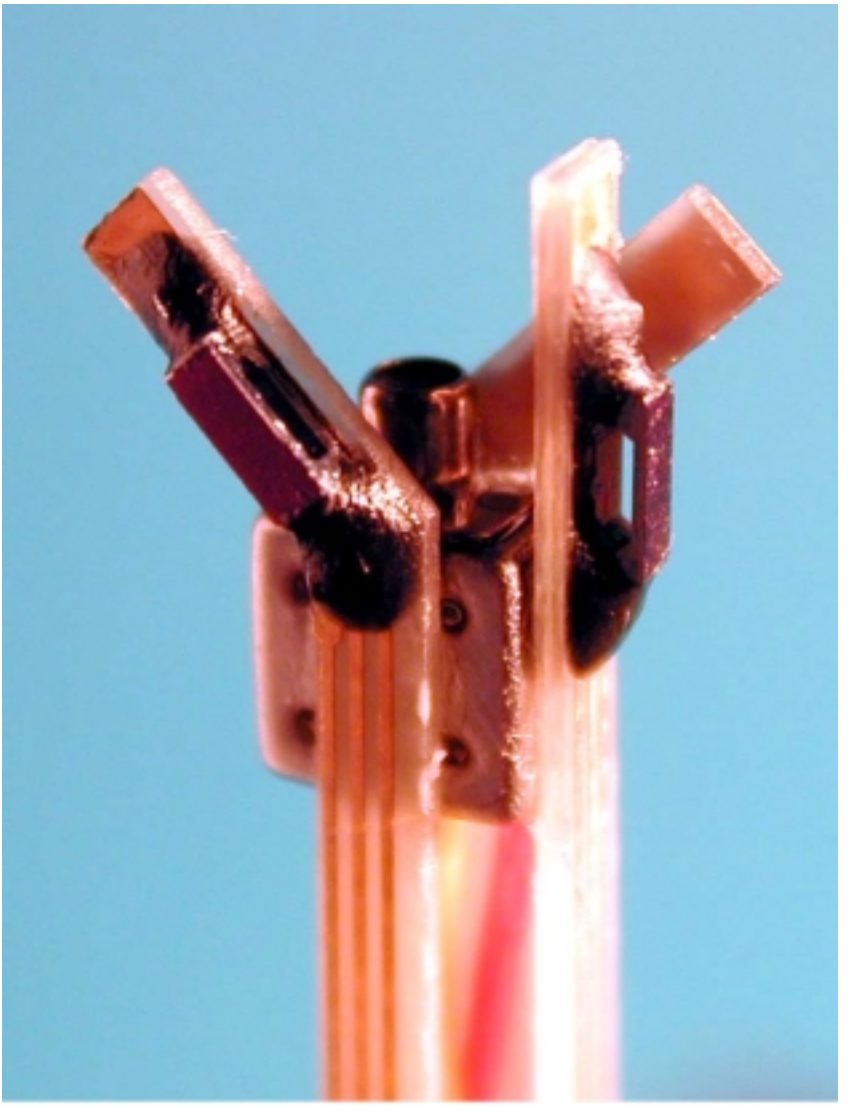

Fig.76: Realisation of an ultra miniature (5×5×5mm) three-dimensional sound (intensity) probe (2001).

3D pu intensity probe

A three-dimensional, miniature sound (intensity) probe was realised using three Microflowns and a miniature microphone. The Microflowns are protected with a package that introduces almost no gain (see Fig. 12) so the signal to noise ratio will be approximately 10dB less than the half-inch Microflown (see Fig. 82). The miniature microphone that is used in this realisation has a 5dB better selfnoise than the miniature microphone that is used in the half-inch sound probe (see Fig. 82). So the signal to noise ratio of the intensity measurement will be about 5dB to 10dB less good as the half-inch sound intensity probe. This ultra miniature three-dimensional sound (intensity) probe will fit in an imaginary 5×5×5mm cube.

Conclusions and discussion

- The experiments show that the particle velocity sensor is a good addition to the existing acoustic measurement devices. A three-dimensional sound intensity probe based on three Microflowns and a sound pressure microphone (p-$\mathbf{u}^3$) has been realised and tested up to 5kHz in an anechoic and reverberation room comparison. The measurement results were satisfactory.
- It showed possible to measure the 3D sound intensity with the use of four discrete sensors.
- An ultra miniature sound (intensity probe) was realised that allows sound (intensity) measurements with a good signal to noise ratio.
- It is also possible to create a 3D Microflown. If a miniature pressure microphone is added a 3D sound intensity probe can be manufactured with the size of a sugar cube.

A three dimensional Microflown

A realisation of the p-u^3 probe is shown in Fig.76. In order to make a fully micro machined sound intensity probe a three dimensional Microflowns is realised. The 3D Microflown has the advantage the physical dimensions are reduced and the sensors are placed orthogonal but several problems have to be sold before it will be ready as a product. The packaging will be the main issue to solve.

The first step in the realisation of the p-$\mathbf{u}^3$ probe is the micro machining part. The special part is solving the paradox of realising a three dimensional probe in the two dimensional (planar) technique. The through-the-wafer etching solves the planar character: $\mathbf{u}_y$ and $\mathbf{u}_z$ Microflowns are made on the surface and the $\mathbf{u}_x$ Microflown is obtained through the wafer, see Fig. 3.

A 300nm low-stress silicon nitride layer is deposited on a 3" silicon wafer by low-pressure chemical vapour deposition. A 200nm platinum/10 nm chromium layer is deposited and patterned on the bottom and on top of the wafer. The chromium layer is used for adhesion purposes.

The silicon nitride layer is patterned (top and bottom) with a reactive ion etching technique. The finalising step of the micro machining is the anisotropic wet chemical etching with a KOH solution.

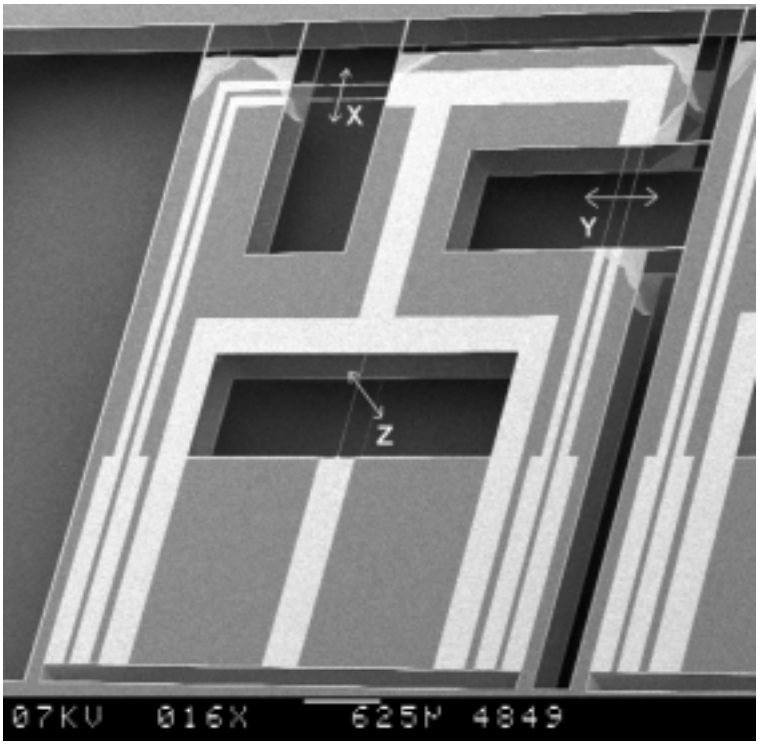

Fig.77: Realisation of a three dimensional Microflown (Erwin Berenschot 1999).

References

[1] W.F. Druyvesteyn et al., A New acoustic measurement probe; the Microflown, IOA London, 1999.

[2] H.-E. de Bree et al., Three-dimensional sound intensity measurements using Microflown particle velocity sensors, Orlando, MEMS ’99.

B4 SIGNAL TO NOISE RATIO OF INTENSITY PROBES

Theory and measurements are presented on the signal to noise ratio of the p**u** sound intensity probe and compared to p-p probe. Measurements of the signals and noise levels are presented.

Introduction

Two types of sound intensity probes (the p-p probe and the p-u probe) will be compared on their noise behaviour. First the noise behaviour of the particle velocity determination will be observed. For comparison a theoretical plane sound wave with an intensity level of 94dB SIL will be offered to both probes. This means that both sound pressure level as particle velocity are 94dB, SPL respectively PVL, see chapter 2.

After the signal to noise (S/N) behaviour of the particle velocity probes is calculated and the S/N of the sound pressure probe is presented, the S/N of the sound intensity determination is observed.

Other aspects of the p-p probe will not be presented extensively in this chapter. In short, the particle velocity is determined from the pressure difference between two closely spaced pressure microphones, see chapter 2. The sound pressure is calculated by the sum of both microphones. Usually the spacing between both probes is 12mm. Because the pressure gradient becomes smaller at lower frequencies the spacing is increased to 5cm in the bandwidth of 20Hz-1kHz. The lowest frequency the sound intensity measurement is still valid is determined by the reactivity of the sound field (the amount of reverberation). The 20Hz measurement is only possible under ideal circumstances (almost anechoic conditions) if the reactivity increases to higher levels, the lowest frequency that can be measured increases to 100-200Hz.

For high frequency measurements (above the 5kHz), the wavelength becomes short compared to the 12mm spacing and therefore the spacing has to be reduced to 6mm. For very high frequencies (10kHz-20kHz), the p-p probe becomes bulky and accurate measurements are not possible anymore.

S/N particle velocity determination (p-p)

First the signal to noise ratio of the particle velocity determination will be calculated. After that, the signal to noise ratio of the p-p sound intensity probe will be presented.

An example of a high quality matched pair is the Brüel & Kjær sound intensity microphone pair B&K 4181 [27]. One of the ½ inch matched microphones has a selfnoise of 20dB(A) and a sensitivity of 11.2 mV/Pa (U_{signal}=11.2mV at 1 Pa (=94dB SPL) pressure at 1 kHz). It will be convenient to express noise values in voltages and in one Hertz bandwidth the noise is than expressed in V/√Hz.

The noise level of the sound pressure probes is: U_{noise}=31.6nV/√Hz at 1 kHz [27]. The microphones do have a 1/f noise behaviour that will be dominant at frequencies under 500Hz [30]. Note that the 1/f noise is "1/f" in the power spectrum. The noise behaviour in the voltage domain is thus proportional to $f^{-½}$. Because the pressure difference is measured to determine the particle velocity, two microphones are used, the total noise will increase by a factor √2.

The signal (that will be used to calculate the particle velocity) is produced by the subtraction of both pressure microphone outputs. Under the condition that the microphone spacing is much smaller than the wavelength (see also chapter 2), For harmonic sound waves ($p(t,x)=p_0 \cos(\omega t-\omega x/c)$), the output of the pressure difference is given by:

$$p(x,t)-p(x+\Delta x,t)=\frac{p(x,t)-p(x+\Delta x,t)}{\Delta x}\Delta x \approx \frac{d}{dx}p(x,t)\Delta x=\frac{\omega}{c}p(x,t)\Delta x \quad (141)$$

For low frequencies (wavelength is small compared to the spacing of the sound probes) the pressure gradient is increasing linearly with the frequency. In Fig. 78 the output of the pressure difference assembly is depicted if a plane wave with a sound intensity level of 94dB is offered to the p-p probe. The signal is calculated for the three spacings that are commonly used.

For frequencies below 500Hz the signal to noise ratio decreases rapidly, Fig. 78. This is because of the presence of 1/f noise of the pressure microphones and due to the fact that the pressure difference signal is decreasing proportional with frequency.

For the particle velocity determination, the signal will be integrated, see Eq. (3) and therefore the frequency response will be flat, the noise

however will also be integrated and therefore the signal to noise ratio therefore will not be altered.

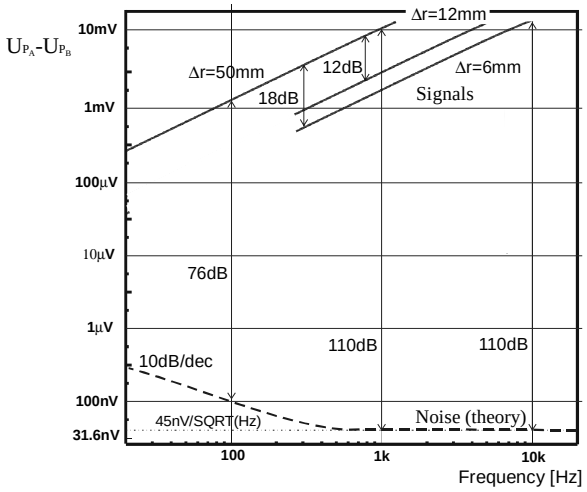

Fig. 78: Calculated noise output voltages in V/√Hz and signals due to 94dB SIL of a pp sound intensity probe.

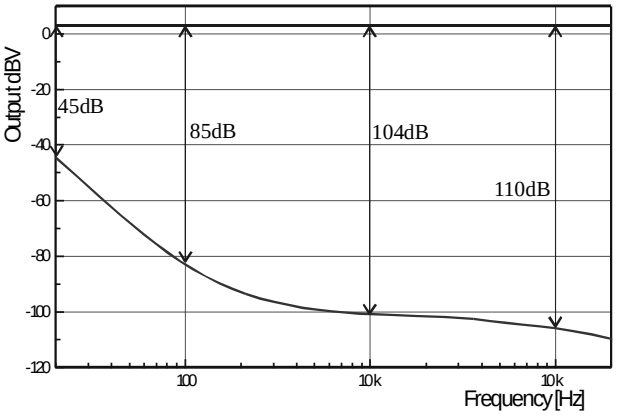

Fig. 79: Measured noise spectrum of a GRAS Type 40AC (½" Free field) in V/√Hz. Signal (14mV/Pa) is calibrated only at 1kHz and is assumed to have a flat frequency response. Both signal as noise are 40dB amplified.

S/N particle velocity determination (p-u)

The Microflown in the half-inch ICP probe is powered with 40mW and the operating temperature 600K (see chapter AIII "*The half-inch ICP Microflown, electronics and performance*"). The measured frequency response and noise spectra are shown in Fig. 23. The frequency behaviour has a low frequency roll-off at 10Hz due to the ICP preamplifier electronics. The noise is measured in a 1Hz bandwidth so the noise spectrum that is presented in Fig. 23 is given in dB(V)/√Hz.

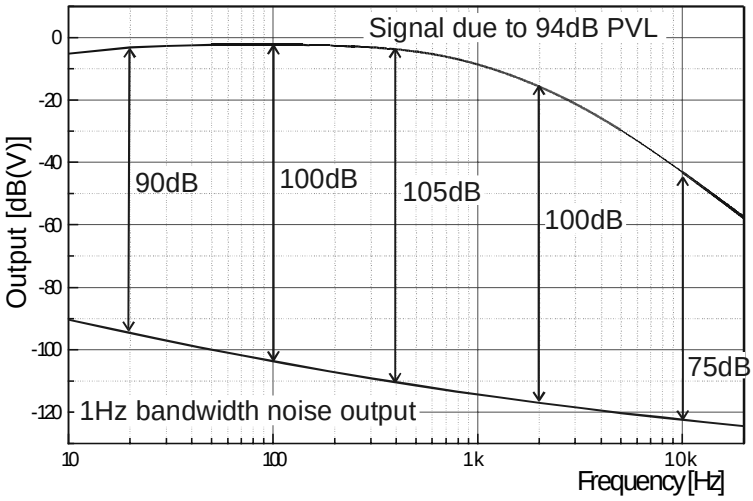

Fig. 80: Measured output of a ½" ICP Probe: the signal is due to 94dB PVL, the noise spectrum is given in dB(V)/√Hz.-

Comparing signal to noise ratios of particle velocity determination

As can be seen in Fig. 78 and Fig. 23 the S/N of the Microflown is higher for frequencies below 300Hz and the signal to noise ratio of the p-p method becomes better for frequencies between the 2kHz to 10kHz. The general conclusion is that the signal to noise ratio of a Microflown is better for lower frequencies (f<300Hz) that the two probes have comparable noise figures in the bandwidth of 300Hz up to 3kHz and that

the p-p probe has a better signal to noise ratio for frequencies in the 3-10kHz bandwidth.

S/N of the sound intensity probes

The signal to noise ratio of the particle velocity determination has been observed, now the sound pressure determination will be examined. Sound intensity is defined as the time averaged product of the sound pressure and the corresponding particle velocity at the same position.

$$I = \frac{1}{T}\int_{o}^{T} p(t)\cdot \boldsymbol{u}(t)dt \tag{142}$$

Normally this product is determined by taking the cross spectrum of both signals but in fact this provides the same results. If a p-p method is applied, the particle velocity is calculated like (see chapter 2):

$$u_r = -\frac{1}{\rho \Delta r}\int (p_a - p_b)dt \tag{143}$$

and the sound intensity determination will become:

$$I_{pp} = -\frac{1}{\rho \Delta r}\frac{1}{T}\int_{o}^{T}(p_a + p_b)\cdot\left(\int_{0}^{t}(p_a - p_b)d\tau\right)dt \tag{144}$$

If the output (signal plus noise) of a sound pressure microphone is described as: $p=\Delta p+\delta p$, using Δp as the signal part and δp as the noise component, Eq. (144) will alter in:

$$I_{pp} = -\frac{1}{\rho \Delta r}\frac{1}{T}\int_{o}^{T}(\Delta p_a + \delta p_a + \Delta p_b + \delta p_b)\cdot\left(\int_{o}^{t}(\Delta p_a + \delta p_a - \Delta p_b - \delta p_b)d\tau\right)dt \tag{145}$$

The time-averaged product of two non-correlating sources is zero (at least in theory) and therefore Eq. (145) will alter in:

$$I_{pp} = I_{pp\,noisefree} + \frac{1}{T}\int\left\{\delta p_a \int \delta p_a dt + \delta p_b \int \delta p_b dt\right\}dt \tag{146}$$

If the noise is of the form: $\delta p = \delta\tilde{p}\,cos\left(\omega t + \tilde{\theta}\right)$, the expression $\delta p_a \int \delta p_a dt$ equals $\delta\tilde{p}^2\, sin\left(2\omega t + 2\tilde{\theta}\right)/\,2\omega$ of which the time average is

zero. Therefore (in theory) the noise of a pp intensity probe should be zero. Of course the same is true for the p**u** intensity measurement:

$$\begin{aligned} I_{pu} &= \frac{1}{T}\int (\Delta p + \delta p)\cdot(\Delta u + \delta u)dt \\ &= I_{pu\,noisefree} + \frac{1}{T}\int \Delta p\,\delta u + \delta p\,\Delta u + \delta u\,\delta p)dt = I_{pu\,noisefree} \end{aligned} \tag{147}$$

In case of the pp sound intensity method, the imaginary part of the cross spectrum is taken, in case of the p**u** sound intensity method, the real part of the cross spectrum is used, see chapter BI "*A comparison of the p-u method with the p-p method*". Either way, the cross spectrum is used and two non-correlating sources will result theoretically in a zero output.

Signal and noise measurements.

For the pu sound intensity method the signal and noise have been measured. In table 1 results are shown for a ½" Bruel & Kjaer pressure microphone (type 4134) and an example of a Microflown. Results have been taken from [25].

Table 1: frequency, in 1/3 octave bands; measured signals in dB of pressure microphone and Microflown sensor.

Freq.	p signal	p noise	U signal	u noise	p S/N	U S/N	p.u signal	p.u noise	p.u S/N
40	49.3	40.8	57.8	42.0	8.5	16.8	53	40.1	12.8
80	63.2	44.2	66.2	34.1	19	32.1	64.6	37.6	27
500	75.5	17.1	65.2	28.7	58.4	36.5	70.4	5.5	64.9
4000	83.3	25.1	48.2	26.8	58.2	5.8	65.2	5.4	59.8

In table 1 measured values are given for the pressure microphone, when a signal was applied to the loudspeaker (p, signal) and when no signal was applied (p, noise). The measured values are in dB, the difference thus being a sort of S/N ratio (signal to noise ratio); the transfer function of the loudspeaker has to be taken into account to obtain the regular S/N.

In Fig. 81 the measured S/N ratio's are plotted for the 1/3 octave bands for the pressure microphone, the Microflown and the intensity (cross correlation).

The general trend is that at high frequencies the S/N ratio of the pressure microphone is better, while at low frequencies the reverse results appear. Besides that the intensity determines the free-field properties of a sound source, the S/N ratio as determined from the intensity can be better than the S/N ratio's of the separate sensors. A similar effect has been demonstrated using two pressure microphones [31].

The S/N ratio of the time averaged product of *p1(t).p2(t)* (*p1(t)* and *p2(t)* are the two microphones signals) is higher than the S/N ratio's of the separate microphones [31]. The reason for this effect is that in the time-averaged product of the two sensor signals the uncorrelated noise is cancelled. For the p-**u** probe this effect is shown in Fig. 81 and table 1. The S/N ratio referring to the intensity determination is always larger than lowest S/N ratio of the p- and u- probe, in some frequency bands it is higher than the S/N ratio's of both sensors.

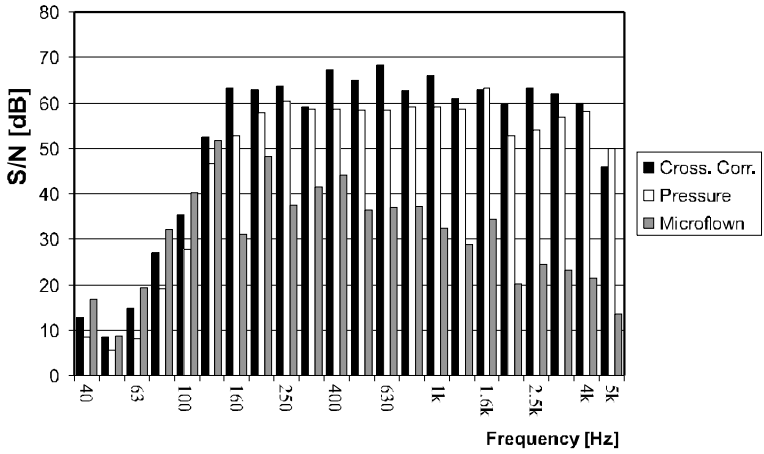

Fig. 81: S/N ratio's of the pressure microphone, Microflown and cross-correlation of both.

Signal to noise ratio ½" p-u probe [32]

The signal to noise ratio (S/N) of a p-**u** sound intensity probe is higher than the S/N of the auto-spectrum of separate probes. This is related to the fact that the cross-spectrum of two non-correlated noise sources (the selfnoise of the microphone and Microflown) is theoretically zero.

Fig. 82 shows the noise level of a calibrated half inch p-**u** probe given in dB SIL and measured in 1Hz bandwidth (sound intensity level has a reference of 1pW; in a plane wave a certain SIL in dB corresponds to the same amount number of SPL in dB).

1/f noise is the cause of the low frequency noise and the reduced signal to noise ratio of the Microflown causes the high frequency noise.

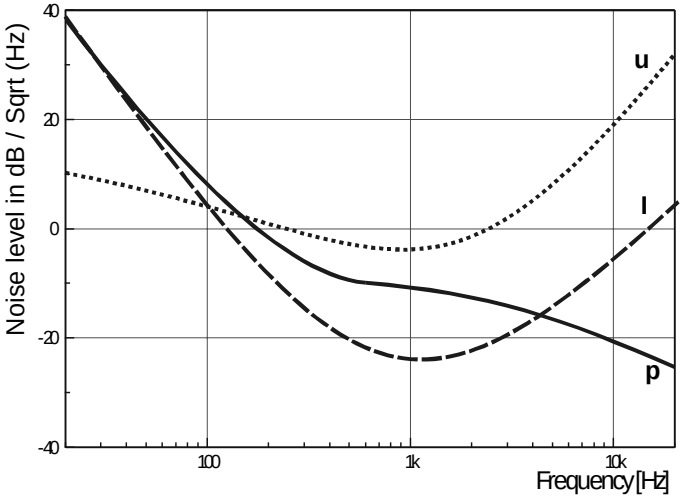

Fig. 82: Measured noise levels of a half-inch p-**u** probe. The noise level of sound pressure microphone (solid line) is given in dB SPL (re. 20µPa)/√Hz, the Microflown (dotted line) is given in dB PVL (re. 20nm/s)/√Hz and the sound intensity signal (dashed line) is given in dB SIL (re. 1pW)/√Hz.

For all frequencies sound levels well below the threshold of hearing can be measured with this ½" p-**u** sound intensity probe, see also Fig. 2.9.

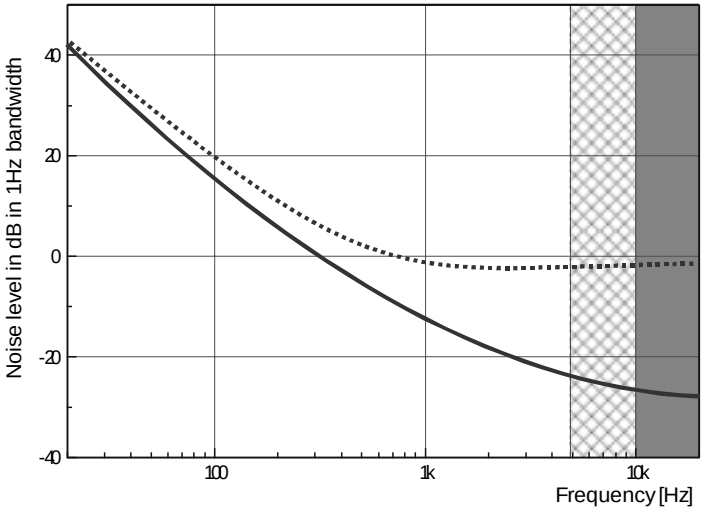

Fig. 83: Measured noise levels of a half-inch p-p probe. The noise levels of the sound pressure microphones (dotted line) are given in dB SPL (re. 20µPa)/√Hz, the sound intensity signal (solid line) is given in dB SIL (re. 1pW)/√Hz. From frequencies above 5kHz ¼" microphones should be used so the intensity determination such as displayed here will not valid anymore. For frequencies above 10kHz no intensity measurement can be performed anymore.

References

[25] W.F. Druyvesteyn et al., A New acoustic measurement probe; the Microflown, IOA London, 1999.

[26] F. Jacobsen, Sound intensity measurement at low levels, Journal of Sound and Vibration, 1993

[27] Brüel & Kjær, Sound Intensity, Technical Review, No. 3, 1982.

[28] Brüel & Kjær, Product Data, Sound Intensity Microphone Pairs - Types 4178 4181, 1995.

[29] Master Catalogue, Electronic Instruments, Brüel & Kjær, may 1989.

[30] G.S.K. Wong, T.F.W. Embleton, AIP Handbook of condenser microphones, Theory, calibration, and measurements, 1994, New York.

[31] O.H. Bjor, Proc. Internoise 97, Budapest pp. 1367.

[32] R. Raangs et al., A low cost sound intensity probe, AES 2001 Amsterdam.

B5 INFLUENCE OF WIND

This chapter is based on the work that has been described in [1]. In the anechoic room the intensity was determined using one sound source (loudspeaker) at a distance of 1.2m and a fan at a distance of 1.0m from the intensity probe; the loudspeaker and the fan were at opposite sides of the probe. When the fan was switched on the wind velocity at the intensity probe was about 3.4 m/sec. First the intensities were measured with a p-p probe and a p-**u** probe with the fan switched off. Then the same measurement was repeated with the fan switched on.

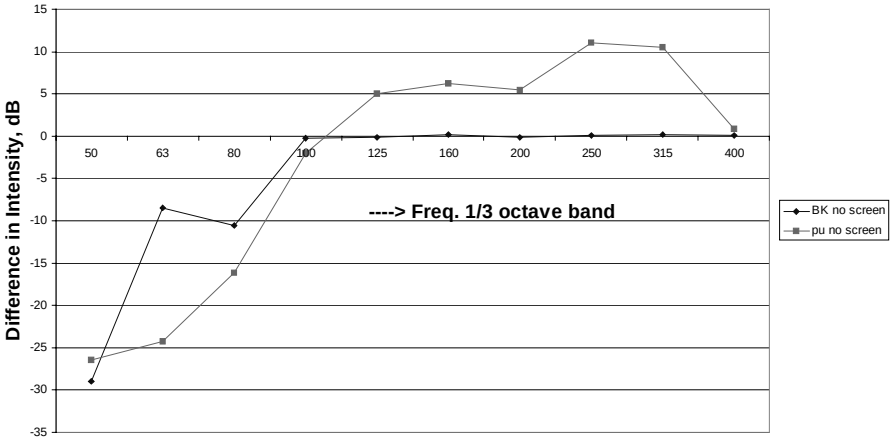

Fig. 84: Influence of wind, no windscreen.

Some experimental results are shown in Fig. 84 and Fig. 85. The abbreviations in the figure refer to the p-p probe (BK) and the p-**u** probe (pu). Horizontally the frequency in 1/3-octave bands is plotted, vertically the measured difference in intensity when the fan is switched on or off. Fig. 84 shows a strong influence of the wind caused by the fan; this influence is stronger for the p-**u** probe and is not acceptable. In order to diminish this influence a windscreen, as delivered by Bruel & Kjaer with the p-p probe is used; see also [1]. In Fig. 85 similar results as in Fig. 84

are shown, but now with the windscreen around the p-p or p-**u** probe. This figure shows that the influence of wind is still stronger for the p-**u** probe than for the p-p probe, but that the effects are for frequencies above 100Hz smaller than 1dB and thus acceptable.

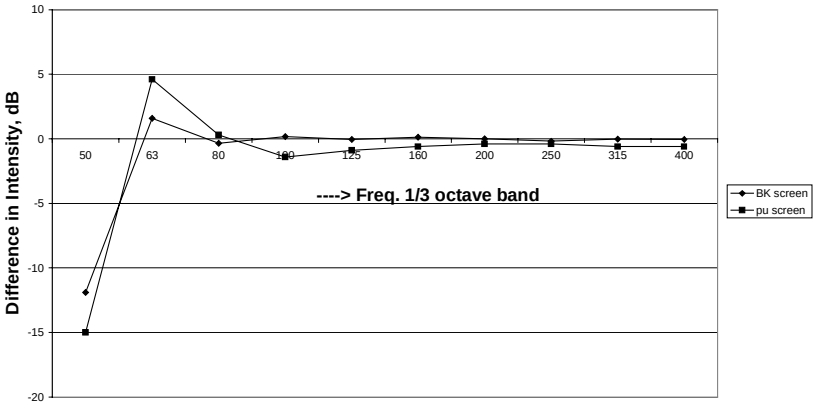

Fig. 85: Influence of wind, including windscreen.

References

[1] W.F. Druyvesteyn and H-E. de Bree, A new sound intensity probe; comparison to the Bruel & Kjaer p-p probe, J. audio Eng. Soc., Vol. 48, 2000 No. 1/2 January/February.

B6 THE U-U SOUND MEASUREMENT

As stated in chapter 2, it is possible to calculate sound pressure with the use of two closely spaced particle velocity microphones. If the sound pressure is calculated and particle velocity is measured sound intensity can be determined. In this chapter theory, realisation and measurements are presented.

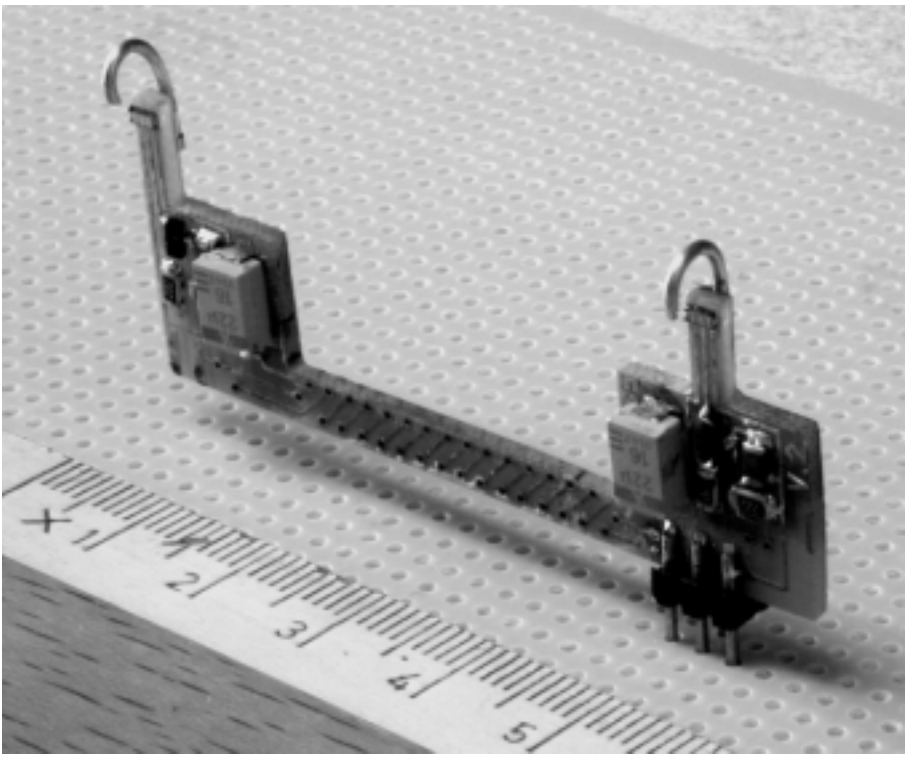

Fig. 86: First realisation of the u-u probe (1996). The spacing between the Microflowns is five centimetres. Cantilever Microflowns are located just below the metal protection wires.

Introduction

This chapter will show the possibility of using two particle velocity microphones (Microflowns) to determine sound pressure. The sound pressure that is derived from two closely spaced Microflowns is directional, which is remarkable since sound pressure is a scalar.

An acoustic wave is described by sound pressure and particle velocity. With the use of two closely spaced Microflowns it is possible to measure the particle velocity gradient, from which sound pressure can be derived.

One possible application is the determination of sound intensity. The (time-integrated) product of the sound pressure and the particle velocity defines sound intensity. Traditionally in sound intensity measurements, one had to measure the particle velocity by using two pressure microphones, because particle velocity microphones did not exist. This method requires two exactly identical pressure microphones, otherwise measuring errors become unacceptable large. A disadvantage is that for different frequency bands a different spacing is required; to measure the complete frequency band the spacing should be altered and the probe re-calibrated three times. This leads to longer measurement times.

When the Microflown is used it becomes a lot easier to measure sound intensity. If a Microflown is combined with a conventional pressure microphone, the sound intensity is easily determined by multiplying both output signals (this is called the "direct or **pu** method", see chapter B1 *"A comparison of the p-u method with the p-p method"*).

A method using two Microflowns to measure sound pressure and sound intensity is treated here, it is called the **u-u** method. At first sight this method has the same disadvantages as the p-p method. However, since the Microflown is made in a batch process it easy to create identical sensors and phase mismatch problems are reduced. When multiple Microflowns are used in one line, altering the spacing is not necessary: one can simply switch Microflowns and thus spacing electronically. Possible difficulty could be the calibration, no effort is spent to solve this.

One fundamental difference of the **u-u** method compared to the p-p method is that the polar pattern of the p-p determined sound intensity is a "figure of eight" (a $\cos\theta$ function). The polar pattern of the sound intensity that is determined using the **u-u** method determined has more directivity, a $\cos^3\theta$ function. This is due to the directivity of the sound pressure determination of the **u-u** method, see paragraph 1.5 "discussion".

Theory

The sound pressure (or just pressure) can be seen as a temporary increase and decrease of the number of particles in a certain small volume at a fixed temperature. When the number of particles increases in this volume, particles have to enter this volume and there has to be movement of particles. This movement is associated with the particle velocity.

The sound intensity in a specified direction, *r*, at a point is the average rate at which sound energy is transmitted through a unit area perpendicular to the specified direction at the point under consideration:

$$I_r = \frac{1}{T}\int_0^T u_r(t)\cdot p(t)dt \ \ [Wm^{-2}] \tag{148}$$

Where $u_r(t)$ represents the particle velocity [m/s] in the specified direction $\mathbf{e_r}$, $p(t)$ [Pa] the sound pressure and T [s] the integration time.

The specific acoustic impedance is defined by ratio of pressure and particle velocity. This impedance is normally a complex value varying in place, indicating that the pressure and velocity are not in phase. The pressure can therefore not be determined by simply multiplying the velocity with a constant value (like for instance the free field impedance).

To obtain the sound pressure, a method in which two closely spaced Microflowns is applied. This method is based upon the linearised equation of mass conservation, see also chapter 2:

$$\frac{\partial\rho(t)}{\partial t} + \rho_0 \cdot \frac{\partial\big(u(t,r)\big)}{\partial r} = 0 \tag{149}$$

Where ρ_0 is the intrinsic density of the medium and c the undisturbed velocity of sound . Some calculations will lead to (see also chapter 2):

$$p(t,r) = -\rho_0 \cdot c^2 \cdot \int \frac{\partial\big(u(t,r)\big)}{\partial r} dt \tag{150}$$

In practice, the particle velocity gradient, $\partial(u(t,r))/\partial r$, can be approximated by measuring the velocities, u_a and u_b, at two closely spaced points, with a separation of Δr. Dividing u_a-u_b, by the distance Δr gives the following estimation for the pressure $\hat{p}(t,R)$ at position R:

$$\hat{p}(t,R) = -\frac{\rho_0 \cdot c^2}{\Delta r} \cdot \int (u_b\big(t, R - \tfrac{1}{2}\Delta r\big) - u_a\big(t, R + \tfrac{1}{2}\Delta r\big))dt \tag{151}$$

This approximation is valid as long as the separation (Δr) is small compared to the wavelength of interest. The particle velocity component of the sound wave is estimated by the sum of both Microflowns divided by two. The sound intensity can be determined by using the block diagram that is depicted in Fig. 87.

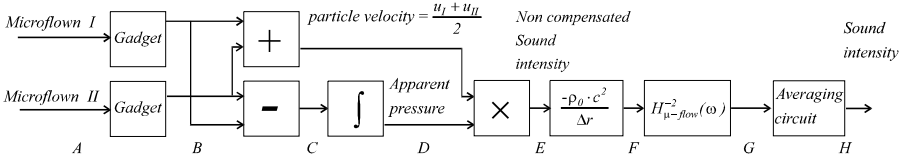

Fig. 87: Block diagram to measure the sound intensity by the use of two particle velocity probes.

Measurements

The Gadget is used as preamplifier, see chapter 4. The transfer function of the Microflown, $H_{\mu\text{-}flown}(\omega)$, is depending on the frequency and shows a low pass characteristic. Only linear operations are performed on the signals of both Microflowns and these have been fabricated in the same batch, which suggests the sensors to be practically identical. Therefore the compensation for this behaviour can be performed after multiplying particle velocity and apparent pressure (point E in Fig. 87). When subtracting (point B..C), the signal is very sensitive to a phase mismatch. By shifting the compensation networks, errors can be avoided.

For the measurement of sound pressure and particle velocity a standing wave tube which is driven by a loudspeaker and is closed rigidly at the end is used. A standing wave is realised, whereby the phase shift between sound pressure and particle velocity is theoretically 90°. The measurement set-up is illustrated in Fig. 88.

u-u probe

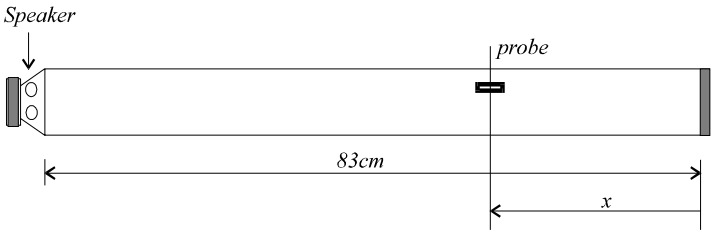

Fig. 88: Standing wave tube.

In the rigidly terminated tube the RMS value of the minima and maximums were measured real time with a pressure microphone and with two closely spaced Microflowns (see Fig. 86).

Now observe point D in Fig. 87: the sum signal of the Microflowns represents the averaged particle velocity and the integrated differential signal of the two Microflowns the sound pressure. The measured results in a standing wave are illustrated in Fig. 89.

As can be seen, at 550 Hz the minima in the pressure are separated by 30 centimetres. The minima in the particle velocity are shifted by ¼ wavelength, which is 15 cm.

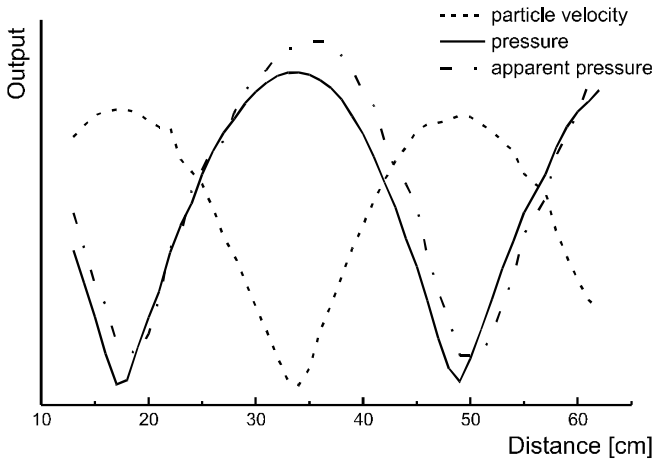

Fig. 89: The minima and maximums in a closed tube of the particle velocity, pressure and apparent pressure.

In an environment without reflections (anechoic conditions), the sound intensity has been measured. The results can be seen in Fig. 90, as expected the sound pressure and the particle velocity do have a phase shift of zero degrees (point D in Fig. 87). The sound intensity before averaging is shown in the lower part of the graph with a solid line. The result of the direct method is also shown (dashed line).

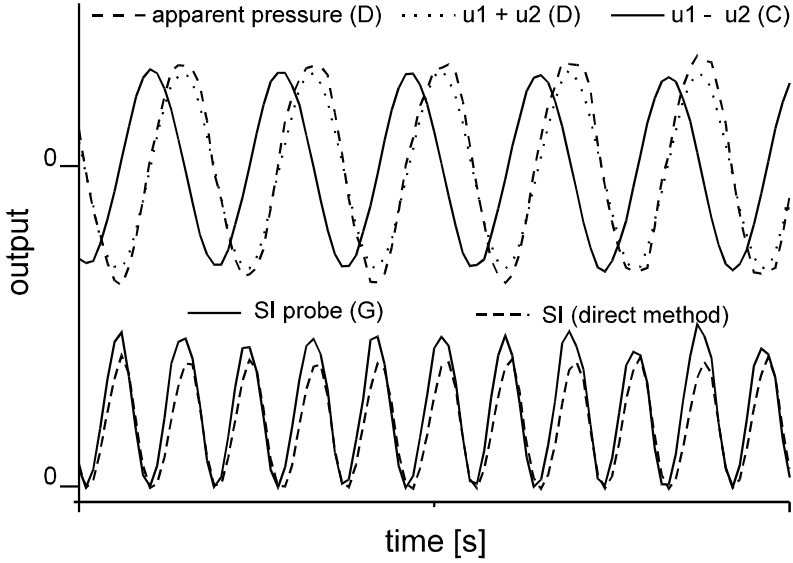

Fig. 90: Instantaneous sound intensity in a tube without reflections. The phase shift between sound pressure and particle velocity is zero degrees. In the lower part of the figure the sound intensity measured with two closely spaced particle velocity probes is depicted by a solid line and the sound intensity measured with the use of a sound pressure probe and a velocity probe is depicted by a dashed line.

Discussion

It is shown that it is possible to measure the apparent pressure and sound intensity with two Microflowns in one direction, it is called the **u-u** method. The method doesn't differ much from the p-p method. Still there is an important difference: the p-p method is based on the linearised equation of momentum conservation. The sound pressure is a scalar

quantity and the pressure gradient determines the particle velocity component in one direction.

The **u-u** method uses the linearised equation of mass conservation. A three-dimensional extension of Eq. 2.14 is the use of the divergence of the particle velocity to calculate the sound pressure:

$$p = -\rho_0 c^2 \cdot \int \frac{\partial u}{\partial x} + \frac{\partial u}{\partial y} + \frac{\partial u}{\partial z} dt \tag{152}$$

So in fact to calculate sound pressure one should use six Microflowns. However, in practical situations the one-dimensional realisation will be sufficient. For the p-p probe and the p-**u** probe the polar pattern is a "figure of eight". Contradictory, one can deduct that the sound pressure determination of the **u-u** method has a "second order figure of eight" polar pattern ($cos^2(\theta)$ shape). (One way to look at this is that pressure gradient generated by (omni directional) pressure microphones results in a figure of eight response and so a gradient generated by elements that have a figure of eight response for them selves will generate the square of a figure of eight response. Another argument is based on the fact that $\cos^2\theta + sin^2\theta = 1$; $\cos(\theta + 90^\circ) = sin\theta$).

A one-dimensional sound intensity probe (in the direction u_x) is described with:

$$I_x = \frac{1}{T}\int_0^T p \cdot u_x dt \tag{153}$$

If the sound pressure is calculated with three dimensional particle velocity probes the expression alters in:

$$\begin{aligned} I_x &= \frac{1}{T}\int_0^T \left(-\rho_0 c^2 \int \frac{\partial u_x}{\partial x} + \frac{\partial u_y}{\partial y} + \frac{\partial u_x}{\partial z} dt \right) \cdot u_x dt \\ &= \frac{-\rho_0 c^2}{T}\int_0^T \left(\underbrace{\int \frac{\partial u_x}{\partial x} dt}_{p_x} \cdot u_x + \underbrace{\int \frac{\partial u_y}{\partial y} dt}_{p_y} \cdot u_x + \underbrace{\int \frac{\partial u_z}{\partial z} dt}_{p_z} \cdot u_x \right) dt \end{aligned} \tag{154}$$

The sound pressure p, a scalar, is composed of three parts: p_x, p_y, p_z associated with the corresponding particle velocity components in the x, y

and z direction. If an intensity probe is constructed with only particle velocity sensors in the x direction, $\boldsymbol{I'}_x$, it detects:

$$I_x^{'} = \frac{-\rho_0 c^2}{T} \int_0^T \underbrace{\int \frac{\partial u_x}{\partial x} dt}_{p_x} \cdot u_x dt \tag{155}$$

Since $u_x = \cos\theta \left| \mathbf{u} \right|$, where θ represents the angle between the axis of measuring and the propagation vector of the sound wave, and $p_x = p \cos^2\theta$, for the detected is found: $I'_x = I \cos^3\theta$. A one-dimensional u-u sound intensity measurement will result in an under estimation of the off-axis intensity.

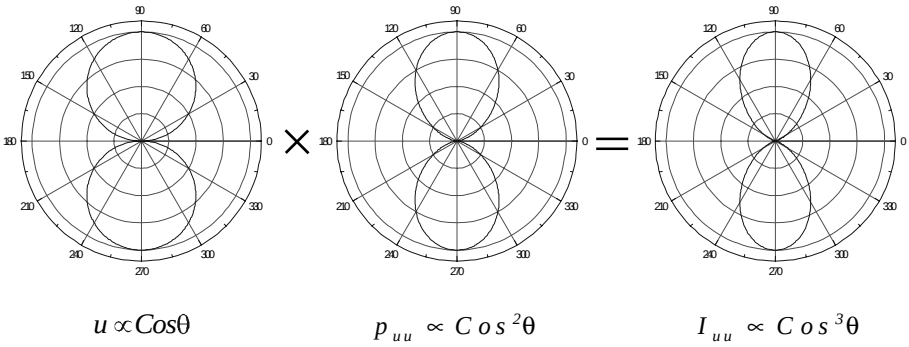

Fig. 91: Polar plot of the particle velocity, calculated pressure and sound intensity of a one-dimensional **u-u** probe.

To be able to measure the apparent pressure and sound intensity in real time at different frequency bandwidths, some electronics have to be implemented. Industrial systems available for sound intensity measurements can easily be remade for the two Microflowns method.

For the pp method, the two pressure microphones are spaces by a spacer. This is a solid tube between the two probes. This spacer serves two purposes: it is easy to create a correct spacing. More important is that it solves the following problem. The position where a sound probe seems to observe sound pressure is not exactly at the membrane but a little bit in front of it. This position is frequency dependent. The spacer keeps the

physical location of the sound probes and position where it seems to be perceived at the same position.

This is not possible for the **u-u** probe. If this is a problem is not clear, this phenomena is not yet studied.

A complete other way of performing a free field measurement in a diffuse field is by the use of two Microflowns that are positioned in perpendicular to each other:

The $\mathbf{u}_{//}$-$\mathbf{u}_{\perp}$ Probe; free-field measurements in a reverberant room.

In chapter 2 it was shown that the directional characteristics (the polar pattern) of a Microflown behaves as u_x=cosθ. This means that if the Microflown sensor is oriented in a plane perpendicular to the direction of the direct-particle velocity, only the contribution of the diffuse sound field is measured. If the second Microflown sensor is positioned in the direction of the direct field (i.e. the free field) then this sensor measures the direct- and the reverberant sound field. It is therefore a sort of sound intensity measurement.

In the anechoic- and the reverberant room the following measurements were performed:

- The sound source (a loudspeaker) is positioned on the x-axis at a distance of 1 meter from the $\mathbf{u}_{//}$-$\mathbf{u}_{\perp}$ probe.
- One probe ($\mathbf{u}_{//}$) is positioned such that $\theta=0^0$ (the resistance wires of the probe are parallel to the y-axis and in the x-y plane).
- For the other probe ($\mathbf{u}_{\perp}$) $\theta=90^0$ (the resistance wires are in the y-z plane).
- Using the same loudspeaker excitation the rms signals from the $\mathbf{u}_{//}$ and $\mathbf{u}_{\perp}$ were measured.

One expects to obtain the following results:

- **Anechoic** room: $\mathbf{u}_{//}^2=\mathbf{u}_0^2$; $\mathbf{u}_{\perp}^2 = 0$, i.e. no reverberant field.
- **Reverberation** room: $\mathbf{u}_{//}^2=\mathbf{u}_0^2+1/3u_{rev.}^2$, $\mathbf{u}_{\perp}^2=1/3\mathbf{u}_{rev}^2$, where $\mathbf{u}_0$ means the free field particle velocity (in the anechoic- and reverberation room) and $\mathbf{u}_{rev}$ means the diffuse particle velocity in the reverberation room; the factor ½ is due to the directional characteristics of the Microflown sensor, see section 3 (integral $\cos^2(\varphi) = 1/3$).

Thus:

$$u_0^2 \equiv u_{//\,anechoic}^2 = u_{//\,reverberation}^2 - u_{\perp reverberation}^2$$

The measured deviations are given in Fig. 92; vertically is plotted "Diff. Free":

$$Diff.Free \equiv 10Log\{u_{//\,anechoic}^2\} - 10Log\{u_{//\,reverberation}^2 - u_{\perp reverberation}^2\}$$

The reverberant field is also plotted in Fig. 92 as "Rev."

$$\mathrm{Re}\,v. \equiv 10Log\{u_{//\,reverberation}^2\} - 10Log\{u_{//\,anechoic}^2\} = 10Log\left\{1 + \frac{u_{rev}^2}{2u_o^2}\right\}$$

Fig. 92 shows that with this simple measurement the direct and diffuse sound field in a reverberant room can be determined.

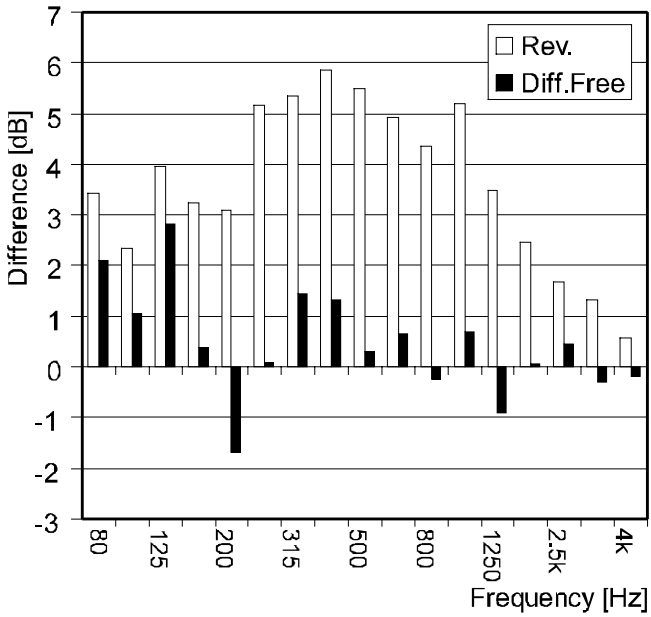

Fig. 92: Difference in free field determined anechoic- and reverberation room, and reverberant field.

u-u probe

The method is under investigation now. More results of this method may be expected in the near future.