A NEW ACOUSTIC MEASUREMENT PROBE; THE MICROFLOWN

W.F. Druyvesteyn, H-E de Bree, M. Elwenspoek
w.f.druyvesteyn@el.utwente.nl; h.e.deBree@el.utwente.nl;
University of Twente, PO Box 217, 7500 AE Enschede, The Netherlands.

0. CONTENTS.

1. INTRODUCTION: WHY A NEW PROBE FOR ACOUSTIC MEASUREMENTS?

Experiments with a new acoustic sensor: **the Microflown**, are described. With this sensor the **particle velocity** of an acoustic disturbance is measured, whereas with standard microphones the sound **pressure** is measured.

In acoustic measurements it is important to determine the direct sound as well as the total sound field. The usual procedure then is to perform the measurements in an anechoic room and to repeat them in a reverberant room. When this is not possible, an alternative method is to measure the sound intensity [1,2]; the commonly used intensity probe consists of two closely spaced identical pressure microphones. From the difference in signal of these two microphones the particle velocity is determined.

A much simpler method is to combine a pressure microphone with a particle velocity sensor: the Microflown. Experiments with this probe are discussed in section 5.1.

When two Microflowns, directed in perpendicular directions are used, simple free field measurements in a reverberant environment can be done; see discussion in section 5.2. For this purpose the directional characteristics of the Microflown are used (see section 3).

Combining one pressure microphone with three Microflowns, directed in the x-, y- and z-direction results in a simple 3-D (3 dimensional) intensity probe (see section 6).

2. 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 8.5x7.2x5.1 m^3, a volume of 227.6 m^3; it is a nonrectangular room. The reverberation time was about 6 (1000 Hz) to 7.8 (250 Hz) 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 20- 5000 Hz. As pressure microphone a ½" Bruel & Kjaer, type 4134, was used, various

Microflowns were used. Most of the Microflowns were encapsulated such that the external configurations are the same as the configurations of the pressure microphone. 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.

3. DESCRIPTION AND PROPERTIES OF THE MICROFLOWN SENSOR.

The Microflown [3] consists of two cantilevers of silicon nitride with an electrically conducting platinum pattern on top of them, see Figure 1 a SEM photo. The size of the cantilevers is 800×40×1 μm^3 (l×w×h). The metal pattern is used as temperature sensor and heater. The silicon nitride layer is used as a carrier for the platinum resistor patterns.

In the fabrication process first a silicon nitride surface layer with a thickness of 1 μm is grown by low-pressure chemical vapour deposition on a silicon substrate, see Figure 1 (a). Then a platinum (0.15 μm) metal layer is sputtered and patterned (Figure 1b). After this the silicon nitride is patterned (plasma etching) to create the cantilever structures (Figure 1c). Anisotropic wet etching will free the sensors. It is possible to make a wafer of more than a thousand Microflowns with exactly the same mechanical and acoustic properties at the same time.

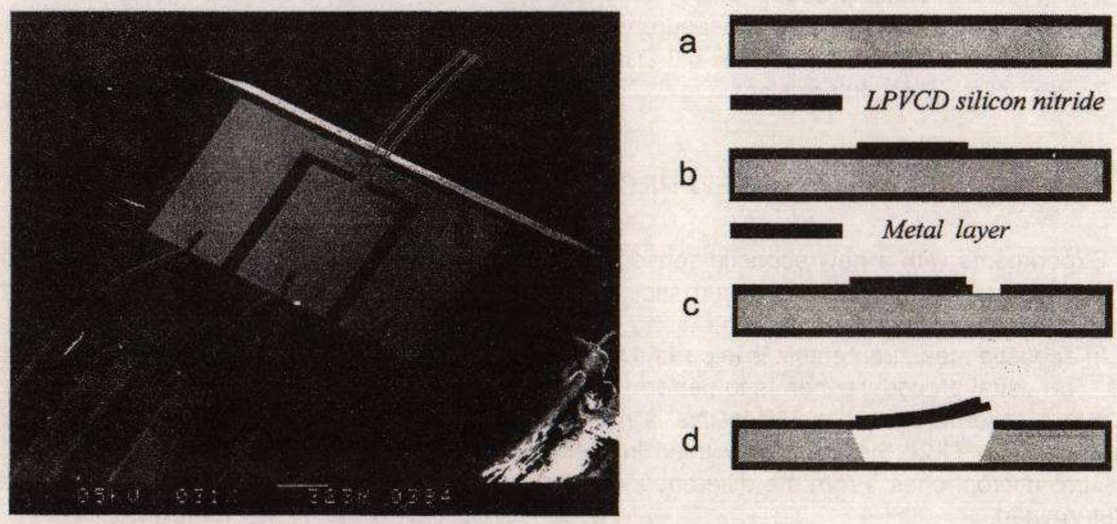
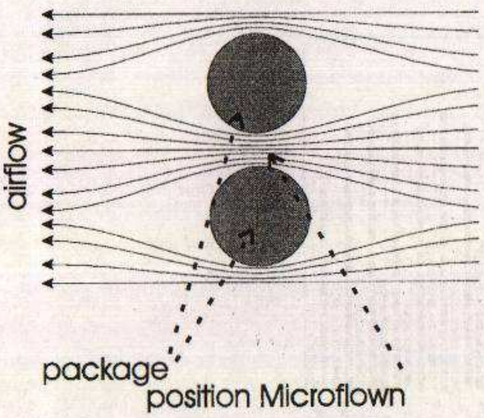


Figure 1 (l): A SEM photograph of a Microflown (r): Production steps.

To protect the fragile sensors the Microflown is encapsulated. The Microflown shown in Figure 6 is encapsulated in a similar form as a ½" Bruel & Kjaer pressure microphone. An attractive aspect of the package as shown in Figure 6, is that the particle velocity level increases when a well-chosen obstacle is near the sensor, see Figure 2.

The sensors are powered by an electrical current, causing them to heat up to an operational temperature of about 200^{0}C to 300^{0}C. If the temperature of the sensor changes the resistance will also change. When particle velocity is present a convective heat transfer of both sensors will cause a temperature drop of both sensors. The upstream sensor, however, will drop more in temperature than the downstream since the downstream sensor is heated by the upstream convective heat loss. The temperature changes of the two sensors are thus different and the resistance changes also. When the differential resistance changes are measured a linear particle velocity sensor is obtained. An example of a time windowed measurement of a plane acoustic wave (in a long tube) is shown in Figure 3.



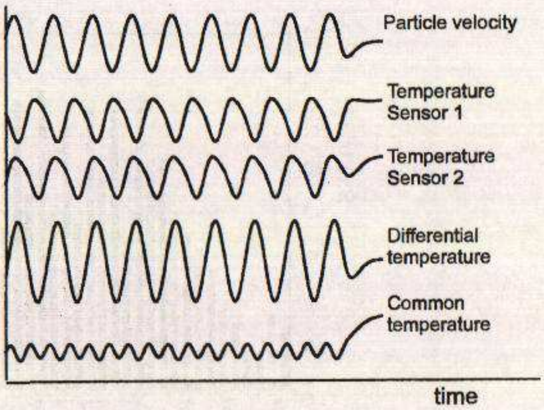


Figure 2 (l): An obstacle will result in a particle velocity gain. Figure 3 (r): Time windowed measurement of the temperatures of the sensors of the Microflown.

4. SINGLE CHANNEL MEASUREMENTS; COMPARISON WITH PRESSURE MICROPHONE.

To make a good comparison between a pressure microphone and a particle velocity sensor is complicated because the physical behaviour and the corresponding electronics of the two sensors are quite different. Therefore some general rough comparisons will be made in this section. In an anechoic- and a reverberant room measurements were performed using a loudspeaker as sound source. In table 1 results are shown for a ½" Bruel & Kjaer pressure microphone (type 4134) and an example of a Microflown.

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	S/N, p	S/N, u	p.u,signal	p.u,noise	S/N, p.u
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.

Similar values are given in column 4,5 and 7 for the Microflown sensor. Values given in column 8, 9 and 10 are discussed in section 5.1. In Figure 4 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 latter is discussed in section 5.1.

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.

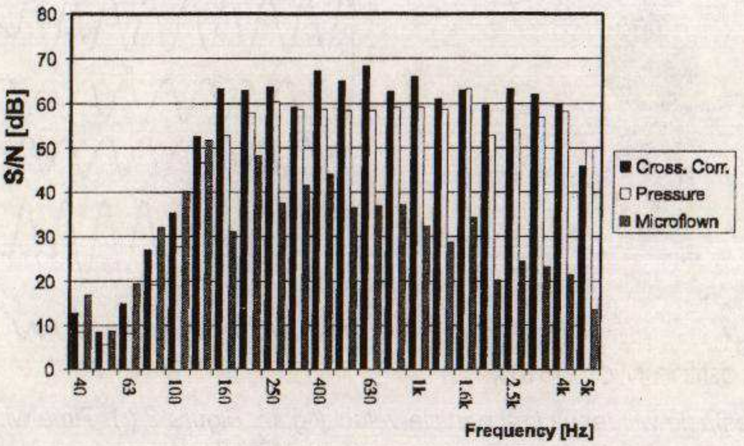


Figure 4: S/N ratio's of the pressure microphone, Microflown and cross-correlation of both.

An important difference between a pressure microphone and a Microflown is the directional characteristic. Independent of frequency the Microflown shows a cos(φ) dependence; the angle φ is defined as follows. Define a vector **n**, perpendicular to the length of the resistance wires of the Microflown sensor and in the plane of these wires; the angle between the particle velocity vector **u** and the vector **n** is defined as φ. Experimental results are shown in figure 8; plotted vertically is the measured signal from the sensors. The curves refer to a cos(φ) dependence, or a |cos(φ)| dependence; the points refer to the measured values for the pressure microphone (▲), Microflown sensor (●) and the intensity (■), which will be discussed in section 5.1. The measurements refer to the 1/3 octave band of 400 Hz.

The Microflown sensor shows a |cos(φ)| dependence. When a dual channel measurement is performed and also the phase difference between for instance the pressure microphone and the Microflown sensor is determined a cos(φ) dependence can be obtained.

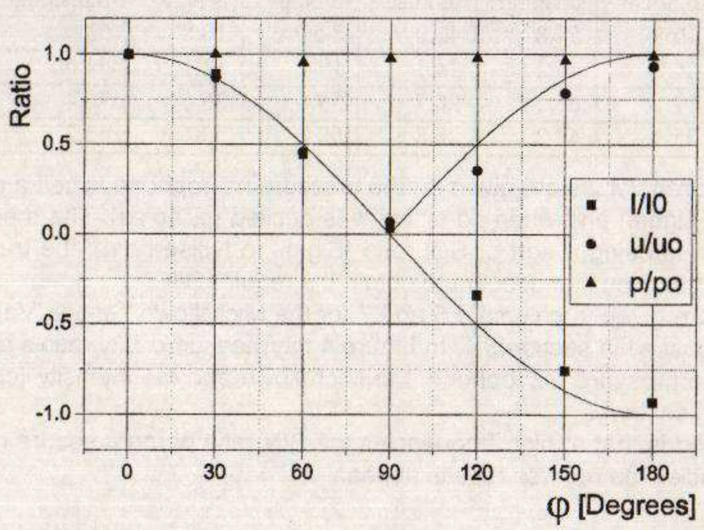


Figure 5. Directional characteristics of a pressure microphone, Microflown and the intensity.

5. DUAL CHANNEL MEASUREMENTS.

5.1 P-U INTENSITY PROBE.

The intensity is defined as the time averaged product of the instantaneous pressure, p(t) and the corresponding instantaneous particle velocity, **u**(t) at the same position:

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

Where the intensity **I** and the velocity **u** are vectors [1].

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 standard intensity probe consists of two closely spaced identical pressure microphones, which are placed "face to face"; different spacers between the microphones have to be used for different frequency ranges. 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. The intensity can be calculated from the **imaginary** part of the cross-spectrum between the two microphone signals, using a dual-channel analyser [2].

A more direct approach would be to combine a pressure microphone with a particle velocity sensor. This has been done successfully and reported at the 104th AES Convention 1998 [4]. Figure 6 shows an example of a used p-**u** intensity probe consisting of a ½" Bruel & Kjaer microphone and a particle velocity sensor.

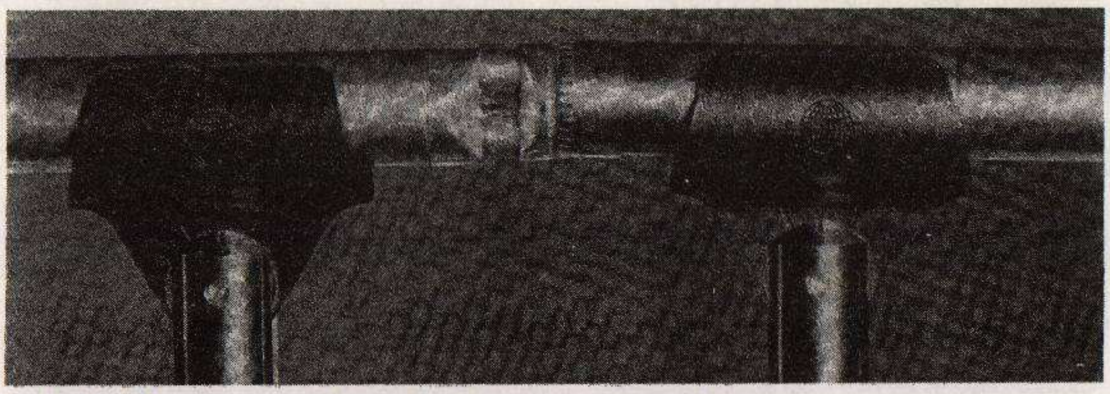


Figure 6: Left the Microflown sensor right the ½" pressure microphone.

Due to the quite different operation of the two sensors an (extra) phase difference occurs. 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 this experiment a phase difference α_0 between the p- and **u** sensor was found, then the intensity measured in another experiment is found from: $\mathbf{I} \propto \cos(\alpha_0).\ \mathrm{Re}(G_{AB}) + \sin(\alpha_0).\ \mathrm{Im}(G_{AB})$; G_{AB} being the cross-spectrum between the microphone- and the Microflown signal [4].

Critical experiments have been performed in the anechoic room as well as in the reverberant room using a p-p intensity probe and a p-**u** intensity probe. It appears that the p-**u** probe is as good as the used p-p probe [4].

In a later report [5] also the influence of wind on the performance of the p-p and p-**u** probe has been investigated. It appears that using a windscreen for both probes the influence of wind is weak.

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.

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 [6].

The S/N ratio of the product of $p_1(t).p_2(t)$ ($p_1(t)$ and $p_2(t)$ are the two microphones signals) is higher than the S/N ratio's of the separate microphones [6]. 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 Figure 4 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.

5.2 $u_{//}$- $u_\perp$ PROBE; FREE-FIELD MEASUREMENTS IN A REVERBERANT ROOM.

In section 4 it was shown that the directional characteristics of a Microflown behaves as cos(φ). 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.

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

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

One expects to obtain the following results:

- **Anechoic** room: $u_{//}^2 = u_0^2$, $u_\perp^2 = 0$, i.e. no reverberant field.
- **Reverberation** room: $u_{//}^2 = u_0^2 + ½.u_{rev.}^2$, $u_\perp^2 = ½.u_{rev}^2$, where u_0 = free field particle velocity (in the anechoic- and reverberation room) and u_{rev} = diffuse particle velocity in the reverberation room; the factor ½ comes from the directional characteristics of the Microflown sensor, see section 3 (integral $\cos^2(\varphi) = ½$).

Thus:

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

The measured deviations are given in Figure 7; 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 Figure 7 as "Rev."

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

Figure 7 shows that with this simple measurement the direct- and diffuse sound field in a reverberant room can be determined.

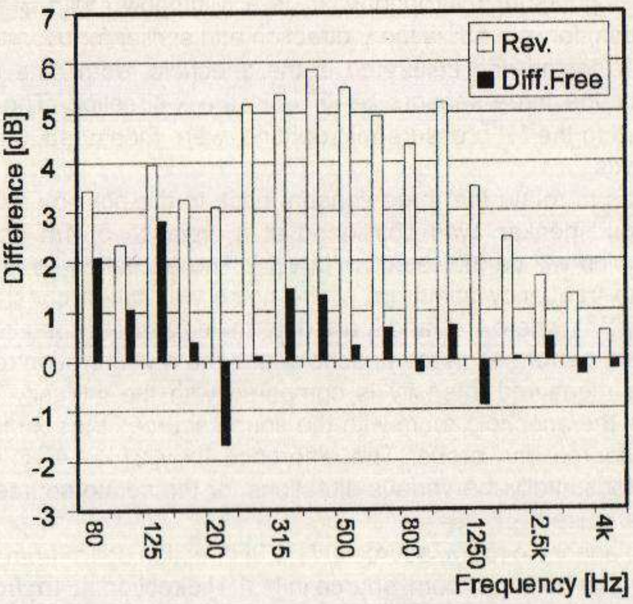


Figure 7: Difference in free field determined anechoic- and reverberation room, and reverberant field.

In conclusion, the section on dual channel measurements: using a combination of a p- and a **u**-sensor; or two **u**-sensors is always advantageous above using just one acoustic sensor, i.e.:

- free field- and diffuse field sound measurements can be performed,
- the S/N ratio is higher.

6. MULTI-CHANNEL MEASUREMENTS; A 3-D INTENSITY PROBE.

Since the one dimensional p-**u** intensity probe is relative simple to realise, as compared to the standard p-p probe, the next step is to make a **three dimensional** intensity probe, consisting of one pressure microphone and three Microflowns. Figure 8 shows the configuration of such an intensity probe. In Figure 8 (r) x, y and z axis are shown.

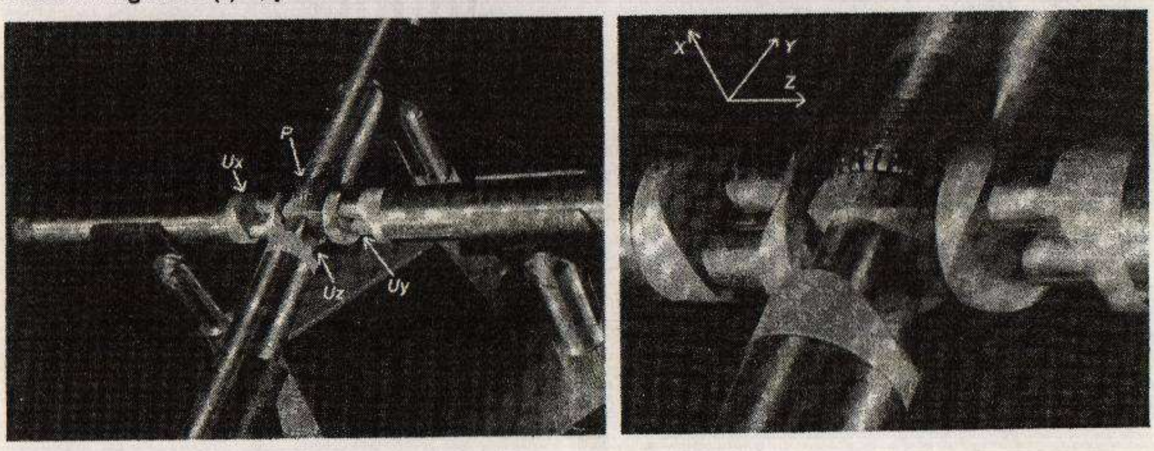


Figure 8: Photo 3-D probe, left overview; right detail.

p= ½" Bruel & Kjaer pressure microphone; u_x is a Microflown with $\varphi = 0^0$ (see section 4 for definition) in the x-direction; for u_y $\varphi = 0^0$ is the y-direction and similar for u_z.

First the sensors are tested and calibrated in the anechoic room. The u_y and u_z sensors are rotated 90^0 such that for the three sensors $\varphi = 0^0$ is 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 sketched in Figure 8. The sound source, a loudspeaker, was positioned at a distance of 1m from the probe; various directions were used, which will be indicated as $[x_i,y_i,z_i]$. The excitation i.e. the current through the loudspeaker coil and the frequency spectrum of the noise was taken constant. The intensity was calculated as $(I_x^2 + I_y^2 + I_z^2)^{1/2}$, where I_x, I_y and I_z are determined as described in section 5.1.

The experiments were performed in the anechoic- and the reverberation room. Some results are shown in Figure 9. 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], in dB, is plotted vertically in the figure. 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* refer to the case where the distance source → probe in the reverberation room was diminished from 1 m to 0.55 m, thus:
- R*,[101] means: reverberation room, source in [1,0,1] direction at **0.55** m from the probe.

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 5-6 dB.

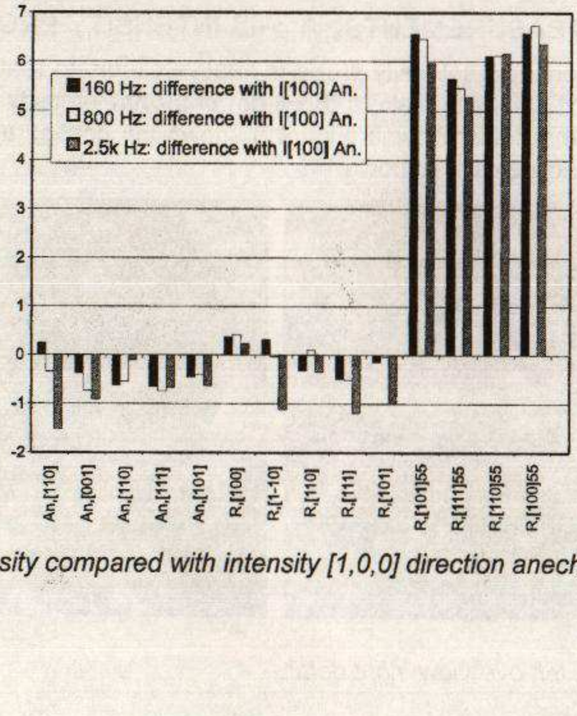


Figure 9: Measured intensity compared with intensity [1,0,0] direction anechoic room.

7. CONCLUSIONS AND FUTURE PLANS.

The experiments show that the particle velocity sensor is a good addition to the existing acoustic measurement devices. Although at low frequencies the S/N ratio is high, the S/N ratio at higher frequencies should be improved in the future.

A combination of a Microflown sensor with a pressure microphone has a number of advantages. A simple (cheap) intensity probe is obtained and the S/N ratio can be higher than the individual S/N ratio's of the two sensors. Also the distance between the two (or more) sensors can be made much smaller than for the case of a p-p intensity probe.

A compact acoustic measurement device consisting of one or more Microflown sensors in combination with a pressure microphone, all in one housing, seems also to be feasible.

In comparison to a 3-D p-p intensity probe [1, pp. 113] the 3-D intensity probe consisting of one pressure microphone and three Microflown sensors is much simpler to manufacture. In the future it seems worthwhile to continue the development on multichannel sensors, as well as a combination of a pressure microphone with Microflown sensors, as well as on arrays of Microflown sensors.

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DEVELOPMENT OF A MEASUREMENT SYSTEM FOR VERY LOW SOUND PRESSURE LEVELS

O-H. Bjor Norsonic AS, P.O.Box 24, N-3421 Lierskogen, NORWAY
G. B. Ese Norsonic AS, P.O.Box 24, N-3421 Lierskogen, NORWAY

Measurement of extremely low sound levels is important for many applications. It may be used to describe the acoustic environment in recording studios, auditory rooms, concert halls and sound-insulated rooms. It may also be used for measurement of noise emissions from quiet machinery and equipment such as lighting-armature.

Sound levels are normally measured by the use of a microphone in connection with a sound level meter or a sound analyser. The lowest sound pressure level that can be measured with this configuration is limited by the inherent noise of the microphone, preamplifier and the instrument. In order to have a transducer which is omnidirectional over the complete frequency range of interest, the size of the transducer has to be limited. A measuring microphone with a diameter of 13 to 25 mm is the normal choice.

Most high-quality sound level meters and analysers with normal microphones have an A-weighted noise floor in the range 12–20 dB. At the cost of less environmental stability, a microphone system with an A-weighted noise floor just below 0 dB has been constructed [2]. The inherent noise of the measuring systems adds to the measured level and reduces the level linearity for levels close to the noise floor. Even after correction for this, the lowest level that can be measured with satisfying linearity is 2–5 dB above the noise floor.

The inherent noise limiting the lower end of the measuring range is normally the random noise in the electronic circuitry and in the resistive parts of the impedances in the transducer itself. This means that if the same acoustic field is measured with another but identical measurement system it will be limited by an identical noise process but the noise signals will be statistically independent and uncorrelated. The real signal from the sound field will, however, be similar and thus fully correlated in the two microphone systems. By adding the output from two such microphones together a 3 dB increase in the signal to noise ratio may be obtained. This process may be continued: Each doubling of the number of microphones brings the resulting noise 3 dB further down. However, in order to reduce the noise level significantly, we will soon end up with an impractical large number of microphones.

For most applications we are not interested in the sound signal itself, just the weighted sound pressure level. In this case we can obtain much better performance by multiplying the signals from two microphones.

Let s_1 and s_2 be the output from two identical microphone systems. In addition to the signal p from the acoustic field, the output signals are assumed to be contaminated by two unwanted components n and e. The unwanted component is divided so that n is the uncorrelated part and e the fully correlated part of the two inherent noise components. All signals are functions of the time t:

$$s_1(t) = p_1(t) + e_1(t) + n_1(t)$$

$$s_2(t) = p_2(t) + e_2(t) + n_2(t)$$

It is assumed that the signals are spectrally weighted in a convenient way like A-weighted or filtered in a bandlimiting filter like an 1/3 octave filter.

If the microphones are placed very close together, like the face-to-face arrangement known from intensity probes, the acoustic signals $p_1(t)$ and $p_2(t)$ may be considered equal and called $p(t)$.

If the signals are multiplied together the expected product will be:

$$E\{s_1(t)s_2(t)\} = p^2(t) + E_0^2$$

where

$$E_0^2 = E\{e_1(t)e_2(t)\}$$

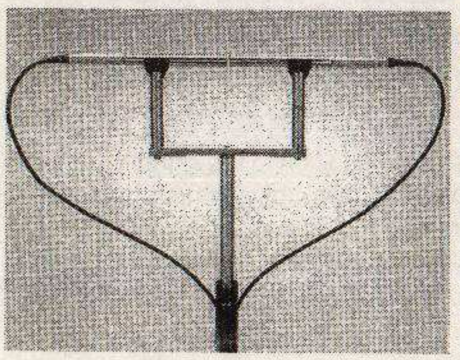
The components $n_1(t)$ and $n_2(t)$ will not contribute because they are uncorrelated. Normally the correlated part of the inherent noise E_0 is very low and will typically consist of parts from common sources like hum components or signals demodulated from unwanted electromagnetic fields.

For practical measurements the expected values of the level L are approximated by taking the time average over the measurement period T:

$$L = 10\log\left[\frac{1}{Tp_0^2}\int_0^T s_1(t)s_2(t)dt\right] = 10\log\left\{\frac{1}{T}\int_0^T\left[\left(\frac{p(t)}{p_0}\right)^2 + \left(\frac{E_0}{p_0}\right)^2\right]dt\right\}$$

where p_0 is the reference sound pressure for level calculations.

The diaphragm of the microphones should be brought close together relative to the wavelength of the highest frequency of interest in order to maintain the omnidirectivity for the microphones. This may be obtained by side-by-side configuration of the two microphones, but in general the best performance is obtained by a face-to face configuration as known from sound intensity instrumentation [4]. The preamplifiers and mounting systems of a sound intensity probe forms a very convenient microphone system. However, the intensity microphones should normally be replaced by more sensitive microphones in order to obtain the lowest inherent noise. Even though the phase response matters for this type of measurement as well, a normal pair of microphones of identical type is sufficiently phase matched for this application. The sensitivity will only be reduced by the square-root of the cosine to the phase difference.



For some applications it may be convenient to increase the spacing between the microphones. This is done to reduce noise from wind and other airflow. In this application it will be necessary to move the microphones further apart in order to reduce the correlation between the noise sources. However, this cannot be done without disturbing the directional response of the microphone system.

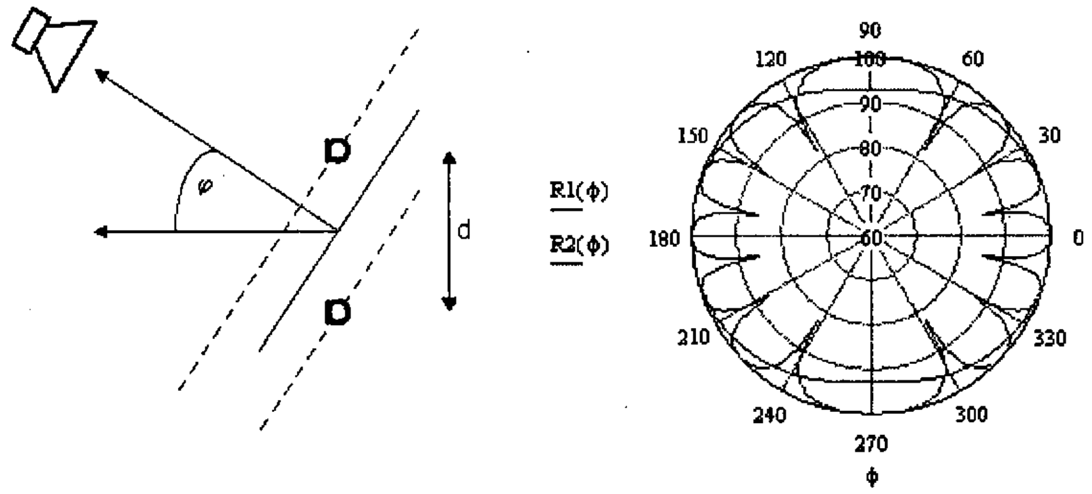
Development of a Measurement System for Very Low Sound Pressure Levels

O-H. Bjor and G. B. Ese Norsonic AS, P.O.Box 24, N-3421 Lierskogen, NORWAY

Let d be the distance between the acoustic centres of the microphones. If ϕ is the angle between the direction of the sound field - assuming plane waves - and the plane perpendicular to the line between the microphones, the directional response in dB will be:

$$L = 10\log[|\cos(k \cdot d \cdot \sin(\phi))|]$$

where k is the wavenumber $2\pi/\lambda$. The figure below shows the arrangement and obtained directional response for d = 0,5 m for the frequencies 150 Hz and 1000 Hz.



Available standard measurement equipment where this sort of signal processing is at hand may be used for the sound level measurement. The figure on next page shows a setup based on two pressure microphones and the Norsonic real time analyser *Nor-840* and the signal processing in the form of a block diagram. The standard software delivered with the analyser for sound intensity measurement is applied in the pressure/particle velocity mode (pu-mode). Normally this mode connects channel 1 to the pressure transducer and channel 2 to the particle velocity transducer. For the described application, however, the other pressure microphone is connected to channel 2. Due to the difference in reference level for a velocity and a pressure transducer, channel 2 must be calibrated with an offset of 52,04 dB to obtain correct readings. The intensity display is used for the read-out.

The preamplifiers from a p-p sound intensity probe may be used for the measurement. For some applications crosstalk from the control lines of the remote control (wherever applicable) may be observed when using this method. The crosstalk is normally well below the inherent noise, but as the noise is removed by averaging, small components normally buried in the noise may be observed. It may be a good practice to check the system by substituting dummy-microphones for the normal microphones. The dummy-microphone has the capacitance of a microphone but no or very low acoustic sensitivity. By this method the acoustic signals are removed and the size of the correlated signals in the two channels may be investigated. It should be noted that the inherent noise in a dummy-microphone is far less than the inherent noise in a real microphone.

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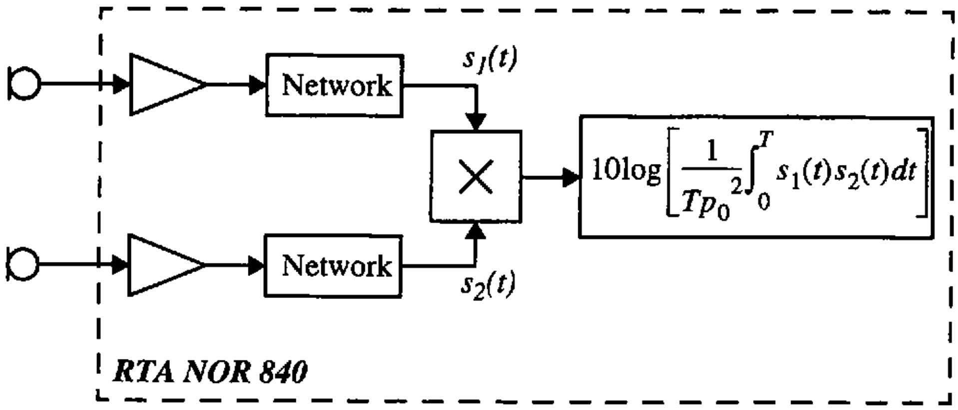
O-H. Bjor and G. B. Ese Norsonic AS, P.O.Box 24, N-3421 Lierskogen, NORWAY



An experimental setup was done with Norsonic real-time analyser 840, two normal preamplifiers Norsonic 1201 supplied with two 1/2" freefield microphones GRAS 40AF. The microphones have a sensitivity of approximately 50 mV per pascal. The normal A-weighted noise floor for these microphones in combination with the analyser was measured to be 15 dB. Nevertheless, by applying the two-microphone method the background A-weighted noise in an anechoic chamber was measured to be –8 dB. This corresponds to an improvement of more than 20 dB compared to the inherent noise of each microphone. The level is more than 10 dB lower than the level obtainable with a one-microphone instrumentation when using the microphone with the lowest noise available. The measured A-weighted noise level of –8 dB was measured in a quiet part of the night with the ventilation system switched off. Nevertheless the level was dominated by low frequency components. It is therefore assumed that this level corresponds to the actual background noise rather than forming a limit to the measurement technique. However, this is not very far from the lower limit set by the thermal noise in air.

The measurement may easily be set up based on standard available units and displayed as spectrally weighted levels or as levels in 1/3 octave bands.

References:

1 Møller, P.K.: *Measurement of Background Noise in Sound Insulated Rooms.* Proceedings of NAS-80

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3 Benson, J.W.: *A real-time blast noise detection and wind noise rejection system.* Noise Control Engineering Journal Vol. 44, No6, Nov/Dec 1996.

4 IEC 1043 (1993): *Electroacoustics - Instruments for the measurement of sound intensity - Measurement with pairs of pressure sensing microphones.*