ACOUSTICS

Summary

In this chapter three matters are discussed. What is sound; how can one detect sound and what acoustic measurements can be performed? Definitions of acoustic quantities are given, and on the basis of the wave equation relations are shown between sound pressure and particle velocity. Operation principles of sound pressure, pressure-gradient and particle velocity microphones are presented. And measurements that require both sound pressure as particle velocity like sound intensity, sound energy and acoustic impedance measurements complete this chapter.

What is sound

Very simply, sound can be observed as a propagating vibration in any substance. This substance can be air, water, wood, or any other material: in fact only in vacuum sound can not propagate. In this chapter sound is observed where it is perceived most: in the air.

Sound is produced by a source, for example a firecracker, travels through the air with a certain velocity (which is about 340m/s or 1225km/h) and can be reflected perhaps a few times by a wall or the ground. An ear or microphone detects it. After having perceived the sound of the firecracker one can distinguish its origin. Besides that, it can be said to be a loud bang and perhaps to be an annoying experience.

This annoying experience, or the quantification “loud”, is a subjective interpretation of sound. If large amounts of people are annoyed by a certain distress factor, regulations will follow. In this case sound measurements to quantify the amount of noise should be performed. Questions like “was it really a loud bang or was it somebody who didn’t like firecrackers?” should be answered in a proper way because of the subjectivity of perceiving sound.

The human ear is very sensitive for small pressure variations in the air. Not all frequencies of these pressure variations are noticed similar.

The human ear is most sensitive for pressure variations of about 3kHz (corresponding to 3000 variations per second). For frequencies of 30Hz the ear is about 200 times less sensitive. The static air pressure is about hundred thousand Pascal (Pa), a pressure variation of one Pa is perceived as very loud at 1kHz, but as a soft tone at 20Hz, see Fig. 2.9.

Sound can be understood as variations of the static air pressure *and* as a velocity of air particles: particle velocity. The following may explain this. Pressure is defined as force per unit area. This force is caused by an amount of air particles at a certain position. So variations in the (sound) pressure are influenced by variations in the number of air particles. If the number of particles in a certain volume changes during time, there must be a particle motion towards that volume and backwards. The velocity of the movement of these particles is called particle velocity. Therefore sound consist always of two parts: the sound pressure and the velocity of the particle motion; particle velocity.

For complete quantification of a sound field, both sound pressure as particle velocity should be measured. In an electrical analogy, sound pressure corresponds to an electric voltage and particle velocity to electric current. The amount of power is quantified in Watts. Multiplication of voltage and current results in electrical power and analogously, sound pressure times particle velocity is associated with sound power in the acoustical domain.

Sound pressure, affected by the number of air particles at a certain position, does not have a direction; sound pressure is a scalar quantity. Contrary, the velocity of air particles does have a certain direction: particle velocity is a vector quantity.

To find out the direction of a sound wave, sound pressure and particle velocity must be measured simultaneously at one location. The product of both provides information about the direction of the sound wave.

The human ear detects sound pressure only. So how is it possible that we can find out where sound is coming from? This has to do with the shape of the ear and with the fact that we have two ears. Actually this is an example of the most elementary array technology. A human head can be seen as two pressure microphones (P_{left}&P_{right}) separated 20cm by a solid (the head), see Fig. 2.1. Here the sound source localisation will be explained very briefly, in [2.2] theory is explained in more detail.

With the use of both ears one can detect the sound level (difference) and one can detect the time difference between perceiving sound in one ear and the other. If it is assumed that the sound source is located left

forward (Fig. 2.1A) and the frequency is not too high, the sound must travel longer to the right ear than to the left ($L_{left}<L_{right}$). Since low frequency sound waves have very long wavelengths, the head is considered a small obstacle. The sound wave tends to bend around small obstacles and therefore the intensity difference between the two ears is little. By the use of this time delay (or more specific, the phase shift) the source is localised.

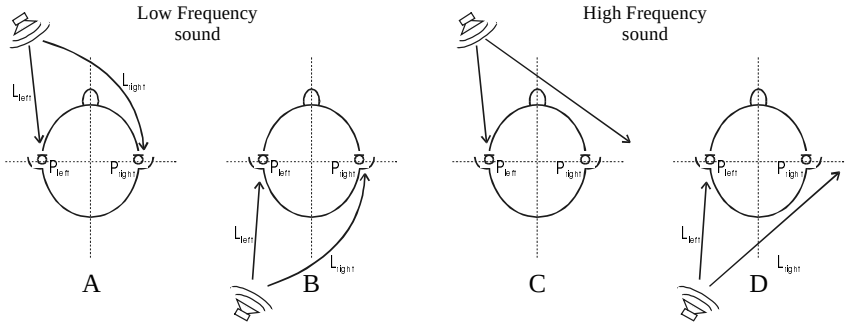

Fig. 2.1: Schematic head perceiving sound and sound waves, represented very basically.

Very simply explained, if the sound source is located left backward (Fig. 2.1B) one detects the same phase shift: the shape of the ears makes the discrimination possible between forward and backward. (Head movements do have also something to do with the source localisation).

Since for higher frequencies the wavelength becomes shorter than the separation of both ears and the phase shift method becomes invalid. Because the head now is larger than the wavelength it becomes an obstacle for sound waves. In situation Fig. 2.1C the left ear will perceive more sound than the right ear: the sound source location is localised by sound level difference. Again, if the sound source is located left backward (Fig. 2.1D) one detects the level difference. The shape of the ears is used again to discriminate between forward and backward.

As stated, the ear can detect sound pressure only and it is impossible to perceive particle velocity. How about acoustic sensors? Well, there are three types of sensors. First, sound pressure sensors, like the ear. Second, sound pressure gradient microphones; this type of acoustic sensor is sensitive for the pressure difference over a distance (like the human head

for low frequencies: two pressure sensors separated by a certain distance). Third, there is only one type of particle velocity sensitive microphone: the Microflown.

Sound pressure microphones detect pressure variations by deflection of a very thin membrane due to the sound pressure difference inside and outside the sensor. It functions similar to the ear. A pressure gradient microphone (also called a velocity microphone) measures the difference in the sound pressure at two spaced places. It has, in a limited bandwidth, a close resemblance to the particle velocity.

To get a better understanding of the relation between the sound pressure, sound pressure gradient and the particle velocity, the wave equation will be examined.

An introduction to acoustics

In this paragraph concepts like sound pressure, particle velocity and their relationship are described. For simple cases like plane waves, solving the one-dimensional wave equation yields the relation between the sound pressure and particle velocity. If the situation becomes more complex, like for instance sound fields near a sound source, the wave equation must be solved three-dimensionally. Only the one-dimensional wave equation is solved here.

Once the sound pressure and the particle velocity are known, the product and the ratio of both can be calculated. The product (associated with the sound intensity) provides information on the amount of acoustic noise and the ratio (the specific acoustic impedance) provides information on the effectiveness of co-operation between the sound pressure and particle velocity.

The sound intensity measurement is a main application of the Microflown. Nowadays this measurement is performed with a probe that consists of two closely spaced pressure microphones, a p-p probe. To comprehend its functioning the one-dimensional wave equation should be understood.

The one-dimensional wave equation

As sound waves travel through matter, each molecule hits another and returns to its original position. The result is that regions of the medium become alternately more dense, being condensations, and less dense, called rarefactions.

Now, observe a small volume ($S \cdot dx$) of air, see Fig. 2.2. Inside the volume there is a certain number of air particles, which can be associated with the air density. At place x particles enter the volume with a certain velocity $u(x,t)$ and particles leave the volume (at place $x+dx$) with a velocity $u(x+dx,t)$: the sound wave is travelling in the x-direction only.

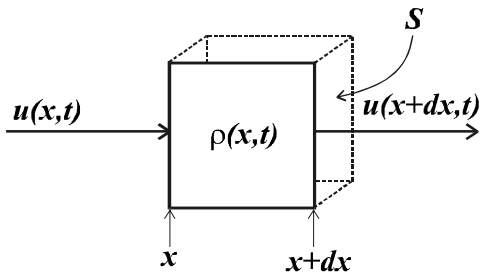

Fig. 2.2: A volume of air through which an acoustic disturbance passes.

The amount of mass that enters the volume equals: $S \cdot \rho(x,t) \cdot u(x,t)$. The amount that leaves the volume: $S \cdot \rho(x+dx,t) \cdot u(x+dx,t)$. Using ρ as the air density. The net rate of mass inflow therefore equals:

$$S\rho(x,t)u(x,t) - S\rho(x+dx,t)u(x+dx,t) = -S\frac{\partial}{\partial x}\big(\rho(x,t)u(x,t)\big)dx \quad (2.1)$$

The principle of mass conservation requires that this net inflow must be balanced by an increase in mass of the element:

$$-S\frac{\partial}{\partial x}\rho(x,t)u(x,t)dx = \frac{d}{dt}S \cdot dx \cdot \rho(x,t) \quad (2.2)$$

The density inside the volume can be described with $\rho(x,t)=\rho_0+\Delta\rho(x,t)$, a constant density ρ_0 plus a time varying part $\Delta\rho(x,t)$. The product of the density and the particle velocity equals: $\rho(x,t) \cdot u(x,t)=u(x,t)\rho_0+\Delta\rho(x,t)\ u(x,t)$. If the density variation can be

considered very small compared to the static density, the latter equation can be simplified into: $\rho(x,t)\cdot u(x,t)\approx u(x,t)\rho_0$. The simplified version of Eq. (2.2) is known as the linearised equation of mass conservation and is only valid if the variation in density is small compared to the density itself:

$$\boxed{\frac{\partial}{\partial t}\Delta\rho(x,t)+\rho_0\frac{\partial}{\partial x}u(x,t)=0} \tag{2.3}$$

The linearised equation of mass conservation can be used to calculate the sound pressure component of a sound wave if the particle velocity is measured at two closely spaced places. This will be clear when Eq. (2.3) is integrated with respect to time:

$$\Delta\rho(x,t)=-\rho_0\int\frac{\partial}{\partial x}u(x,t)dt\approx-\rho_0\int\frac{u(x+\Delta x,t)-u(x,t)}{\Delta x}dt \tag{2.4}$$

For "normal" sound levels (less than 135dB) it is assumed that the increase in pressure and increase in density are linearly related by $p(t,x)=c^2\,\Delta\rho(t,x)$. And thus:

$$p(x,t)=-\frac{\rho_0 c^2}{\Delta x}\int u(x+\Delta x,t)-u(x,t)dt \tag{2.5}$$

With the use of two closely spaced (spacing Δx) particle velocity probes (the separation distance must be much smaller than the wavelength) the (directional) sound pressure can be determined. In chapter BVII "*The **u-u** sound intensity measuring method*" the pressure determination with the use of two Microflowns is described in more detail.

The same small volume ($S\cdot dx$) of air is observed again, see Fig. 2.3. Now, for this volume the Newton's second law, force equals mass times acceleration, (**F**=m**a**) is solved.

For the acceleration (the time derivative of the velocity) of the particles in the volume element $S\cdot dx$ can be written:

$$\frac{d}{dt}u(x,t)=\frac{\partial u(x,t)}{\partial x}\frac{\partial x}{\partial t}+\frac{\partial}{\partial t}u(x,t)=\frac{\partial u(x,t)}{\partial x}u(x,t)+\frac{\partial}{\partial t}u(x,t) \tag{2.6}$$

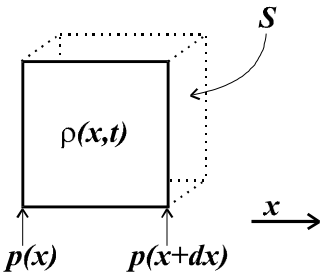

Fig. 2.3: A volume of air through which an acoustic disturbance passes.

For a harmonic particle velocity $u(x,t)\cdot\partial u(x,t)/\partial x$ is much (u/c) smaller then $\partial u(x,t)/\partial t$ and therefore Eq. (2.6) will simplify:

$$\frac{du(x,t)}{dt} \approx \frac{\partial u(x,t)}{\partial t} \tag{2.7}$$

The mass of the volume is given by: $S\cdot dx\cdot\rho_0$. The net force in the x-direction equals: $S\cdot p(x,t)$- $S\cdot p(x+dx,t)$. This net force equals the product of mass times and acceleration of the volume:

$$S\cdot p(x,t) - S\cdot p(x+dx,t) = \underbrace{-\frac{d}{dx}p(x,t)\cdot S\cdot dx}_{F} = \underbrace{S\cdot dx\cdot\rho_0}_{m}\underbrace{\frac{\partial}{\partial t}u(x,t)}_{a} \tag{2.8}$$

The latter can be simplified into:

$$\boxed{\rho_0\frac{d}{dt}u(x,t) + \frac{d}{dx}p(x,t) = 0} \tag{2.9}$$

This relationship is known as the linearised equation of momentum conservation and is only valid if the particle velocity u is small compared to the velocity of sound c.

Analogous to the calculation of sound pressure using the **u-u** method, the linearised equation of momentum conservation can be applied to deduce an expression for the particle velocity (by use of two closely spaced pressure microphones, called the p-p method):

$$u(x,t) = -\frac{1}{\rho_0}\int\frac{d}{dx}p(x,t)dt \approx -\frac{1}{\Delta x\cdot\rho_0}\int p(x+\Delta x,t) - p(x,t)dt \tag{2.10}$$

If the separation distance is much smaller than the wavelength the particle velocity can be determined. This is called the p-p method and will be dealt with in more detail at the end of this chapter (section "*Measuring sound intensity*").

Differentiating Eq. (2.3) with respect to time and Eq. (2.9) with respect to *x* yields:

$$\frac{\partial^2 p(x,t)}{\partial x^2} - \frac{\partial^2 \Delta\rho(x,t)}{\partial t^2} = 0 \tag{2.11}$$

The sound pressure and density variations are linearly related: ($p(x,t)=c^2\,\Delta\rho(x,t)$). By substituting this in Eq. (2.11), the one-dimensional wave equation is achieved:

$$\frac{\partial^2 p(t,x)}{\partial x^2} - \frac{1}{c^2}\frac{\partial^2 p(t,x)}{\partial t^2} = 0 \tag{2.12}$$

This wave equation must be satisfied by acoustic waves propagating in the medium. However, in order to find solutions to this equation, the form that the sound field must take must be specified in more detail. Boundary conditions must be specified in order to find the form of the sound field.

As an example the wave equation will be solved for a spherical sound field radiated from a point source. One can observe the sound intensity to find a suitable function for p. Since the surface of the sphere around the point source increases with the square of the radial distance, the sound intensity, the sound power per unit area, decreases with $1/r^2$. The sound power is a product of the sound pressure and the particle velocity. This product is a function of $1/r^2$ and the sound pressure and the particle velocity are linear related, therefore they will both be a function of $1/r$. The radiation from the point source is spherically symmetric and the wave equation for spherically symmetric wave propagation is given by (see appendix):

$$\frac{\partial^2 (r\cdot p)}{\partial r^2} - \frac{1}{c^2}\frac{\partial^2 (r\cdot p)}{\partial t^2} = 0 \tag{2.13}$$

Since the new function $r\cdot p$ should obey the wave equation Eq. (2.13), of which the solution is of the form $Ae^{i(kx\pm\omega t)}$, ($k=\omega/c$ known as the wave number) the pressure p should be a harmonic outwards-travelling wave proportional to $1/r$:

$$p(r,t) = \mathrm{Re}\left\{Ae^{i(wt-kr)} \cdot r^{-1}\right\} \tag{2.14}$$

Substitution of Eq. (2.14) in Eq. (2.13) shows that this function of the sound pressure obeys the wave equation. The linearised equation of momentum conservation can be used to find the particle velocity that is associated with the sound pressure:

$$u(r,t) = \frac{-1}{\rho_0}\int \frac{d}{dr}\, p(r,t)dt = \frac{A}{i\omega\rho_0}\left(\frac{1}{r} + ik\right)\frac{e^{i(\omega t-kr)}}{r} \tag{2.15}$$

The specific acoustic impedance as a function of the radial distance:

$$z(r) = \frac{p(r,t)}{u_r(r,t)} = \rho_0 c\left(\frac{ikr}{ikr+1}\right) \tag{2.16}$$

Eq. (2.16) shows that, if the distance to the sound source is small, for spherical waves the specific acoustic impedance is proportional to *k*, and thus for the frequency. In practice this means that relative close to the source the acoustic impedance is lower than ρc and frequency dependent. Lower frequencies will be amplified for the particle velocity level but not for the sound pressure level. This is called the nearfield effect, see chapter A5: "*Calibration Methods*".

As an example of the difference between sound pressure and particle velocity, a burst of two sound waves are generated in a tube in opposite directions, see Fig. 2.4.

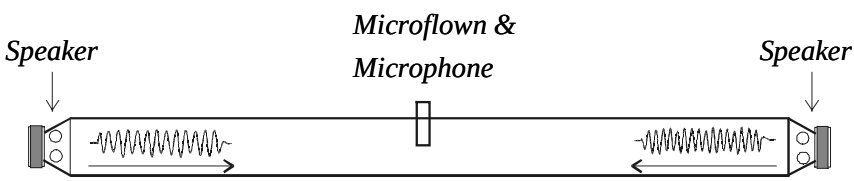

Fig. 2.4: Measurement set-up to demonstrate the difference between sound pressure and particle velocity.

One speaker generates a 150Hz burst and the other a 200Hz burst. The Microflown and microphone are situated in the middle measuring both sound waves travelling in opposite directions (upper two curves in Fig. 2.5).

As can be seen, the output of both microphones is different. If the two signals are subtracted the result will be the leftwards-travelling wave and the sum will result in the rightwards-travelling wave. This is because the particle velocity has a phase shift compared to the sound pressure if the direction is altered from forward to backward, see Fig. 2.6. If both Microflown as microphone have the same sensitivity, the sum of both sensors will result in a one-directional sensor assembly, see also chapter D3: *"unidirectional microphones based on Microflown Technology"*.

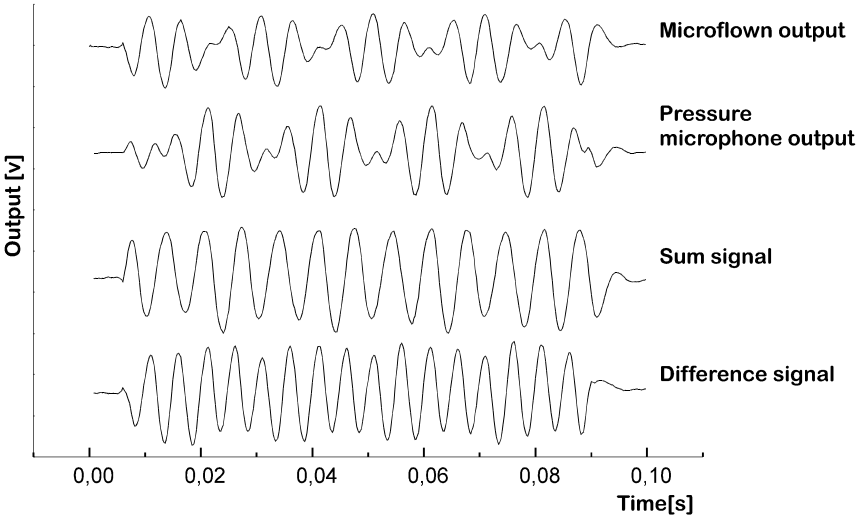

Fig. 2.5: Measured signals of a Microflown and a microphone; their sum and difference.

The determined sound pressure and particle velocity are mostly not in phase and do not have the same sound level value. Only under free field conditions (far from the source where the sound waves are plain and no reflections are present), both sound pressure and particle velocity are completely in (A) or out phase (B) and the measured sound levels will be the same.

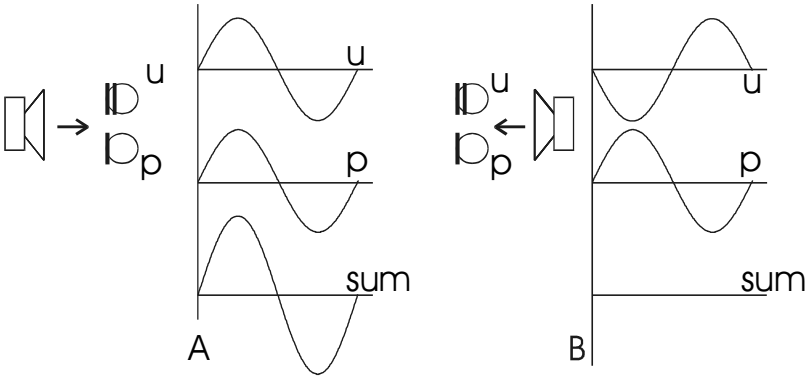

Fig. 2.6: Measured signals of a Microflown and a microphone; their sum and difference.

Definitions

In acoustics a particle is defined as a volume of air that is small compared to the dimensions of our measuring device or the wave length of the acoustic wave and large compared to the molecular mean free path ($5 \cdot 10^{-8} m$) [2.1]. The minimal wavelength in air being about 17 millimetre, the measuring device can be for instance an eardrum (5 millimetre) or a Microflown (100 micrometer). A particle may be interpreted as a cube of air with dimensions of 1μm×1μm×1μm.

The *sound pressure*, *p(t)*, is the incremental change from the static pressure at a given instant caused by the presence of the sound wave. Pressure can be associated with the number of air particles. The *particle velocity* ***u****(t)*, is the velocity of a given infinitesimal part of the medium (the above mentioned particle) at a given instant due to a sound wave only. Note that sound pressure is a scalar and particle velocity is a vector quantity.

The human ear is sensitive for sound pressure in the *frequency* (*f*) range from 20Hz up to 20kHz. Sound waves with frequencies higher than 20kHz are called ultra sound and sound waves with frequencies lower than 20Hz are called infra sound. Low frequency sound waves have frequencies lower than 200Hz.

The *wavelength* λ, is given by the speed of propagation of the wave, *c*, divided by the frequency of vibration:

$$\lambda = \frac{c}{f}[m] \tag{2.17}$$

In air acoustic wavelengths extend over a range from 17 millimetres (f=20kHz) to 17 meters (f=20Hz).

The *specific acoustic impedance* at a point in a sound field is defined as the ratio of the complex amplitude of an individual frequency component of sound pressure at that point, to the complex amplitude of the associated component of particle velocity. It is a complex number, giving the magnitude and phase of the ratio of the individual frequency components. It has an important bearing on energy flow in a sound field because the phase relation-ship between the pressure (a force-like quantity) and the fluid velocity clearly indicates the 'effectiveness of their co-operation', in the same way that the power factor does for voltage and current in an electrical circuit [2.7].

$$Z_s = \frac{p}{u}[Ns/m^3] \tag{2.18}$$

Note that the specific acoustic impedance is a complex figure and not a vector quantity.

The measurement of the specific acoustic impedance is used for example to determine the sound absorbing characteristics of materials like glass wool or foams. These measurements are often performed in an impedance tube. In the tube a one-dimensional acoustic field is generated with a speaker. The plane waves are reflected and absorbed by the sample material at the end of the tube. The reflection coefficient can be calculated from the impedance. Various methods to measure this impedance are presented in chapter C2: *"The acoustic impedance tube"*.

The *characteristic impedance* is the ratio of the effective sound pressure at a given point to the effective particle velocity at that point in a free, plane, progressive sound wave. It is equal to the product of the density ρ, of the medium times the speed of sound c in the medium ($\rho c \approx 435 Nsm^{-3}$). A *plane wave* is an acoustic disturbance in which the pressure is uniform in the plane direction normal to the direction of propagation. When measured far from the acoustic source in the *free field* one obtains a plane wave.

The *sound intensity* in a specified direction 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 considered. It is thus a

vector quantity; defined as the time averaged product of the sound pressure (a scalar quantity, *p*) and the corresponding particle velocity (a vector quantity, ***u***) at the same position.

$$\boldsymbol{I}[W/m^2] = \lim_{T\to\infty} \frac{1}{T}\int_0^T \boldsymbol{u}(t)\cdot p(t)dt \tag{2.19}$$

Sound energy density E is defined as the sum of potential energy *U* and the kinetic energy *K*, both per unit volume [2.7]. The potential energy *U* is associated with the sound pressure and is given by:

$$U = \frac{|p|^2}{2\rho_0 c^2} \tag{2.20}$$

The kinetic energy density *K* is associated with the particle velocity and is given by:

$$K = \tfrac{1}{2}\rho_0 |u|^2 \tag{2.21}$$

Since nowadays apart from the sound pressure, particle velocity can be determined instantly (a unique feature of the Microflown) acoustic quantities as sound intensity, acoustic impedance and sound energy density can be measured instantaneous and on one position in space.

Sound intensity is associated with the product of sound pressure and particle velocity and quantifies the amount of sound. The specific acoustic impedance is related with the ratio of both, and is a useful quantity to determine for example the reflection or absorption of matter. Sound energy density quantifies how much energy is “stored” in an acoustic wave, sound intensity quantifies how much sound energy is transported and specific acoustic impedance quantifies the possibility for sound energy to be transported.

For most purposes, the absolute numbers obtained from measurements have little significance themselves; they are almost always compared to a reference, and are usually quoted as: levels re. …" that reference. As in many physical sciences, the units and reference levels are derived from common observed phenomena. The zero-level of sound pressure, for example, is not a physical zero (that is the absence of sound pressure); rather, it is something of an average "threshold of hearing" at about 1kHz for humans. The physical pressure associated with this level is very small (20 micro Pascal). About one million times greater sound pressure tends

to cause considerable discomfort. Since the sound intensity generally varies with the square of the sound pressure, this (audio) range represents a ratio of one trillion (10^{12}) to one in energy [2.9]!

The most known *sound pressure level*, given in decibels, is defined as twenty times the logarithm to the base ten of the ratio of the measured sound pressure of this sound to a reference effective sound pressure of 20 micro Pascal.

$$SPL = 20Log_{10}\frac{p}{p_{ref.}}(re. = 20\mu Pa)[dB] \tag{2.22}$$

The *particle velocity level*, also given in decibels, has a reference effective particle velocity of 50 nano-meter per second. It can be argued that the reference should be 20μPa/ρc.

$$PVL = 20Log_{10}\frac{u}{u_{ref}}(re. = 50nm \cdot s^{-1})[dB] \tag{2.23}$$

The *sound intensity level* has a reference intensity of one pico-Watt.

$$\boldsymbol{S}IL = 10Log_{10}\frac{I}{I_{ref}}(re. = 1pW)[dB] \tag{2.24}$$

The ratio between the reference sound pressure and the reference particle velocity equals roughly the characteristic imepdance.

Various values of the magnitudes mentioned quantities are given in Table 2.1. It this case it is assumed that the specific acoustic impedance equals the characteristic impedance, the values of PVL and SIL are equal to the SPL.

Table 2.1: Some acoustic values

Description	*level* [dB]	*p* [Pa]	***u*** [m/s]	***I*** [W/m^2]
Artillery fire, airplane	140	200	0.45	100
Rock concert	120	20	0.045	1
Reference level	94	1	2.5×10^{-3}	2.5×10^{-3}
Vacuum cleaner	80	0.2	4.5×10^{-4}	1×10^{-4}
Conversation, face-to-face	60	0.02	4.5×10^{-5}	1×10^{-6}
Whispering	40	2×10^{-3}	4.5×10^{-6}	1×10^{-8}
Inside empty theater	20	2×10^{-4}	4.5×10^{-7}	1×10^{-10}
Threshold of hearing	0	2×10^{-5}	4.5×10^{-8}	1×10^{-12}

Measuring sound

Transducers are devices that convert one quantity into another. Membrane microphones are acoustic transducers that are commonly applied to convert sound into electrical signals. The sound pressure "pushes" a membrane forward and backwards and the deflection of the membrane is detected electrically. When this membrane closes a cavity the microphone detects sound pressure. In a construction in which both sides of the membrane are subject to the acoustic wave, the microphone is sensitive for sound pressure gradient.

Sound, however, consist of two elements: sound pressure and particle velocity. A pressure microphone is constructed to measure sound pressure whereas the Microflown enables the direct and reliable measurement of particle velocity.

To determine the amount of noise, sound intensity has to be measured [2.7]. Two sensors are required for the determination of sound intensity. The most obvious way is measuring sound pressure (p) with a microphone and particle velocity ($\boldsymbol{u}$) with a Microflown, and then multiply and time integrate both signals. This is called the p-**u** method. Different methods use closely spaced microphones (p-p method) or Microflowns (**u**-**u** method). Sound intensity is associated with the product of sound pressure and particle velocity and another basic acoustic measurement is related to the ratio of both acoustic quantities: the determination of the specific acoustic impedance. Sound energy is associated by the square of both sound pressure as particle velocity.

Measuring sound pressure

The detection of sound pressure of a microphone is similar to the way the human ear perceives sound. A membrane closes of a cavity, inside the cavity the pressure equals the static air-pressure and pressure variations outside the cavity cause a movement of the membrane. This movement is detected. In this paragraph the functioning of a capacitive microphone is explained.

All practical microphones detect the (sound pressure induced) deflection, or movement, of a membrane. There are several ways to implement this.

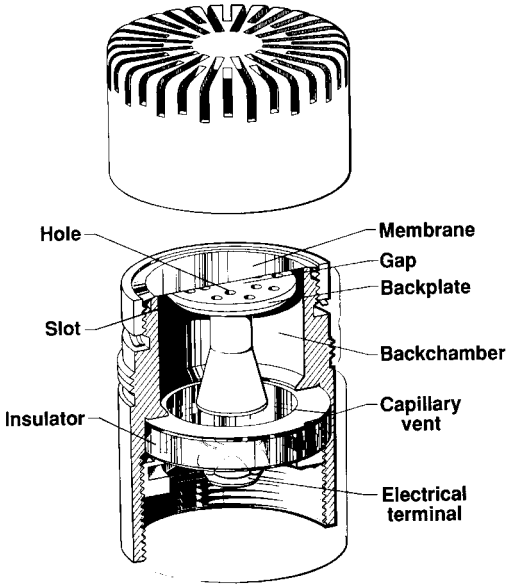

Fig. 2.7: A cross-section of a condenser microphone.

Electromagnetic microphones consist of a coil connected to the membrane. The movement of the membrane causes the coil to move in a magnetic field resulting in an electrical signal. The first who developed this type of microphone was the German teacher Phillipp Reis [2.3] in the year 1860. Since then the development of sound pressure measurements began. The capacitive microphone detects the acoustic pressure variations by means of a condenser [2.4]. The membrane acts as one plate of the capacitor whereas a perforated-rigid-back-electrode is the other plate. A deflection of the membrane causes a variation of the capacitance. This type of microphone needs an additional (bias) voltage for operation and is called the condenser microphone. To avoid external biasing it is possible to provide one of the plates with an electret, a electrically charged layer, creating a so-called electret microphone. Capacitive microphones have good noise properties (or signal to noise ratio), which is the most important parameter of a microphone. The first condenser (electret) microphone was built in 1918 [2.4]. The construction of today's measurement condenser microphone is illustrated in Fig. 2.7.

Although several different ways to detect deflection of a membrane exist, the previous mentioned are the most common. To get a better understanding of the sound pressure measurement, the condenser microphone is investigated further.

Two electrical conducting plates separated by an insulator will create a condenser. The condenser microphone consists on an electrical conducting, stretched membrane over an electrical conducting backplate forming together, with the air gap (insulator), a condenser. Sound pressure excites the membrane, changing the capacitance of the microphone. This capacitance is electrically measured. The performance of a condenser microphone depends upon the design of its electrical and mechanical systems.

A high polarisation voltage, large membrane radius, small air gap, and low membrane tension favour the sensitivity. Sensitivity is a microphone's electrical output caused by a sound pressure given in Volts per Pascal.

The bandwidth, the range of frequencies within which a microphone yields a usable output signal, is favoured by a high membrane tension, small membrane radius and low surface mass density (implying a thin membrane). The response of a microphone "rolls-off" at low frequencies (frequencies lower than 1 to 125Hz) due to the presence of a capillary vent, see Fig. 2.7. The purpose of this vent is to provide a path for pressure equalisation on both sides of the membrane following a change in ambient pressure. However, this vent supplies also a path for low-frequency acoustical waves to reach the backside of the membrane and thus suppresses the low-frequency response of the microphone.

The selfnoise is the electrical output of a microphone (in an absolute silent environment) caused by noise, expressed in a quantity corresponding to a certain sound level. Resistive damping of the membrane by the air layer and the holes in the backplate (see Fig. 2.7) in the air gap causes the thermal noise of a condenser microphone. For lower frequencies, the preamplifier 1/f-noise dominates. The thermal noise of the microphone is a strong inverse function of the membrane radius. If the sensitivity of the microphone is low the preamplifier noise becomes more dominant.

The preamplifier is needed to convert the very high impedance of the capacitive microphone to a lower output impedance. The voltage gain of these preamplifiers is usually unity.

The non-linear relationship between the capacitance and the membrane displacement is the main reason for the harmonic distortion at high sound pressure levels. A high membrane tension and small membrane radius will increase the upper sound pressure limit.

A polarisation voltage causes an attracting force between the membrane and the backplate. If the polarisation voltage is chosen too high, the membrane will collapse to the backplate.

A polar pattern (or microphone directivity) expresses the sensitivity to the angle of incidence of the sound field. A sound pressure microphone should be omni-directional which implies a constant sensitivity for any angle of incidence of the sound field. The larger the membrane radius (and thus physical size) the more the microphone behaves as an obstacle in the sound field. Due to this effect the directivity of the microphone increases for higher frequencies. Another cause of directivity of sound pressure microphones is related to the diameter. When the wavelength is becoming smaller than the diameter, and the angle of incidence sound wave is perpendicular to the membrane, one part of the membrane is puled and another part is pushed. The sensitivity will be reduced due to this phenomenon.

Generally a microphone with a half-inch membrane radius and 200V polarisation voltage is chosen for general measuring purposes. This is because the microphone has a relatively low selfnoise (20dB(A)), a sufficiently large bandwidth (20kHz) and good omni-directional properties (3dB variation at 8kHz). The microphone can be used for sound levels up to 160dB.

Table 2.2: General microphone specifications related to the membrane radius

Membrane radius	Sensitivity [mV/Pa]	Bandwidth [kHz]	Selfnoise dB(A)	Upper limit [3% distortion]
1"	50	10	10	150 dB SPL
½"	12,5	24	22	160 dB SPL
¼"	4	100	36	170 dB SPL
1/8"	1	160	55	180 dB SPL
1/10" (Knowles)	30	20	27	100 dB SPL

Fig. 2.8 shows how mechanical properties effect the frequency response and dynamic range and table 2.2 shows how dynamic range, bandwidth and sensitivity depend on microphone size. When the membrane diameter decreases, the maximum sound pressure level a microphone can withstand, and the noise floor will increase. Sensitivity is

proportional to the diaphragm area. In actual microphone design, other parameters may be changed along with the membrane radius so that particularly useful compromises in bandwidth, dynamic range, and sensitivity can be achieved. The upper four microphones are used for measurement purposes and the Knowles is used for audio pick-up.

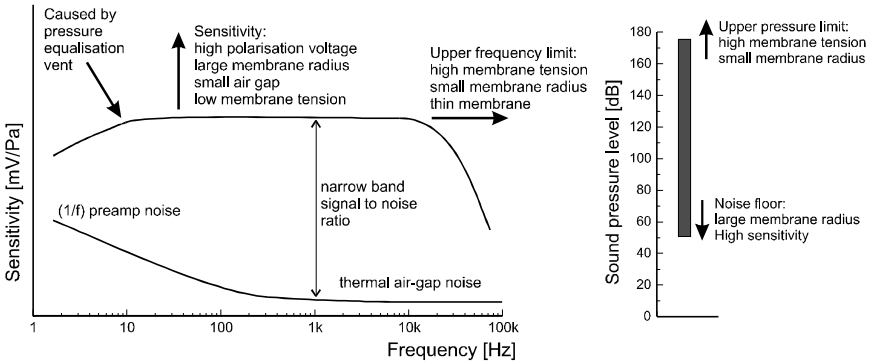

Fig. 2.8: The effect of mechanical properties on acoustical specs of condenser microphones.

The cylindrical microphone (Knowles Electronics) for instance has a diameter of 0.1". This microphone is used for hearing aid purposes. The selfnoise of this miniature microphone is 27dB(A), the effective* bandwidth however is limited to approximately 20kHz (1/8” B&K has a bandwidth of 160kHz), and the upper limit of the sound pressure is 100dB SPL. The sensitivity is 30mV/Pa. The high sensitivity and low sound-pressure upper-limit suggest low membrane tension. The lower membrane tension will result in a different mechanical design i.e. air layer and holes in the backplate; therefore the selfnoise due to the resistive damping of the membrane by the air layer and the holes in the backplate is lower than the 1/8” B&K. Due to the low membrane tension the sensitivity is high and the preamplifier noise becomes less dominant.

* A Helmoltz resonator (created by the housing) that amplifies the high frequency response probably extends the bandwidth somewhat.

The Human ear

The human ear is sensitive for sound pressure. Hearing is a subjective response to sound pressure. At very low frequencies or very high pressure-levels, additional sensations are experienced but normally the ear is the acoustic receiver. It receives the vibrations on the eardrum, multiplies them by means of small bones in the middle ear, and transmits the vibrations through a fluid to nerve endings within the inner ear. These nerves transmit an impulse to the brain that analyses and translates the acoustic impulse into a concept or situation. In other words the essential difference between the detection of sound with a pressure microphone and hearing is that hearing is the combination of two pressure sensors *and* a processing unit. A microphone can only detect the sound pressure, interpretation must be done afterwards.

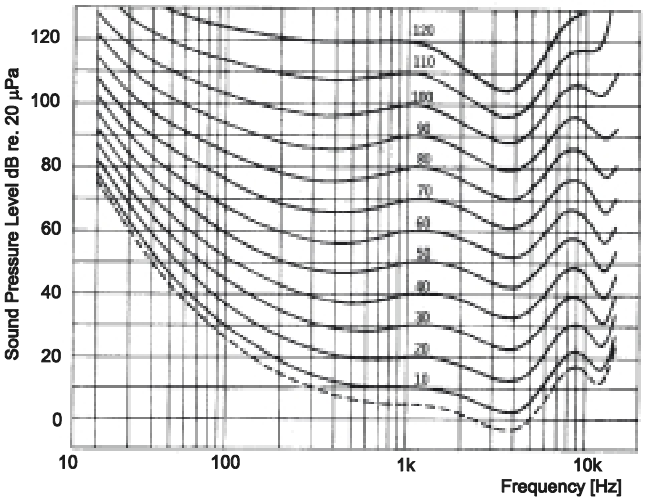

Fig. 2.9: Equal loudness contours of the human hearing.

Loudness is the physical response to sound. At any given frequency between 20Hz and 20kHz the loudness varies directly with the sound pressure, but not in a simple straight-line manner. In Fig. 2.9 the equal loudness contours are shown, they represent the sound pressure level necessary at each frequency to produce the same loudness response in the

average listener. As can be seen loudness is a very non-linear physical response. Humans are quite deaf at low frequencies. At 20Hz the level must be almost 70dB higher (ten million times as much sound energy) as at 2kHz to produce hearing. Another non-linearity appears at low frequencies: if, for example, the sound level is rising from 80dB to 90dB, humans perceive a 20dB increase of the sound pressure.

To convert sound measurements performed with a microphone in one figure representing loudness, the “A”-weighing curve is introduced. This “A”-weighing curve is displayed in chapter 4 and is similar to the inverse of the equal loudness contour of 0dB, as shown in Fig. 2.9.

Measuring sound pressure gradient

Now the condenser microphone is explained, the way in which sound pressure gradient is determined is easy to conceive. The difference between a sound pressure microphone and a (sound pressure) gradient microphone is that the gradient microphone has two (in stead of one) acoustic entrances separated with a distance L, see Fig. 2.10A. The membrane will deflect due to a sound pressure difference over the membrane. (In fact a pressure microphone detects a pressure gradient over the membrane as well, but in the back chamber the pressure is constant).

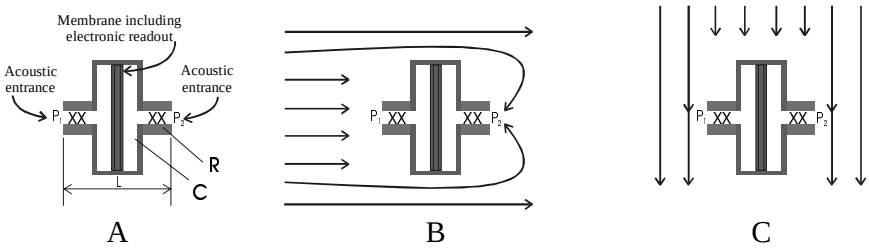

Fig. 2.10: A schematic cross-section of a pressure gradient microphone.

If a sound wave enters in the sensitive direction, see Fig. 2.10B, first the wave reaches the input P_1 and a while later input P_2. In this way the pressure gradient is determined; the output of this type of microphone will be linear with respect to: $p(x)-p(x+\Delta x)$. Observing Eq. (2.10), it is not surprising that this type of microphone is also called “velocity

microphone" since the particle velocity can be calculated by a pressure difference of two closely spaced microphones.

The velocity microphone will not be sensitive in the direction perpendicular to the sensitive direction, see Fig. 2.10C. The sound wave reaches both sides of the membrane at the same time and therefore the subtraction $p(x)-p(x+\Delta x)$ is zero.

It can be deduced that the sensitivity as function of the angle entry (the polar pattern) is a cosine function, the response is plotted in Fig. 2.11. At the right side this cosine function is shown. This shape of polar pattern is often called a "figure of eight". Normally a logarithm representation is preferred, it is shown at the left side.

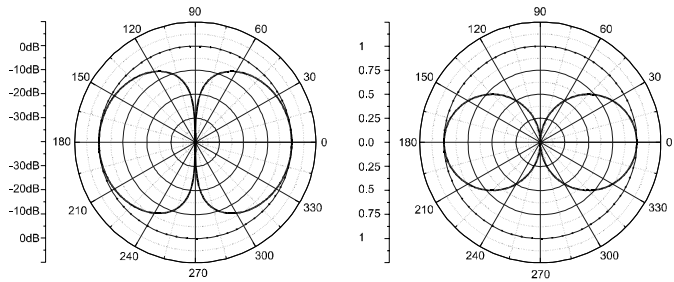

Fig. 2.11: Response of a sound pressure microphone (dashed line; omni directional) and a sound pressure gradient microphone (solid line; figure of eight).

For harmonic sound waves ($p(t,x)=p_0 \cos(\omega t-\omega x/c)$), the frequency response of a gradient microphone is not constant; the response increases linear with frequency:

$$\begin{aligned} p(x,t)-p(x+\Delta x,t) &= \frac{p(x,t)-p(x+\Delta x,t)}{\Delta x}\Delta x \\ &\approx \frac{d}{dx}p(x,t)\Delta x = \frac{\omega}{c}p(x,t)\Delta x \end{aligned} \tag{2.25}$$

When the frequency increases so much that such that half the wavelength equals the separation distance, the sensitivity becomes maximal: two times higher than if the membrane was used in one-input configuration (regular pressure microphone). After this maximum the sensitivity drops dramatically, see Fig. 2.12. To show this effect, the frequency response due to a separation distance of 5cm is plotted: as can

be seen the sensitivity is at its maximum at 3.3kHz; ($\frac{1}{2}\lambda=\frac{1}{2}\cdot c/f=0.5\times 330/3300=5cm$).

The response should be integrated, to create a useful output signal (flat frequency response). Simply said, in the frequency domain integrating is the same as multiplying the sensitivity with 1/f.

Integrating can quite easily be achieved with an electronic circuit. In this case, for decreasing frequencies the signal to noise ratio will decrease.

It is also possible to use acoustic elements to achieve the integrating. Very simply explained, a dissipating element *R* is placed between the acoustic inputs and the membrane (that is stretched very loose), indicated with "xxx" in Fig. 2.10 and a large acoustic chamber creating a compliance *C* are together creating an acoustic RC low pass filter (integrator). The resulting frequency response will be reasonable flat from 250Hz up to 16kHz. Due to the lesser damping of the pressure gradients' membrane, the inherent noise of the capsule is much lower than a sound pressure sensitive microphone. However, the frequency response is filtered by the acoustic low-pass filter and therefore the selfnoise of a pressure gradient microphone is a few dB worse than a pressure microphone. For audio applications where signal to noise values should be as high as possible, acoustic integration is used whereas for measurement purposes where linearity is the most important issue, electronic integration is employed. See also chapter D2: *"Add-on Microflown for a high end pressure gradient microphone"*.

Integration of the output of the 5cm-separation configuration results in a flat frequency response from low frequencies up to 3.3kHz. For frequencies higher than 3.4kHz the response will be useless.

To achieve a useful output signal for high frequencies the separation distance should not exceed 1cm. But as can be seen in see Fig. 2.12 for low frequencies the signal to noise ratio is very poor compared to a regular pressure microphone.

As stated, the output signal is integrated to create a flat frequency response. When corrected electronically afterwards, the noise is also integrated and the signal to noise ratio therefore is not affected by this correction. The selfnoise of a sound pressure gradient microphone (pair) that makes use of electrical correction afterwards is therefore always worse than a sound pressure microphone. When corrected acoustically beforehand, the frequency response will lack the first decade (20Hz-250Hz) and the noise figures will be only a few decibels worse than

sound pressure microphones. A good sound pressure gradient microphone (Schoeps MK8) has a selfnoise of 19dB(A) and a good sound pressure microphone (Schoeps MK2H) has a selfnoise of 13dB(A) to 15dB(A).

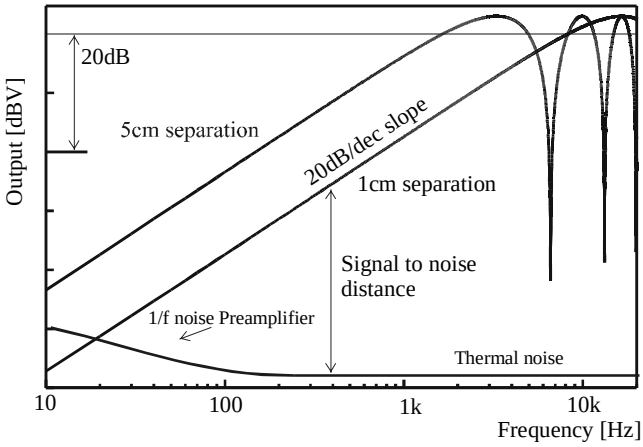

Fig. 2.12: Response of a pressure gradient microphone with an acoustic input separation of 5cm and 1cm.

Measuring particle velocity

The direct measurement of particle velocity is quite new. Until the invention of the Microflown in 1994, the particle velocity could only be measured indirectly. Here three velocity detecting methods will be described: the ribbon microphone, and ultrasonic detection method and the Microflown.

As stated before, the sound pressure gradient shows a great resemblance to the particle velocity. After all, Eq. (2.10) states that the pressure gradient is linear proportional to the particle velocity. However in Fig. 2.12 one can see that the sensitivity of a pressure gradient microphone increases proportional with the frequency. For zero Hertz (a DC or constant particle flow) their sensitivity is zero. The Microflown However originates from a mass flow sensor; a sensor to measure DC-flow. The fast response time and improved sensitivity is the only difference between a mass-flow sensor and a Microflown.

The general behaviour of pressure gradient microphones is the lower the frequency the lower sensitivity. From zero Hertz up to it corner frequency a Microflown has constant sensitivity and signal to noise ratio.

Unlike this crucial difference there are also similarities. For instance the polar pattern is the same for a Microflown and a pressure gradient microphone, see Fig. 2.11. The near field effect (see above "*the one-dimensional wave equation*") is also noticed in the same manner.

Another type of "velocity" microphone is the so-called ribbon microphone [2.10]. The operation principle of this type of microphone is again based on pressure gradient. In this case an aluminium foil (the ribbon) is placed under a low tension in a strong magnetic field, see Fig. 2.13. The difference in sound pressure between the two sides drives the ribbon and the velocity of the ribbon has the same response as the pressure gradient microphone (see Fig. 2.12). The output voltage due to the motion of the ribbon however is proportional to the product of the flux density, the length and the velocity of the ribbon. The result is a "flat" frequency response. (The ribbon moving in a magnetic field has an integrating character, whereas the pressure gradient microphone integrating the integration of the signal have to be performed electronically afterwards or acoustically on forehand).

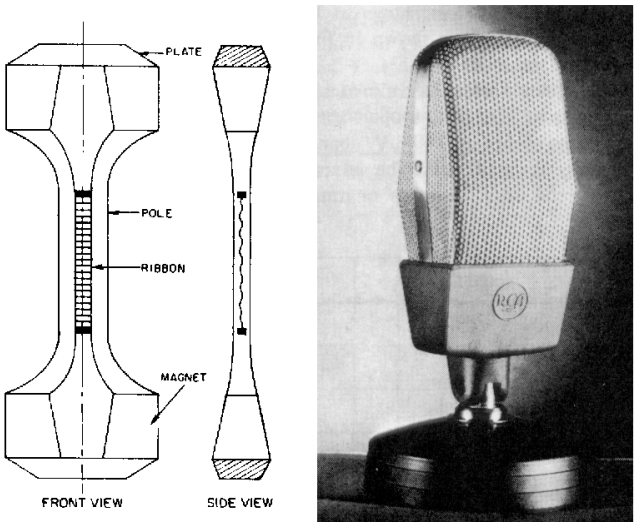

Fig. 2.13: A ribbon type of microphone. Left: schematic overview; right: the RCA BK11A velocity microphone [2.10].

The ribbon velocity microphone is a pressure gradient type of microphone that has (thus) no DC sensitivity. The selfnoise of such microphones is quite low since there is (of course) no thermal noise of the airgap. The main noise source is thermal noise of the electrical resistance of the ribbon.

A complete other way of sensing particle velocity is the principle of ultrasonic transduction: two parallel ultrasonic beams are launched in opposite directions. The travelling time from the transmitter to the receiver is linear proportional to the speed of sound in the air. When the air is moving this movement should be added to this speed. The probe consists of two transmitter-receiver pairs that are positioned opposite directions, see Fig. 2.14. The *time difference* in travelling time of the ultrasonic sound waves is proportional to the particle velocity; this type of velocity probe is also capable to measure DC flow. It is a true particle velocity sensor but it is a distributed sensor: it doesn't determine the particle velocity in one spot.

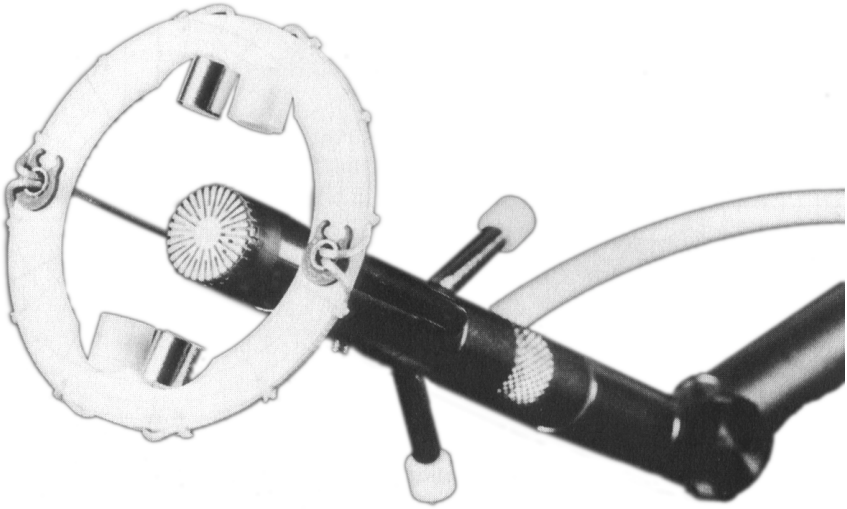

Fig. 2.14: Type 216 p-u intensity probe (Norwegian electronics).

This probe however is not used very often since it is very sensitive for DC flows as for example wind or movements of the probe. Furthermore it is, just like the p-p probe, a distributed sensor with the problems associated with this type of sensing (for example maximal frequency limited by the spacing). At last, due to it physical dimensions, the probe is difficult to calibrate.

The Microflown is an acoustical sensor that is based on a thermal principle. It consists of two very closely spaced thin wires of silicon nitride with an electrically conducting platinum pattern on top of them. A SEM photograph of a cantilever type of Microflown is depicted in Fig. 2.15.

The Microflown consists of a silicon die with platinum electrical connections on top of it and two, heated cantilever temperature sensors. The die is broken apart from its neighbours, this can be seen at the side. At the top of the die one can see two sensors sticking out. The electrical connecting wirebonds are also visible.

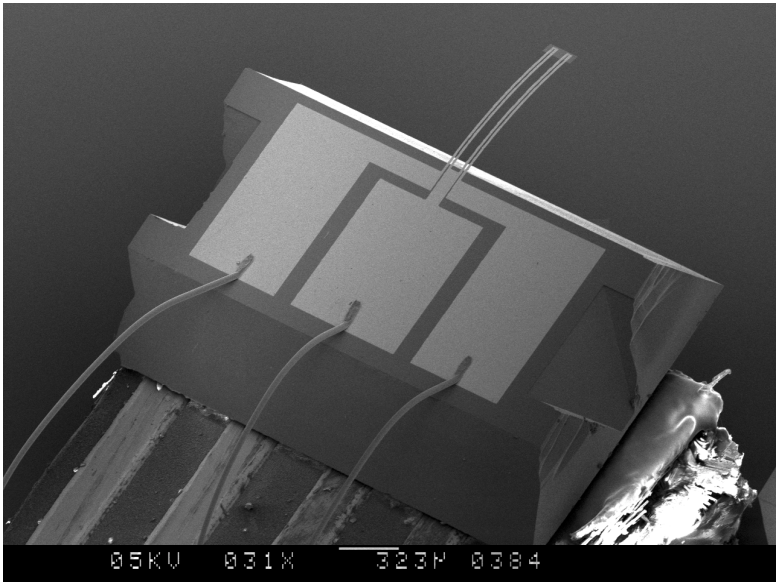

Fig. 2.15: Scanning electron microscope photo of a cantilever type of Microflown. The two wires that are sticking out are the actual Microflown, the three squares are the “bondpads” that are used to make electrical connections and the three wires in the lower part of the picture are wirebonds: Aluminium wires that are 80 micron in diameter. See further chapter 1 and 3.

The size of the two wires is 1000×10×0.5 μm (l×w×h). The metal pattern is used as temperature sensor *and* heater. The silicon nitride layer is used as a mechanical carrier for the platinum resistor patterns. The sensors are powered by an electrical current, causing the sensors to heat up. The temperature difference of the two cantilevers is linear dependent on the particle velocity level.

Fabrication methods of the Microflown will be considered in chapter 3, in this chapter the operation principle will be explained briefly. A more detailed mathematical model of the Microflown model is presented in chapter A1: *"model of the Microflown"*. The two squares S1 and S2 in Fig. 2.16 represent the two temperature sensors of the Microflown. The temperature sensors are implemented as platinum resistors and are powered by an electrical current dissipating an electrical power P_{el}, causing it to heat-up. And an increase of the temperature of the sensors leads to an increase of the resistance as well. If no particle velocity is present the sensors will have a typical operational temperature of about 200ºC to 400ºC and all the heat is transferred in the surrounding air ($q_{stat.}$). When particle velocity is present, it alters the temperature distribution around the resistors. The temperature difference of the two sensors quantifies the particle velocity.

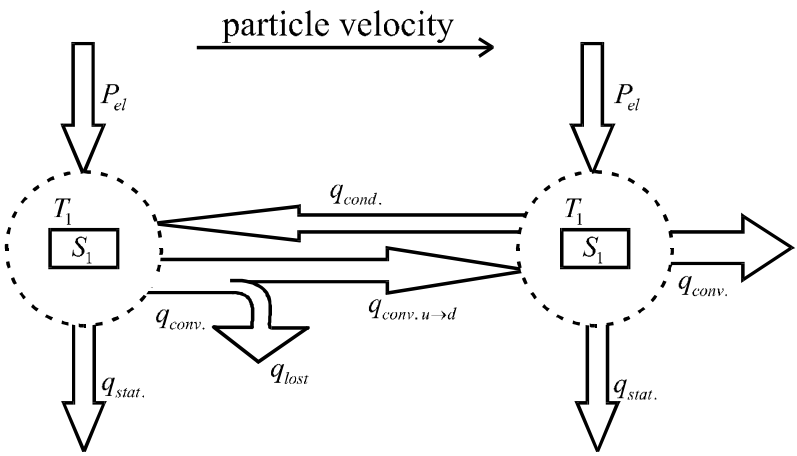

Fig. 2.16: Schematic overview of the heat flows around a Microflown.

Particle velocity causes a convective heat transfer of both sensors (q_{conv}) which will create a temperature drop of both sensors. The upstream sensor however, drops more in temperature than the downstream sensor since the downstream sensor is heated by the upstream convective heat loss ($q_{conv\ u \rightarrow d}$), see Fig. 2.16. A temperature difference is the result. This temperature difference is proportional to the particle velocity.

Not all the convective heat loss of S_1 will be transferred to S_2, a certain percentage will be lost (q_{lost}). This percentage will rise if the sensors are positioned further apart from each other. If, on the other hand, the sensors are brought closer together another phenomenon will become dominant. The particle velocity induced temperature difference will cause a conductive heat flow ($q_{cond.}$) in the opposite direction. This feedback heat flow will temper the sensitivity. The closer the sensors are placed, the more conductive heat flow will take its effect. Temperature variations due to an acoustical wave of the two sensors of the Microflown are shown in Fig. 4.1.

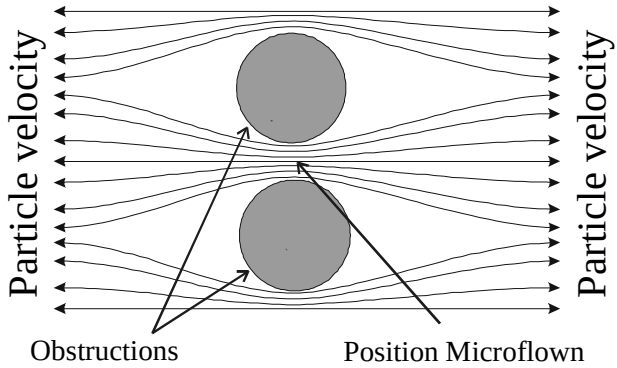

Fig. 2.17: A well-chosen package will result in a particle velocity gain (top view).

At higher frequencies the sensitivity of the Microflown is decreasing. This high frequency roll-off that is caused by the time it takes heat to travel from one wire to the other can be estimated by a first order low pass frequency response that has a (diffusion) corner frequency (f_{diff}) in the order of 700Hz. The second high frequency roll-off is caused by the heat capacity (or thermal mass) and shows an exact first order low pass behaviour that has a heat capacity corner frequency ($f_{heat\ cap}$) in the order of 2kHz to 8kHz for modern Microflowns. These effects are explained in more detail in chapter A1: *"Model of the Microflown"*.

To protect its fragile sensors, the Microflown has to be packaged. An attractive aspect of the packaging is that the particle velocity level increases when a well-chosen obstacle is placed near the sensors, see Fig. 2.17 and Fig. 2.18. The so-called *package gain* is caused due to a "channelling" of the particle velocity, this can result in an increase of the particle velocity level of a factor 3 up to 10 times (+10dB to +30dB).

Since the noise has a thermal electrical origin it is not be affected by the package (gain). The selfnoise of a packaged Microflown therefore can be expected 10dB to 20dB less than a non-packaged Microflown.

It is very difficult to compare all individual properties of the Microflown with a pressure microphone. The sensitivity of the Microflown is defined as the electrical output due to 1m/s particle velocity input, see chapter 4. The output however is not an electrical voltage but a relative differential resistance variation. The sensitivity of sound pressure microphones is given in mV/Pa. 1Pa sound pressure corresponds to a sound pressure level of 94dB. Therefore, for easy comparison, the sensitivity of the Microflown is often given mV per 94dB (PVL).

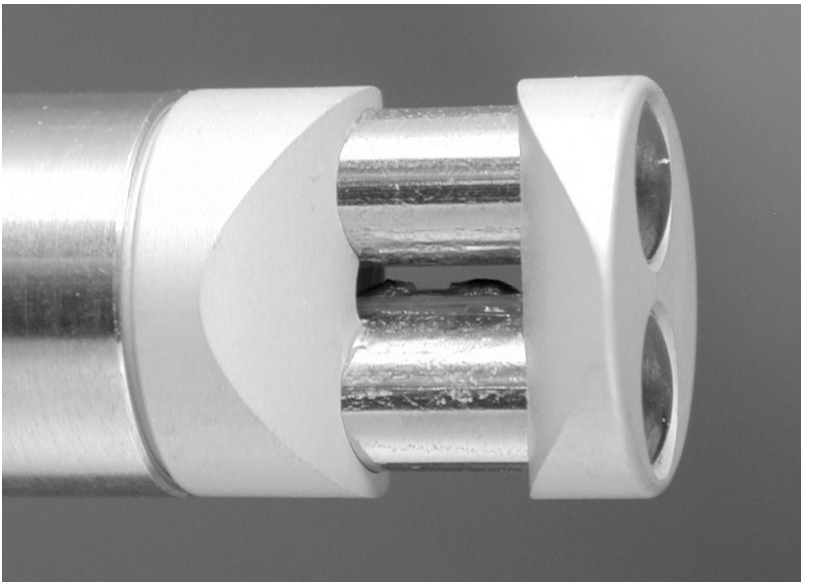

Fig. 2.18: An example of a package with a gain of +9 to +12dB: the half-inch particle velocity probe (1998).

The bandwidth is defined as the range of frequencies within which a microphone delivers a usable output signal. The Microflown has a low-pass behaviour. The output however is still usable for measurement purposes and when corrected also for audio purposes.

The upper particle velocity limit for a Microflown can be compared with the upper sound pressure limit for a microphone but this is not one of the very important properties. The best way to compare the Microflown with a pressure microphone is by comparing the equivalent noise level in relation to the size. The “A”-weighed level for speech applications and the narrow band (for instance Bw=1Hz) level for measuring purposes.

Similar to pressure microphones the selfnoise of the Microflown improves if the physical dimensions increase. In chapter 4 it is made clear that the selfnoise is related to the inverse of the square root length (or dissipated power) of the Microflown. So twice as long will result in a $\sqrt{2}$ (3dB) improvement of the selfnoise. A well-packaged 1mm long Microflown has an overall size that can be compared with the *Knowles Electronics* 0.1" cylindrical microphone. The selfnoise of this miniature microphone is 27dB(A).

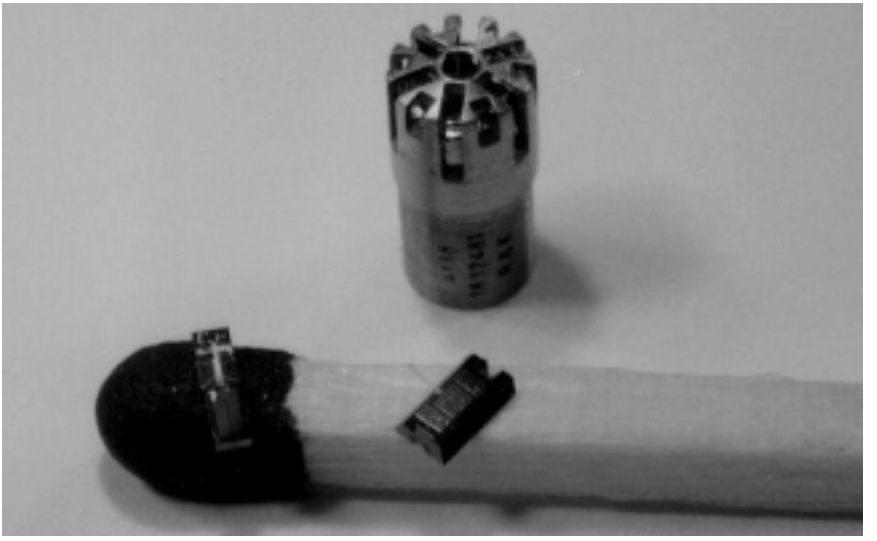

Fig. 2.19: A B&K 1/8” condenser microphone, a cantilever and a bridge type of Microflown.

The Microflown will have a selfnoise of 40dB(A), so can be considered five times as bad. The power dissipation of the Microflown will be much higher: 80mW compared with a 40µW dissipation of the Knowles microphone.

There are three realisations of the Microflown, a cantilever, a bridge type and a through-the-wafer type, see also chapter 3. The bridge and through-the-wafer type of Microflown have a better mechanical stability, since the sensors are connected at two sides. This allows to make the sensors thinner. The corner frequency of the bridge and through-the-wafer type of Microflown type ($f_{heat\ cap}=2kHz$) therefore is somewhat higher than a cantilever type ($f_{heat\ cap}=400Hz$). The Microflown can be used above its corner frequency, for speech purposes the low-pass frequency behaviour should be corrected electrically or with the use of acoustic elements, see chapter D1: "*the pressure Microflown*". For measurement purposes the correction can also be performed digitally afterwards.

Measuring sound intensity

The *sound intensity* in a specified direction 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 considered. This measurement allows obtaining information about noise sources independent on the environment or other noise sources. It is thus a vector quantity; defined as the time averaged product of the sound pressure (a scalar quantity, p) and the corresponding particle velocity (a vector quantity, $\boldsymbol{u}$) at the same position, see Eq. (2.19).

The most obvious way is to use a sound pressure microphone and a particle velocity microphone for this measurement however the direct measurement of particle velocity in one place (by use of the Microflown) is quite new. Traditionally sound-intensity is measured by the use of two closely spaced and (phase) matched pressure microphone. The method is based upon Newton's second law, that is the *equation of motion*:

$$\rho \cdot \frac{\partial u_r}{\partial t} = -\frac{\partial p}{\partial r} \tag{2.26}$$

Where u_r is the particle velocity in one direction r. Since the pressure gradient is proportional to the particle acceleration, the particle velocity

can be obtained by integrating the pressure gradient in respect to time, see also Eq. (2.10).

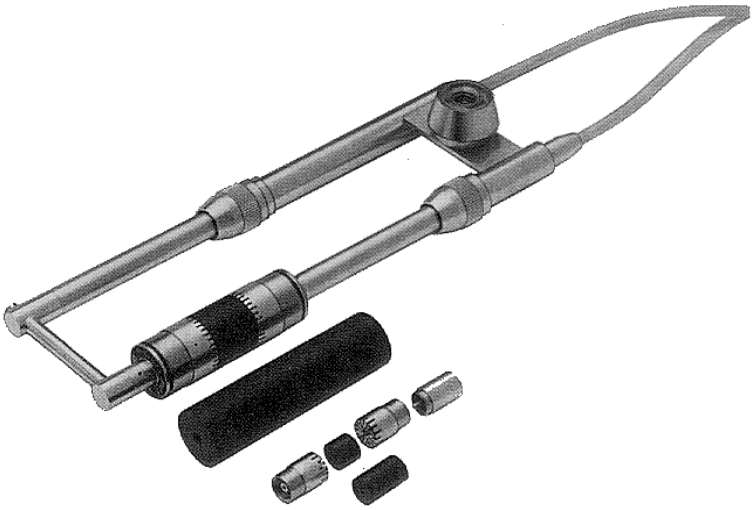

Fig. 2.20: A photograph of a B&K sound acoustic intensity probe.

In practice, the pressure gradient is approximated by measuring the pressure at two points at a close distance Δr.

$$u_r = -\frac{1}{\rho \Delta r}\int (p_a - p_b)\,dt \tag{2.27}$$

This approximation is valid as long as the separation, Δr, is small compared with the wavelength. If the frequency increases the spacing should be chosen smaller. In practice three spacings are being used: 5cm, 12mm and 6mm. A practical sound intensity probe consist of two closely spaced pressure microphones, as for instance the Brüel & Kjær intensity probe fitted with two ½" pressure microphones, see Fig. 2.20.

The block diagram of the measuring electronics of a sound intensity probe is shown in Fig. 2.21.

Estimating the particle velocity by a pressure difference method has a few disadvantages. If the intensity is not constant over Δr, for instance performing nearfield measurements, an estimation error is made. The phase mismatch between the two measuring channels (including the microphones) will be the limiting factor at the low end of the bandwidth.

This is due to the fact that the difference signal of both signals is used to estimate the particle velocity. For low frequencies the wavelength becomes very large compared to the spacing so a phase mismatch becomes more important. For more details about the pros and cons of p-p and p-**u** sound intensity measurements see chapter B1: "*A comparison of the p-u method with the p-p method*".

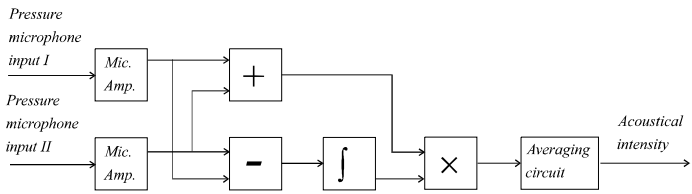

Fig. 2.21: The block diagram of how to obtain the acoustical intensity from two pressure microphone signals.

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