SOUND-MEASURING INSTRUMENTATION, CH. 9

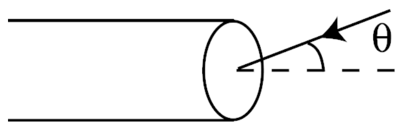
The most common sound measurements are measurements of the sound pressure or the sound power of a sound source. Another quantity which might be measured is the sound intensity (which is needed in some types of sound power measurements, but not all). To find the sound intensity one needs to measure the sound pressure as well as the particle velocity – but the particle velocity is often measured indirectly from two measurements of the sound pressure. Yet another type of measurement is the impedance, which can be used to characterize a surface. Also an impedance measurement requires knowledge of the sound pressure and particle velocity – but also here we can manage with just sound pressure measurements.

So, sound pressure measurements are really very central to all sound measurements, and then we require a transducer which converts sounds pressure into an electrical signal – the microphone.

In underwater measurements, the situation is very similar, but the most common transducer is called a hydrophone. More on that in later chapters.

MICROPHONES, 9.3 – 9.6

Microphones are transducers which transform a sound pressure *p* (most common) or particle velocity *u* into an electric voltage. Microphones always have a directivity that can be described with a directivity function $D(\theta,\varphi)$. Many microphones are rotationally symmetrical, which means that the directivity function is solely a function of the angle θ:



$$e_{out} \propto \begin{cases} p \cdot D(\theta) \\ u_x \cdot D(\theta) \end{cases}$$

> Note: In this course we don't go into the details of a microphone's directivity, and most measurement microphones are intended to be omnidirectional, that is, $D(\theta) = 1$. However, when microphones are used in speech applications, or for music recordings, then the directivity is highly important. This will be covered more in detail in the course Audio Technology.

APPLICATIONS - REQUIREMENTS

All microphones need to fulfill certain requirements. Some of these requirements are similar for all microphones whereas others differ between applications. Some general requirements are:

• Flat frequency response. Since the microphones have some inevitable directivity, there are two distinct classes, free-field and diffuse-field microphones, see below.

• Adequate dynamic range, i.e., the microphone should be capable of transducing loud enough sounds without clipping, and it should have a low enough (electronic) self-noise that the weakest sounds of interest can be measured.

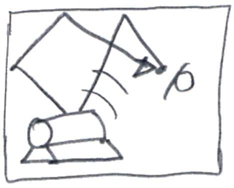
Anechoic / free field measurements (outdoors)

- Measurement of frequency response/sensitivity/directivity of a sound source or loudspeaker
- Measurement of the sound power of a sound source (9.18, 9.19)
- Noise level measurements

Special requirements for the microphone:

- Flat frequency response for at least one direction. These microphones are called free-field microphones.

Indoor measurements

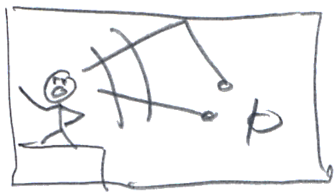


- Measurement of the sound power of a sound source (9.20-9.23)
- Measurement of acoustical characteristics of a room

Special requirements for the microphone:

- Flat frequency response, on average, for all directions. These microphones are called diffuse-field microphones.

Recordings

• Recordings of speech, music in "dry" (≈ anechoic) rooms or performance spaces – concert halls etc.

Special requirements for the microphone:
Especially important for recording microphones: The proper directivity and a low self-noise.

Note: Some microphones may deliberately have a non-flat frequency response. This means that the timbre / spectrum is affected in a way desired by the recording technician / performer.

Speech communication

• Telephones, hands-free systems, teleconferencing microphones, hearing aids
• Sound reinforcement

Special requirements for the microphone:
• High directivity to suppress reverberation and background noise.
• Small size (for telephones and hearing aids), robust, cheap.

Special applications

• Recordings from a distance – interviews, bird songs, etc
These often require especially high directivity, which can be achieved by array techniques – many microphones in an array, or a single microphone with special tubes or channels, the so-called shotgun microphone. A parabolic reflector can also yield a high directivity – at least when the parabolic reflector is larger than the wavelength of the sound wave.

MICROPHONE TYPES

A few different microphone types exist, based on their different transducing principles. All these microphones give an output voltage which is propotional to the sound pressure (over a limited frequency range):

$$e_{out} \propto p$$

Sidenote: This relationship is achieved indirectly. The actual transduction principle translates the velocity, excursion, or acceleration of a little membrane into an electronic voltage. The microphone is then constructed so that $x_{membrane}$, $u_{membrane}$ or $a_{membrane}$ is proportional to the sound pressure!

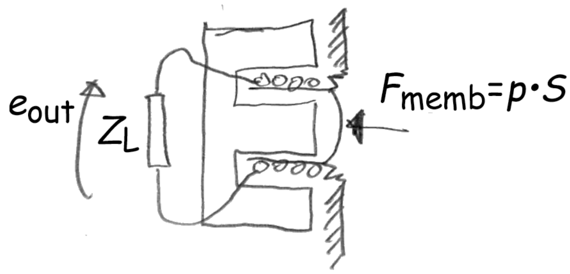
Type	Transduction equation	Properties (FR = frequency response)
Dynamic (moving coil) microphone	$e_{out} = Bl \cdot u_{membrane}$	Robust. Not as flat FR as a condenser mic.
Condenser (capacitor) microphone	$e_{out} = \dfrac{E_0}{X_0} x_{membrane}$	Expensive. Not as robust. Flattest FR. Stable over time and temperature.
Electret microphone	$e_{out} = \dfrac{Q_0}{C_0 X_0} x_{membrane}$	Inexpensive. Flat FR. Not as stable over time and temperature as the condenser microphone.

There are also ribbon microphones (same transduction principle as moving coil microphones) and piezo-electric (or crystal) microphones which have its own transduction principle based on the piezoelectric principle. Old telephones used to have carbon microphones, which used yet another transduction principle – variable resistance. A very new type of microphone is the so-called Microflown sensor, which uses a type of hot wire anemometer, measuring the velocity of the air movements around two very thin wires with a very high precision.

Related transducers are accelerometers, but they measure the vibrations of a surface.

DYNAMIC MICROPHONE

A dynamic microphone is a reversed electrodynamic loudspeaker. A simplified drawing illustrates that a small membrane (of area S) is forced to vibrate by the incident sound wave, $F_{memb} = p \cdot S$. The membrane is glued to a small cylinder on which a coil is wound. The cylinder with the coil sits in a narrow slit of a permanent magnet. Then the transduction equation $e_{out} = Bl \cdot u_{membrane}$ tells us what the output voltage will be, via the magnetic flux density B and the length l of the coil.

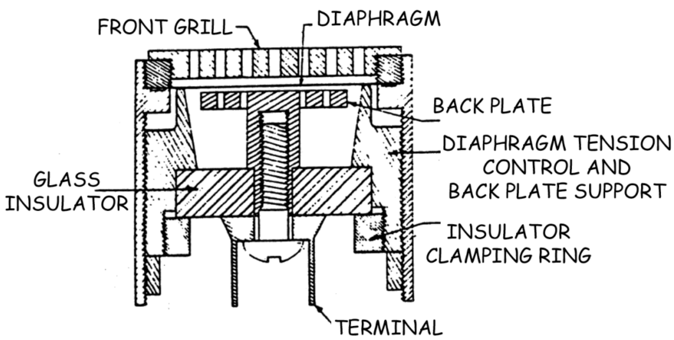


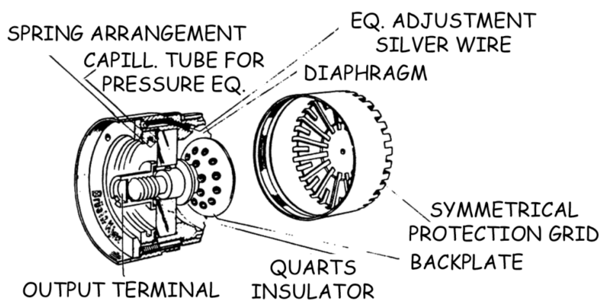
Z_L = input impedance of the amplifier

Note! We say that $F_{memb} = p \cdot S$. This is valid if the microphone is small compared to the wavelength so that $\int p \, dS \approx p \cdot S$!

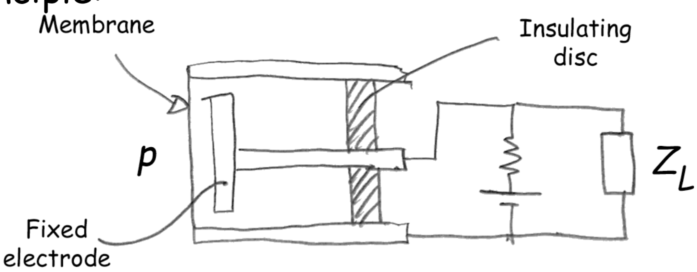
CONDENSER MICROPHONE

Structure:





Working principle:



A DC-voltage, E_0, is connected to the capacitor. E_0 is typically 200 V for measurement microphones and 48 V for studio microphones.

A resistor, R_0, (with high resistance) is connected in series with the DC-voltage resulting in a constant charge on the membrane.

A force, $F = p \cdot S$, will in the same way as for the dynamic microphone lead to vibrations of the very light moving electrode (which is a very thin metallic membrane).

Note 1: The dynamic microphone has a source impedance which is determined by the resistance and inductance of the coil. This can typically be 150 or 300 ohms (resistive). This is low enough that the microphone can be connected to the input of a microphone amplifier via a long cable.

Note 2: The condenser microphone has a source impedance which is capacitive, and that is not suitable for connecting via a cable to a microphone amplifier. Therefore, all condenser microphones have a preamplifier right at the microphone, to yield a suitable source impedance for connections via long cables. This preamplifier can use the polarizing voltage of 200 V or 48 V.

Note 3: Electret microphones are similar to condenser microphones but have a permanent charge on their membrane, so that no polarizing voltage is needed. They still require some kind of preamplifier next to the microphone though.

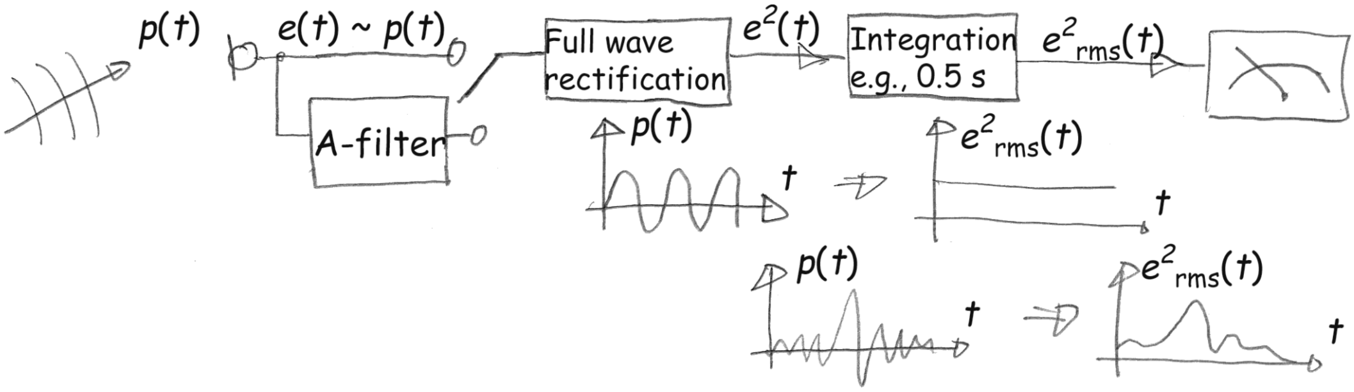
SENSITIVITY, 9.4

The sensitivity of a microphone is usually given in V/Pa (in Europe), or V/μBar (in the US).

Example: A common measurement microphone is the diffuse-field microphone Brüel & Kjær 4942 which has a sensitivity of 50 mV/Pa.

SOUND LEVEL METER, 9.7 – 9.9

A sound level meter calculates the rms-value of the sound pressure, and then converts it to the logarithmic sound pressure level, L_p:



The processing is done digitally, so that the signal is converted by an A/D converter as soon as the signal has "arrived" from the microphone pre-amplifier.

Note1! The digital processing of a sound level meter only handles numbers, and can (of course) not tell us what the sound pressure level is without a calibration. A calibration is done by exposing the microphone to a known sound pressure level. Calibrators are special types of miniature loudspeakers which generate a sound wave with a frequency of 1000 Hz and a sound pressure rms amplitude of 1 Pa inside a small cavity where the microphone fits very tightly (so that an almost airtight volume is created).

Check: what sound pressure level does a sound pressure rms amplitude of 1 Pa correspond to?

Demonstration: measurement of SPL as function of distance from a sound source.

A,B,C-WEIGHTED SOUND PRESSURE LEVEL

In the drawing above, it is indicated that an "A-filter" could be switched in. By using this filter, we will get the A-weighted SPL, in dBA. There are also B- and C-weighting filters, and they give dBB and dBC values. These filters, which try to "mimic" the frequency response of the ear, were presented in chapter 3, and will be further discussed in chapter 10.

EQUIVALENT SOUND PRESSURE LEVEL, 9.9

Most sounds are varying with time, that is, their rms-value isn't constant over time. (A sinusoidal signal gives a sound pressure which varies rapidly up and down each cycle, but its' rms signal will stay constant!) Therefore, one must decide how to quantify the time-varying strength. Sound level meters often have a number of different time-integration choices (some simple meters have just a single fixed setting). The integration time is the time constant of a first-order low-pass filter.

- "Impulse" – integration time = 35 ms
- "Fast" – integration time = 125 ms
- "Slow" – integration time = 1 s

The so-called equivalent level is related – but is based on simply the total average (of p2) over a certain time, whereas Impulse, Fast, Slow uses a low-pass filtering of the variations (of p2)!

$$L_{eq.} = 10 \cdot \log \frac{\frac{1}{T}\int_0^T p_{rms}^2 dt}{p_{ref}^2} = 10 \cdot \log 10\left(\frac{1}{T}\int_0^T 10^{L_p/10} dt\right)$$

So, we are simply calculating the average during, e.g., a workday of 8 hours. At workplaces in Norway, an equivalent level of 85 dBA is the maximum permissible value unless hearing protectors are used.

NB! Since the equivalent level is the average over a longer time, you can expose yourself to a higher level (than 85 dBA) during shorter moments as long as you compensate that with longer periods of lower levels.

OCTAVE BAND FILTERS, 9.11 – 9.12, 9.16

Many regulations specify the maximum A-weighted equivalent sound preussure level during an eight-hour workday, during a 24-hour day, or during a year. Such measurements often don't give enough information about the cause of a noise problem. More detailed measurements are done either using FFT analysis, which can give a high frequency resolution, or octave-band measurements or third-octave band measurements. More advanced sound level meters can often offer 1/1-octave or 1/3 octave band analysis of the sound. The octave bands are standardized band-pass filters, the center frequencies of which are shown in the table below:

Center frequencies [Hz]	
Octave bands	Third-octave bands
…	…
	50
63	63
	80
	100
125	125
	160
	200
250	250
	315
	400
500	500
	630
	…

Note that the center frequencies of every third third-octave band is an octave band center frequency!

The upper frequency of each band can be found by taking

$$f_{n,lower} = \sqrt{f_{n,center} \cdot f_{n-1,center}}$$
$$f_{n,upper} = \sqrt{f_{n,center} \cdot f_{n+1,center}}$$

Example: The octave band of 250 Hz has the lower and upper cut-off frequencies $\sqrt{125 \cdot 250}\ Hz \approx 177\ Hz$ and $\sqrt{250 \cdot 500}\ Hz \approx 354\ Hz$.

MEASUREMENT OF STOCHASTIC SIGNALS, 9.16

Many sound signals are of a stochastic nature, but we are interested in the average (over time) p2-value, such as in the equivalent SPL. Then the uncertainty in our estimate of the p_{rms} value will depend on how long we are averaging, and what bandwidth we are measuring over:

$$\varepsilon_{p-rms} = \frac{1}{2\sqrt{B \cdot T_{measure}}}$$ where B = bandwidth in Hz.

Example: If we want to measure the sound pressure level for a random noise signal in the octave band of 250 Hz, and we measure during 1 second – how large is our uncertainty in dB?

The bandwidth of the 250 Hz octave band is $250\left(\sqrt{2}-1/\sqrt{2}\right)$ Hz ≈ 177 Hz.

Then $\varepsilon_{p-rms} \approx 1/2\sqrt{177\cdot 1} \approx 0.038$, which means that the true rms-value with 95% probability is within 1±2*0.038 (for a normal distribution, approx. 95% of the values are within ±2 standard deviations!). This translates to a <u>level uncertainty</u> of 20*log10(1-2*0.038) to 20*log10(1+2*0.038) ≈ [-0.7 dB to 0.6 dB] relative to the true long-term average.

SUGGESTED PROBLEMS, P. 210

1, 2, 6, 7, 10.

TERMINOLOGY

<u>English</u>	<u>Norsk</u>
Sound pressure	Lydtrykk
SPL = sound pressure level	Lydtrykksnivå
Particle velocity	Partikkelhastighet
Sound power	Lydeffekt
Transducer	Omvandler
Anechoic	Ekkofri
Sensitivity	Følsomhet
Condenser microphone	Kondensatormikrofon
Sound level meter	Lydnivåmåler
Equivalent SPL	Ekvivalentnivå